Avaya Solution & Interoperability Test Lab

Application Notes for Teledex iPhone LD4200 and ND2200 Series with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Teledex iPhone™ which were compliance tested with Avaya Communication Manager and Avaya SIP Enablement Services. The overall objective of the interoperability compliance testing is to verify Teledex iPhone functions in an environment that is comprised of Avaya Communication Manager, Avaya SIP Enablement Services, as well as various Avaya SIP and H.323 IP Telephones.

Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab at the request of the Solutions Marketing Team.

TABLE OF CONTENTS

1. Introduction

These Application Notes describe the procedures for configuring Teledex iPhone™ which were compliance tested with Avaya Communication Manager and Avaya SIP Enablement Services. The overall objective of the interoperability compliance testing is to verify Teledex iPhone functions in an environment that is comprised of Avaya Communication Manager, Avaya SIP Enablement Services, as well as various Avaya SIP and H.323 IP Telephones.

Teledex iPhone SIP LD4200 and ND2200 Series phones are SIP endpoints designed for hotel environment. Teledex iPhone integrates into a SIP environment, providing the cost control benefits of managing one network for both voice and data services to guest rooms. During the compliance test, two types of Teledex iPhone, ND2210S and LD4210S, were evaluated. The ND2210S and LD4210S SIP phones utilize the same firmware and provide the same functionality. However, the LD4210S SIP phone consists of a 5.6-inch color touch screen display, while the ND2210S SIP phone does not have a display.

These Application Notes assume that Avaya Communication Manager and Avaya SES are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document.

1.1. Supported Features

Table 1 gives a summary of the features supported and tested with Teledex iPhone. Some features are supported locally at the telephone, while others are only available with Avaya SIP Enablement Services and Avaya Communication Manager. Some Avaya Communication Manager features shown in **Table 1** are invoked by using Teledex iPhone VoIP Phone Configuration Portal, or by dialing a Feature Name Extension (FNE). Speed dial button on the telephone can be programmed to an FNE.

Features	Teledex iPhone SIP LD4200	Teledex iPhone SIP ND2200	Notes
Basic Calling Features			
Extension to extension call	X	X	
Intercept tones/displays	X	X	
Call Waiting	X	X	Enable Multiple Line Appearance in the VoIP Phone Configuration Portal
Do not Disturb	X	X	Enable Do Not Disturb feature in the VoIP Phone Configuration Portal
Speed Dial Buttons	X	X	
Message Waiting Support	X	X	
Call Transfer	X	X	Enable Call Transfer in the VoIP Phone Configuration Portal. Use FLASH button on the phone to transfer calls.
Other Features			
Call Hold	X	X	
Music on Hold	X	X	
Call Forwarding Unconditional	X	X	Configure Feature Name Extension (FNE)
Call Forward Busy	X	X	Configure FNE
Call Forward No Answer	X	X	Configure FNE
Conference – 3rd party added	X	X	Configure FNE
Conference – 3rd party joins	X	X	Configure FNE
Call Park/Unpark	X	X	Configure FNE
Call Pickup	X	X	Configure FNE
Automatic Redial	X	X	Configure FNE
Last Number Dialed	X	X	Configure FNE
Priority Call	X	X	Configure FNE
Send All Calls	X	X	Configure FNE
Send All Calls Cancel	X	X	Configure FNE
Transfer to Voice Mail	X	X	Configure FNE

Table 1

2. Network Topology

Figure 1 illustrates a sample configuration consisting of Avaya S8720 Servers controlling G650 Media Gateways, an Avaya SIP Enablement Services (SES) server, and Teledex iPhone, Avaya 4626 Series H.323 IP Telephones, Avaya 9600 Series SIP IP Telephones, and Avaya 2420 Series Digital Telephones.

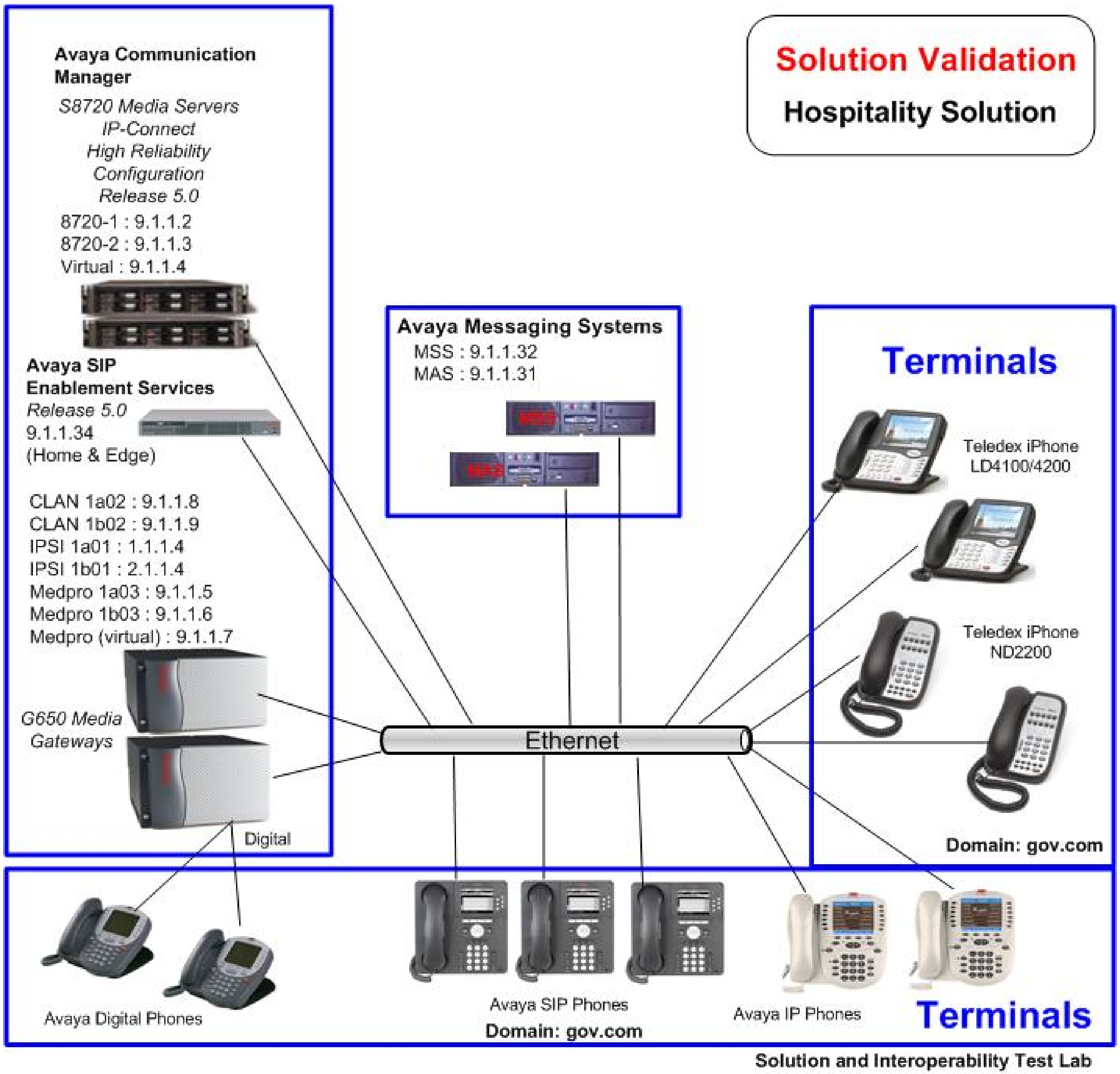


Figure 1 – Avaya Hospitality Solution Reference Configuration

3. Equipment and Software Validated

The following equipment and software were used for the sample configuration:

Device Description	Versions Tested
Avaya Communication Manager - S8720 Servers	Release 5.0 (R015x.00.0.825.4)
Avaya G650 Media Gateway - IPSI (TN2312BP) - CLAN (TN799DP) - MedPro (TN2602AP)	 - HW15 FW039 - HW01 FW156 - HW02 FW033
Avaya SES (Combined Home-Edge)	Release 5.0 (825.31)
Avaya 4626 Series H.323 Telephones	R2.4
Avaya 9600 Series SIP Telephones	R2.2.0.7
Avaya 2420 Digital Telephones	N/A
Avaya Modular Messaging	Release 3.1
Teledex iPhone SIP LD4100/4200 Series SIP ND2200 Series	Boot Version 2.02 Boot Build Date: July 22, 2008 App Version: 1.10.02-080722 App Build Date: July 22, 2008

4. Configure Avaya Communication Manager

This section describes the procedure for setting up a SIP trunk between Avaya Communication Manager and Avaya SES. The steps include setting up an IP codec set, an IP network region, an IP node name, a signaling group, a trunk group, and a SIP station. Before a trunk can be configured, it is necessary to verify if there is enough capacity to set up an additional trunk. The highlights in the following screens indicate the values used. Default values may be used for all other fields.

These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. Avaya SIP telephones and Teledex iPhone SIP LD4200 and ND200 Series phones are configured as off-PBX telephones in Avaya Communication Manager.

4.1. Capacity Verification

Enter the **display system-parameters customer-options** command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses. If not, contact an authorized Avaya account representative to obtain additional licenses. Each Teledex iPhone will use one Off-PBX Telephone license.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V15                              Software Package: Standard
       Location: 1                             RFA System ID (SID): 1
       Platform: 6                             RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 44000 347
                                      Maximum Stations: 36000 148
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 400   3
                    Maximum Off-PBX Telephones -   OPS: 400   16
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 400   0




        (NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 2**, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   40
           Maximum Concurrently Registered IP Stations: 12000 34
             Maximum Administered Remote Office Trunks: 8000  0
Maximum Concurrently Registered Remote Office Stations: 12000 0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 100   4
                   Maximum Video Capable IP Softphones: 100   4
                       Maximum Administered SIP Trunks: 100   20
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   2
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

        (NOTE: You must logoff & login to effect the permission changes.)
```

4.2. IP Codec Set

This section describes the steps for administering an IP codec set in Avaya Communication Manager. This IP codec set is used in the IP network region for communications between Avaya Communication Manager and Avaya SES. Enter the **change ip-codec-set <c>** command, where **c** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 4.3** when configuring an IP network region to specify which audio codecs may be used within and between network regions. In the sample configuration, only one network region is used.

For integration with Avaya Communication Manager, enter G.711MU. Retain all other default field values.

```
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.711MU          n          2        20
 2:
 3:
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

4.3. Configure IP Network Region

This section describes the steps for administering an IP network region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- Authoritative Domain – This should match the SIP Domain value on Avaya SES, in **Section 5.1**. In the sample configuration, **gov.com** was used.
- Codec Set – Enter the IP codec set number as provisioned in **Section 4.2**.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: gov.com
    Name: Main Region - HQ
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 48      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 48       Use Default Server Parameters? y
        Video PHB Value: 34
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 4      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
```

4.4. Configure IP Node Name

This section describes the steps for setting an IP node name for Avaya SES in Avaya Communication Manager. Enter the **change node-names ip** command, and add a node name for Avaya SES along with its IP address. The CLAN board (in the case of an Avaya S8300 Server, Processor-Ethernet, procr) will be used as well in subsequent steps in these Application Notes.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
CLAN-01A02          9.1.1.8
SES1                9.1.1.34
```

4.5. Configure SIP Signaling Group

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES. Enter the **add signaling-group <s>** command, where **s** is an available signaling group, and configure the following:

- Group Type – Set to **sip.**
- Near-end Node Name **–** Set to **CLAN-01A02** as displayed in **Section 4.4**.
- Far-end Node Name – Set to the Avaya SES name configured in **Section 4.4**.
- Far-end Network Region – Set to the region configured in **Section 4.3**.
- Far-end Domain **–** This should match the SIP Domain value in **Section 5.1**. In the sample configuration, **gov.com** was used.
- Direct IP-IP Audio Connections – Set it to "y".
- IP Audio Hairpinning **–** Set it to "y".

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

 Group Number: 2             Group Type: sip
                       Transport Method: tls

         IP Video? n




   Near-end Node Name: CLAN-01A02            Far-end Node Name: SES1
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
       Far-end Domain: gov.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): 120
```

4.6. Configure SIP Trunk Group

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES. Enter the **add trunk-group <t>** command, where **t** is an unallocated trunk group, and configure the following:

- Group Type – Set to **sip**.
- Group Name – Enter a descriptive name.
- TAC– Set to any available trunk access code that is valid in the provisioned dial plan.
- Signaling Group – Set to the Group Number field value configured in **Section 4.5**.
- Number of Members **–** Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used, but still within the maximum number allowed (see **Section 4.1**).
- Service Type – Set to **tie**.

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

Group Number: 2                   Group Type: sip          CDR Reports: y
  Group Name: CM to SES                  COR: 1        TN: 1        TAC: 102
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                        Signaling Group: 2
                                                      Number of Members: 20
```

On **Page 5** of the trunk-group form, verify that all trunk group members are assigned, as shown below.

```
display trunk-group 2                                           Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/20
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  20

       Port             Name
  1: T00024             CM to SES
  2: T00025             CM to SES
  3: T00026             CM to SES
  4: T00027             CM to SES
  5: T00028             CM to SES
  6: T00029             CM to SES
  7: T00030             CM to SES
  8: T00031             CM to SES
  9: T00032             CM to SES
 10: T00033             CM to SES
 11: T00034             CM to SES
 12: T00035             CM to SES
 13: T00036             CM to SES
 14: T00037             CM to SES
 15: T00038             CM to SES
```

4.7. Configure Feature Name Extensions

The Feature Name Extensions (FNEs) can be defined using the **change off-pbx-telephone feature-name-extensions** command. This command is used to support both OPS and Extension to Cellular. The fields that have been populated reflect the features tested during compliance testing as per **Table 1**.

```
change off-pbx-telephone feature-name-extensions set 1         Page   1 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
                     Set Name: Teledex-Phones

     Active Appearance Select: 48800
          Automatic Call Back: 48801
   Automatic Call-Back Cancel: 48802
             Call Forward All: 48803
  Call Forward Busy/No Answer: 48804
          Call Forward Cancel: 48805
                    Call Park: 48806
        Call Park Answer Back: 48807
                 Call Pick-Up: 48808
         Calling Number Block:
       Calling Number Unblock:
         Conference on Answer:
        Directed Call Pick-Up: 48812
        Drop Last Added Party:
    Exclusion (Toggle On/Off):
   Extended Group Call Pickup: 48815
       Held Appearance Select:
```

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

       Idle Appearance Select: 48818
           Last Number Dialed: 48819
         Malicious Call Trace:
  Malicious Call Trace Cancel:
          Off-Pbx Call Enable:
         Off-Pbx Call Disable:
                Priority Call: 40016
               Send All Calls: 48822
        Send All Calls Cancel: 48823
          Transfer On Hang-Up:
       Transfer to Voice Mail: 48825
      Whisper Page Activation:
```

4.8. Specify Class of Service

Use the **change cos** command to set the appropriate service permissions to support OPS features (shown in bold). In the sample configuration, a COS of 1 was used. On Page 2, set the value of **VIP Caller** to "y" only if all calls made by telephones with this COS should be priority calls. Priority call indication (e.g., distinctive ring and display of "Priority") is only supported on Avaya Digital and IP telephones.

```
change cos                                                      Page   1 of   2
CLASS OF SERVICE

                                0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 Auto Callback                  n  y  y  n  y  n  y  n  y  n  y  n  y  n  y  n
 Call Fwd-All Calls             n  y  n  y  y  y  n  y  y  n  y  y  y  n  n  y
 Data Privacy                   n  n  n  n  n  y  y  y  y  n  y  n  n  y  y  y
 Priority Calling               n  y  n  n  n  y  n  n  n  y  y  y  y  y  y  y
 Console Permissions            n  y  n  n  n  n  n  n  n  n  y  n  n  n  n  n
 Off-hook Alert                 n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                    n  n  n  n  n  y  y  n  n  n  n  n  n  n  n  n
 Restrict Call Fwd-Off Net      y  y  y  y  y  y  y  y  y  y  n  y  y  y  y  y
 Call Forwarding Busy/DA        n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Personal Station Access (PSA)  y  y  y  y  y  y  y  y  y  y  n  y  y  n  n  n
 Extended Forwarding All        n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding B/DA       n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Trk-to-Trk Transfer Override   y  y  y  y  y  y  y  n  n  n  n  n  n  n  n  n
 QSIG Call Offer Originations   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Contact Closure Activation     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

```
change cos                                                      Page   2 of   2
                                CLASS OF SERVICE

                                0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 VIP Caller                     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

4.9. Specify Class of Restriction

Use the **change cor** command to enable applicable calling features. To use the Directed Call Pickup feature, the **Can Use Directed Call Pickup** and **Can Be Picked Up By Directed Call Pickup** fields must be set to “y” for the affected stations. In the sample configuration, the telephones were assigned to COR 10.

```
change cor 10                                                   Page   1 of  23
                              CLASS OF RESTRICTION

               COR Number: 10
          COR Description: Teledex Stations

                      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             Hear VDN of Origin Annc.? n
        Send ANI for MFE? n               Add/Remove Agent Skills? 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
```

```
change cor 10                                                   Page   4 of  23
                              CLASS OF RESTRICTION

CALLING PERMISSION (Enter "y" to grant permission to call specified COR)

  0? y      15? y      30? y      44? y      58? y      72? y      86? y
  1? y      16? y      31? y      45? y      59? y      73? y      87? y
  2? y      17? y      32? y      46? y      60? y      74? y      88? y
  3? y      18? y      33? y      47? y      61? y      75? y      89? y
  4? y      19? y      34? y      48? y      62? y      76? y      90? y
  5? y      20? y      35? y      49? y      63? y      77? y      91? y
  6? y      21? y      36? y      50? y      64? y      78? y      92? y
  7? y      22? y      37? y      51? y      65? y      79? y      93? y
  8? y      23? y      38? y      52? y      66? y      80? y      94? y
  9? y      24? y      39? y      53? y      67? y      81? y      95? y
 10? y      25? y      40? y      54? y      68? y      82? y      96? y
 11? y      26? y      41? y      55? y      69? y      83? y      97? y
 12? y      27? y      42? y      56? y      70? y      84? y      98? y
 13? y      28? y      43? y      57? y      71? y      85? y      99? y
 14? y      29? y
```

4.10. Add Coverage Path

Use the **add coverage path** command to configure the coverage path to be used for the voice messaging hunt group, which is group “h7” in the sample configuration. The default values can be used for the **COVERAGE CRITERIA**.

```
add coverage path 1
                                 COVERAGE PATH

                   Coverage Path Number: 1
                                                  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: 2
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h7             Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

4.11. Configure Hunt Group

Enter **add hunt-group h**, where **h** is an unused hunt group number. The following fields were configured for the compliance test:

- Group Name – Provide a descriptive name of the group.
- Group Extension – Provide the hunt group extension.

```
add hunt-group 7                                              Page   1 of  60
                                  HUNT GROUP

            Group Number: 7                                 ACD? n
              Group Name: MM                              Queue? n
         Group Extension: 44444                          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:
```

On Page 2, the following fields were configured for the compliance test.

- Message Center – Set to **qsig-mwi**.
- Voice Mail Number – Set to **44444**.
- Routing Digits (e.g. AAR/ARS Access Code) **- 8**.

```
add hunt-group 7                                              Page   2 of  60
                                  HUNT GROUP

                       LWC Reception: none          AUDIX Name:



                      Message Center: qsig-mwi
                     Send Reroute Request: y
                   Voice Mail Number: 44444
Routing Digits (e.g. AAR/ARS Access Code): 8          Provide Ringback? n
                TSC per MWI Interrogation? n
```

4.12. Configure SIP Endpoints

Enter **add station s**, where **s** is an extension valid in the provisioned dial plan. Assign the same extension as the media server extension administered in Avaya SES. Use "6408D+" for the **Station Type,** "X" for the **Port**, and be sure to include the **Coverage Path** for voice messaging or other hunt group if applicable. Use the **COS** and **COR** values administered in the previous sections. Enter the user name in the **Name** field. Use defaults for the other fields on Page 1.

```
add station 44025                                             Page   1 of   5
                                     STATION

Extension: 44025                         Lock Messages? n               BCC: 0
     Type: 6408D+                        Security Code:                  TN: 1
     Port: X                           Coverage Path 1: 1               COR: 10
     Name: Teledex ND1 Line 1          Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 2          Personalized Ringing Pattern: 1
             Data Module? n                      Message Lamp Ext: 44025
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english

          Survivable COR: internal                 Media Complex Ext:
   Survivable Trunk Dest? y                             IP SoftPhone? n
                                                 Remote Office Phone? n
                                                            IP Video? n
```

On Page 2, note the following:

- If this telephone will have a bridged appearance for another telephone (see Page 3 for this station), then **Bridged Call Alerting** should be set to "y", so that this telephone will ring when the other telephone is called. Note that no other operational behaviors of the bridged appearance feature apply to SIP telephones (e.g., off-hook indication, bridge-on, etc.).
- By default, the last call appearance is reserved for outgoing calls from the telephone. If call waiting is to be locally configured on the telephone, set the **Restrict Last Appearance** field to "n", so that a second call to that extension will be presented at the telephone rather than sent to the coverage path (e.g., voice messaging).
- Select "qsig-mwi" for **MWI Served User Type**.

```
add station 44025                                           Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                    Idle Appearance Preference? n
 Per Button Ring Control? n                  Bridged Idle Line Preference? n
   Bridged Call Alerting? n                      Restrict Last Appearance? n
  Active Station Ringing: single

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic                   Audible Message Waiting? n
    MWI Served User Type: qsig-mwi             Display Client Redirection? n
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 44025                        IP Audio Hairpinning? n
```

On Page 4 under the heading **BUTTON ASSIGNMENTS**, fill in the number of call appearances ("call-appr" buttons) that are to be supported for the telephone. In the configuration example, the Teledex iPhone has been configured with 3 call appearances, which can handle 3 calls (i.e., one active plus two calls waiting).

```
add station 44025                                           Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                          Headset? n
       Jack:                                          Speaker? n
      Cable:                                         Mounting: d
      Floor:                                      Cord Length: 0
   Building:                                        Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                          5:
 2: call-appr                          6:
 3: call-appr                          7:
 4:                                    8:
```

Enter the **add off-pbx-telephone station-mapping** command and configure the following:

- Station Extension – Enter the extension configured above.
- Application – Set to **OPS**.
- Phone Number – Enter the number that Teledex iPhone will use for registration and call termination. In the example below, the Phone Number is the same as the Station Extension, but is not required to be the same.
- Config Set – Set to **1**, which contains the default values.
- Trunk Select **–** Set to the trunk group number configured in **Section 4.6**.

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

 Station        Application Dial   CC  Phone Number      Trunk        Config
 Extension                  Prefix                       Selection    Set
 44025             OPS          -      44025              2           1
                                -
```

On Page 2, change the **Call Limit** to match the number of “call-appr” entries in the **add station** form. Also make sure that **Mapping Mode** is set to “both”.

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

 Station       Call       Mapping      Calls       Bridged        Location
 Extension     Limit      Mode         Allowed     Calls
 44025          6       both           all         none
```

Repeat add station and add off-pbx-telephone station-mapping as necessary to configure additional SIP endpoint extensions.

The following screen shows the OPS stations created during testing.

```
list off-pbx-telephone station-mapping                               Page   1

                    STATION TO OFF-PBX TELEPHONE MAPPING

Station         Appl   CC  Phone Number     Config Trunk   Mapping      Calls
Extension                                   Set    Select  Mode         Allowed

44025           OPS        44025            1  /   2       both         all
44026           OPS        44026            1  /   2       both         all
44027           OPS        44027            1  /   2       both         all
44028           OPS        44028            1  /   2       both         all
44029           OPS        44029            1  /   2       both         all
```

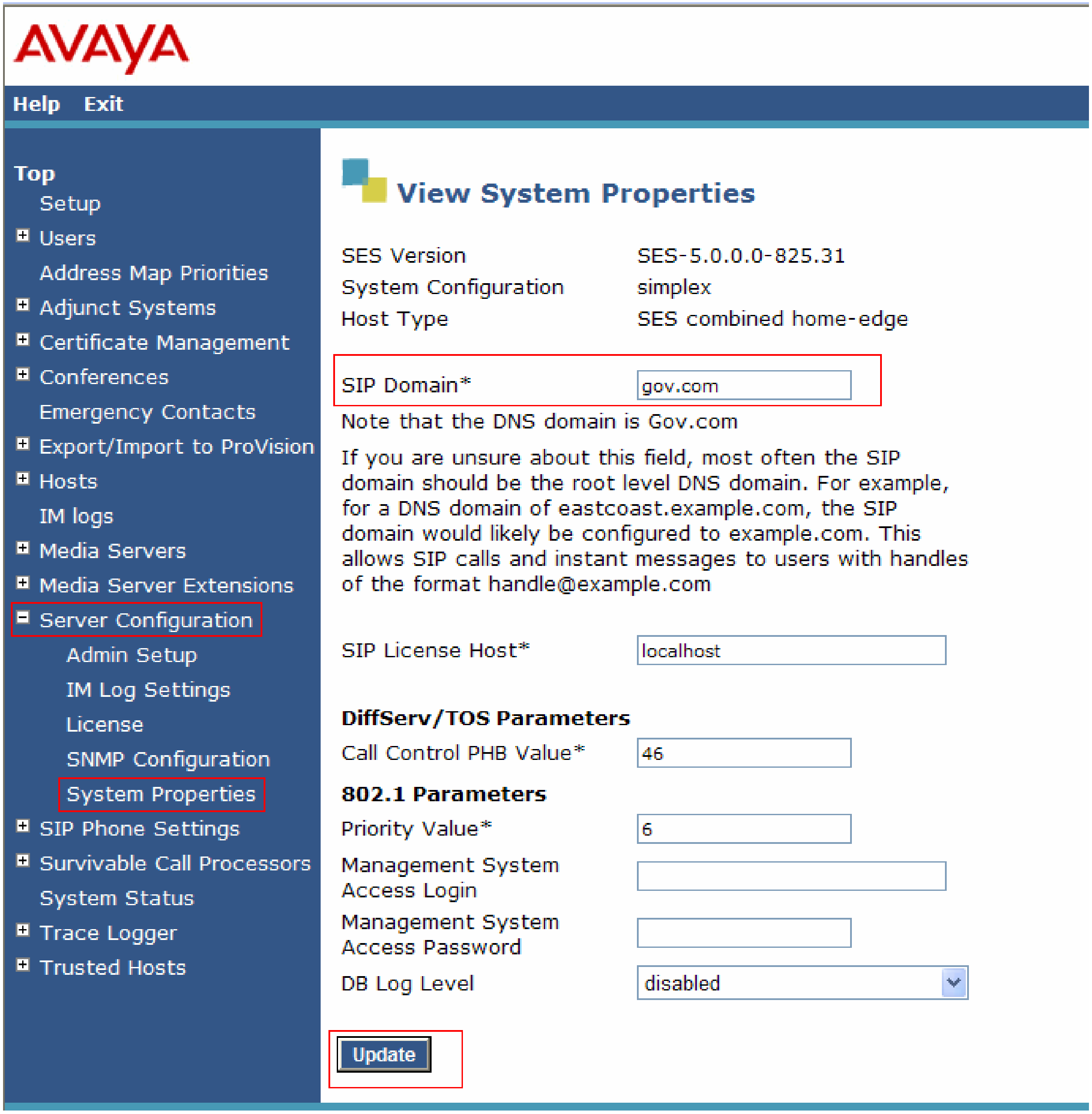
5. Configure Avaya SES

This section describes the steps for creating a SIP trunk between Avaya SES and Avaya Communication Manager. SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. Teledex iPhone™ will register with Avaya SES using the SIP user accounts. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

5.1. Configure SES Server Properties

Launch a web browser, enter **https://<IP address of SES server>/admin** in the URL, and log in with the appropriate credentials. Click on the **Launch SES Administration Interface** link upon successful log in.

In the **Integrated Management SIP Server Management** page, select the **Server Configuration → System Properties** link from the left pane of the screen. Verify the SIP Domain matches the Far-end Domain field value configured for the signaling group in Avaya Communication Manager in **Section 4.5**. Click on the **Update** button if a field change was necessary.

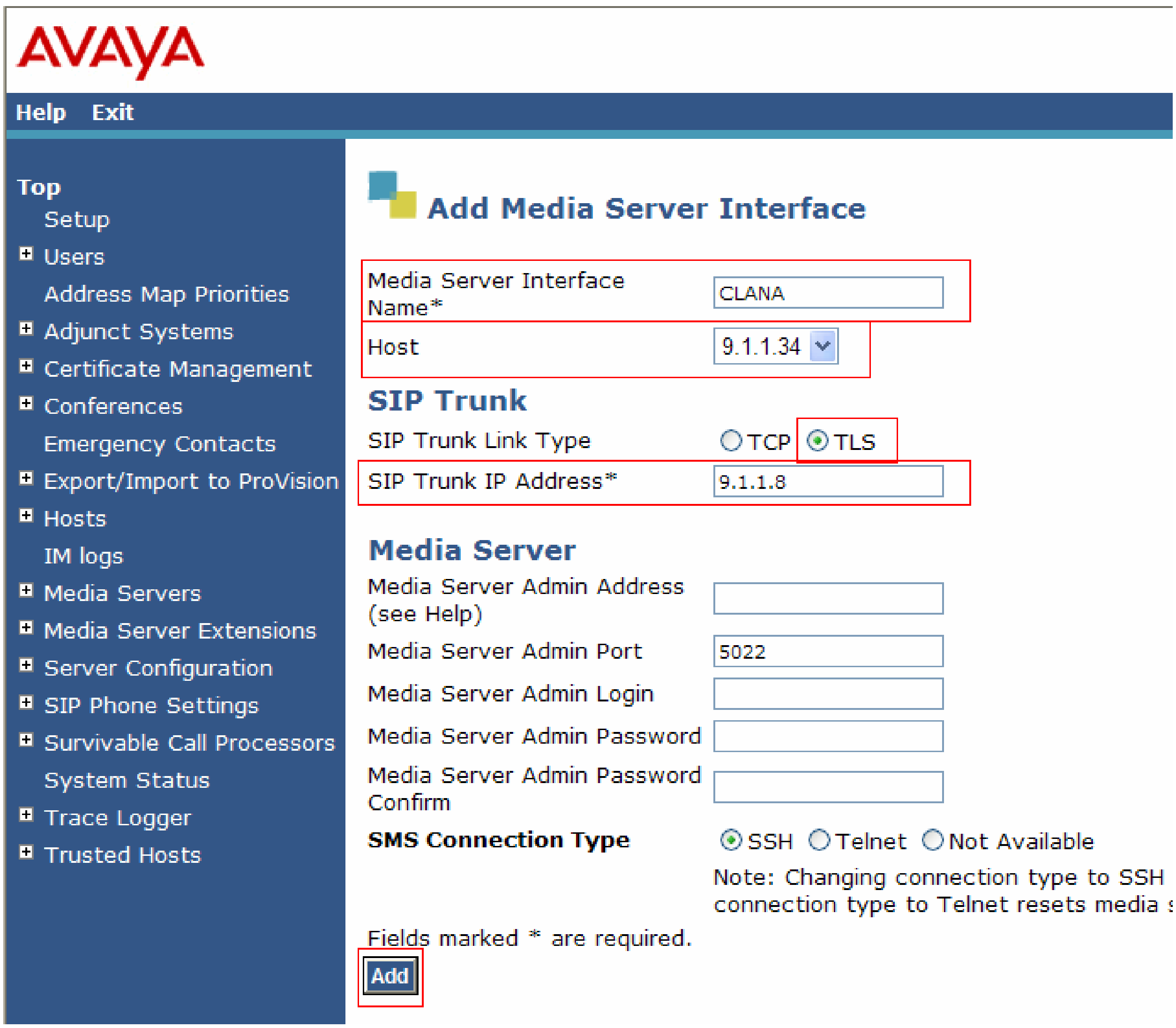


5.2. Configure Media Server Interface

This section provides steps to add SIP-enabled media servers to the SIP domain. In the **Integrated Management SIP Server Management** page, select the **Media Servers → Add** link from the left pane of the screen. The following screen shows the Add Media Server Interface page. The highlighted fields were configured for the compliance test:

- Media Server Interface Name – Enter a descriptive name for the media server interface.
- Host – From the drop-down list of IP addresses, select the IP address of the Avaya SES server to be associated with the Media Server interface.
- SIP Trunk Link Type – Select **TLS**.
- SIP Trunk IP Address – Enter the IP address for the media server's CLAN (or procr) IP interface that terminates the SIP link from Avaya SES (see **Section 4.4**).

Click **Add** when finished.

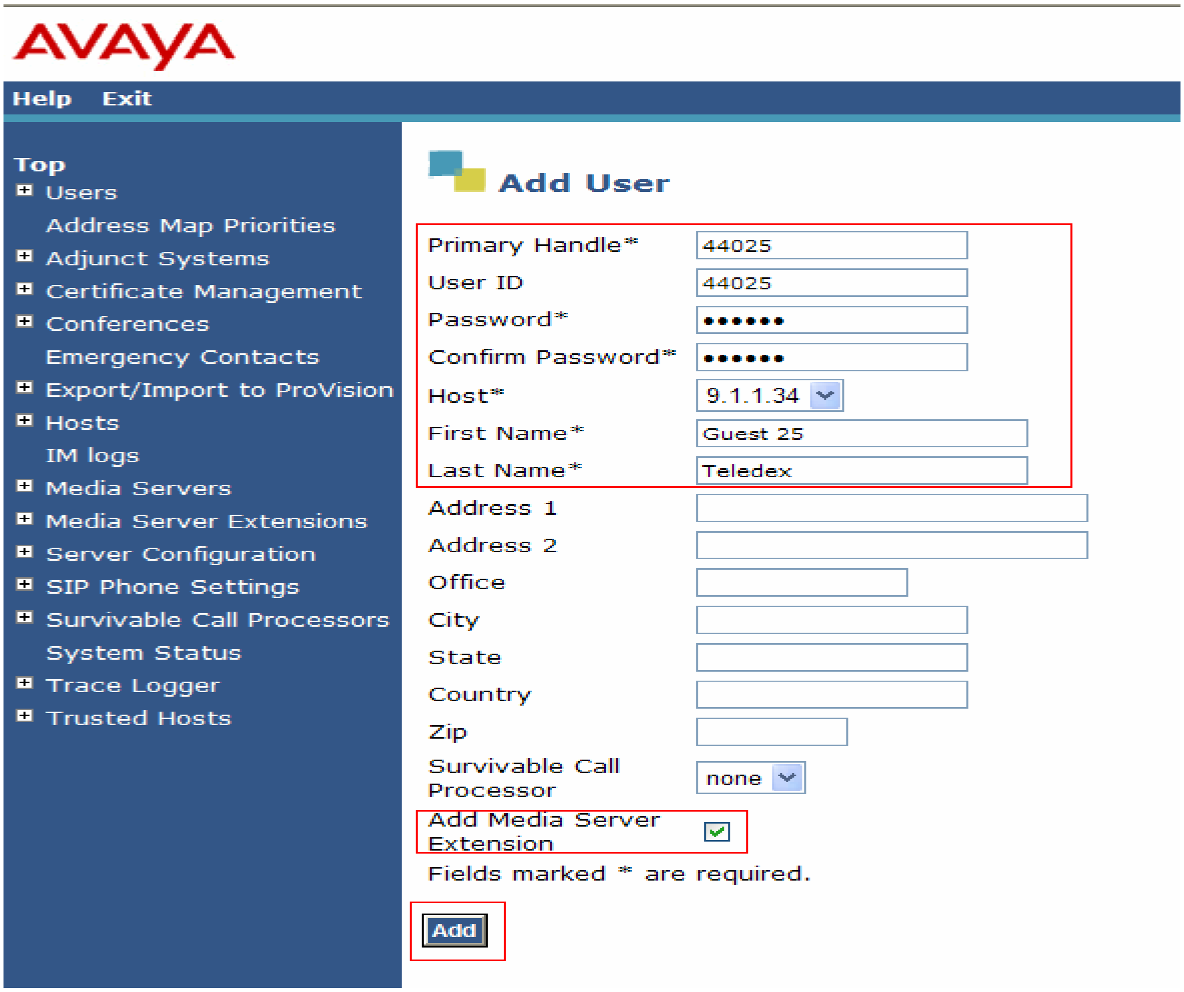


5.3. Configure Users

This section provides steps to add users to be administered in the Avaya SES database. In the Integrated Management SIP Server Management page, select the **Users → Add** link from the left pane of the screen. The highlighted fields were configured for the compliance test:

- Primary Handle – Enter the phone number of iPhone. This number was configured in **Section 4.12**.
- User ID – Set to any descriptive name.
- Password / Confirm Password – Enter a password of at least 6 alphanumeric characters; both field entries must match exactly. Note the password entered in the screen below. This will be needed in **Section 6.2**, (for the Phone Password field).
- Host – From the drop-down list of IP addresses, select the host serving the domain for this user. The IP address of the current server is selected by default.
- First Name – Enter the first name of the user in alphanumeric characters.
- Last Name – Enter the last name of the user in alphanumeric characters.
- Add Media Server Extension – Select this field to associate a new extension number with this user in the database. The Add Media Server Extension screen will be displayed next, after this user profile has been added.

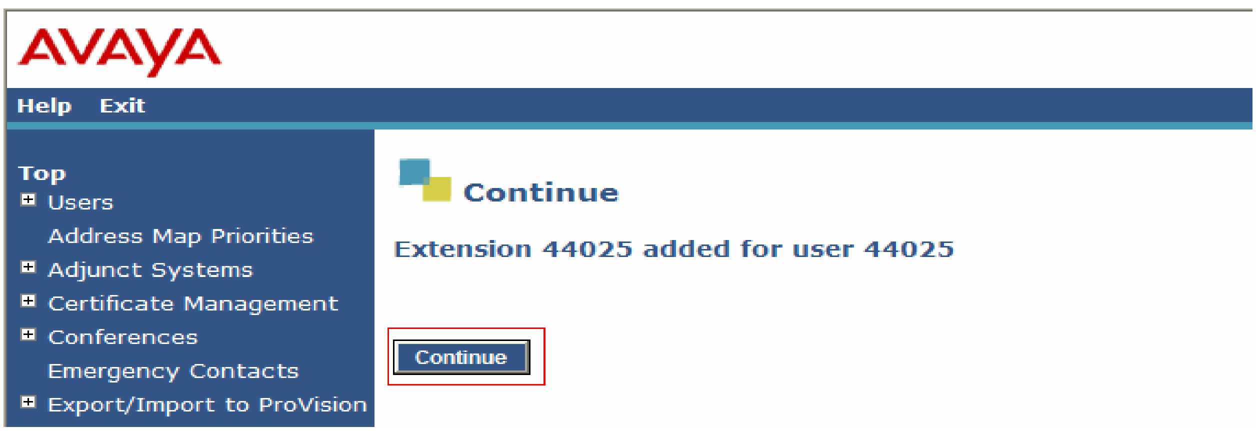
Click **Add** when finished.



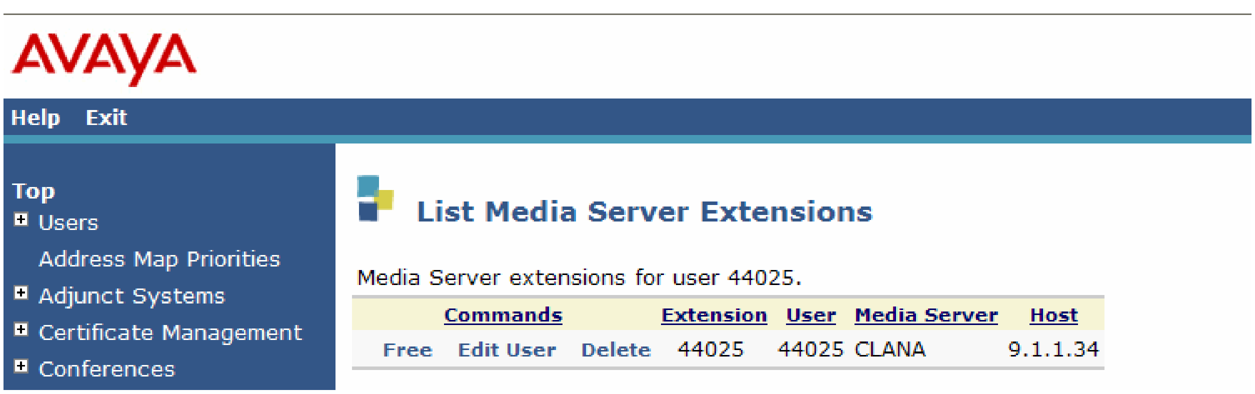
Click on the **Continue** button.

At the next screen, enter the numeric telephone extension to be created in the database. This should match the Phone Number entry on the off-pbx-telephone station-mapping form in **Section 4.12**. Select the extension's media server from the drop-down list. Click on the **Add** button.

Click on the **Continue** button.



Following screen will be displayed which will list the above created Media Server Extension.

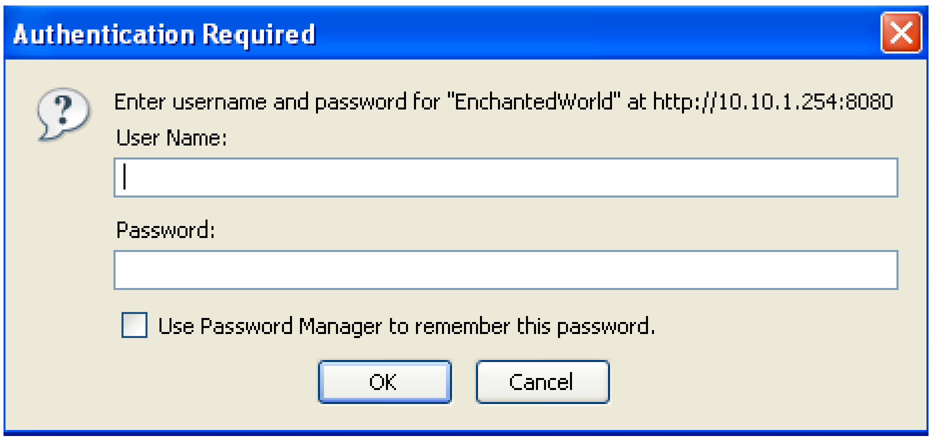


Repeat Steps in **Section 5.3** for every SIP endpoint created in **Section 4.12**.

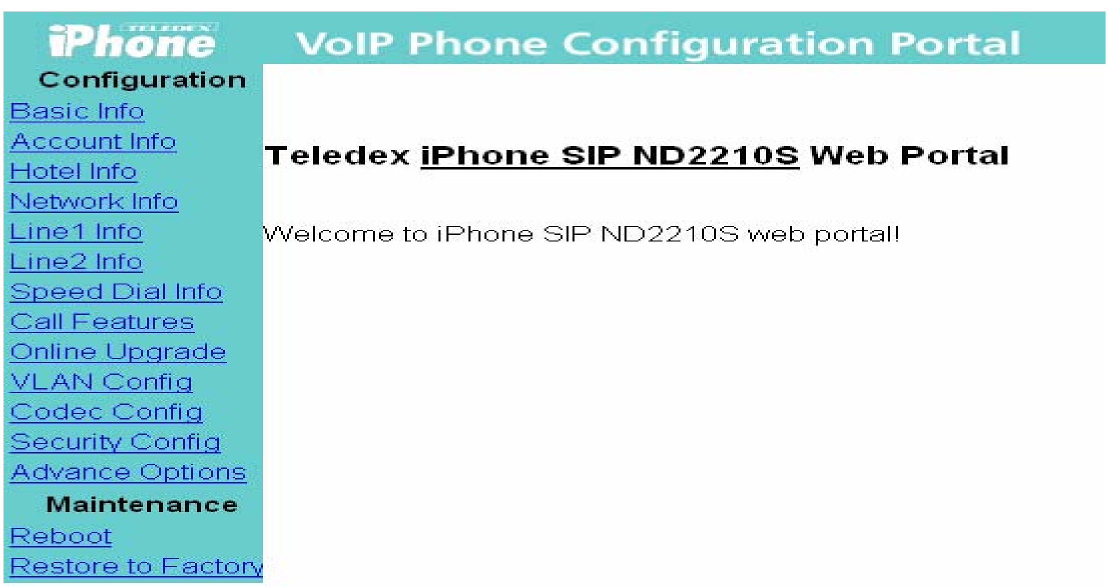
6. Configure Teledex iPhone

This section describes the steps to configure Teledex iPhone. SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. Teledex iPhone will register with Avaya SES using the SIP user accounts. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

The default address of the Teledex iPhone as shipped from the factory is 10.10.1.254. Launch a web browser, enter **http://<10.10.1.254:8080>** in the URL, and log in with the appropriate credentials for accessing the iPhone Configuration Web Client page.



The Teledex iPhone Configuration Portal will be displayed after successfully log in.



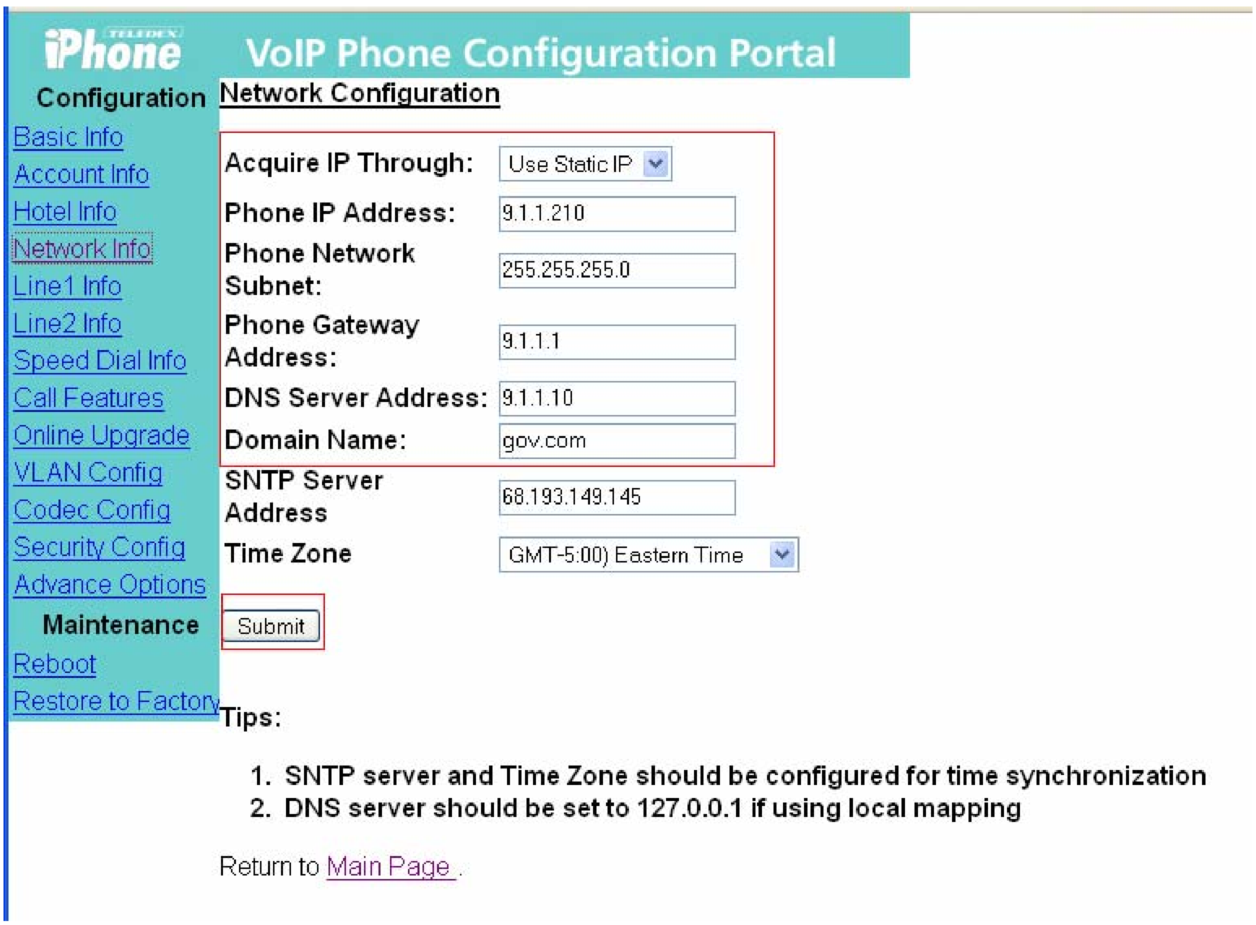
6.1. Configure Network Info

From the Teledex iPhone Configuration Portal, select **Network Info** from the left side menu.

Configure the following fields:

- Acquire IP Through – Select **Use Static IP** from the drop down menu. Teledex IPhone supports both static and DHCP, however static IP addresses are recommended in the Hospitality environment.
- Phone IP Address – Enter the IP address of the iPhone.
- Phone Network Subnet – Enter the subnet mask of the network that the iPhone is located.
- Phone Gateway Address – Enter the default gateway IP address of the iPhone.
- DNS Server Address – Enter the DNS server IP address.
- Domain Name – Enter the SIP domain configured in **Section 5.1**.

Click on **Submit** button.



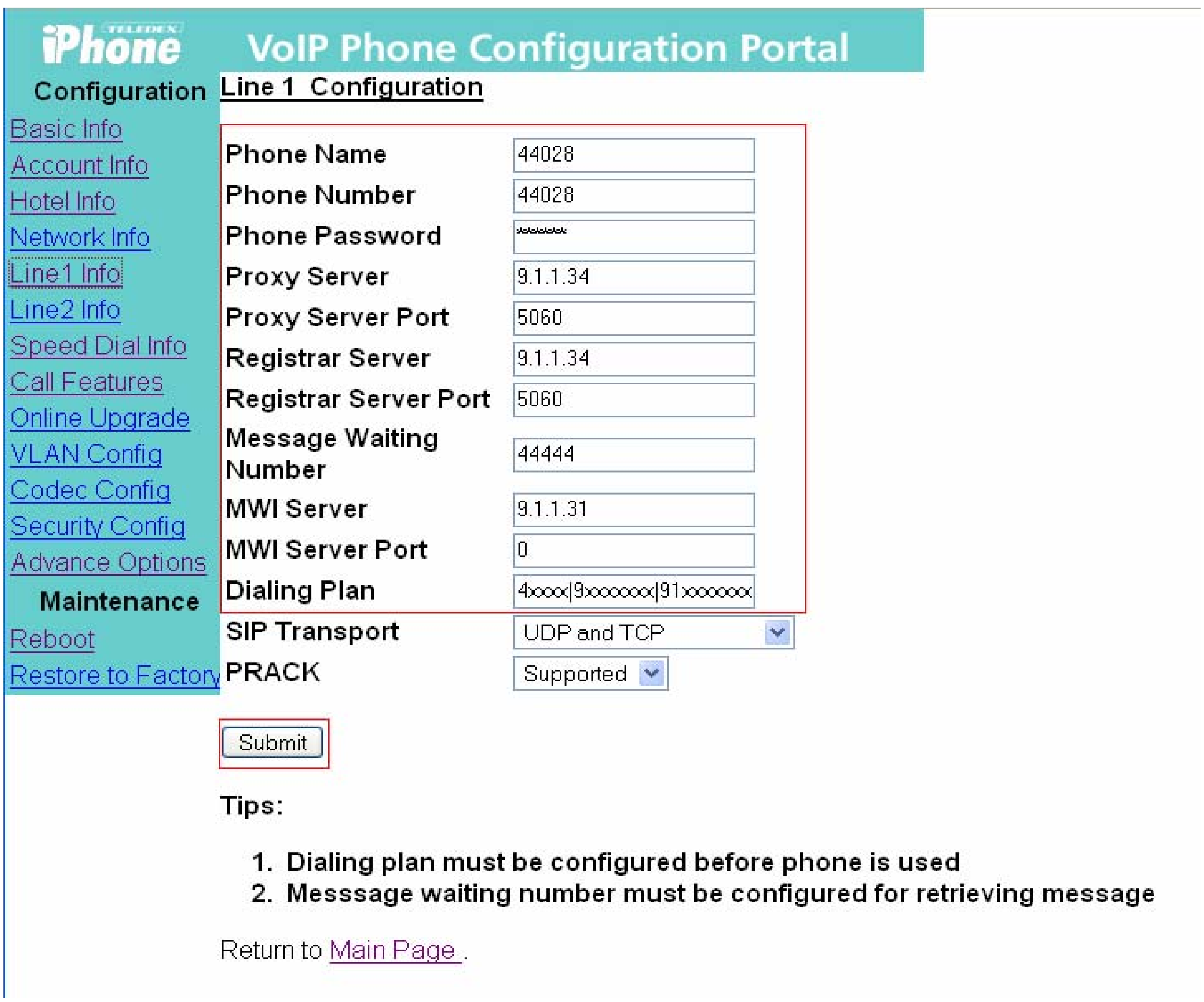
6.2. Configure Line 1 Info

From the Teledex iPhone Configuration Portal, select **Line Info 1** from the left side menu.

Configure the following fields:

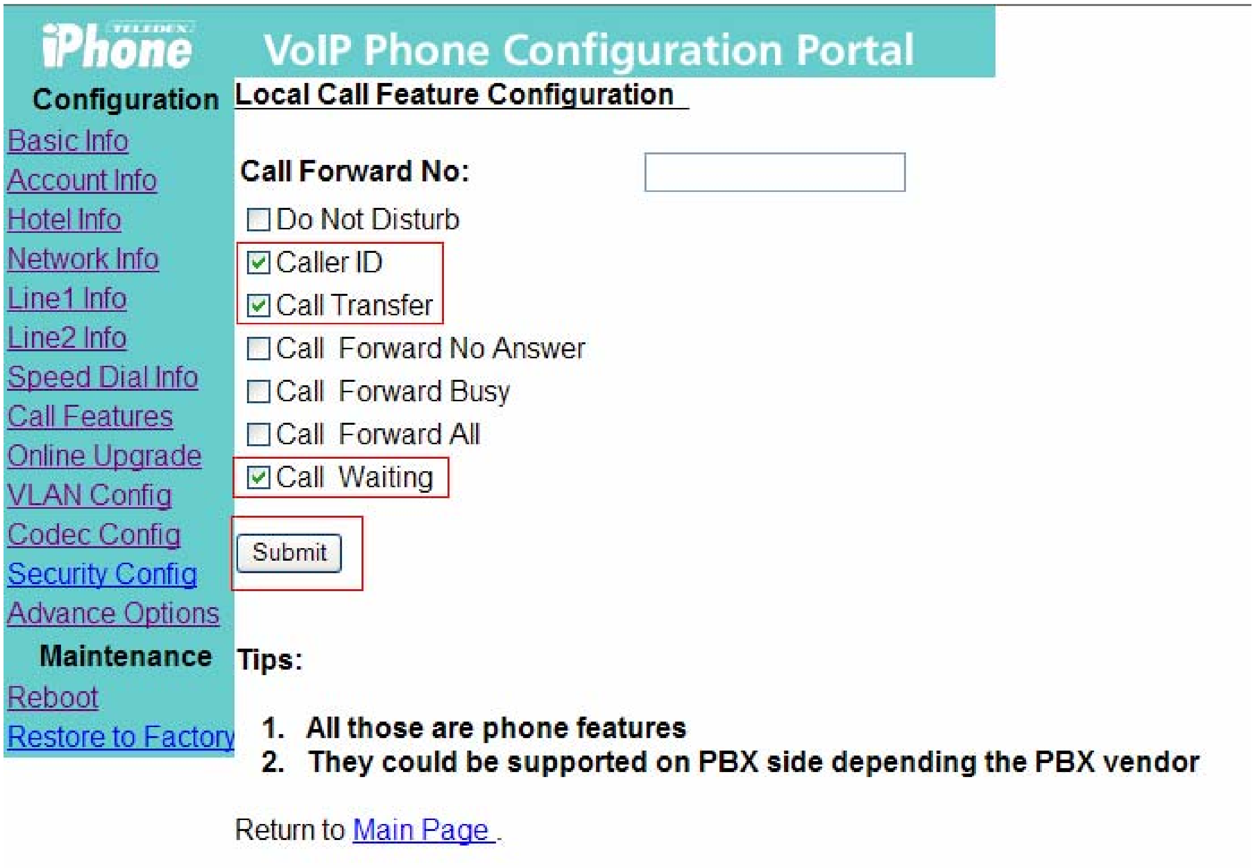
- Phone Name – Enter a descriptive phone name
- Phone Number – Enter the iPhone extension number configured in **Section 5.3**.
- Phone Password – Enter the iPhone extension password configured in **Section 5.3**.
- Proxy Server – Enter the SIP server IP address.
- Proxy Server Port – Enter 5060.
- Registrar Server – Enter the SIP server IP address.
- Registrar Server Port – Enter 5060.
- Message Waiting Number – Enter the Voice Mail Number configured in **Section 4.11**.
- Dialing Plan – If needed, modify the default dialing string.

Click on **Submit** button to save the Line 1 configuration.



6.3. Configure Call Features

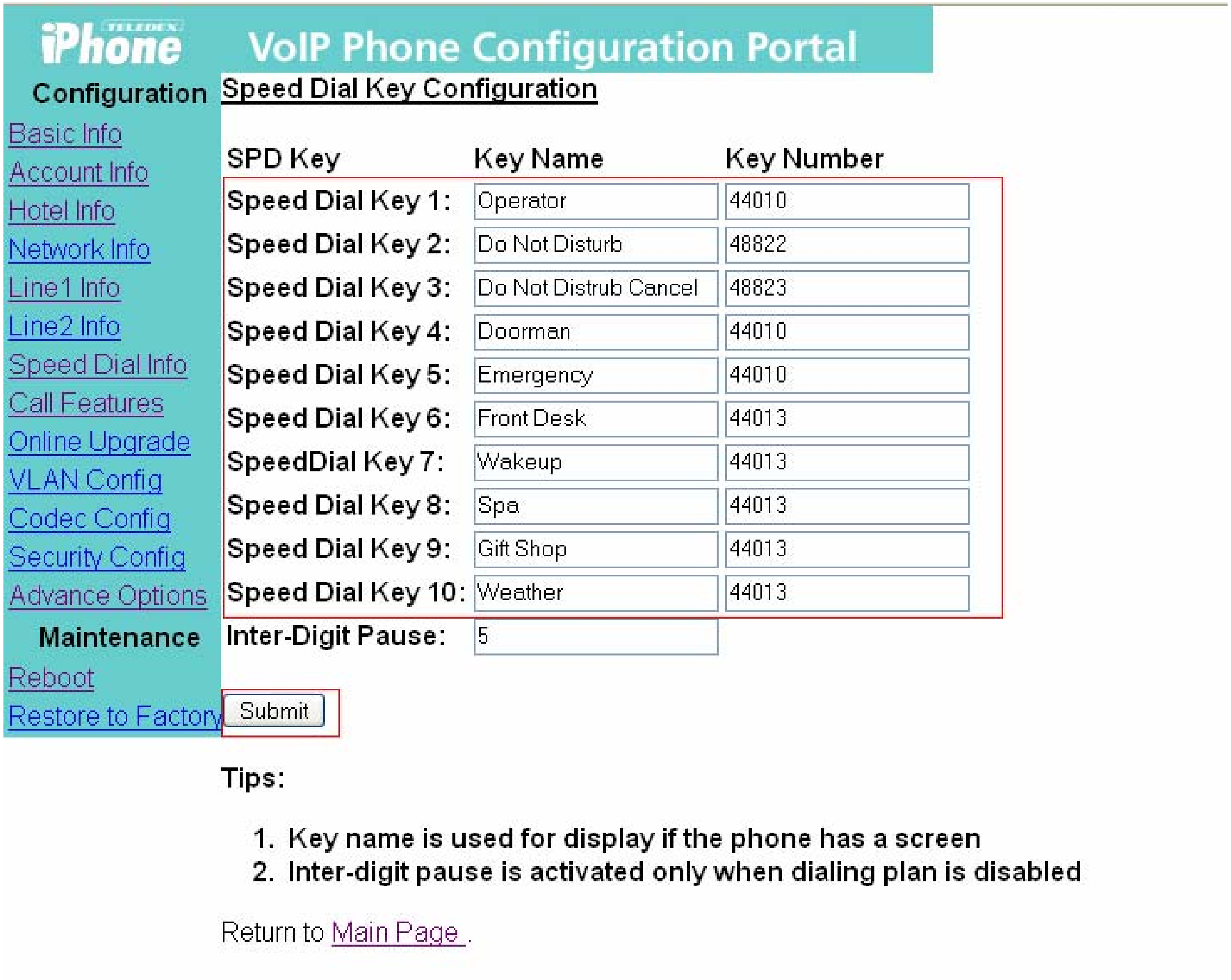
From the Teledex iPhone Configuration Portal, select **Call Features** from the left side menu. Enable Caller ID, Call Transfer and Call Waiting and click on **Submit** button to save.



6.4. Configure Auto Dial Keys

From the Teledex iPhone Configuration Portal, select **Speed Dial Info** from the left side menu to configure the speed dial feature. Provide an extension or Feature Name Extension as configured in **Section 4.7** for each Speed Dial Key field.

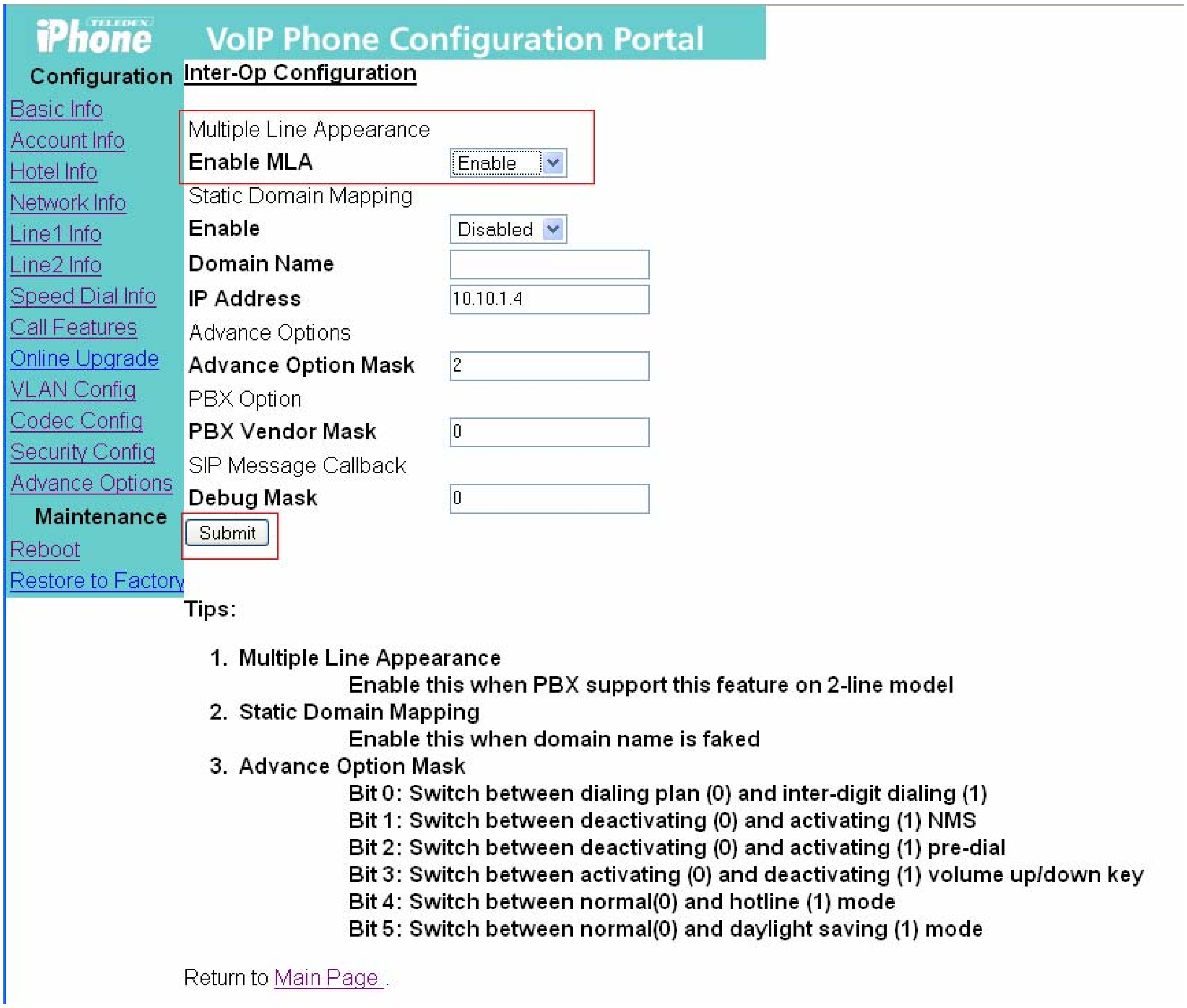
Click on the **Submit** button to save the Speed Dial Key configuration.



6.5. Enable Multiple Line Appearances

From the Teledex iPhone Configuration Portal, select **Advance Options** from the left side menu.

Select Enable from the drop down menu for Multiple Line Appearance Enable MLA. Click on **Submit** button.



6.6. Reboot iPhone

The Teledex iPhone needs to be rebooted after modifying any configurations. From the left side menu, select **Reboot**. Click on **Reboot** button.

7. Interoperability Compliance Testing

The interoperability compliance testing included basic feature and serviceability testing.

The Hospitality solution consisting of Teledex iPhone SIP LD4200 and ND2200 Series phones was successfully tested with Avaya Communication Manager and Avaya SIP Enablement Services. Refer to **Table 1** for all the supported features for the Teledex iPhones which were tested as part of the compliance testing.

The serviceability testing focused on verifying the ability of Teledex iPhone to recover from adverse conditions, such as:

- Server interchanges / Reset
- Disconnect/reconnect of Ethernet cable to Teledex iPhones.

7.1. General Test Approach

All test cases were performed manually. The general approach was to register the Teledex iPhone to Avaya SES, place outbound calls, and receive inbound calls. Serviceability failures were simulated by disconnecting cables, and by executing reset system commands from the Avaya Communication Manager System Access Terminal interface.

7.2. Test Results

Basic calling features worked which included extension to extension call, call hold, do not disturb, call forwarding, and conference.

Few observations were made during testing which are noted below:

1. Using the FNE to make a priority call from Teledex iPhone to another Teledex iPhone is delivered as a normal call, i.e. does not get the priority call display and ringtones. Using the FNE to make a priority call from Teledex iPhone to Avaya IP/SIP phone is delivered as a priority call, i.e. the call display and ringtone indicate that the call is a priority call.

2. Dialing the Teledex iPhone extension when Send All Calls (SAC) feature is activated on the Teledex iPhone (via the FNE) will ring the Teledex iPhone once before it goes to the voice mailbox.

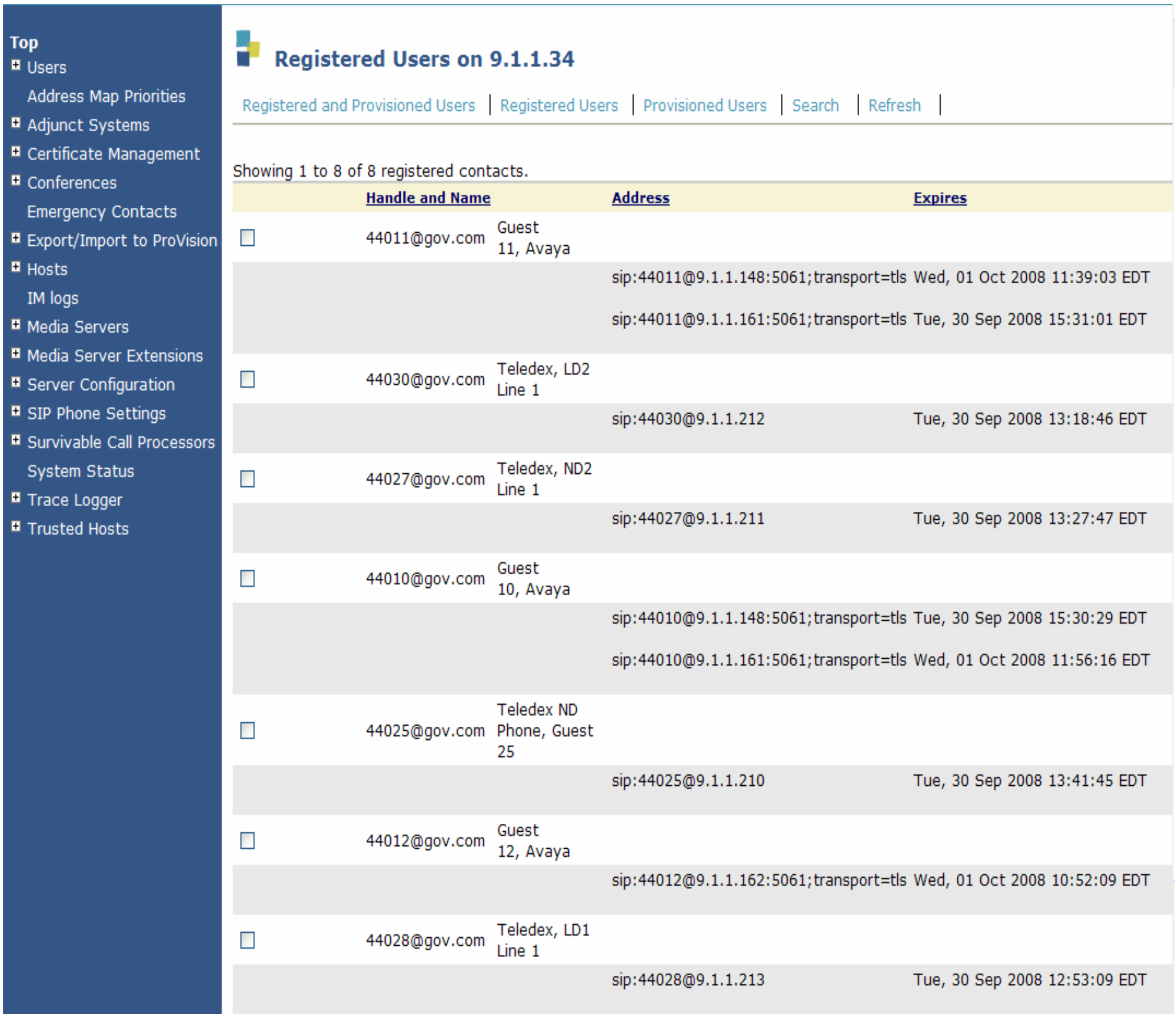
3. Cannot use Presence or Instant Messaging on the Teledex iPhones LD4200 and ND2200. These features only work with Avaya SIP phones.

4. The FLASH button on the Teledex iPhone SIP LD4200 phone is on the touch screen. At times, the user has to press hard to activate the feature.

8. Verification and Troubleshooting

This section provides the tests that can be performed to verify proper configuration of Avaya Communication Manager, Avaya SES, and Teledex iPhones:

1. In the Avaya SES **Integrated Management SIP Server Management** page, select the **Users → Registered Users** link from the left pane of the screen. Verify all SIP endpoints are registered.



2. Using a network protocol analyzer, verify correct REGISTER messages are exchanged between Avaya SES and Teledex iPhones.

9. Support

Technical support for Teledex iPhones can be obtained by contacting via the support link at iphonesupport@teledex.com or by calling the support telephone number at 408-574-2661.

10. Conclusion

These Application Notes describe the configuration steps required for Teledex iPhones to interoperate with Avaya Communication Manager and Avaya SIP Enablement Services. All feature and serviceability test cases were completed.

11. Terminology

AWU	Auto Wake-UP
DND	Do Not Disturb
FAC	Feature Access Code
FNE	Feature Name Extension
PMS	Property Management System
SES	SIP Enablement Services
SMS	Short Messaging Services

12. Additional References

Avaya documentation can be located at http://support.avaya.com

[1] *Administrators Guide for Avaya Communication Manager*, Document 03-300509, Issue 4.0, Release 5.0, Jan 2008.
http://support.avaya.com/elmodocs2/comm_mgr/r5.0/03-300509_4.pdf

[2] *Installing, Administrating, Maintaining, and Troubleshooting SIP Enablement Services*, Document 03-600768, Jan 2008.
http://support.avaya.com/elmodocs2/sip/03_600768_5.pdf

The following document was provided by Teledex.

[3] Teledex iPhone SIP ND2200 Series IP Phone User's Guide.

[4] Teledex iPhone SIP LD4100/4200 Series IP Phone User's Guide.

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.