



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.....	3
1.1.	Supported Features.....	3
2.	Network Topology.....	5
3.	Equipment and Software Validated.....	6
4.	Configure Avaya Communication Manager.....	6
4.1.	Capacity Verification.....	7
4.2.	IP Codec Set.....	8
4.3.	Configure IP Network Region.....	8
4.4.	Configure IP Node Name.....	9
4.5.	Configure SIP Signaling Group.....	9
4.6.	Configure SIP Trunk Group.....	10
4.7.	Configure Feature Name Extensions.....	11
4.8.	Specify Class of Service.....	12
4.9.	Specify Class of Restriction.....	13
4.10.	Add Coverage Path.....	14
4.11.	Configure Hunt Group.....	14
4.12.	Configure SIP Endpoints.....	15
5.	Configure Avaya SES.....	18
5.1.	Configure SES Server Properties.....	18
5.2.	Configure Media Server Interface.....	20
5.3.	Configure Users.....	21
6.	Configure Teledex iPhone™.....	24
6.1.	Configure Network Info.....	25
6.2.	Configure Line 1 Info.....	26
6.3.	Configure Call Features.....	27
6.4.	Configure Auto Dial Keys.....	28
6.5.	Enable Multiple Line Appearances.....	29
6.6.	Reboot iPhone.....	30
7.	Interoperability Compliance Testing.....	30
7.1.	General Test Approach.....	30
7.2.	Test Results.....	31
8.	Verification and Troubleshooting.....	31
9.	Support.....	32
10.	Conclusion.....	32
11.	Terminology.....	33
12.	Additional References.....	33

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 – 3 rd party added	X	X	Configure FNE
Conference – 3 rd 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.

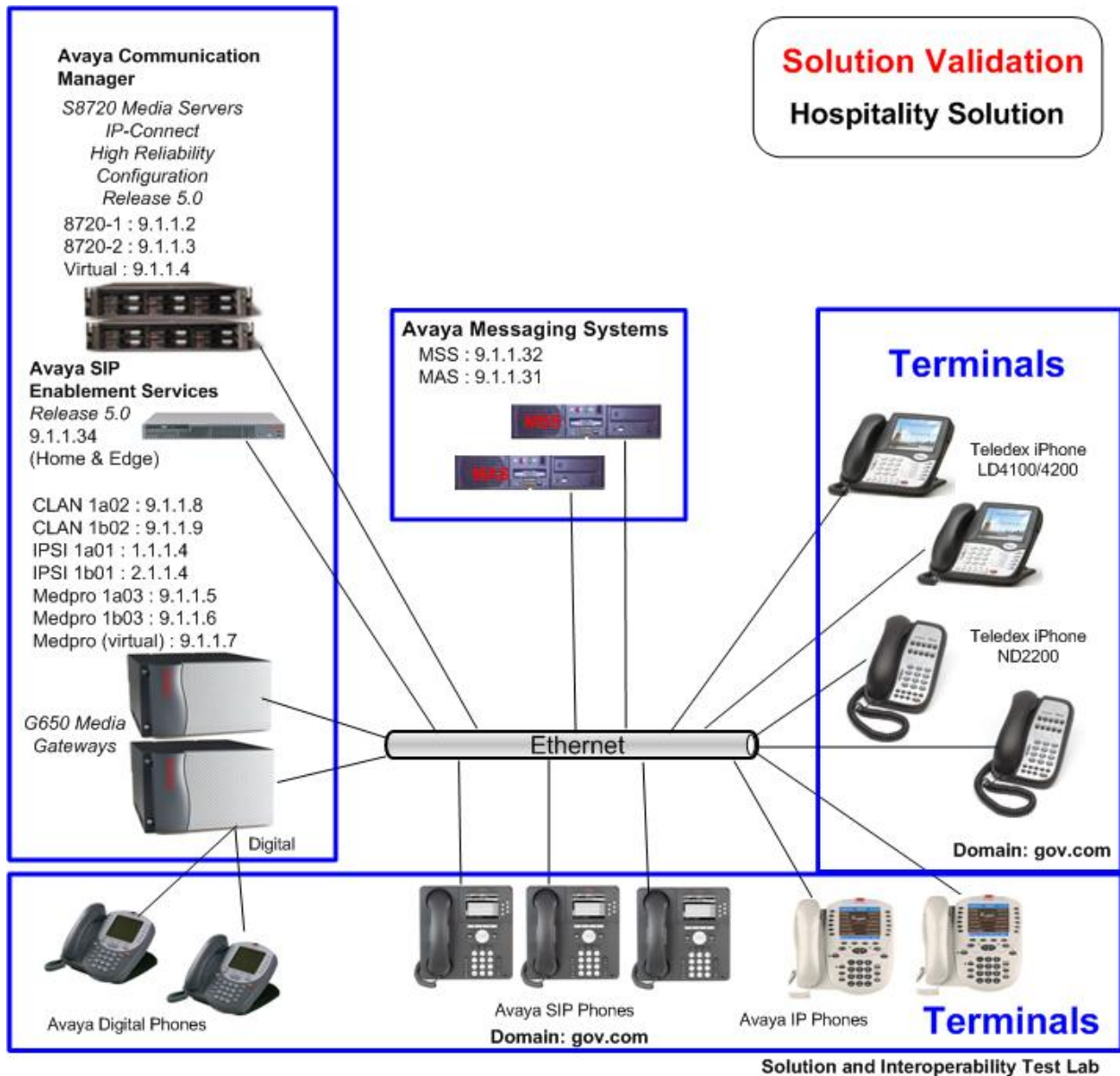


Figure 1 – Avaya Hospitality Solution Reference Configuration

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
  Codec Set: 1
  UDP Port Min: 2048
  UDP Port Max: 65535
  Intra-region IP-IP Direct Audio: yes
  Inter-region IP-IP Direct Audio: yes
  IP Audio Hairpinning? y
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 48
  Audio PHB Value: 48
  Video PHB Value: 34
  RTCP Reporting Enabled? y
  RTCP MONITOR SERVER PARAMETERS
  Use Default Server Parameters? y
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 6
  Audio 802.1p Priority: 6
  Video 802.1p Priority: 4
  AUDIO RESOURCE RESERVATION PARAMETERS
  RSVP Enabled? n
H.323 IP ENDPOINTS
  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 Domain: gov.com                Far-end Network Region: 1

                                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
                                Next Path Number:
                                Hunt after Coverage? n
                                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
                                HUNT GROUP
                                Page 1 of 60
                                Group Number: 7
                                Group Name: MM
                                Group Extension: 44444
                                Group Type: ucd-mia
                                TN: 1
                                COR: 1
                                Security Code:
                                ISDN/SIP Caller Display:
                                ACD? n
                                Queue? n
                                Vector? n
                                Coverage Path:
                                Night Service Destination:
                                MM Early Answer? n
                                Local Agent Preference? n
```

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

  Loss Group: 2                                       Time of Day Lock Table:
  Data Module? n                                     Personalized Ringing Pattern: 1
  Speakerphone: 2-way                               Message Lamp Ext: 44025
  Display Language: english                         Mute Button Enabled? y

  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 Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config 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 Extension	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location
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 Extension	Appl	CC	Phone Number	Config Set	Trunk Select	Mapping Mode	Calls 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.

AVAYA Integrated Management
Standard Management Solutions

Help Log Off

 SES Administration	The Administration Web Interface allows you to administer this SES server.	Launch SES Administration Interface
Maintenance	The Maintenance Web Interface allows you to maintain, troubleshoot, and configure the media server.	Launch Maintenance Web Interface

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.

AVAYA

Help Exit

Top

- Setup
- Users
 - Address Map Priorities
- Adjunct Systems
- Certificate Management
- Conferences
- Emergency Contacts
- Export/Import to ProVision
- Hosts
 - IM logs
- Media Servers
- Media Server Extensions
- Server Configuration**
 - Admin Setup
 - IM Log Settings
 - License
 - SNMP Configuration
 - System Properties**
- SIP Phone Settings
- Survivable Call Processors
 - System Status
- Trace Logger
- Trusted Hosts

View System Properties

SES Version	SES-5.0.0.0-825.31
System Configuration	simplex
Host Type	SES combined home-edge
SIP Domain*	<input type="text" value="gov.com"/>

Note that the DNS domain is Gov.com

If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

SIP License Host*	<input type="text" value="localhost"/>
-------------------	--

DiffServ/TOS Parameters

Call Control PHB Value*	<input type="text" value="46"/>
-------------------------	---------------------------------

802.1 Parameters

Priority Value*	<input type="text" value="6"/>
Management System Access Login	<input type="text"/>
Management System Access Password	<input type="text"/>
DB Log Level	<input type="text" value="disabled"/>

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.

The screenshot displays the 'Add Media Server Interface' configuration page in the Avaya Integrated Management System. The page is divided into a left navigation pane and a main content area. The left pane contains a 'Top' menu with various system management options, including 'Media Servers'. The main content area is titled 'Add Media Server Interface' and contains several sections of configuration fields:

- Media Server Interface Name***: A text input field containing 'CLANA'.
- Host**: A dropdown menu showing '9.1.1.34'.
- SIP Trunk Link Type**: Radio buttons for 'TCP' and 'TLS', with 'TLS' selected.
- SIP Trunk IP Address***: A text input field containing '9.1.1.8'.
- Media Server**: A section with fields for 'Media Server Admin Address (see Help)', 'Media Server Admin Port' (5022), 'Media Server Admin Login', 'Media Server Admin Password', and 'Media Server Admin Password Confirm'.
- SMS Connection Type**: Radio buttons for 'SSH', 'Telnet', and 'Not Available', with 'SSH' selected.

Below the configuration fields, there is a note: 'Note: Changing connection type to SSH connection type to Telnet resets media :'. At the bottom of the page, there is a note: 'Fields marked * are required.' and a blue 'Add' button.

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.

Top

- ▣ Users
 - Address Map Priorities
- ▣ Adjunct Systems
- ▣ Certificate Management
- ▣ Conferences
 - Emergency Contacts
- ▣ Export/Import to ProVision
- ▣ Hosts
 - IM logs
- ▣ Media Servers
- ▣ Media Server Extensions
- ▣ Server Configuration
- ▣ SIP Phone Settings
- ▣ Survivable Call Processors
 - System Status
- ▣ Trace Logger
- ▣ Trusted Hosts

Add User

Primary Handle*	<input type="text" value="44025"/>
User ID	<input type="text" value="44025"/>
Password*	<input type="password" value="•••••"/>
Confirm Password*	<input type="password" value="•••••"/>
Host*	<input type="text" value="9.1.1.34"/> ▼
First Name*	<input type="text" value="Guest 25"/>
Last Name*	<input type="text" value="Teledex"/>
Address 1	<input type="text"/>
Address 2	<input type="text"/>
Office	<input type="text"/>
City	<input type="text"/>
State	<input type="text"/>
Country	<input type="text"/>
Zip	<input type="text"/>
Survivable Call Processor	<input type="text" value="none"/> ▼
Add Media Server Extension	<input checked="" type="checkbox"/>

Fields marked * are required.

Click on the **Continue** button.

Top

- ▣ Users
 - Address Map Priorities
- ▣ Adjunct Systems
- ▣ Certificate Management
- ▣ Conferences
 - Emergency Contacts
- ▣ Export/Import to ProVision

Continue

User ID 44025 added.

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.

AVAYA

Help Exit

Top

- Users
 - Address Map Priorities
- Adjunct Systems
- Certificate Management
- Conferences
 - Emergency Contacts
- Export/Import to ProVision
- Hosts
 - IM logs

Add Media Server Extension

Add Media Server extension for user 44025.

Extension

Media Server

Fields marked * are required.

Add

Click on the **Continue** button.

AVAYA

Help Exit

Top

- Users
 - Address Map Priorities
- Adjunct Systems
- Certificate Management
- Conferences
 - Emergency Contacts
- Export/Import to ProVision

Continue

Extension 44025 added for user 44025

Continue

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

AVAYA

Help Exit

Top

- Users
 - Address Map Priorities
- Adjunct Systems
- Certificate Management
- Conferences

List Media Server Extensions

Media Server extensions for user 44025.

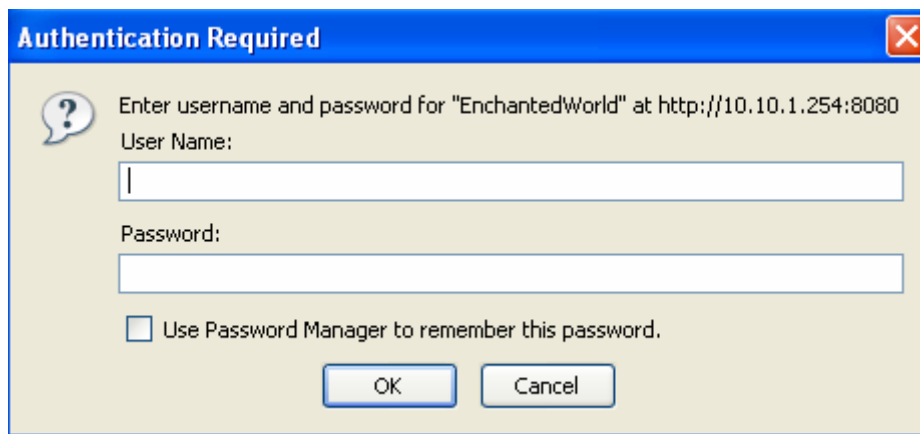
Commands	Extension	User	Media Server	Host
Free Edit User Delete	44025	44025	CLANA	9.1.1.34

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

6. Configure Teledex iPhone

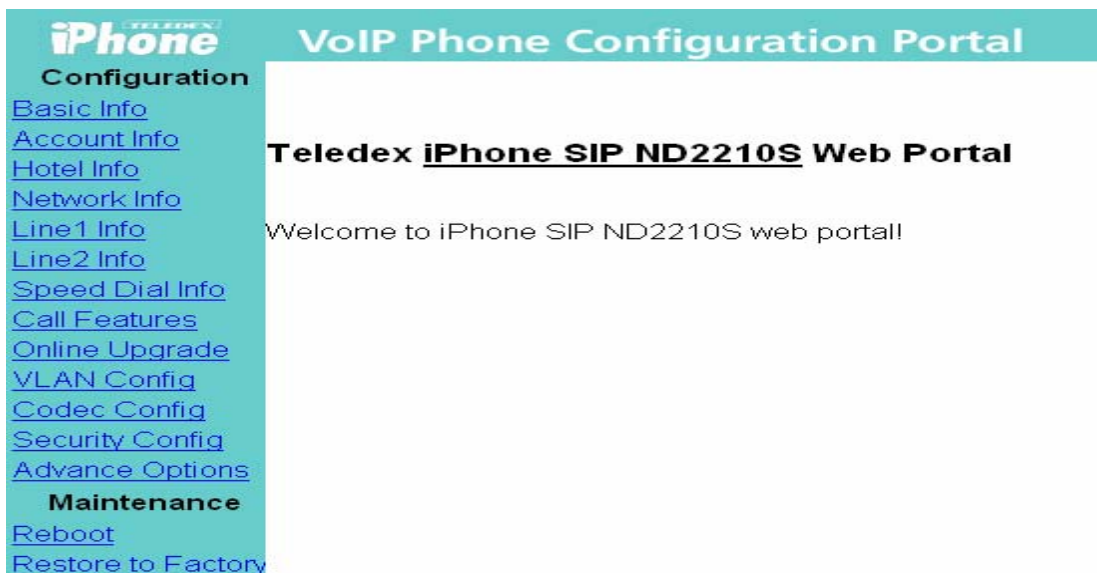
This section describes the steps to configure Teledex iPhone. SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. Teledex iPhone will register with Avaya SES using the SIP user accounts. The highlights in the following screens indicate 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 image shows a standard Windows-style dialog box titled "Authentication Required". It has a blue header bar with a red close button in the top right corner. The main area is light beige and contains a question mark icon in a speech bubble. The text reads: "Enter username and password for 'EnchantedWorld' at http://10.10.1.254:8080". Below this are two input fields: "User Name:" and "Password:". At the bottom, there is a checkbox labeled "Use Password Manager to remember this password." and two buttons: "OK" and "Cancel".

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



The image shows the "Teledex iPhone VoIP Phone Configuration Portal". The top header is teal with the "iPhone" logo and the text "VoIP Phone Configuration Portal". On the left, there is a teal sidebar with a "Configuration" section containing links for "Basic Info", "Account Info", "Hotel Info", "Network Info", "Line1 Info", "Line2 Info", "Speed Dial Info", "Call Features", "Online Upgrade", "VLAN Config", "Codec Config", "Security Config", and "Advance Options". Below this is a "Maintenance" section with links for "Reboot" and "Restore to Factory". The main content area is white and features the title "Teledex iPhone SIP ND2210S Web Portal" and the message "Welcome to iPhone SIP ND2210S web portal!"

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.

The screenshot shows the 'VoIP Phone Configuration Portal' for a Teledex iPhone. The left sidebar contains a menu with the following items: Configuration, Basic Info, Account Info, Hotel Info, Network Info (highlighted), Line1 Info, Line2 Info, Speed Dial Info, Call Features, Online Upgrade, VLAN Config, Codec Config, Security Config, Advance Options, Maintenance, Reboot, and Restore to Factory. The main content area is titled 'Network Configuration' and contains the following fields:

Acquire IP Through:	Use Static IP
Phone IP Address:	9.1.1.210
Phone Network Subnet:	255.255.255.0
Phone Gateway Address:	9.1.1.1
DNS Server Address:	9.1.1.10
Domain Name:	gov.com
SNTP Server Address:	68.193.149.145
Time Zone:	GMT-5:00 Eastern Time

Below the form is a 'Submit' button. Underneath, there is a 'Tips:' section with two numbered items:

1. SNTP server and Time Zone should be configured for time synchronization
2. DNS server should be set to 127.0.0.1 if using local mapping

At the bottom, there is a link to 'Return to Main Page'.

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.

iPhone VoIP Phone Configuration Portal

Configuration Line 1 Configuration

[Basic Info](#)
[Account Info](#)
[Hotel Info](#)
[Network Info](#)
[Line1 Info](#)
[Line2 Info](#)
[Speed Dial Info](#)
[Call Features](#)
[Online Upgrade](#)
[VLAN Config](#)
[Codec Config](#)
[Security Config](#)
[Advance Options](#)
Maintenance
[Reboot](#)
[Restore to Factory](#)

Phone Name	<input type="text" value="44028"/>
Phone Number	<input type="text" value="44028"/>
Phone Password	<input type="password" value="xxxxxxxx"/>
Proxy Server	<input type="text" value="9.1.1.34"/>
Proxy Server Port	<input type="text" value="5060"/>
Registrar Server	<input type="text" value="9.1.1.34"/>
Registrar Server Port	<input type="text" value="5060"/>
Message Waiting Number	<input type="text" value="44444"/>
MWI Server	<input type="text" value="9.1.1.31"/>
MWI Server Port	<input type="text" value="0"/>
Dialing Plan	<input type="text" value="4xxx 9xxxxxx 91xxxxxx"/>
SIP Transport	<input type="text" value="UDP and TCP"/>
PRACK	<input type="text" value="Supported"/>

Tips:

1. Dialing plan must be configured before phone is used
2. Message waiting number must be configured for retrieving message

Return to [Main Page](#) .

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.

iPhone (TELEDUX)
VoIP Phone Configuration Portal

Configuration Local Call Feature Configuration

[Basic Info](#)
[Account Info](#)
[Hotel Info](#)
[Network Info](#)
[Line1 Info](#)
[Line2 Info](#)
[Speed Dial Info](#)
[Call Features](#)
[Online Upgrade](#)
[VLAN Config](#)
[Codec Config](#)
[Security Config](#)
[Advance Options](#)

Maintenance
[Reboot](#)
[Restore to Factory](#)

Call Forward No:

Do Not Disturb
 Caller ID
 Call Transfer
 Call Forward No Answer
 Call Forward Busy
 Call Forward All
 Call Waiting

Tips:

1. All those are phone features
2. They could be supported on PBX side depending the PBX vendor

Return to [Main Page](#).

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.

The screenshot shows the 'Speed Dial Key Configuration' page in the Teledex iPhone VoIP Phone Configuration Portal. The page has a teal header with the 'iPhone' logo and the title 'VoIP Phone Configuration Portal'. A navigation menu on the left lists various configuration options, with 'Speed Dial Info' highlighted. The main content area contains a table with three columns: 'SPD Key', 'Key Name', and 'Key Number'. There are 10 rows for speed dial keys and one row for 'Inter-Digit Pause'. A 'Submit' button is located at the bottom left of the configuration area.

SPD Key	Key Name	Key Number
Speed Dial Key 1:	Operator	44010
Speed Dial Key 2:	Do Not Disturb	48822
Speed Dial Key 3:	Do Not Disturb Cancel	48823
Speed Dial Key 4:	Doorman	44010
Speed Dial Key 5:	Emergency	44010
Speed Dial Key 6:	Front Desk	44013
Speed Dial Key 7:	Wakeup	44013
Speed Dial Key 8:	Spa	44013
Speed Dial Key 9:	Gift Shop	44013
Speed Dial Key 10:	Weather	44013
Inter-Digit Pause:	5	

Tips:

1. Key name is used for display if the phone has a screen
2. Inter-digit pause is activated only when dialing plan is disabled

Return to [Main Page](#).

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.

The screenshot shows the 'iPhone VoIP Phone Configuration Portal' interface. On the left is a navigation menu with categories: Configuration, Maintenance, and Reboot. Under Configuration, 'Advance Options' is selected. The main content area is titled 'Inter-Op Configuration' and contains the following settings:

- Multiple Line Appearance: Enable MLA (dropdown menu set to 'Enable')
- Static Domain Mapping: Enable (dropdown menu set to 'Disabled')
- Domain Name: (text input field)
- IP Address: 10.10.1.4
- Advance Options: Advance Option Mask (text input field set to '2')
- PBX Option: PBX Vendor Mask (text input field set to '0')
- SIP Message Callback: Debug Mask (text input field set to '0')

A 'Submit' button is located at the bottom of the configuration form. Below the form is a 'Tips:' section with three numbered items:

- 1. Multiple Line Appearance**
Enable this when PBX support this feature on 2-line model
- 2. Static Domain Mapping**
Enable this when domain name is faked
- 3. Advance Option Mask**
 - Bit 0: Switch between dialing plan (0) and inter-digit dialing (1)
 - Bit 1: Switch between deactivating (0) and activating (1) NMS
 - Bit 2: Switch between deactivating (0) and activating (1) pre-dial
 - Bit 3: Switch between activating (0) and deactivating (1) volume up/down key
 - Bit 4: Switch between normal(0) and hotline (1) mode
 - Bit 5: Switch between normal(0) and daylight saving (1) mode

At the bottom, there is a link: Return to [Main Page](#).

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.

The screenshot shows the 'VoIP Phone Configuration Portal' for a Teledex iPhone. On the left is a navigation menu with categories: 'Configuration' (containing links for Basic Info, Account Info, Hotel Info, Network Info, Line1 Info, Line2 Info, Speed Dial Info, Call Features, Online Upgrade, VLAN Config, Codec Config, Security Config, and Advance Options) and 'Maintenance' (containing links for Reboot and Restore to Factory). The main content area is titled 'Reboot The Phone' and includes the instruction 'Press the *Reboot* button to reboot the phone.' with a 'Reboot' button highlighted by a red box. Below this is a 'NOTICE' section with the text 'Once you change settings, Reboot phone from here!!' in blue. A 'Tips:' section follows with the point '1. Reboot will not change anything on the phone'. At the bottom, there is a link to 'Return to [Main Page](#).'

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.

Top

- ▣ Users
- Address Map Priorities
- ▣ Adjunct Systems
- ▣ Certificate Management
- ▣ Conferences
- Emergency Contacts
- ▣ Export/Import to ProVision
- ▣ Hosts
- IM logs
- ▣ Media Servers
- ▣ Media Server Extensions
- ▣ Server Configuration
- ▣ SIP Phone Settings
- ▣ Survivable Call Processors
- System Status
- ▣ Trace Logger
- ▣ Trusted Hosts

Registered Users on 9.1.1.34

[Registered and Provisioned Users](#) | [Registered Users](#) | [Provisioned Users](#) | [Search](#) | [Refresh](#)

Showing 1 to 8 of 8 registered contacts.

	Handle and Name	Address	Expires
<input type="checkbox"/>	44011@gov.com Guest 11, Avaya	sip:44011@9.1.1.148:5061;transport=tls	Wed, 01 Oct 2008 11:39:03 EDT
		sip:44011@9.1.1.161:5061;transport=tls	Tue, 30 Sep 2008 15:31:01 EDT
<input type="checkbox"/>	44030@gov.com Teledex, LD2 Line 1	sip:44030@9.1.1.212	Tue, 30 Sep 2008 13:18:46 EDT
<input type="checkbox"/>	44027@gov.com Teledex, ND2 Line 1	sip:44027@9.1.1.211	Tue, 30 Sep 2008 13:27:47 EDT
<input type="checkbox"/>	44010@gov.com Guest 10, Avaya	sip:44010@9.1.1.148:5061;transport=tls	Tue, 30 Sep 2008 15:30:29 EDT
		sip:44010@9.1.1.161:5061;transport=tls	Wed, 01 Oct 2008 11:56:16 EDT
<input type="checkbox"/>	44025@gov.com Teledex ND Phone, Guest 25	sip:44025@9.1.1.210	Tue, 30 Sep 2008 13:41:45 EDT
<input type="checkbox"/>	44012@gov.com Guest 12, Avaya	sip:44012@9.1.1.162:5061;transport=tls	Wed, 01 Oct 2008 10:52:09 EDT
<input type="checkbox"/>	44028@gov.com Teledex, LD1 Line 1	sip:44028@9.1.1.213	Tue, 30 Sep 2008 12:53:09 EDT

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.

©2008 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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.