



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Yealink T-22 SIP Phones with Avaya Aura[®] Communication Manager and Avaya Aura[®] Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for the Yealink T-22 SIP phone to interoperate with Avaya Aura[®] Communication Manager and Avaya Aura[®] Session Manager.

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

1. Introduction

These Application Notes describe the steps required to connect Yealink T-22 Handset to a SIP infrastructure consisting of Avaya Aura[®] Session Manager and Avaya Aura[®] Communication Manager. The Yealink T-22 Handset is a display handset with 3 line appearances. Also described is how Avaya Aura[®] Communication Manager features can be made available in addition to the standard features supported in the T-22 handset. In this configuration, the Off-PBX Stations (OPS) feature set is extended from Avaya Aura[®] Communication Manager to the T-22 Handset, providing the T-22 Handset with enhanced calling features.

2. General Test Approach and Test Results

To verify interoperability of Yealink T-22 handset with Communication Manager and Session Manager, calls were made between T-22 handset and Avaya SIP, H.323 and Digital stations using various codec settings and exercising common PBX features. The telephony features were activated and deactivated using pre-programmed buttons. Yealink T-22 handset passed all compliance testing with all scenarios resulting in the expected outcome.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of T-22 handset with Session Manager
- Calls between T-22 handset and Avaya SIP, H.323, and digital stations
- G.711 and G729 codec support
- PBX features including Multiple Call Appearances, Hold, Transfer, and Conference
- Proper system recovery after an T-22 handset restart and loss of IP connection
- Correct T-22 handset behavior during Session Manager and Communication Manager failovers and simulated network failures

2.2. Test Results

During testing the Yealink T-22 handset completed all scenarios with results in all cases as expected.

2.3. Support

Technical support from Yealink can be obtained through the following:

Phone: +44 (0)161 763 2060

E-mail: sales@yealink.co.uk.

Web: <http://www.yealink.co.uk>

3. Reference Configuration

The diagram illustrates an enterprise site with an Avaya SIP-based network, including a Session Manager, S8800 Media Server running Communication Manager with a G650 Media Gateway, and Avaya IP endpoints. The enterprise site also contains one T-22 handset and one T-28 handset used to verify call functionality between Yealink handsets. The SIP handsets are registered with Session Manager and are configured as endpoint users. Communication Manager extends the telephony functionality that is supported by the SIP-based T-22 device through the use of Feature Name Extensions (FNEs).

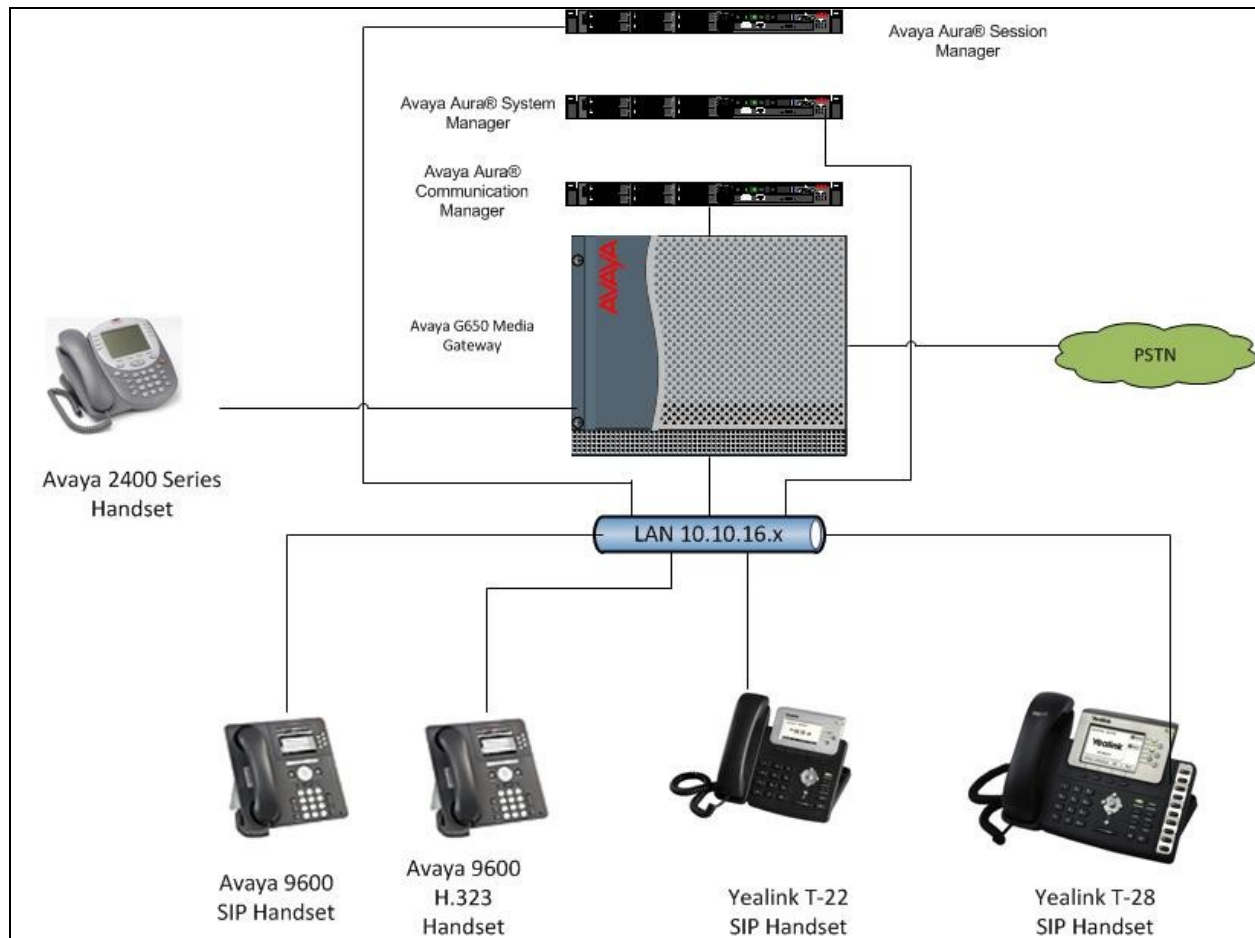


Figure 1: T-22 with Avaya SIP Solution

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 Media Server with G650 Media Gateway	Avaya Aura® Communication Manager 6.0.1 (R16x.0.0.345.0-18444)
Avaya S8800 Media Server	Avaya Aura® Session Manager 6.1 (Build 6.1.0.0.610023)
Avaya S8800 Media Server	Avaya Aura® System Manager 6.1 (Build 6.1.0.4.5072-6.1.4.113)
Avaya 9600 Series Handsets	2.6.4.0 (SIP)
Avaya 9600 Series Handsets	3.1 (H.323)
Avaya 2420 Handsets	NA
Yealink T-22 Handset	7.60.23.5
Yealink T-28 Handset	2.60.23.5

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring the T-22 handset as an Off-PBX Station (OPS) and configuring a SIP trunk between Communication Manager and Session Manager. Use the System Access Terminal (SAT) to configure Communication Manager. Log in with the appropriate credentials. The configuration steps described are also applicable to other Linux-based Avaya Servers and Media Gateways running Avaya Aura® Communication Manager.

5.1. Verify System Capacity

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 1**, verify that the **Maximum Off-PBX Telephones** allowed in the system is sufficient. One OPS station is required per T-22 handset.

di splay system-parameters customer-options		Page	1 of 10
OPTIONAL FEATURES			
G3 Version: V15	Software Package: Standard		
Location: 2	RFA System ID (SID): 1		
Platform: 6	RFA Module ID (MID): 1		
		USED	
Platform Maximum Ports:		48000	282
Maximum Stations:		36000	48
Maximum XMOBILE Stations:		0	0
Maximum Off-PBX Telephones - EC500:		200	0
Maximum Off-PBX Telephones - OPS:		200	18
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 **System-Parameters Customer-Options** form, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient.

display system-parameters customer-options		Page 2 of 10
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	200	0
Maximum Concurrently Registered IP Stations:	18000	1
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 Stations:	0	0
Maximum Video Capable IP Softphones:	0	0
Maximum Administered SIP Trunks:	300	138
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0
Maximum Number of DS1 Boards with Echo Cancellation:	100	0
Maximum TN2501 VAL Boards:	128	0
Maximum Media Gateway VAL Sources:	0	0
Maximum TN2602 Boards with 80 VoIP Channels:	128	0
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.)		

5.2. Define System Features

Use the **change system-parameters features** command to administer system wide features for SIP endpoints. These are all standard Communication Manager features that are also available to OPS stations. On **Page 17** set **Whisper Page Tone Given To: all**

change system-parameters features		Page 17 of 18
FEATURE-RELATED SYSTEM PARAMETERS		
INTERCEPT TREATMENT PARAMETERS		
Invalid Number Dialed Intercept Treatment: tone		
Invalid Number Dialed Display:		
Restricted Number Dialed Intercept Treatment: tone		
Restricted Number Dialed Display:		
Intercept Treatment On Failed Trunk Transfers? n		
WHISPER PAGE		
Whisper Page Tone Given To: all		
6400/8400/2420J LINE APPEARANCE LED SETTINGS		
Station Putting Call On Hold: green		wink
Station When Call is Active: steady		
Other Stations When Call Is Put On Hold: green		wink
Other Stations When Call Is Active: green		
Ringing: green		flash
Idle: steady		
Pickup On Transfer? y		

5.3. Define the Dial Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions, OPS Feature Name Extensions (FNEs), and Feature Access Codes (FACs). Feature Access Code (FAC) must also be specified for the corresponding feature. In the sample configuration, telephone extensions are four digits long and begin with **1**, FNEs are also four digits beginning with **1**, and the FACs have formats as indicated with a **Call Type** of **fac**.

change dialplan analysis						Page 1 of 12		
DIAL PLAN ANALYSIS TABLE								
Location: all						Percent Full: 1		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
0	1	ext	7	4	ext			
1	4	ext	88	4	ext			
2	4	udp	89	4	ext			
3005	8	udp	9	1	fac			
3015	9	udp	*	3	fac			
31	4	udp	#	3	fac			

5.4. Define Feature Access Codes (FACs)

A FAC (feature access code) should be defined for each feature that will be used via the OPS FNEs. Use **change feature-access-codes** to define the required access codes. The FACs used in the sample configuration are shown in bold.

change feature-access-codes		Page 1 of 9
FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1 Access Code:		
Abbreviated Dialing List2 Access Code:		
Abbreviated Dialing List3 Access Code:		
Abbreviated Dial - Prgm Group List Access Code:		
Announcement Access Code:		
Answer Back Access Code: *24		
Attendant Access Code:		
Auto Alternate Routing (AAR) Access Code: 4		
Auto Route Selection (ARS) - Access Code 1: 9		Access Code 2:
Automatic Callback Activation: *25		Deactivation: #25
Call Forwarding Activation Busy/DA: *21 All: *20		Deactivation: #20
Call Forwarding Enhanced Status: Act:		Deactivation:
Call Park Access Code: *26		
Call Pickup Access Code: *27		
CAS Remote Hold/Answer Hold-Unhold Access Code:		
CDR Account Code Access Code:		
Change COR Access Code:		
Change Coverage Access Code:		
Conditional Call Extend Activation:		Deactivation:
Contact Closure Open Code:		Close Code:

FEATURE ACCESS CODE (FAC)

Contact Closure Pulse Code:

Data Origination Access Code:

Data Privacy Access Code:

Directed Call Pickup Access Code: *28

Directed Group Call Pickup Access Code:

Emergency Access to Attendant Access Code:

EC500 Self-Administration Access Codes:

Enhanced EC500 Activation:

Deactivation:

Enterprise Mobility User Activation:

Deactivation:

Extended Call Fwd Activate Busy D/A All:

Deactivation:

Extended Group Call Pickup Access Code:

Facility Test Calls Access Code:

Flash Access Code:

Group Control Restrict Activation:

Deactivation:

Hunt Group Busy Activation:

Deactivation:

ISDN Access Code:

Last Number Dialed Access Code: *29

Leave Word Calling Message Retrieval Lock:

Leave Word Calling Message Retrieval Unlock:

FEATURE ACCESS CODE (FAC)

Leave Word Calling Send A Message:

Leave Word Calling Cancel A Message:

Limit Number of Concurrent Calls Activation:

Deactivation:

Malicious Call Trace Activation:

Deactivation:

Meet-me Conference Access Code Change:

Message Sequence Trace (MST) Disable:

PASTE (Display PBX data on Phone) Access Code:

Personal Station Access (PSA) Associate Code:

Dissociate Code:

Per Call CPN Blocking Code Access Code: *34

Per Call CPN Unblocking Code Access Code: *35

Posted Messages Activation:

Deactivation:

Priority Calling Access Code: *30

Program Access Code:

Refresh Terminal Parameters Access Code:

Remote Send All Calls Activation:

Deactivation:

Self Station Display Activation:

Send All Calls Activation: *31**Deactivation: #31**

Station Firmware Download Access Code:

```

change feature-access-codes                                     Page 4 of 9
                                FEATURE ACCESS CODE (FAC)
                                Station Lock Activation:      Deactivation:
                                Station Security Code Change Access Code:
                                Station User Admin of FBI Assign: Remove:
                                Station User Button Ring Control Access Code:
                                Terminal Dial-Up Test Access Code:
                                Terminal Translation Initialization Merge Code: Separation Code:
                                Transfer to Voice Mail Access Code: *32
                                Trunk Answer Any Station Access Code:
                                User Control Restrict Activation: Deactivation:
                                Voice Coverage Message Retrieval Access Code:
                                Voice Principal Message Retrieval Access Code:
                                Whisper Page Activation Access Code: *33

                                PIN Checking for Private Calls Access Code:
                                PIN Checking for Private Calls Using ARS Access Code:
                                PIN Checking for Private Calls Using AAR Access Code:

```

5.5. Define Feature Name Extensions (FNEs)

The OPS FNEs can be defined using the **change off-pbx-telephone feature-name-extensions** command. The following screens show in bold the FNEs defined for use with the sample configuration.

```

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

    Active Appearance Select: 1700
    Automatic Call Back: 1701
    Automatic Call-Back Cancel: 1702
    Call Forward All: 1703
    Call Forward Busy/No Answer: 1704
    Call Forward Cancel: 1705
    Call Park: 1706
    Call Park Answer Back: 1707
    Call Pick-Up: 1708
    Calling Number Block: 1709
    Calling Number Unblock: 1710
    Conditional Call Extend Enable: 1711
    Conditional Call Extend Disable: 1712
    Conference Complete: 1713
    Conference on Answer: 1714
    Directed Call Pick-Up: 1715
    Drop Last Added Party: 1716

```



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

Page 2 of 2

```

Exclusion (Toggle On/Off): 1717
Extended Group Call Pickup:
  Held Appearance Select: 1718
  Idle Appearance Select: 1719
  Last Number Dialed: 1720
Malicious Call Trace:
Malicious Call Trace Cancel:
  Off-Pbx Call Enable:
  Off-Pbx Call Disable:
    Priority Call: 1725
    Recall: 1726
    Send All Calls: 1727
    Send All Calls Cancel: 1728
    Transfer Complete: 1729
    Transfer On Hang-Up: 1730
    Transfer to Voice Mail: 1731
    Whisper Page Activation: 1732

```

5.6. Configure Class of Service (COS)

Use the **change cos** command to set the appropriate service permissions to support OPS features (shown in bold). For the sample configuration a COS of **1** was used.

```
change cos
```

Page 1 of 2

	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	y	y	n	y	n
Call Fwd-All Calls	n	y	n	y	y	n	n	y	y	n	n	y	y	n	n	y
Data Privacy	n	n	n	n	n	y	y	y	y	n	n	n	n	y	y	y
Priority Calling	n	y	n	n	n	n	n	n	n	y	y	y	y	y	y	y
Console Permissions	n	y	n	n	n	n	n	n	n	n	n	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	n	n	n	n	n	n	n	n	n	n	n
Restrict Call Fwd-Off Net	y	n	y	y	y	y	y	y	y	y	y	n	y	y	y	y
Call Forwarding Busy/DA	n	y	n	n	n	n	n	n	n	n	n	y	n	n	n	n
Personal Station Access (PSA)	n	y	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Extended Forwarding All	n	y	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Extended Forwarding B/DA	n	y	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Trk-to-Trk Transfer Override	n	y	n	n	n	n	n	n	n	n	n	y	n	n	n	n
QSIG Call Offer Originations	n	n	n	n	n	n	n	n	n	n	n	y	n	n	n	n
Contact Closure Activation	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n

5.7. Configure Class of Restriction (COR)

Use the **change cor 1** command to enable applicable calling features. To use the Directed Call Pickup feature, the **Can Be Picked Up By Directed Call Pickup** and **Can Use Directed Call Pickup** fields must be set to **y**. In the sample configuration the handsets were assigned to COR 1.

change cor 1		Page 1 of 23
CLASS OF RESTRICTION		
COR Number: 1		
COR Description: Default		
FRL: 0		
APLT? y		
Can Be Service Observed? y		
Calling Party Restriction: none		
Can Be A Service Observer? y		
Called Party Restriction: none		
Partitioned Group Number: 1		
Forced Entry of Account Codes? n		
Priority Queuing? n		
Direct Agent Calling? n		
Restriction Override: all		
Facility Access Trunk Test? n		
Restricted Call List? n		
Can Change Coverage? n		
Access to MCT? y		
Fully Restricted Service? n		
Group II Category For MFC: 7		
Send ANI for MFE? n		
MF ANI Prefix:		
Automatic Charge Display? n		
Hear System Music on Hold? y		
PASTE (Display PBX Data on Phone)? y		
Can Be Picked Up By Directed Call Pickup? y		
Can Use Directed Call Pickup? y		
Group Controlled Restriction: inactive		

5.8. Add Stations

Unlike previous versions of Session Manager the Station Features and button assignments can be added using the Endpoint Editor in System Manager. This method was used in this test configuration and procedure can be found In **Section 6.9**

5.8.1. Verify Off PBX Station Mapping

Following completion of the procedures in **Section 6.9** use the **display off-pbx-telephone station-mapping** command to verify that SIP Endpoints added to Session Manager in **section 6.9** have been administered in Communication Manager. The example below shows that **Station Extension 1318** uses the **Application OPS**.

display off-pbx-telephone station-mapping						Page	1 of	3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION								
Station	Application	Dial	CC	Phone Number	Trunk	Config	Dual	
Extension		Prefix			Selection	Set	Mode	
1318	OPS	-		1318	aar	1		

5.9. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the C-LAN board in the Avaya G650 Media Gateway and the Session Manager IP address. The host names will be used throughout the other configuration screens of Communication Manager.

```
change node-names ip
                                     IP NODE NAMES
      Name      IP Address
AES522         10.10.16.25
CLAN          10.10.16.31
CM521          10.10.16.23
Gateway        10.10.16.1
MedPro         10.10.16.32
61sysmgr       10.10.16.56
61sesmgr       10.10.16.54
SM61         10.10.16.201
default        0.0.0.0
procr          10.10.16.47
procr6         ::
( 16 of 16   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**. By default, **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used. This codec set will be used for calls routed over the SIP trunk to Session Manager as **ip-network region 1** is specified in the SIP signaling group.

```
change ip-network-region 1
                                     Page 1 of 19
                                     IP NETWORK REGION
      Region: 1
Location: 1      Authoritative Domain: avaya.com
      Name: Default Region
MEDIA PARAMETERS
      Codec Set: 1      Intra-region IP-IP Direct Audio: yes
                        Inter-region IP-IP Direct Audio: yes
      UDP Port Min: 2048      IP Audio Hairpinning? y
      UDP Port Max: 8001
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
```

In the **IP Codec Set** form, select the audio codec's supported for calls routed over the SIP trunk. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set 1 was specified in IP Network Region 1 shown above. Multiple codecs may be specified in the **IP Codec Set** form in order of preference; the example below includes **G.711A** (a-law), **G.711MU** (mu-law) and **G.729**.

change ip-codec-set 1

Page 1 of 2

IP Codec Set

Codec Set: 1

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1: G.711A	n	2	20
2: G.711MU	n	2	20
3: G.729	n	2	20
4:			
5:			
6:			
7:			

Media Encryption

1: none

2:

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form shown as follows:

- Set the **Group Type** field to **sip**
- Set the **Transport Method** to the desired transport method; **tcp** (transport control protocol) or **tls** (Transport Layer Security). **Note:** for transparency tcp was used during this compliance test but the recommended method is tls.
- Specify the node names for the C-LAN board in the G650 Media Gateway and the active Session Manager node name as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These values are taken from the **IP Node Names** form shown above.
- Ensure that the recommended port value of **5060** for tcp is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields **Note:** If tls is used then the recommended port value is 5061.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is **avaya.com**. This domain is specified in the Uniform Resource Identifier (URI) of the “SIP To Address” in the INVITE message. Misconfiguring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.
- If calls to/from SIP endpoints are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to **y**.
- The **DTMF over IP** field should be set to the default value of **rtp-payload**. Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 6		Page 1 of 1	
SIGNALING GROUP			
Group Number: 6	Group Type: sip		
	Transport Method: tcp		
IMS Enabled? n			
IP Video? n			
Near-end Node Name: CLAN1	Far-end Node Name: 61sesmgr		
Near-end Listen Port: 5060	Far-end Listen Port: 5060		
	Far-end Network Region: 1		
Far-end Domain: avaya.com			
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n		
	RFC 3389 Comfort Noise? n		
DTMF over IP: rtp-payload	Direct IP-IP Audio Connections? y		
Session Establishment Timer(min): 3	IP Audio Hairpinning? y		
Enable Layer 3 Test? n	Direct IP-IP Early Media? n		
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6		

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to SIP Handsets. Enter a descriptive name in the **Group Name** field. Set the **Group Type** field to **sip**. Enter a **TAC** code compatible with the Communication Manager. Set the **Service Type** field to **tie**, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

add trunk-group 6		Page 1 of 21	
TRUNK GROUP			
Group Number: 6	Group Type: sip	CDR Reports: y	
Group Name: SES OPS	COR: 1	TN: 1	TAC: 506
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Signaling Group: 6	
		Number of Members: 30	

On **Page 3** of the trunk group form, set the **Numbering Format** field to **private**. This field specifies the format of the calling party number sent to the far-end.

add trunk-group 6		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private		UUI Treatment: service-provider	
		Replace Restricted Numbers? y	
		Replace Unavailable Numbers? y	
Show ANSWERED BY on Display? y			

Configure the **Private Numbering** form to send the calling party number to the far-end. Add entries so that local stations with a 4-digit extension beginning with **13, 15 and 16** and whose calls are routed over SIP trunk group **6** have the number sent to the far-end for display purposes.

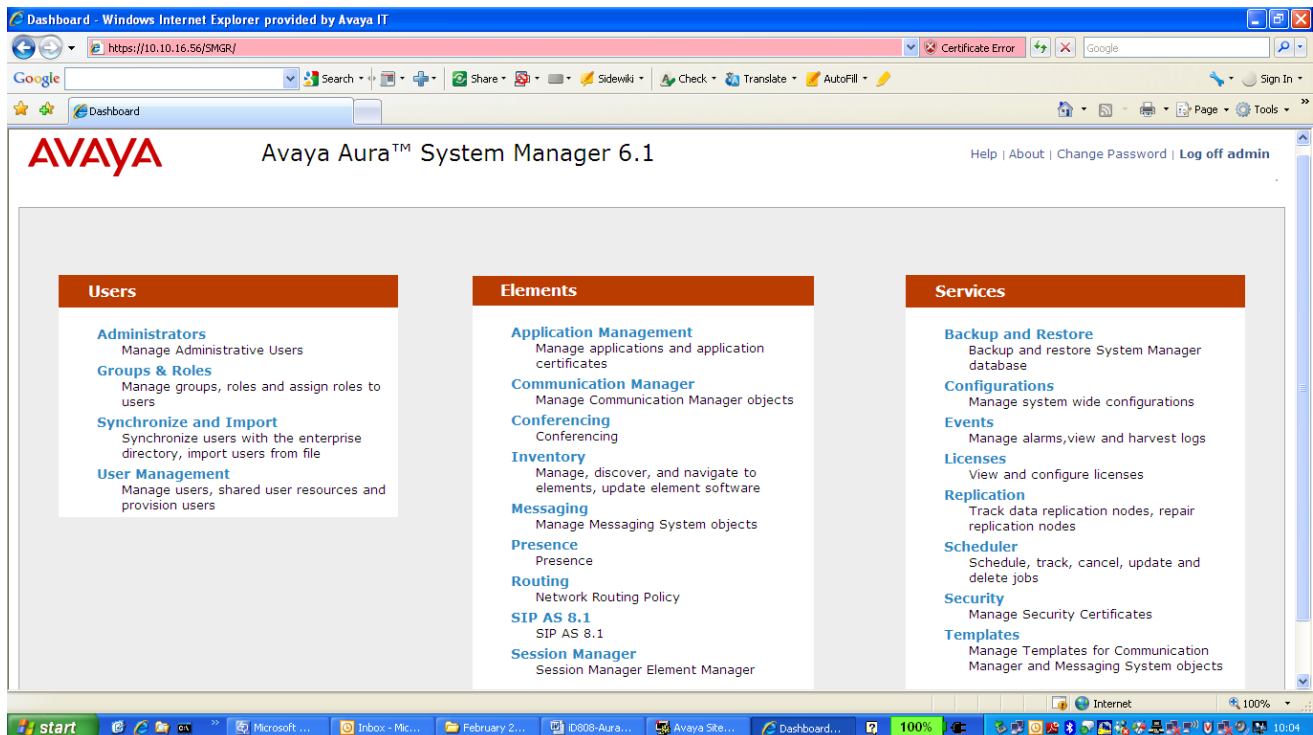
change private-numbering 0		Page 1 of 2	
NUMBERING - PRIVATE FORMAT			
Ext Len	Ext Code	Trk Grp(s)	Private Prefix
4	13	6	
4	15	6	
4	16	6	
		Total Len	
		4	Total Administered: 3
		4	Maximum Entries: 540
		4	

6. Configure Avaya Aura® Session Manager

This section covers the administration of Session Manager. Session Manager is configured via an internet browser using the System Manager web interface. It is assumed that Session Manager software has already been installed. For additional information on installation tasks refer to [4].

6.1. Logging in to Avaya Aura® System Manager

To access the administration web interface, enter **http://<ip-addr>/SMGR** as the URL in an Internet browser. Where <ip-addr> is the IP address of smgr on System Platform. Log in with the appropriate credentials. The main screen is displayed, as shown below.



6.2. Verify System Properties

From the main screen of the web interface, choose **Session Manager** from the **Elements** section. Verify that a green tick shows under **Tests Passed**, **Security Module** is **Up** and **Service State** is set to **Accept New Service**.

The screenshot shows the 'Session Manager Dashboard' in the Avaya Aura System Manager. The left sidebar contains a navigation menu with options like Dashboard, Session Manager, Administration, and various configuration sections. The main content area displays the 'Session Manager Instances' table. The table has columns for Session Manager, Type, Alarms, Tests Pass, Security Module, Service State, Entity Monitoring, Active Call Count, Registrations, and Version. The row for '61sesmgr' shows 'Core' type, '8/0/1' alarms, a green checkmark for 'Tests Pass', 'Up' for 'Security Module', and 'Accept New Service' for 'Service State'. The 'Service State' cell is highlighted with a red border.

Session Manager	Type	Alarms	Tests Pass	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Version
61sesmgr	Core	8/0/1	✓	Up	Accept New Service	0/2	0	4	6.1.0.0.610

Next go to **Routing** from the **Elements** section of the main screen and select **Domains**. Check the domain administered.

The screenshot shows the 'Domain Management' page in the Avaya Aura System Manager. The left sidebar contains a navigation menu with options like Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area displays the 'Domain Management' table. The table has columns for Name, Type, Default, and Notes. The row for 'avaya.com' shows 'sip' type and a checkbox for 'Default'. The 'Default' checkbox is checked.

Name	Type	Default	Notes
avaya.com	sip	<input checked="" type="checkbox"/>	

6.3. Add Location

Select **Routing** (not shown) from the **Elements** section of the main screen and chose **Locations**. Click on the new button (not shown) and add a **Name** and **IP Address Pattern** for the Location in the format shown under **Location Patterns**. Click on the **Commit** button to save.

Location Details Commit

Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth.
See Session Manager -> Session Manager Administration -> Global Setting

General

* **Name:**
Notes:

Overall Managed Bandwidth

Managed Bandwidth Units:
Total Bandwidth:

Per-Call Bandwidth Parameters

* **Default Audio Bandwidth:**

Location Pattern

1 Item | Filter: 0

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.10.16.*	<input type="text"/>

6.4. Create a SIP entity

From the **Elements** section of the main screen choose **Routing**. From the left hand side menu choose **SIP Entities**. Click on **New** and enter a **Name** and **FQDN or IP Address** for the Session Manager Security Module. Select **Type** as **Session Manager** and **Location** as the Session Manager Location created in **Section 6.3**.

SIP Entity Details [Commit]

General

* Name: 61sesmgr

* FQDN or IP Address: 10.10.16.201

Type: Session Manager

Notes:

Location: SessionMGR

Outbound Proxy:

Time Zone: Etc/GMT

Credential name:

SIP Link Monitoring: Use Session Manager Configuration

SIP Link Monitoring

Add the **Protocol** and **Port** information to the **Port** section of the SIP Entity details screen below. The entity link section will automatically populate after the link is added in **section 6.5**. Click **Commit** to save the changes.

Entity Links

Add Remove

2 Items Refresh Filter: Enable

SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
61sesmgr	TCP	* 5060	Commgr	* 5060	<input checked="" type="checkbox"/>
61sesmgr	TLS	* 5061	SBC6	* 5061	<input checked="" type="checkbox"/>

Select : All, None

Port

Add Remove

3 Items Refresh Filter: Enable

Port	Protocol	Default Domain	Notes
5060	TCP	avaya.com	
5060	UDP	avaya.com	
5061	TLS	avaya.com	

Select : All, None

* Input Required [Commit] [Cancel]

A Communication Manager SIP Entity must be added also with an appropriate **Name** and the **FQDN or IP Address** of the CLAN checked in **Section 5.9 Protocol** and **Port** details are added in the same way as the previous screen.

Home / Elements / Routing / SIP Entities - SIP Entity Details

SIP Entity Details

General

* Name: Commgr

* FQDN or IP Address: 10.10.16.31

Type: CM

Notes:

Adaptation:

Location: SessionMGR

Time Zone: Etc/GMT

Override Port & Transport with DNS SRV:

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

Commit

6.5. Add an Entity link

From the **Routing** menu choose **Entity Links**, choose an appropriate **Name** and then choose the entities added in **section 6.4**, the **Protocol** used (TCP used in this example) and the **Port** the protocol communicates on. Click on the **Commit** button to save.

Home / Elements / Routing / Entity Links - Entity Links

Entity Links

1 Item | Refresh

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* ToCM	* 61sesmgr	TCP	* 5060	* Commgr	* 5060	<input checked="" type="checkbox"/>	

* Input Required

Commit

6.6. Add Avaya Aura® Communication Manager Managed Element

From the **Elements** section of the main screen chose **Inventory** and then **Manage Elements**. Click the **New** (not shown) button and enter a valid **Name**, **Type** as **CM** and the SAT IP address in the **Node** field. Click on **Commit** to save.

The screenshot shows the 'Edit CM: CommsMGR' form. The left sidebar contains a tree view with 'Inventory' expanded, showing 'Manage Elements', 'Discovered Inventory', 'Discovery', 'Management', and 'Synchronization'. The main content area has a breadcrumb 'Home / Elements / Inventory / Manage Elements - Edit CM' and a 'Commit' button. Below the breadcrumb are tabs for 'Application' and 'Attributes'. The 'Application' tab is active, showing a form with the following fields: 'Name' (CommsMGR), 'Type' (CM), 'Description' (empty text area), and 'Node' (10.10.16.47). A red box highlights the 'Name', 'Type', and 'Node' fields.

6.7. Add Routing Policy

From the **Elements** section of the main screen chose **Routing** and then **Routing Policies**. Click on the **New** button and add a **Name** for the policy. Select the Communication Manager entity as a Destination under **SIP Entity as Destination**.

The screenshot shows the 'Routing Policy Details' form. The left sidebar contains a tree view with 'Routing' expanded, showing 'Domains', 'Locations', 'Adaptations', 'SIP Entities', 'Entity Links', 'Time Ranges', 'Routing Policies', 'Dial Patterns', 'Regular Expressions', and 'Defaults'. The main content area has a breadcrumb 'Routing Policy Details' and a 'Commit' button. Below the breadcrumb are tabs for 'General' and 'SIP Entity as Destination'. The 'General' tab is active, showing a form with the following fields: 'Name' (ToCM), 'Disabled' (checkbox), and 'Notes' (empty text area). A red box highlights the 'Name' field. The 'SIP Entity as Destination' tab is also visible, showing a 'Select' button and a table with the following data:

Name	FQDN or IP Address	Type	Notes
Commgr	10.10.16.31	CM	

Add the **Dial Patterns** for non SIP stations and PSTN routing. A **Pattern** to be dialed and **Min**, **Max** digits are entered. Click on the **Commit** button to save.

Dial Patterns

[Add](#) [Remove](#)

1 Item | [Refresh](#) Filter:

<input type="checkbox"/>	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
<input type="checkbox"/>	0141	10	14	<input type="checkbox"/>	-ALL-	-ALL-	

Select : All, None

Regular Expressions

[Add](#) [Remove](#)

0 Items | [Refresh](#) Filter:

<input type="checkbox"/>	Pattern	Rank Order	Deny	Notes
--------------------------	---------	------------	------	-------

* Input Required [Commit](#)

6.8. Add Application and Application Sequence

Select **Session Manager** from the **Elements** section of the main screen and choose **Application Configuration** → **Application**. Click on the **New** button (not shown) and enter an appropriate **Name**, Select the CM SIP Entity added in **Section 6.4** and the Communication Manager Managed Element added in **Section 6.6** as **CM System for SIP Entity**. Click on the **Commit** button to save.

Session Manager | Home / Elements / Session Manager / Application Configuration / Applications - Applications

Application Editor [Commit](#)

Application

*Name

*SIP Entity

*CM System for SIP Entity [Refresh](#) [View/Add CM Systems](#)

Description

Application Attributes (optional)

Name	Value
Application Handle	<input type="text"/>
URI Parameters	<input type="text"/>

* Required [Commit](#)

Next, choose **Application Sequences** and click the **New** button (not shown). Add a **Name** and select the Application added above to interact with the Communication Manager Entity.

Application Sequence Editor Commit

Application Sequence

*Name

Description

Applications in this Sequence

Move First Move Last Remove

1 Item

<input type="checkbox"/>	Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
<input type="checkbox"/>	1	add	Commgr	<input checked="" type="checkbox"/>	

Select : All, None

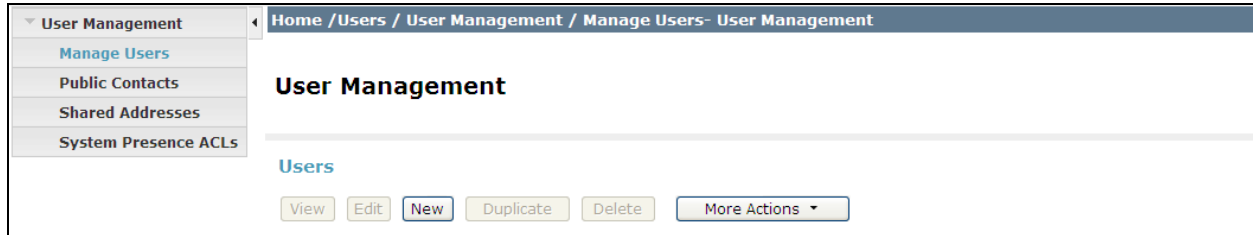
Available Applications

2 Items | Refresh Filter

	Name	SIP Entity	Description
+	add	Commgr	
+	SBCadd	SBC6	

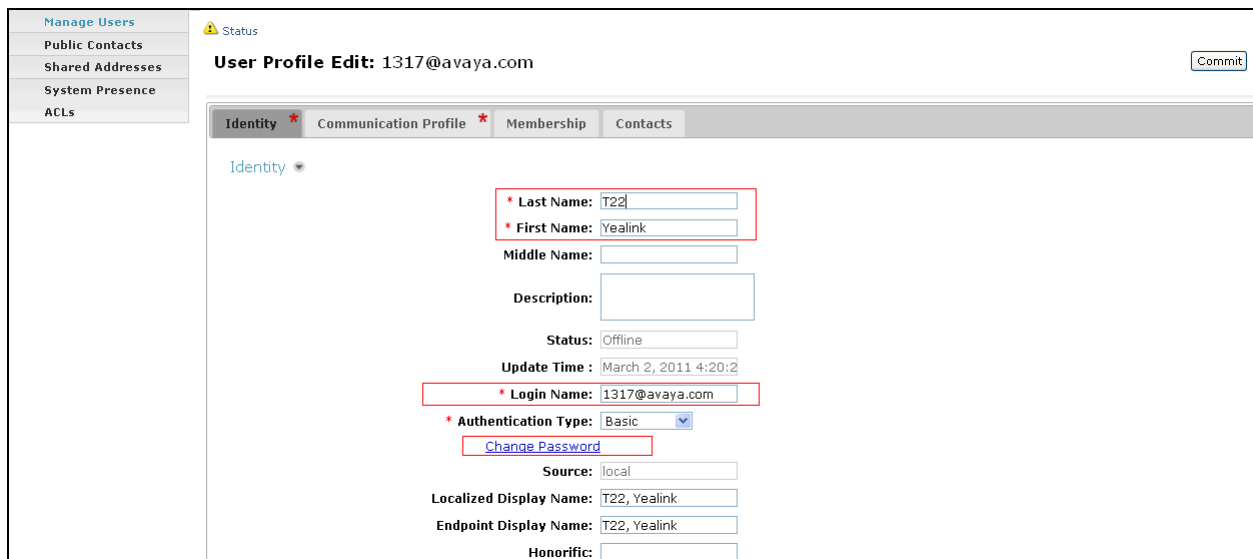
6.9. Add User

From the **User** section of the main screen choose **User Management** and then choose **Manage Users** from the menu. Click **New** to add a user.



The screenshot shows the 'User Management' interface. On the left, there is a sidebar with 'User Management' selected, and a sub-menu with 'Manage Users', 'Public Contacts', 'Shared Addresses', and 'System Presence ACLs'. The main area has a breadcrumb 'Home /Users / User Management / Manage Users- User Management' and a title 'User Management'. Below the title, there is a 'Users' section with buttons for 'View', 'Edit', 'New', 'Duplicate', 'Delete', and 'More Actions'.

Under the **Identity** tab fill in the required information. The **Login Name** field contains the fully qualified name in the form <user>@<sip domain>. The **Password** in this section is purely for user log in and is not the passcode used to log in the phone.



The screenshot shows the 'User Profile Edit' form for '1317@avaya.com'. The 'Identity' tab is selected. The form contains the following fields: 'Last Name' (T22), 'First Name' (Yealink), 'Middle Name' (empty), 'Description' (empty), 'Status' (Offline), 'Update Time' (March 2, 2011 4:20:2), 'Login Name' (1317@avaya.com), 'Authentication Type' (Basic), 'Source' (local), 'Localized Display Name' (T22, Yealink), 'Endpoint Display Name' (T22, Yealink), and 'Honorific' (empty). There is a 'Change Password' link and a 'Commit' button.

Under the **Communication Profile** tab enter the **Communication Profile Password** as the passcode used to log in the handset.



The screenshot shows the 'Communication Profile' tab in the 'User Profile Edit' form. The 'Communication Profile Password' field is visible with a red box around it. Below the password field, there are buttons for 'New', 'Delete', 'Done', and 'Cancel'. At the bottom, there is a table with a single row labeled 'Primary' and a 'Select : None' option.

Still on the **Communication Profile** tab move down to **Communication Address** and click on the **New** button. Enter the **Type** as **Avaya SIP** and the **Fully Qualified Address** the same as on the Identity tab.

Communication Address

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP

* Fully Qualified Address: 1317 @ avaya.com

Add

Move down and select **Session Manager Profile**. Fill in the details with the **Primary Session Manager** as the SIP entity added in **Section 6.4**. Fill in the **Application Sequences** as the Application Sequence added in **Section 6.8**. Fill in the **Home Location** as the Location added in **Section 6.3**

Session Manager Profile

* Primary Session Manager 61sesmgr

Primary	Secondary	Maximum
21	0	21

Secondary Session Manager (None)

Primary	Secondary	Maximum

Origination Application Sequence app seq

Termination Application Sequence app seq

Survivability Server (None)

* Home Location SessionMGR

Move down and select **Endpoint Profile**. Fill in the System as the Communication Manager Managed Element added in **Section 6.6**. Add the **Extension**, **Set Type** and **Port** as required and tick the **Delete Endpoint on Unassign of Endpoint from User or on Delete User** box.

Note: Endpoint editor can be used to administer features and buttons but this was not required in this instance.

Endpoint Profile

* System CommsMGR

* Profile Type Endpoint

Use Existing Endpoints ☐

* Extension 1317 Endpoint Editor

Template DEFAULT_9630SIP_CM_6_0

Set Type 9630SIP

Security Code

* Port 800047

Voice Mail Number

Delete Endpoint on Unassign of Endpoint from User or on Delete User. ☒

7. Configure Yealink T-22 Handset

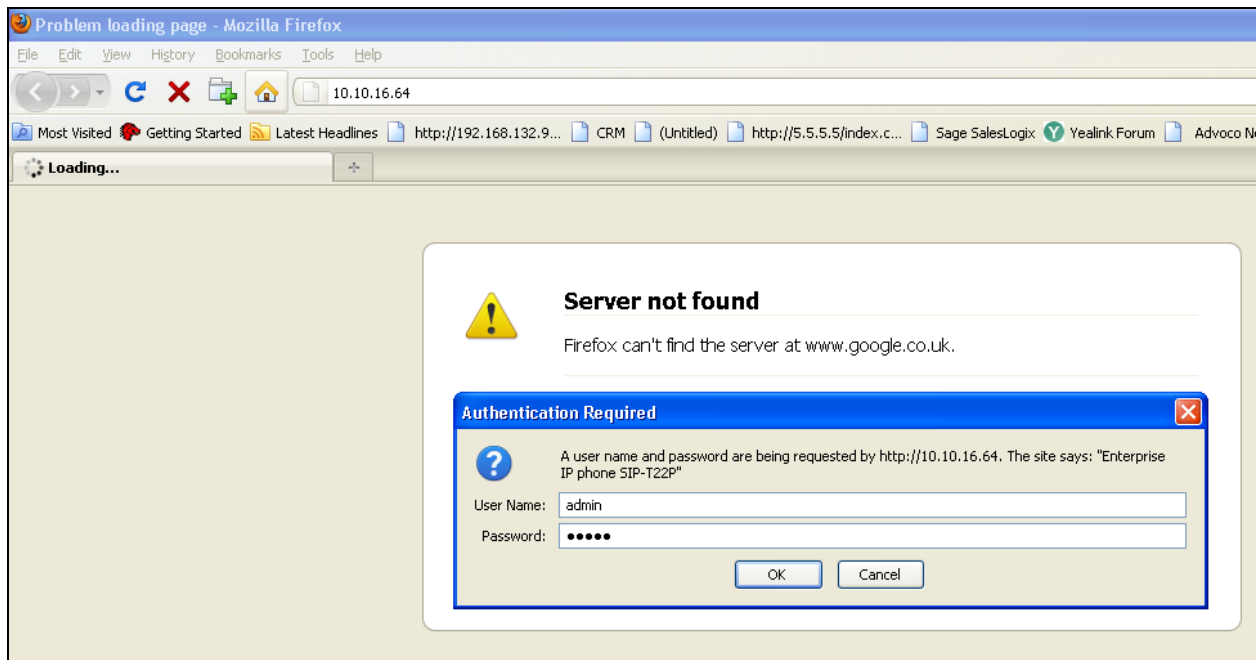
This section covers the administration of the Yealink T-22 Handset device. The Yealink T-22 is configured via an Internet browser using the integral web interface. To access the web interface the IP Address of the device is entered into the browser command line. The Yealink T-22 by default is set to obtain an IP Address by DHCP.

7.1. Determining device IP Address

Press OK button on the keypad of the phone to enter the status page and find out the IP address of IP phone. Enter it (for example <http://10.10.16.64>) into the address bar of web browser. The default administrator's login name and password are **admin/admin**.

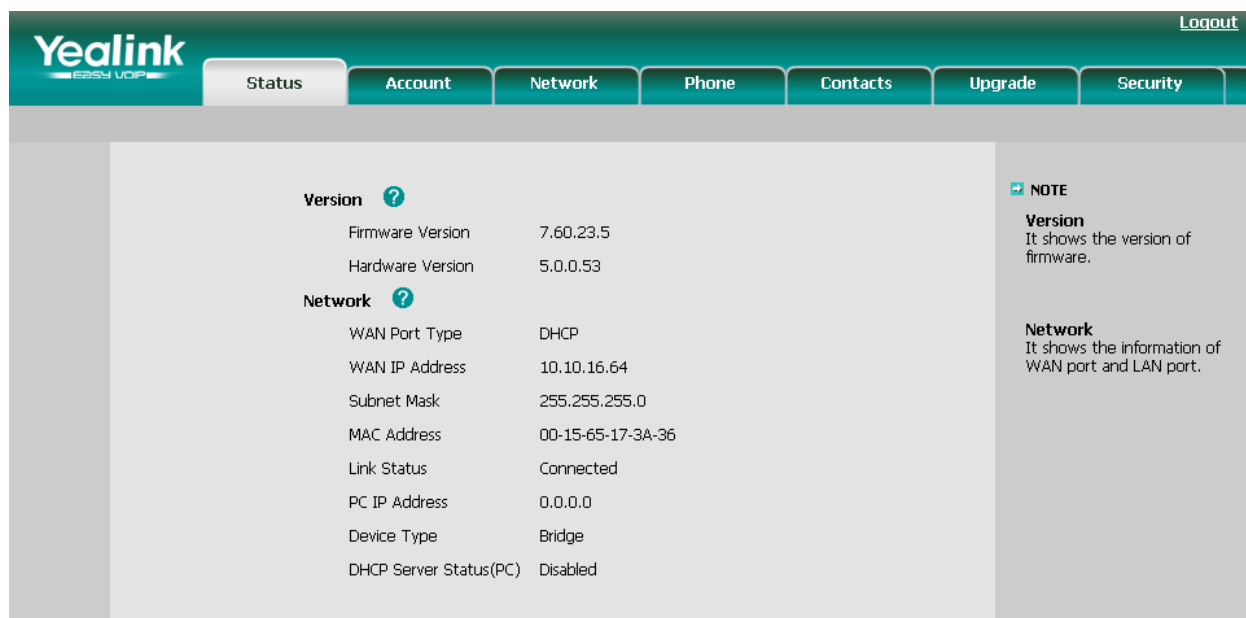
7.2. Configuring via the Web Browser

Enter the Yealink T-22 IP Address (for example <http://10.10.16.64>) into the address bar of web browser. The default login name and password are both **admin**.



7.3. Status Screen

The Status Screen will be shown following a successful log in.



The screenshot shows the Yealink web interface with the 'Status' tab selected. The interface has a green header with the Yealink logo and a 'Logout' link. Below the header is a navigation bar with tabs: Status, Account, Network, Phone, Contacts, Upgrade, and Security. The main content area is divided into two sections. The left section contains two expandable sections: 'Version' and 'Network'. The 'Version' section shows 'Firmware Version' as 7.60.23.5 and 'Hardware Version' as 5.0.0.53. The 'Network' section shows various network parameters: WAN Port Type (DHCP), WAN IP Address (10.10.16.64), Subnet Mask (255.255.255.0), MAC Address (00-15-65-17-3A-36), Link Status (Connected), PC IP Address (0.0.0.0), Device Type (Bridge), and DHCP Server Status(PC) (Disabled). The right section contains a 'NOTE' box with two entries: 'Version' (It shows the version of firmware.) and 'Network' (It shows the information of WAN port and LAN port.).

Version	
Firmware Version	7.60.23.5
Hardware Version	5.0.0.53

Network	
WAN Port Type	DHCP
WAN IP Address	10.10.16.64
Subnet Mask	255.255.255.0
MAC Address	00-15-65-17-3A-36
Link Status	Connected
PC IP Address	0.0.0.0
Device Type	Bridge
DHCP Server Status(PC)	Disabled

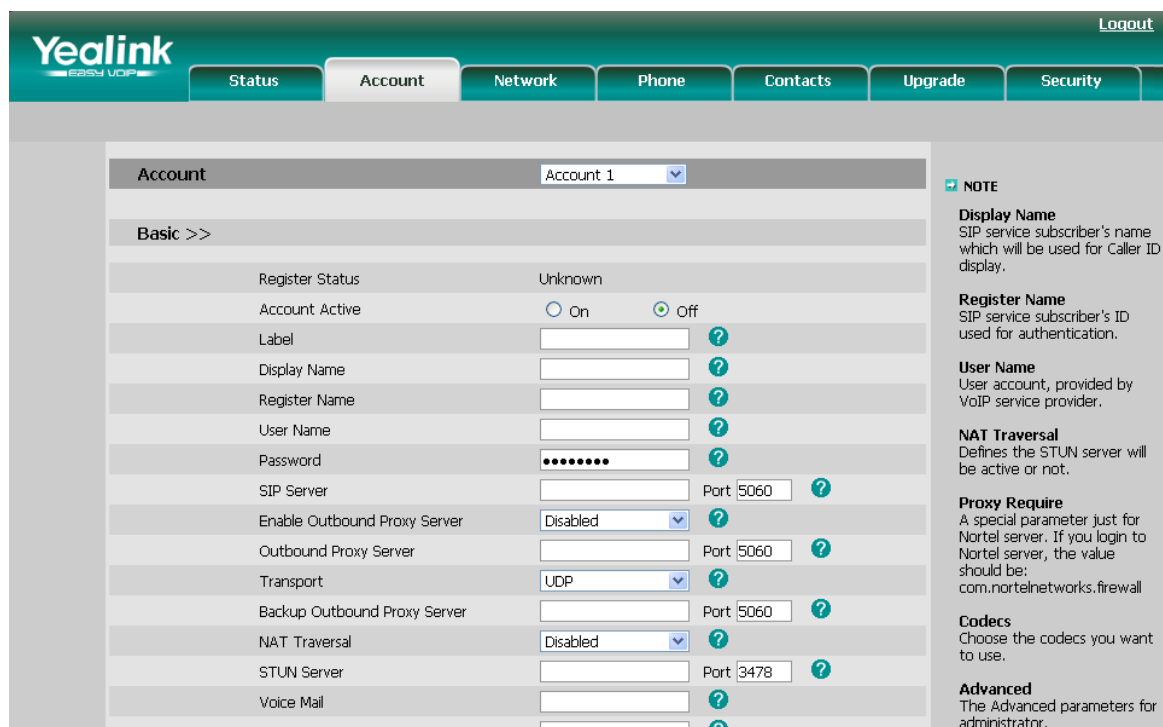
NOTE

Version
It shows the version of firmware.

Network
It shows the information of WAN port and LAN port.

7.4. Account Configuration

Click on the tab labeled **Account**



The screenshot shows the Yealink web interface with the 'Account' tab selected. The interface has a green header with the Yealink logo and a 'Logout' link. Below the header is a navigation bar with tabs: Status, Account, Network, Phone, Contacts, Upgrade, and Security. The main content area is divided into two sections. The left section contains an 'Account' section with a dropdown menu showing 'Account 1'. Below this is a 'Basic >>' section with various configuration options: Register Status (Unknown), Account Active (On/Off), Label, Display Name, Register Name, User Name, Password, SIP Server, Enable Outbound Proxy Server (Disabled), Outbound Proxy Server, Transport (UDP), Backup Outbound Proxy Server, NAT Traversal (Disabled), STUN Server, Voice Mail, and Proxy Require. The right section contains a 'NOTE' box with several entries: 'Display Name' (SIP service subscriber's name which will be used for Caller ID display.), 'Register Name' (SIP service subscriber's ID used for authentication.), 'User Name' (User account, provided by VoIP service provider.), 'NAT Traversal' (Defines the STUN server will be active or not.), 'Proxy Require' (A special parameter just for Nortel server. If you login to Nortel server, the value should be: com.nortelnetworks.firewall), 'Codecs' (Choose the codecs you want to use.), and 'Advanced' (The Advanced parameters for administrator.).

Account	
Account	Account 1

Basic >>

Register Status	Unknown
Account Active	<input type="radio"/> On <input checked="" type="radio"/> Off
Label	<input type="text"/>
Display Name	<input type="text"/>
Register Name	<input type="text"/>
User Name	<input type="text"/>
Password	<input type="password"/>
SIP Server	<input type="text"/> Port 5060
Enable Outbound Proxy Server	Disabled
Outbound Proxy Server	<input type="text"/> Port 5060
Transport	UDP
Backup Outbound Proxy Server	<input type="text"/> Port 5060
NAT Traversal	Disabled
STUN Server	<input type="text"/> Port 3478
Voice Mail	<input type="text"/>
Proxy Require	<input type="text"/>

NOTE

Display Name
SIP service subscriber's name which will be used for Caller ID display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be: com.nortelnetworks.firewall

Codecs
Choose the codecs you want to use.

Advanced
The Advanced parameters for administrator.

7.4.1. Enter the Account details

Enter the account details as highlighted in blue in the image below to match the settings in the Session Manager added in **Section 6.9**. Press the **Confirm** button at the bottom of the page to save the changes and if the details have been entered correctly the **Register Status** will be **Registered** as highlighted in Red in the image below.

Yealink
EASY VoIP

Status Account Network Phone Contacts Upgrade Security

Account Account 1

Basic >>

Register Status	Registered
Account Active	<input checked="" type="radio"/> On <input type="radio"/> Off
Label	T22-1317 ?
Display Name	T22-1317 ?
Register Name	1317 ?
User Name	1317 ?
Password ?
SIP Server	10.10.16.201 Port 5060 ?
Enable Outbound Proxy Server	Disabled ?
Outbound Proxy Server	Port 5060 ?
Transport	UDP ?
Backup Outbound Proxy Server	Port 5060 ?
NAT Traversal	Disabled ?
STUN Server	Port 3478 ?
Voice Mail	1699 ?
Proxy Require	?

NOTE

Display Name
SIP service subscriber's name which will be used for Caller ID display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be: com.nortelnetworks.firewall

Codecs
Choose the codecs you want to use.

Advanced
The Advanced parameters for administrator.

Click the **Advanced** option under the **Account** tab and enter the voicemail message waiting settings as highlighted in blue in the image below. If Voicemail is to be used as part of the Yealink T-22 setup the setting: **SubscribeMWIToVM** must be set to **Disabled** to enable the device to register to the voicemail system as the Account Number.

The screenshot shows the Yealink web interface with the 'Account' tab selected. The 'Advanced' settings section is expanded, showing various configuration options. The 'SubscribeMWIToVM' setting is highlighted with a blue box and is set to 'Disabled'.

Account	
Account 1	
Basic >>	
Codecs >> ?	
Advanced >>	
UDP Keep-alive Message	Enabled ?
UDP Keep-alive Interval(seconds)	30
Login Expire(seconds)	3600 ?
Local SIP Port	5060 ?
RPort	Disabled ?
SIP Session Timer(seconds) T1	0.5 ?
SIP Session Timer(seconds) T2	4
SIP Session Timer(seconds) T4	5
Subscribe Period(seconds)	1800 ?
DTMF Type	RFC2833 ?
How to INFO DTMF	Disabled
DTMF Payload(Scope:96~255)	101
100 reliable retransmission	Disabled ?
Enable Precondition	Disabled ?
Subscribe Register	Disabled ?
Subscribe for MWI	Enabled ?
MWI Subscription Period(Scope:0~84600)(seconds)	3600
SubscribeMWIToVM	Disabled

NOTE

Display Name
SIP service subscriber's name which will be used for Caller ID display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be: com.nortelnetworks.firewall

Codecs
Choose the codecs you want to use.

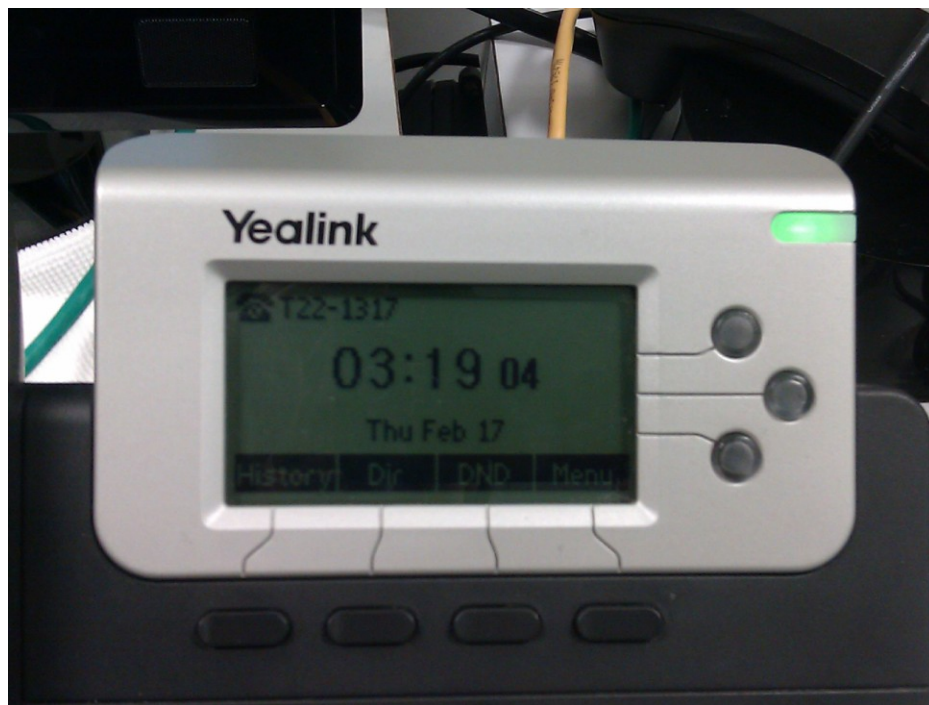
Advanced
The Advanced parameters for administrator.

8. Verification Steps

All features were tested using the sample configuration. The following steps can be used to verify and/or troubleshoot installations in the field. Verify that the T-22 handset has successfully registered with Session Manager, from the Dashboard select **Session Manager** from the **Elements** section and choose **System Status** → **User Registrations** this will display a list of registered user's on Session Manager as shown below. The **Address** and **IP Address** fields are populated when the handset has successfully registered.

System Status	<input type="checkbox"/>	► Show	1320@avaya.com	1320@avaya.com	Yealink	VP2009	SessionMGR	10.10.16.51:5062	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
SIP Entity	<input type="checkbox"/>	► Show	---	1310@avaya.com	i808	Turret1	SessionMGR	---	<input type="checkbox"/>	<input type="checkbox"/>		<input type="checkbox"/>
Monitoring	<input type="checkbox"/>	► Show	1321@avaya.com	1321@avaya.com	Yealink2	VP2009	SessionMGR	10.10.16.59:5062	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
Managed Bandwidth	<input type="checkbox"/>	► Show	1315@avaya.com	1315@avaya.com	Yealink	T18	SessionMGR	10.10.16.70:5062	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
Usage	<input type="checkbox"/>	► Show	---	1311@avaya.com	i808	Turret2	SessionMGR	---	<input type="checkbox"/>	<input type="checkbox"/>		<input type="checkbox"/>
Security Module	<input type="checkbox"/>	► Show	1316@avaya.com	1316@avaya.com	Yealink	T20	SessionMGR	10.10.16.57:5062	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
Status	<input type="checkbox"/>	► Show	---	1523@avaya.com	iTurret2	Privacy2	SessionMGR	---	<input type="checkbox"/>	<input type="checkbox"/>		<input type="checkbox"/>
Registration	<input type="checkbox"/>	► Show	1317@avaya.com	1317@avaya.com	Yealink	T22	SessionMGR	10.10.16.64:5062	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
Summary	<input type="checkbox"/>	► Show	---	1522@avaya.com	iTurret2	Privacy1	SessionMGR	---	<input type="checkbox"/>	<input type="checkbox"/>		<input type="checkbox"/>
User Registrations	<input type="checkbox"/>	► Show	1318@avaya.com	1318@avaya.com	Yealink	T26	SessionMGR	10.10.16.66:5062	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>

The picture below shows that the T-22 Handset is registered with Session Manager. The handset name is shown on the display. When the handset fails to register the display shows **No Service**.



9. Conclusion

These Application Notes have described the administration steps required to use Yealink T-22 handsets with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Both basic and extended feature sets were covered in the interoperability testing.

10. Additional References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administering Avaya Aura® Communication Manager*, 9th august 2010, Document Number 03-300509.
- [2] *Avaya Extension to Cellular User Guide Avaya Aura® Communication Manager*, Nov 2009
- [3] *SIP Support in Avaya Aura® Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.
- [4] *Installing and configuring Avaya Aura® Session Manager*, 5th January 2011, Document Number 03-603473.
- [5] *Session Initiation Protocol Service Examples draft-ietf-sipping-service-examples-15*, Internet-Draft, 11th July 2008, available at <http://tools.ietf.org/html/draft-ietf-sipping-service-examples-15>

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