



**Avaya Solution & Interoperability Test Lab**

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## **Application Notes for Teledex iPhones and Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Teledex iPhones 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.

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

# 1. Introduction

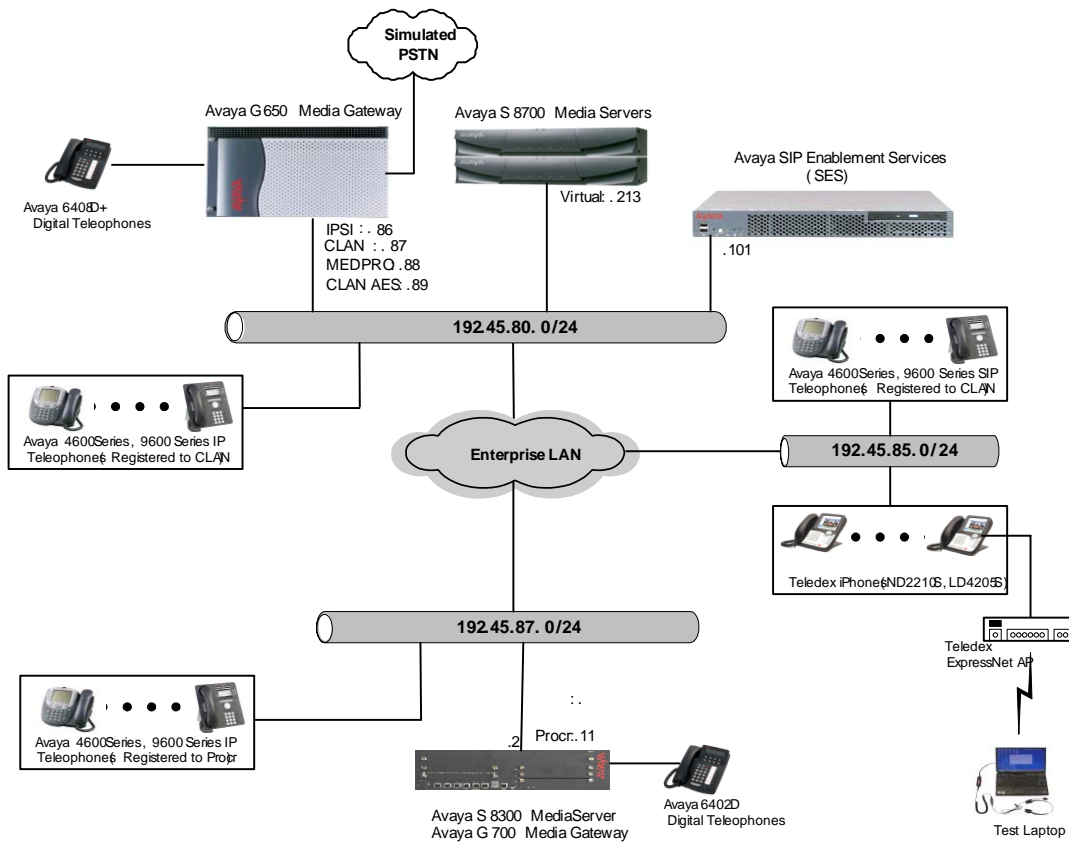
These Application Notes describe the procedures for configuring Teledex iPhones which were compliance tested with Avaya Communication Manager and Avaya SIP Enablement Services. The overall objective of the interoperability compliance testing is to verify Teledex iPhones functions in an environment that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

Teledex iPhone is a SIP endpoint, and designed for the specific needs of the 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 iPhones, ND2210S and LD4205S, were evaluated. The ND2210S and LD4205S SIP phones utilize the same firmware and provide the same functionality. However, the LD4205S SIP phone consists of a 5.6-inch color touch screen display, while the ND2210S SIP phone does not have a display. Each type of Teledex iPhones comes with an embedded access point (AP), providing the data network service. The following features were tested and verified.

- Inbound calls
- Outbound calls
- Call transfers
- Call forwards
- Call conferences
- Speed dialing keys
- Do not disturb
- Message waiting indicator (MWI)
- VLAN tagging

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. For further details on configuration steps not covered in this document, consult [3].

**Figure 1** illustrates a sample configuration consisting of an Avaya S8300 Server, an Avaya G700 Media Gateway, an Avaya SIP Enablement Services (SES) server, and Teledex iPhones. The solution described herein is also extensible to other Avaya Media Servers and Media Gateways. Avaya S8700 Servers with an Avaya G650 Media Gateway were included in the test to provide an inter-switch scenario. For completeness, Avaya 4600 Series SIP IP Telephones, Avaya 4600 Series H.323 IP Telephones, Avaya 9600 Series SIP IP Telephones, Avaya 9600 Series H.323 IP Telephones, and Avaya 6400 Series Digital Telephones, are included in the sample configuration to demonstrate interoperability with the Teledex iPhones. The analog PSTN telephone is also included to demonstrate calls routed by Avaya Communication Manager between Teledex iPhone and the PSTN.



**Figure 1: Avaya DeveloperConnection Compliance Test Configuration**

## 2. Equipment and Software Validated

The following equipment and software were used for the test configuration.

Equipment		Software/Firmware
Avaya S8700 Server		<ul style="list-style-type: none"> <li>Avaya Communication Manager 4.0 (R014x.00.0.730.5 with patch 13566)</li> <li>Avaya Communication Manager 4.0.1 (R014x.00.1.731.2)</li> </ul>
Avaya G650 Media Gateway		-
	TN2312BP IP Server Interface	HW11 FW030
	TN799DP C-LAN Interface	HW01 FW017
	TN2302AP IP Media Processor	HW20 FW108
	TN2602AP IP Media Processor	HW02 FW007
Avaya S8300 Server with Avaya G700 Media Gateway		<ul style="list-style-type: none"> <li>Avaya Communication Manager 4.0 (R014x.00.0.730.5 with patch 13566)</li> <li>Avaya Communication Manager 4.0.1 (R014x.00.1.731.2)</li> </ul>
Avaya SIP Enablement Services		<ul style="list-style-type: none"> <li>SES04.0-04.0.032.0</li> <li>SES04.0-04.0.033.6</li> </ul>
Avaya 4600 and 9600 Series SIP Telephones		
	4610 (SIP)	2.2.2
	4620 (SIP)	2.2.2
	9630 (SIP)	1.0.13.1
Avaya 4600 and 9600 Series IP Telephones		
	4620 (H.323)	2.7
	4625 (H.323)	2.7
	9640 (H.323)	1.2.1
Avaya 6408D+ Digital Telephone		-
Teledex iPhone LD4205S		1.1
Teledex iPhone ND2210S		1.1

## 3. Configure the 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 a list of IP codec set, an IP network region, IP node name, a signaling group, a trunk group, a SIP station, a coverage path, and a hunt group. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

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

### 3.1. 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.

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

G3 Version: V14
Location: 1                               RFA System ID (SID): 1
Platform: 7                               RFA Module ID (MID): 1

                                USED
Platform Maximum Ports: 900          95
Maximum Stations: 450                17
Maximum XMOBILE Stations: 0          0
Maximum Off-PBX Telephones - EC500: 50 0
Maximum Off-PBX Telephones - OPS: 100 9
Maximum Off-PBX Telephones - PBFMC: 0 0
Maximum Off-PBX Telephones - PVFMC: 0 0
Maximum Off-PBX Telephones - SCCAN: 0 0
```

On Page 2 of the OPTIONAL FEATURES form, 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          18
Maximum Concurrently Registered IP Stations: 50 3
Maximum Administered Remote Office Trunks: 0    0
Maximum Concurrently Registered Remote Office Stations: 0 0
Maximum Concurrently Registered IP eCons: 0     0
Max Concur Registered Unauthenticated H.323 Stations: 0 0
Maximum Video Capable H.323 Stations: 5        0
Maximum Video Capable IP Softphones: 5         0
Maximum Administered SIP Trunks: 100          50

Maximum Number of DS1 Boards with Echo Cancellation: 0 0
Maximum TN2501 VAL Boards: 0                  0
Maximum Media Gateway VAL Sources: 0          0
Maximum TN2602 Boards with 80 VoIP Channels: 0 0
Maximum TN2602 Boards with 320 VoIP Channels: 0 0
Maximum Number of Expanded Meet-me Conference Ports: 0 0
```

### 3.2. IP Codec Set

This section describes the steps for administering a codec set in Avaya Communication Manager. This 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 3.3 for configuring IP network region to specify which codec sets may be used within and between network regions. For the compliance testing, G.711MU and G.729B were used.

```
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: G.729B      n           2       20
3:
4:
5:
6:
7:

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

### 3.3. 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 – Set to the **testroom.com**. This should match the SIP Domain value on Avaya SES, in Section 4.1.
- Intra-region IP-IP Direct Audio – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Communication Manager or Avaya SES in the same IP network region.
- Codec Set – Set the codec set number as provisioned in Section 3.2.
- Inter-region IP-IP Direct Audio – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Communication Manager or Avaya SES in different IP network regions.

```
change ip-network-region 1                               Page 1 of 19
                                                         IP NETWORK REGION
Region: 1
Location: Authoritative Domain: testroom.com
Name:
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? n
UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                               RTCP Reporting Enabled? y
Call Control PHB Value: 46                             RTCP MONITOR SERVER PARAMETERS
Audio PHB Value: 46                                   Use Default Server Parameters? y
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5                             AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
```

### 3.4. Configure IP Node Name

This section describes the steps for setting the 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.

```
change node-names ip                                     Page 1 of 2
                                                    IP NODE NAMES
  Name          IP Address
  CLAN          192.45.80.87
  IA770         192.45.87.12
  S8300CDR     192.45.88.11
  S8300G250    192.45.82.11
  S8300G350    192.45.81.11
  SIPServer    192.45.80.101
  chung        10.1.1.1
  default      0.0.0.0
  procr        192.45.87.11
```

### 3.5. Configure SIP Signaling

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SIP Enablement Services. 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 **procr** as displayed in Section 3.4.
- Far-end Node Name - Set to the Avaya SES name configured in Section 3.4.
- Far-end Network Region - Set to the region number configured in Section 3.3.
- Far-end Domain - Set to **testroom.com**. This should match the SIP Domain value in Section 4.1.

```
add signaling-group 1                                     Page 1 of 1
                                                    SIGNALING GROUP
  Group Number: 1          Group Type: sip
                          Transport Method: tls

  Near-end Node Name: procr          Far-end Node Name: SIPServer
  Near-end Listen Port: 5061        Far-end Listen Port: 5061
  Far-end Domain: testroom.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? n
  Enable Layer 3 Test? n
  Session Establishment Timer(min): 3
```



### 3.6. Configure SIP Trunk

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 the Group Type field value configured in Section 3.5.
- Group Name – Enter a descriptive name.
- TAC (Trunk Access Code) – Set to any available trunk access code.
- Signaling Group – Set to the Group Number field value configured in Section 3.5.
- 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.

**Note:** Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.

```
add trunk-group 1                                     Page 1 of 21
                                                    TRUNK GROUP
Group Number: 1                                     Group Type: sip          CDR Reports: y
Group Name: to SES                                COR: 1                  TN: 1              TAC: 115
Direction: two-way                               Outgoing Display? n
Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                                Auth Code? n
                                                    Signaling Group: 1
                                                    Number of Members: 50
```

### 3.7. Configure SIP Endpoint

This section describes the steps for administering Off PBX Stations (OPS) stations in Avaya Communication Manager and associating the OPS station extensions with the telephone numbers of Teledex iPhones. Enter **add station s**, where **s** is an extension valid in the provisioned dial plan. The following fields were configured for the compliance test.

- Type – Set to **6408D+**.
- Port – Set to **X**.
- Coverage Path 1 – Set to **99**. This feature is needed to test the MWI feature.
- Name – Enter a descriptive name.

Repeat this step as necessary to configure additional SIP endpoint extensions.

```
add station 27001                                     Page 1 of 5
                                                    STATION
Extension: 27001                                     Lock Messages? n          BCC: 0
Type: 6408D+                                         Security Code:           TN: 1
Port: X                                               Coverage Path 1: 99      COR: 1
Name: SIP 27001                                       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: 27001
Display Language: english                            Mute Button Enabled? y
Survivable COR: internal                               Media Complex Ext:
Survivable Trunk Dest? y                             IP SoftPhone? n
```

On Page 2, set the MWI Served User Type field to **qsig-mwi**.

```
add station 27001                                     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? y
Active Station Ringing: single
H.320 Conversion? n                                  Per Station CPN - Send Calling Number?
Service Link Mode: as-needed
Multimedia Mode: basic
MWI Served User Type: qsig-mwi                       Display Client Redirection? n
                                                    Select Last Used Appearance? n
                                                    Coverage After Forwarding? s
Emergency Location Ext: 27001                         Direct IP-IP Audio Connections? y
                                                    IP Audio Hairpinning? n
```

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

- Station Extension – Set the extension of the OPS station as 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.
- Trunk Selection – Set to the trunk group number configured in Section 3.6.
- Config Set – Set to 1

```
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
27001	OPS	-		27001	1	1

### 3.8. Configure the Coverage Path

Enter **add coverage path c**, where **c** is an unused coverage path number. In the compliance test, the Point1 field under the COVERAGE POINTS section is set to **h99**.

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

Coverage Path Number: 99

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: h99                      Rng:                      Point2:  
Point3:    Point4:  
Point5:    Point6:

### 3.9. 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 99                                     Page 1 of 60
                                                    HUNT GROUP
Group Number: 99                                     ACD? n
Group Name: Audix                                    Queue? n
Group Extension: 70000                               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 **70000**
- Routing Digits (e.g. AAR/ARS Access Code) - **8**

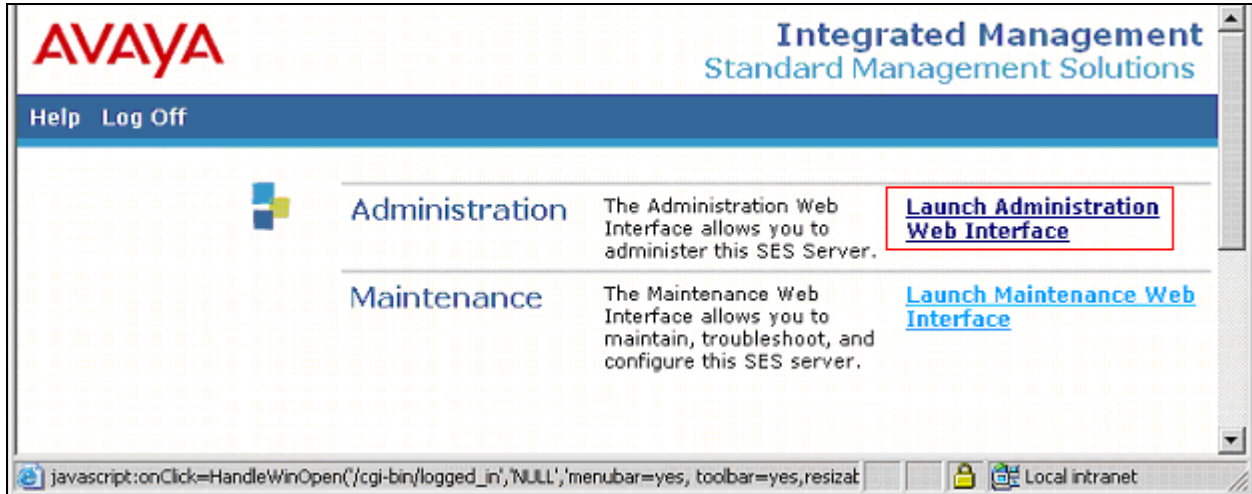
```
add hunt-group 99                                     Page 2 of 60
                                                    HUNT GROUP
LWC Reception: none                                  AUDIX Name:
Message Center: qsig-mwi
Send Reroute Request: y
Voice Mail Number: 70000
Routing Digits (e.g. AAR/ARS Access Code): 8         Provide Ringback? n
TSC per MWI Interrogation? n
```

### 4. Configure the 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.

## 4.1. 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 Administration Web Interface** link upon successful login.



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 on Avaya Communication Manager in Section 3.5. Click on the **Update** button, after the completion.

**AVAYA** Integrated Management SIP Server Management  
Server: 192.45.80.101

Help Exit

**View System Properties**

SES\_Version SES-4.0.0.0-032.0  
System Configuration simplex  
Host Type home/edge

SIP Domain\*

Note that the DNS domain is: testroom.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

License Host\*

Management System Access Login

Management System Access Password

**DiffServ/TOS Parameters**

Call Control PHB Value\*

**802.1 Parameters**

Priority Value\*

**Network Properties**

Local IP 192.45.80.101  
Local Name SIPServer-1.testroom.com  
Logical IP 192.45.80.101  
Logical Name SIPServer-1.testroom.com  
Gateway IP Address 192.45.80.1

**Redundant Properties**

Management Device SAMP

Fields marked \* are required.

## 4.2. 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 alphabetized drop-down list of names, select the name of the SES server that you want to associate with the Avaya Communication Manager Media server's SIP trunking.
- SIP Trunk Link Type – Select one of the listed protocols to be used for the SIP link between the media server and the specified host:
  - TCP (Transport Control Protocol)
  - TLS (Transport Link Security) -- this is the default protocol, which is selected for all servers.
- SIP Trunk IP Address – Enter the IP address for the media server's procr (or CLAN) IP interface that terminates the SIP link from SES.

Click **Add** when finished.

The screenshot shows the 'Add Media Server Interface' configuration page in the Avaya Integrated Management SIP Server Management interface. The page title is 'Add Media Server Interface'. The form contains the following fields and options:

- Media Server Interface Name\***: Text input field containing 'G700'.
- Host**: Dropdown menu showing '192.45.80.101'.
- SIP Trunk Link Type**: Radio buttons for 'TCP' and 'TLS', with 'TLS' selected.
- SIP Trunk IP Address\***: Text input field containing '192.45.87.11'.
- Media Server** section:
  - Media Server Admin Address (see Help): Text input field.
  - Media Server Admin Login: Text input field.
  - Media Server Admin Password: Text input field.
  - Media Server Admin Password Confirm: Text input field.
- SMS Connection Type**: Radio buttons for 'SSH' and 'Telnet'.
- A note below the fields states: 'Fields marked \* are required.'
- A red box highlights the **Add** button at the bottom left of the form area.

### 4.3. 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 Teledex iPhones. This number was configured in Section 3.7.

- 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.
- Host – From the drop-down list of names, select the host serving the domain for this user. The hostname 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.

**AVAYA** Integrated Management  
SIP Server Management  
Server: 192.45.80.101

Help Exit Update

**Add User**

Primary Handle\* 27001

User ID 27001

Password\* \*\*\*\*\*

Confirm Password\* \*\*\*\*\*

Host\* 192.45.80.101

First Name\* SIP

Last Name\* 27001

Address 1

Address 2

Office

City

State

Country

Zip

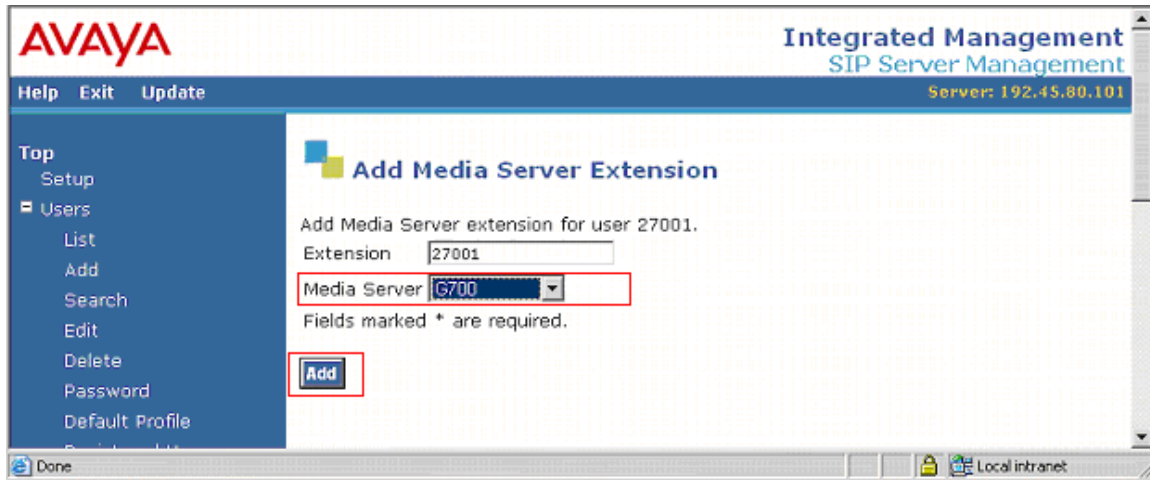
Add Media Server Extension

Fields marked \* are required.

**Add**



From the next screen, enter the numeric telephone extension you want to create in the database. Select the extension's media server from the drop-down list. Click on the **Add** button.



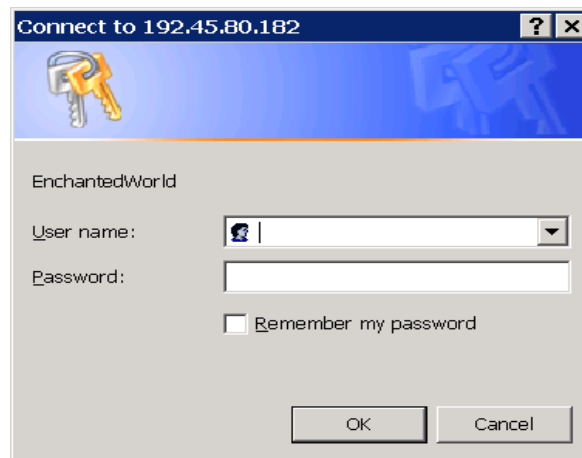
## 5. Configure Teledex iPhone

This section only focuses on the interface between the Avaya SES server and the iPhone. In the iPhone Configuration Web Client page, the following fields were configured:

- Network Info
- Line 1 Info
- Line 2 Info

### 5.1. Configure Network Info

Launch a web browser, enter <http://<IP address of iPhone>:8080> in the URL, and log in with the appropriate credentials for accessing the iPhone Configuration Web Client page.



In the iPhone Configuration Web Client page, select the **Network Info** link from the left pane to configure the network configuration.

The following highlighted fields were configured for the compliance test:

- Acquire IP Through – Select **Use Static IP**, using the drop-down list.
- 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.
- Domain Name – Enter the SIP domain configured in Section 4.1.

Click on the **Submit** button to save the network configuration.

The screenshot shows the 'TELEDEX iPhone™ Configuration Web Client' interface. The 'Network Configuration' section is active, with a sidebar on the left containing navigation links. The configuration fields are as follows:

Field	Value
Acquire IP Through :	Use Static IP
Phone IP Address:	192.45.80.182
Phone Network Subnet:	255.255.255.0
Phone Gateway Address:	192.45.80.1
DNS Server Address:	0.0.0.0
Domain Name:	testroom.com
SNTP Server Address	193.125.143.173
Time Zone	GMT-5:00 Eastern Time

At the bottom of the configuration area, there is a 'Submit' button and a link to 'Return to Main Page'.

## 5.2. Configure Line 1 Info

In the iPhone Configuration Web Client page, select the **Line 1 Info** link from the left pane to configure the line 1.

The following highlighted fields were configured for the compliance test.

- Phone Name – Enter a descriptive name.
- Phone Number – Enter the iPhone extension number configured in Section 4.3.
- Phone Password – Enter the iPhone extension password configured in Section 4.3.
- Proxy Server – Enter the SIP server IP address.
- Registrar Server – Enter the SIP server IP address.
- Message Waiting Number – Enter Voice Mail Number configured in Section 3.9.
- Dialing Plan – Enter dialing plan to define dialing pattern, and check whether it is consistent with that of the PBX.

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

Configuration	Line 1 Configuration
<a href="#">Phone Info</a>	Phone Name <input type="text" value="27002"/>
<a href="#">Hotel Info</a>	Phone Number <input type="text" value="27002"/>
<a href="#">Network Info</a>	Phone Password <input type="text" value="123456"/>
<a href="#">Line 1 Info</a>	Proxy Server <input type="text" value="192.45.80.101"/>
<a href="#">Line 2 Info</a>	Proxy Server Port <input type="text" value="0"/>
<a href="#">Speed Dial Keys</a>	Registrar Server <input type="text" value="192.45.80.101"/>
<a href="#">Call Features</a>	Registrar Server Port <input type="text" value="0"/>
<a href="#">Online Upgrade</a>	Message Waiting Number <input type="text" value="70000"/>
<a href="#">Security Option</a>	MWI Server <input type="text" value="0.0.0.0"/>
<a href="#">VLAN Config</a>	MWI Server Port <input type="text" value="0"/>
<a href="#">InterOp Config</a>	Dialing Plan <input type="text" value="[0-8]xxxx 91 xxxxxxxxxxxx"/>
<b>Maintenance</b>	<input type="button" value="Submit"/>
<a href="#">Reset</a>	
<a href="#">Restore to Factory</a>	

## 5.3. Configure Line 2 Info

In the iPhone Configuration Web Client page, select the **Line 2 Info** link from the left pane to configure the line 2. Line 2 utilizes a different extension than Line 1. Except for the phone number, configuration steps are the same as Line 1.

## 5.4. Configure Speed Dial Key

In the iPhone Configuration Web Client page, select the **Speed Dial Keys** link from the left pane to configure the speed dial feature. Provide an extension for each Speed Dial Key field.

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

Speed Dial Key	Extension
Speed Dial Key 1:	27005
Speed Dial Key 2:	27005
Speed Dial Key 3:	27005
Speed Dial Key 4:	27005
Speed Dial Key 5:	27005
Speed Dial Key 6:	27006
Speed Dial Key 7:	27006
Speed Dial Key 8:	27006
Speed Dial Key 9:	27006
Speed Dial Key 10:	70000

Timing Pause: 5

## 6. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability. The focus of the interoperability compliance testing was primarily on verifying call establishment on Teledex iPhones. Teledex iPhones operations such as inbound calls (Line1 and Line2), outbound calls (Line1 and Line2), mute, hold, transfer, forward, conference, speed dial keys, do not disturb, bridging, message waiting indicator (MWI), and VLAN tag. Teledex iPhone interactions with Avaya SIP Enablement Services (SES), Avaya Communication Manager, and Avaya SIP, H.323, and digital telephones were verified. The serviceability testing introduced failure scenarios to see if Teledex iPhones can recover from failures.

### 6.1. General Test Approach

The general test approach was to place calls to and from Teledex iPhones and exercise basic telephone operations. The main objectives were to verify that:

- Teledex iPhone successfully register with Avaya SES.

- Successfully establish calls between Teledex iPhone and Avaya SIP, H.323, and digital telephones controlled by Avaya SES or Avaya Communication Manager.
- Successfully establish calls between the Teledex iPhone and PSTN telephones through Avaya Communication Manager.
- Teledex iPhone successfully handle concurrent calls on its two lines.
- Teledex iPhone successfully negotiate the right codec.
- Teledex iPhone successfully hold a call, transfers a call, and establishes a three party conference call.

For serviceability testing, failures such as cable pulls and hardware resets were applied.

## 6.2. Test Results

All test cases passed successfully.

The test objectives of Section 6.1 were verified. For serviceability testing, the Teledex iPhones operated properly after recovering from failures such as cable disconnects, and resets of the Teledex iPhones, the Avaya SES server, and Avaya Communication Manager. Teledex iPhones successfully negotiated the codec to be used.

The following observations were made during testing:

- Teledex iPhones fail to shuffle either between Teledex iPhones or with Avaya SIP endpoints. When Avaya SIP endpoint initiates a call to Teledex iPhone, shuffling is done properly. However, when Teledex iPhone initiates a call to an Avaya SIP endpoint or to another Teledex iPhone, the Teledex endpoint does not shuffle.

Teledex will address and resolve the above observation with future firmware releases. Contact Teledex ([www.teledex.com](http://www.teledex.com)) for further updates.

## 7. Verification Steps

The following steps may be used to verify the configuration:

The following steps may be used to verify the configuration:

- Verify that Teledex iPhones successfully register with the Avaya SES server by following the **Users -> Registered Users** link on the SES Administration Web Interface.
- Place calls to and from Teledex iPhones and verify that the calls are successfully established with two-way talk path.
- While calls are established, Enter **status trunk <t>** command, where **t** is the SIP trunk configured in Section 3.6.

## 8. Support

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

## 9. Conclusion

Teledex iPhones were compliance tested with Avaya Communication Manager (Version 4.0) and Avaya SES (Version 4.0). Teledex iPhones (Version 1.1) functioned properly for feature and serviceability. Teledex iPhones successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features like three-way conference, transfers, hold, etc.

During the compliance test, Avaya Communication Manager 4.0 (R014x.00.0.730.5 with patch 13566) and the Avaya SES 4.0 (SES04.0-04.0.032.0) were utilized to finish the test. After the compliance test was completed, the test team retested and verified, with selected test cases, utilizing Avaya Communication Manager 4.0.1 (R014x.00.1.731.2) and the Avaya SES (SES04.0-04.0.033.6) since those became the GA version.

## 10. Additional References

The following Avaya product documentation can be found at <http://support.avaya.com>

[1] *Feature Description and Implementation for Avaya Communication Manager*, Release 4.0, Issue 5, February 2007, Document Number 555-245-205.

[2] *Application Enablement Services Administration and Maintenance Guide*, Release 4.0, Issue 6, February 2007, Document Number 02-300357.

The following document was provided by Teledex.

[3] iPhone ND2210S VoIP Network Terminal User's Guide

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