



## Avaya Solution & Interoperability Test Lab

---

# Application Notes for AEi Communications MT-752 SIP Telephone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

### Abstract

These Application Notes describe the steps required to integrate AEi Communications MT-752 SIP Telephones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. MT-752 SIP Telephones serve the hospitality industry and provide the following features: speakerphone, hold, redial, message waiting indicator (MWI), and programmable buttons. In the compliance test, MT-752 SIP Telephones successfully registered with Session Manager, established calls with the PSTN and other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya Aura® Communication Manager Feature Access Codes (FACs) and Feature Name Extensions (FNEs).

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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 integrate AEi Communications MT-752 SIP Telephones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. MT-752 SIP Telephones serve the hospitality industry and provide the following features: speakerphone, hold, redial, message waiting indicator (MWI), and programmable buttons. In the compliance test, MT-752 SIP Telephones successfully registered with Session Manager, established calls with the PSTN and other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya Aura® Communication Manager Feature Access Codes (FACs) and Feature Name Extensions (FNEs).

## 2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between AEi Communications MT-752 SIP telephones and Avaya SIP and H.323 telephones. Basic telephony features, such as hold, mute, call transfer, conference, and hospitality features including wake up calls and updating housekeeping status for a guest's room were also exercised. In addition, other extended telephony features, such as call forwarding and call pickup were also exercised using FNEs.

The serviceability testing focused on verifying that the AEi Communications MT-752 SIP telephone comes back into service after re-connecting the Ethernet connection or rebooting the SIP phone.

### 2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of MT-752 with Session Manager.
- Calls between MT-752 and Avaya SIP and H.323 telephones with Direct IP-IP Media (Shuffling) enabled and disabled.
- Support of multiple incoming and outgoing calls.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features including hold, mute, redial, blind and attended call transfers, and attended 3-party conferences.
- Long call duration and long hold duration.

- Extended telephony features using Communication Manager FNEs and FACs, such as Hospitality Wakeup calls, Housekeeping Status Access Codes, Call Forwarding and Call Pickup.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve voicemail messages.
- Use of programmable buttons on MT-752.
- Proper system recovery after a restart of the MT-752 and loss of IP connectivity.

## 2.2. Test Results

All test cases passed with the following observations noted:

- The MT-752 SIP telephone does not support blind conference. Attended conference is supported.
- The MT-752 SIP telephone does not support “Pause” in the speed dial buttons.
- The MT-752 SIP telephone does not support # or \* in the dial plan.

## 2.3. Support

For technical support on the AEi Communications MT-752 SIP Telephone, contact AEi Communications Support via phone, email, or website.

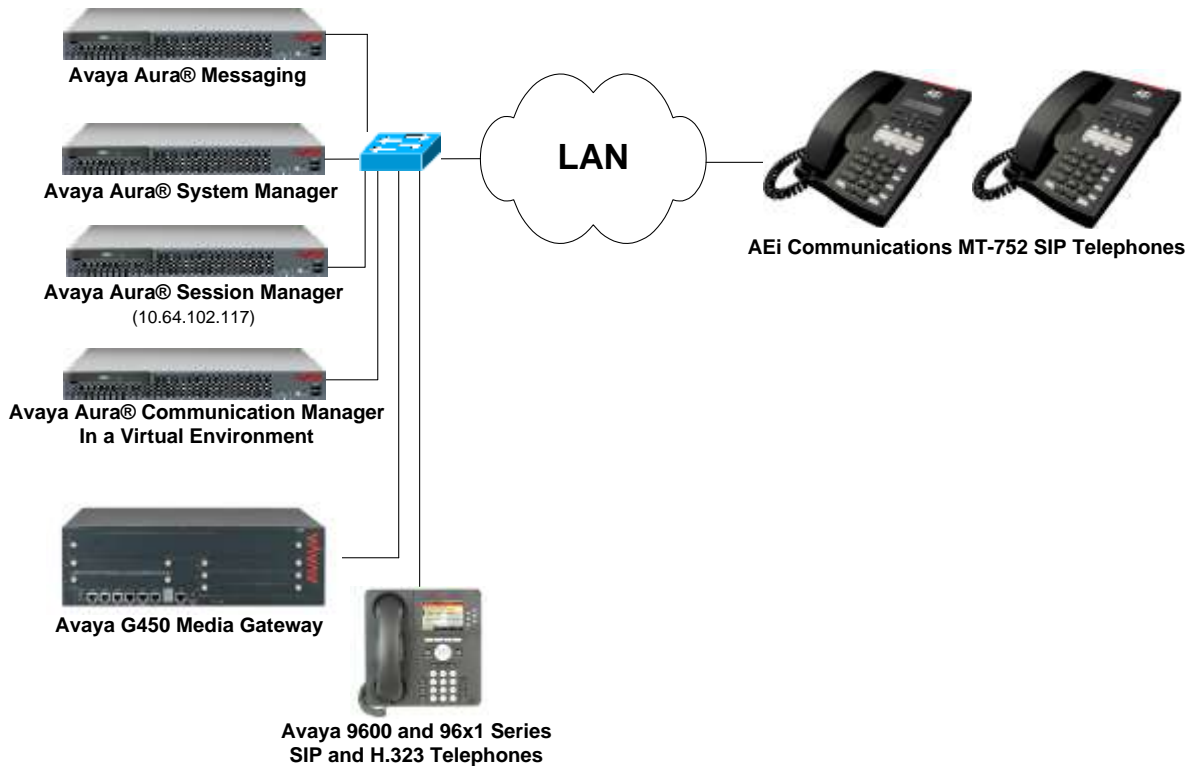
- **Phone:** 1 (650) 552-9416
- **Email:** [sales@aeicomunications.com](mailto:sales@aeicomunications.com)
- **Web:** <http://www.aeicomunications.com/contact.html>

### 3. Reference Configuration

**Figure 1** illustrates a sample configuration with an Avaya SIP-based network that includes the following Avaya products:

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Media Gateway and Avaya Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Aura® Messaging served as the voicemail system.
- Avaya 9600 Series SIP and H.323 Telephones.
- AEi Communications MT-752 SIP Telephones.

AEi Communications MT-752 SIP Telephones registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.



**Figure 1: Avaya SIP Network with AEi Communications MT-752 SIP Telephones**

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager in a Virtual Environment with an Avaya G450 Media Gateway	7.0.0.2 SP 2 (R017x.00.0.441.0 with Patch 22684)
Avaya Aura® Media-Server	7.7.0.226
Avaya Aura® System Manager	7.0 (Build No. 7.0.0.016266-7.0.9.912 Software Update Revision No: 7.0.0.0.3929)
Avaya Aura® Session Manager	7.0.0.0.700007
Avaya Aura® Messaging	6.3.2 SP 2 Patch 3
Avaya 9600 Series IP Deskphones	S3.250A (H.323)
Avaya 96x1 Series IP Deskphones	7.0.0.39 (SIP)
AEi Communications MT-752 SIP Telephone	13751.16.1.01d

## 5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify Communication Manager license
- Administer IP Network Region and IP Codec Set
- Administer Hospitality Features

Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

**Note:** It is assumed that basic configuration of the Communication Manager has already been completed, such as the SIP trunk to Session Manager. The SIP station configuration for AEi Communications MT-752 is configured through Avaya Aura® System Manager in **Section 6.2**.

### 5.1. Verify License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                               Page 1 of 12
                                OPTIONAL FEATURES

G3 Version: V17                                                         Software Package: Enterprise
Location: 2                                                             System ID (SID): 1
Platform: 28                                                            Module ID (MID): 1

                                USED
                                Platform Maximum Ports: 6400 24
                                Maximum Stations: 2400 9
                                Maximum XMOBILE Stations: 2400 0
Maximum Off-PBX Telephones - EC500: 9600 0
Maximum Off-PBX Telephones - OPS: 9600 6
Maximum Off-PBX Telephones - PBFMC: 9600 0
Maximum Off-PBX Telephones - PVFMC: 9600 0
Maximum Off-PBX Telephones - SCCAN: 0 0
                                Maximum Survivable Processors: 313 0

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

## 5.2. Administer IP Network Region and IP Codec Set

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, **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 G450 Media Gateway or Avaya Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

```

change ip-network-region 1                                     Page 1 of 20
                                                           IP NETWORK REGION
    Region: 1
Location: 1          Authoritative Domain: avaya.com
    Name:                               Stub Network Region: n
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
    Call Control PHB Value: 46
        Audio PHB Value: 46
        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 type supported for calls routed over the SIP trunk to the MT-752 SIP telephone. 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. The default settings of the **IP Codec Set** form are shown below. The MT-752 SIP telephone supports G.711 and G.729 codecs.

```

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:

```

### 5.3. Administer Hospitality Features

This section covers the configuration of two hospitality features: wakeup calls and housekeeping status. A hotel guest may enter the wake up feature access code (FAC) followed by the time for the wakeup call in *hhmm* format, where *hh* is the hour and *mm* is the minute. The housekeeping status of a hotel room may be changed by dialing the housekeeping status access code from the hotel room phone.

#### 5.3.1. Administer Feature Name Extensions (FNEs)

Prior to dialing the wakeup call, the SIP user must first receive dial tone from Communication Manager. This is achieved by first dialing the **Idle Appearance Select FNE** configured as shown below. Afterwards, the wakeup call access code may be dialed. The housekeeping status access codes may be dialed directly without dialing the **Idle Appearance Select FNE**.

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

  Exclusion (Toggle On/Off): 78117
  Extended Group Call Pickup:
  Held Appearance Select: 78118
  Idle Appearance Select: 78119
  Last Number Dialed: 78120
  Malicious Call Trace:
  Malicious Call Trace Cancel:
  Off-Pbx Call Enable:
  Off-Pbx Call Disable:
```

#### 5.3.2. Administer Features Access Codes (FACs)

In the **Feature Access Code (FAC)** form configure the **Automatic Wakeup Call Access Code** and the **Housekeeping Status (Client Room) Access Codes**, as needed, as shown below. The FACs should comply with the dial plan.

```
change feature-access-codes          Page 8 of 10
FEATURE ACCESS CODE (FAC)
  Hospitality Features

    Automatic Wakeup Call Access Code: *70
    Housekeeping Status (Client Room) Access Code: *71
    Housekeeping Status (Client Room) Access Code: *72
    Housekeeping Status (Client Room) Access Code: *73
    Housekeeping Status (Client Room) Access Code: *74
    Housekeeping Status (Client Room) Access Code: *75
    Housekeeping Status (Client Room) Access Code: *76
    Housekeeping Status (Station) Access Code: *77
    Housekeeping Status (Station) Access Code: *78
    Housekeeping Status (Station) Access Code: *79
    Housekeeping Status (Station) Access Code: *80
    Verify Wakeup Announcement Access Code: *81
    Voice Do Not Disturb Access Code:
```



### 5.3.3. Allow Wake-up Calls

In the **Hospitality** form, enable **Room Activated Wakeup With Tones**. Communication Manager will prompt the user with tones when enabling a wakeup call. For example, a 3-burst confirmation tone will be played to prompt the user to enter the wakeup time.

```

change system-parameters hospitality                               Page 2 of 3
                                HOSPITALITY

Dual Wakeups? n      Daily Wakeup? n      VIP Wakeup? n

                                Room Activated Wakeup With Tones? y
Time of Scheduled Wakeup Activity Report:
Time of Scheduled Wakeup Summary Report:
Time of Scheduled Emergency Access Summary Report:
                                Announcement Type: silence

Length of Time to Remain Connected to Announcement: 30
Extension to Receive Failed Wakeup LWC Messages:
Routing Extension on Unavailable Voice Synthesis:
Display Room Information in Call Display? n
Automatic Selection of DID Numbers? n
Custom Selection of VIP DID Numbers? n
                                Number of Digits from PMS:
                                PMS Sends Prefix? n
Number of Digits in PMS Coverage Path: 3
                                Digit to Insert/Delete:
  
```

### 5.3.4. Allow Housekeeping Status Updates

To allow housekeeping to change the housekeeping status of a guests room by dialing the appropriate access code, **Client Room** must be enabled on the COS assigned to the SIP phone. In this example, **Client Room** was enabled for COS 1, which was assigned to the AEi MT-752 phone.

```

change cos-group 1                                             Page 1 of 2
CLASS OF SERVICE          COS Group: 1      COS Name:

                                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  n  n  y  y  n  n  y  y  n  n  y
Data Privacy              n  y  n  n  n  y  y  y  n  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  n  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  y n  n  n  n  n  n  n  n  n  n  n  n  n  n
Restrict Call Fwd-Off Net  y  y  y  y  y  y  y  y  y  y  y  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) n  n  n  n  n  n  n  n  n  n  n  n  n  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 n  n  n  n  n  n  n  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
  
```

## 6. Configure Avaya Aura® Session Manager

This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Administer SIP User

**Note:** It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for the MT-752 SIP telephone.

### 6.1. Launch System Manager

Access the System Manager Web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of the System Manager server. Log in using the appropriate credentials.

AVAYA  
Aura System Manager 7.0

Recommended access to System Manager is via FQDN.

[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

User ID:

Password:

[Change Password](#)

**Supported Browsers:** Internet Explorer 9.x, 10.x or 11.x or Firefox 36.0, 37.0 and 38.0.

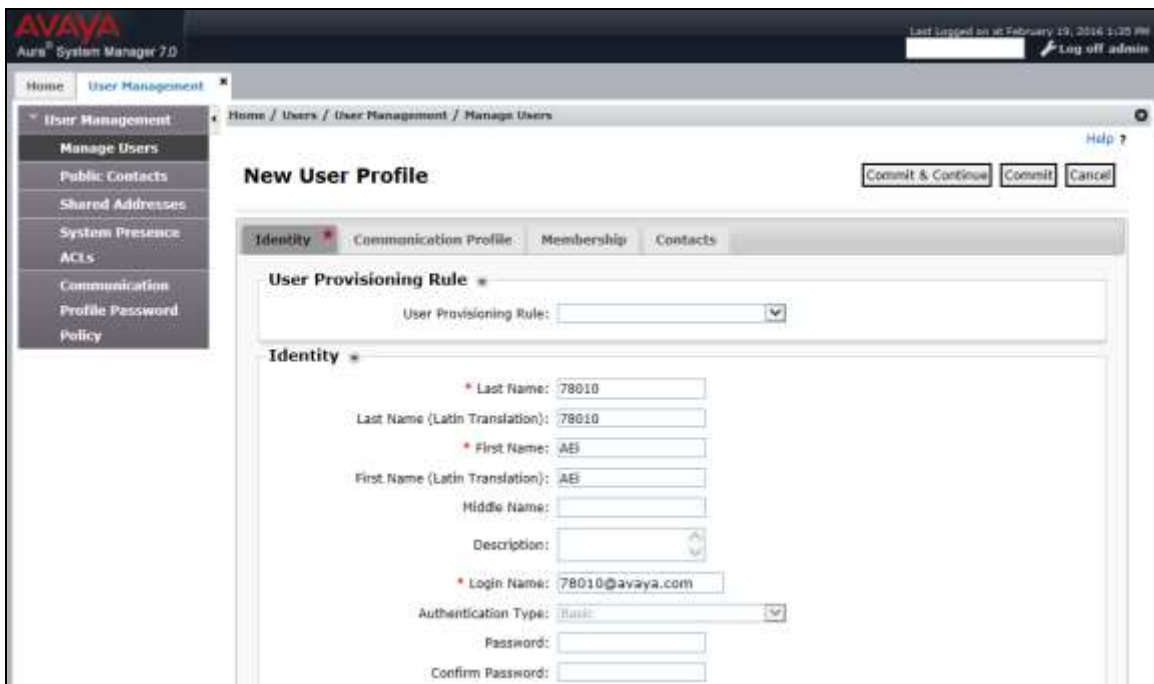
## 6.2. Administer SIP User

In the subsequent screen (not shown), select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **New** to add a user.



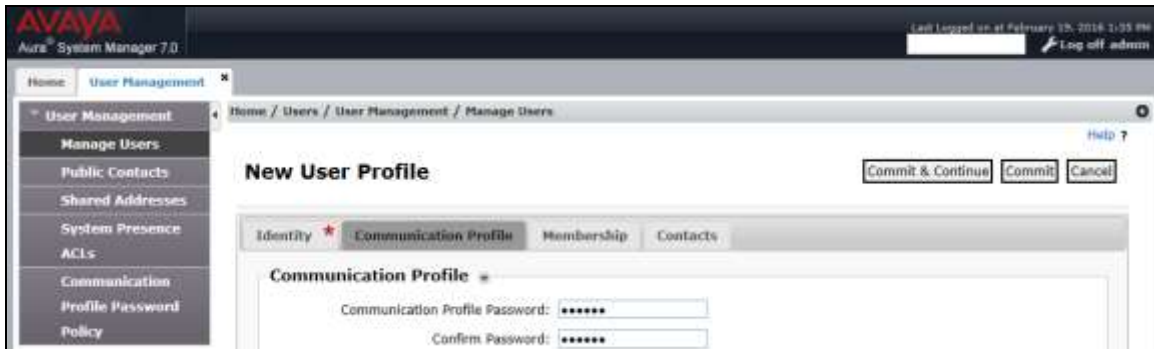
### 6.2.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter “<ext>@<domain>”, where “<ext>” is the desired MT-752 SIP extension and “<domain>” is the applicable SIP domain name from **Section 5.2**. For **Password** and **Confirm Password**, enter the appropriate credentials for System Manager. Retain the default values in the remaining fields.



## 6.2.2. Communication Profile

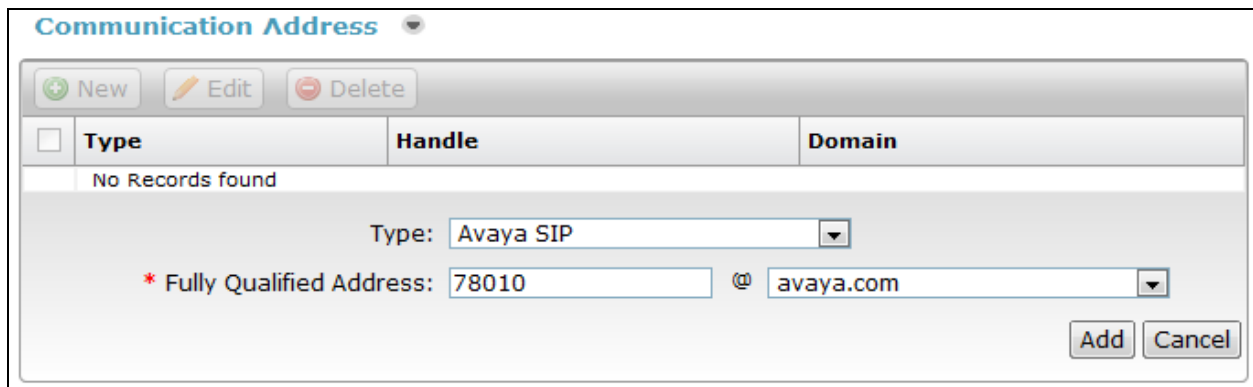
Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration.



The screenshot shows the 'New User Profile' form in the Avaya Aura System Manager 7.0 interface. The 'Communication Profile' tab is active, and the 'Communication Profile Password' and 'Confirm Password' fields are visible, both containing masked characters. The 'Commit & Continue', 'Commit', and 'Cancel' buttons are also present.

## 6.2.3. Communication Address

In the **Communication Address** sub-section, click **New** to add a new entry. The **Communication Address** sub-section is updated with additional fields as shown below. For **Type**, retain “Avaya SIP”. For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.2.1**. Click **Add**.



The screenshot shows the 'Communication Address' form. The 'Type' dropdown is set to 'Avaya SIP'. The 'Fully Qualified Address' field is populated with '78010' and '@ avaya.com'. The 'Add' and 'Cancel' buttons are visible.

Type	Handle	Domain
No Records found		

Type: Avaya SIP

\* Fully Qualified Address: 78010 @ avaya.com

Add Cancel

## 6.2.4. Session Manager Profile

Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

**Session Manager Profile** ▾

**SIP Registration**

\* Primary Session Manager 

Primary	Secondary	Maximum
7	0	7

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices  ▾

Block New Registration When Maximum Registrations Active?

**Application Sequences**

Origination Sequence  ▾

Termination Sequence  ▾

**Call Routing Settings**

\* Home Location  ▾

Conference Factory Set  ▾

**Call History Settings**

Enable Centralized Call History?

## 6.2.5. CM Endpoint Profile

Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.2.1**. For **Template**, select *9600SIP\_DEFAULT\_CM\_7\_0*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click **Commit** to save the configuration (not shown).

**CM Endpoint Profile** ▾

\* System  ▾

\* Profile Type  ▾

Use Existing Endpoints

\* Extension  **Endpoint Editor**

\* Template  ▾

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle  ▾

Calculate Route Pattern

Sip Trunk

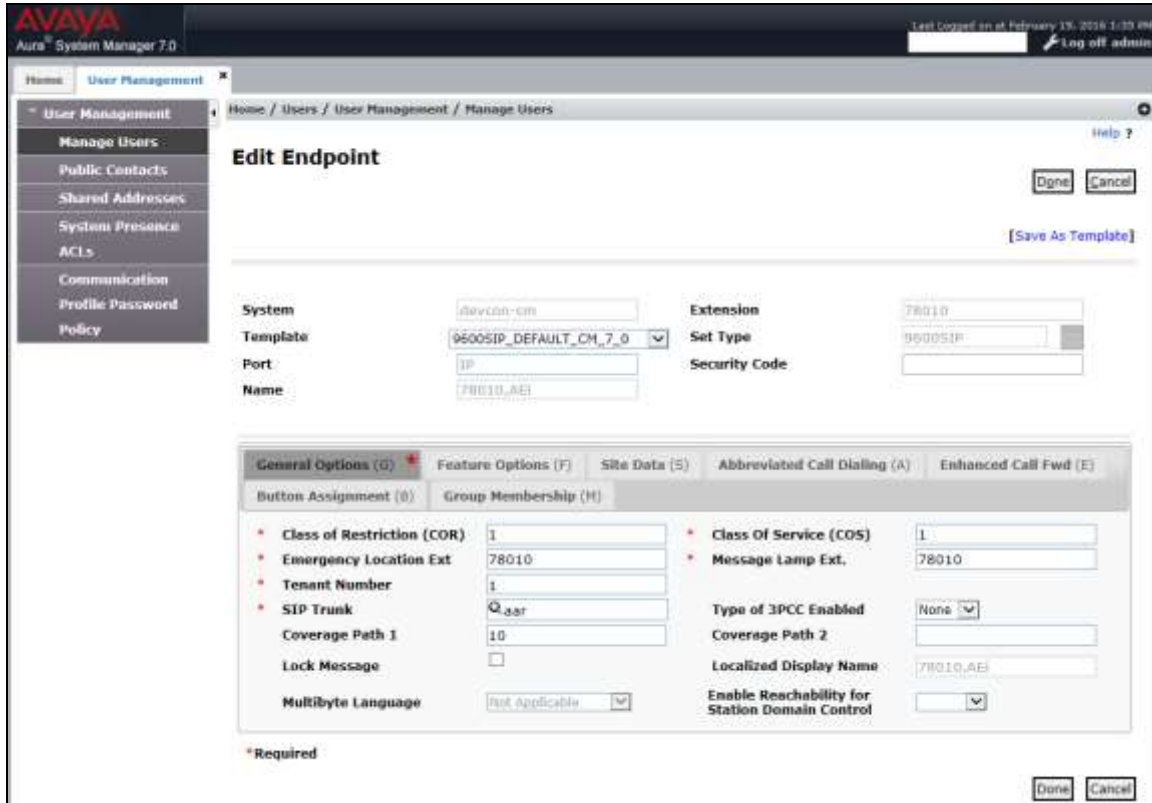
Enhanced Callr-Info display for 1-line phones

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

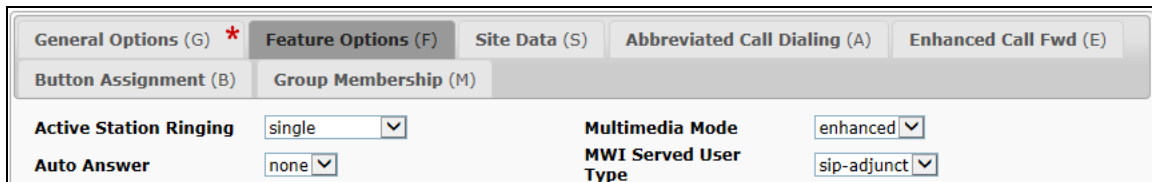
Override Endpoint Name and Localized Name

Allow H.323 and SIP Endpoint Dual Registration

In the **CM Endpoint Profile** sub-section, click the **Endpoint Editor** button to display the page below. In the **General Options** tab, specify that coverage path that points to the voicemail system in the **Coverage Path 1** field. This provides voicemail coverage for the SIP user. In this example, coverage path 10 was used.



In the **Feature Options** tab, set the **MWI Served User Type** field to *sip-adjunct*. This allows MWI to be enabled for the SIP user. The voicemail system was connected via SIP to Session Manager.



## 7. Configure AEi Communications MT-752 SIP Telephone

Access the MT-752 web interface by using the URL “https://ip-address:8000” in an Internet browser window, where “ip-address” is the IP address of the SIP phone. Log in using the appropriate credentials and then click **Login**.



### 7.1. Administer LAN Port Settings

Select **Network** → **LAN Port Settings** in the left pane and configure the SIP phone’s network settings as shown below. In this example, a static IP is configured. Click **Submit**.

**Note:** Initially, the SIP phone’s default IP address was used, as seen via its local menu display, to communicate with the phone.



LAN Port Setting	
IP Type :	<input checked="" type="radio"/> Static IP <input type="radio"/> DHCP Client <input type="radio"/> PPPoE
IP Address :	192.168.100.190
Netmask :	255.255.255.0
Gateway :	192.168.100.1
Primary DNS :	192.168.1.1
Secondary DNS :	0.0.0.0
Mac Address :	00:0e:43:d1:62:59



## 7.2. Administer SIP Accounts

Navigate to **SIP Accounts** in the left pane and click on the # of an available entry to add a SIP account.



**Web Configuration**

### SIP Accounts

You could set information of service domians in this page.

SIP Accounts					
#	Display Name	Registration Server	Status	Registration	Select
<a href="#">1</a>	78010	10.64.102.117	registered	Enable	<input type="checkbox"/>
<a href="#">2</a>					<input type="checkbox"/>
<a href="#">3</a>					<input type="checkbox"/>
<a href="#">4</a>					<input type="checkbox"/>
<a href="#">5</a>					<input type="checkbox"/>
<a href="#">6</a>					<input type="checkbox"/>
<a href="#">7</a>					<input type="checkbox"/>
<a href="#">8</a>					<input type="checkbox"/>
<a href="#">9</a>					<input type="checkbox"/>
<a href="#">10</a>					<input type="checkbox"/>

Navigate to the **SIP Proxy** webpage as shown below. Under the **Basic SIP Proxy Settings** section, configure the following parameters.

- **Transport:** Set to *UDP*.
- **Registration:** Select *Enable*.
- **Send Register:** Select *Enable*.
- **Registration ID:** Specify the Registration ID (e.g., *78010*, the SIP extension).
- **Display Name:** Specify the Display Name (e.g., *78010*, the SIP extension).
- **Authentication Name:** Specify the SIP extension of the MT-752 SIP telephone (e.g., *78010*).
- **Password:** Specify the SIP password configured in **Section 6.2.2**.

- **Registration Server:** Set to the Session Manager IP address (e.g., *10.64.102.117*).
- **Proxy Server:** Set to the Session Manager IP address (e.g., *10.64.102.117*).
- **Proxy Address:** Set to the Session Manager IP address (e.g., *10.64.102.117*).
- **Voice Mail:** Specify the voicemail pilot number (e.g., *25000*).
- **Expire Time:** Specify the frequency of SIP re-registrations (e.g., *300*).
- **DTMF Type:** Select *RFC2833*.
- **MWI:** Set to *Enable*.

**Web Configuration**

**SIP Account Settings**

You could set information of service domains in this

SIP Account 1	
Transport :	UDP ▾
Registration :	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Send Register :	Enable ▾
Registration ID :	78010
Display Name :	78010
Authentication Name :	78010
Password :	••••••
Registration Server :	10.64.102.117
Proxy Server :	10.64.102.117
Proxy Address :	10.64.102.117
Voice Mail :	25000
Expire Time :	300 ▾
DTMF Type :	RFC2833 ▾
Send KeepAlive :	<input type="radio"/> On <input checked="" type="radio"/> Off
MWI :	Enable ▾
RFC2543 Call Hold Support :	Disable ▾
Status: <span style="color: red;">registered</span>	
<input type="button" value="Submit"/> <input type="button" value="cancel"/>	

### 7.3. Administer Global SIP Settings

Navigate to **Global SIP Settings** → **Port Settings** and verify the SIP Listen Port being used (e.g., 5060).

Port Settings	
SIP Listen Port:	5060 (10~65533)
RTP Port Base:	10000 (10~65533)

Submit Reset

Navigate to **Global SIP Settings** → **Codec Settings** to verify the codec priority. In this example, the codec priority is G.711 followed by G.729.

Codec Priority	
First Priority :	G.711 u-law
Second Priority :	G.711 a-law
Third Priority :	G.729
Fourth Priority :	G.723.1

RTP Packet Length	
G.711 Frame Size :	20 ms
G.723 Frame Size :	60 ms
G.729 Frame Size :	20 ms

Submit

Navigate to **Global SIP Settings** → **Other Settings**. The max-SE field specifies the session refresh interval. This field should not be less than 180.

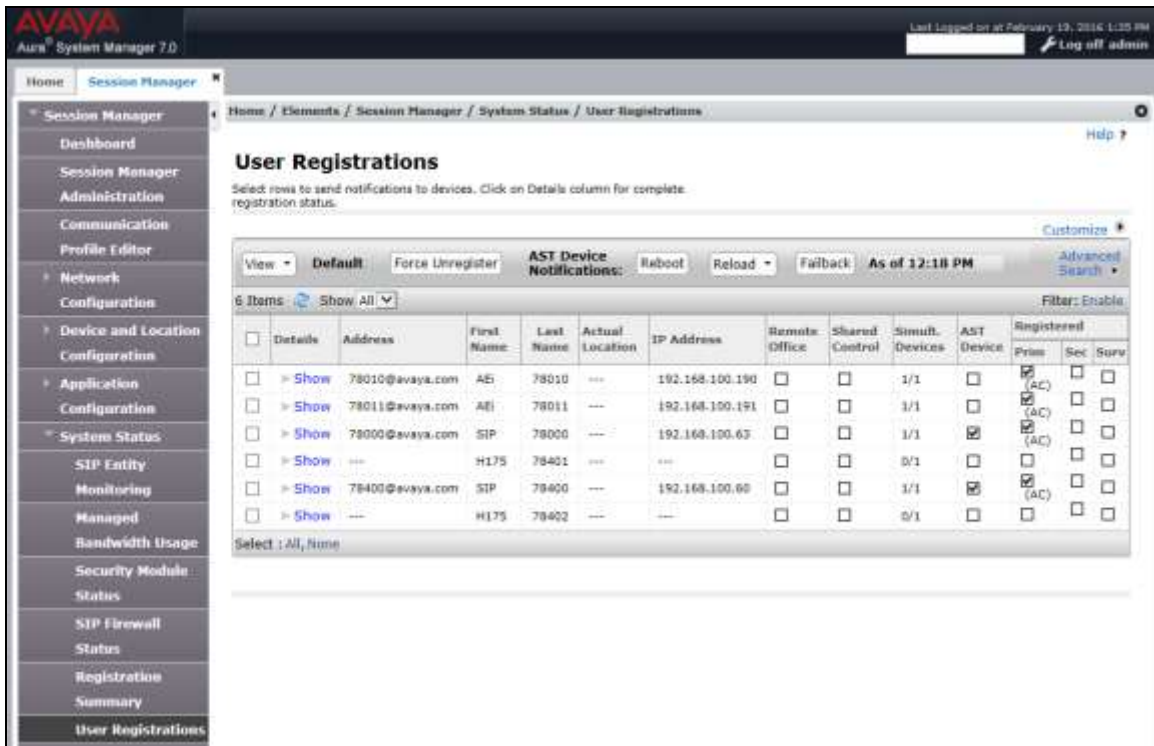
Navigate to the **Phone Settings** → **Dial Plan** to specify the dial plan. This allows the MT-752 SIP telephone to dial a number as soon as it recognizes a numbering format from the dial plan. Otherwise, the inter-digit timer has to expire or the user need to dial # to indicate end of digits.

No.	Switch	Rule	Account	Select
<a href="#">1</a>	Enable	7xxxx	78010	<input type="checkbox"/>
<a href="#">2</a>	Enable	91xxxxxxxxxxx	78010	<input type="checkbox"/>
<a href="#">3</a>	Enable	25000	78010	<input type="checkbox"/>

## 8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the AEi Communications MT-752 SIP Telephone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the MT-752 SIP Telephone has successfully registered with Session Manager. In System Manager, navigate to **Elements** → **Session Manager** → **System Status** → **User Registrations** to check the registration status. The SIP registration status can also be seen in the SIP Account page of the MT-752 web interface seen in **Section 7.2**.



The screenshot shows the Avaya Aura System Manager web interface. The main content area is titled "User Registrations" and displays a table of registered users. The table has the following columns: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, Registered (Prim, Sec, Surv). The table shows 6 items, all of which are registered.

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered
<input type="checkbox"/> > Show	78010@avaya.com	AEI	78010	---	192.168.100.190	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/> > Show	78011@avaya.com	AEI	78011	---	192.168.100.191	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/> > Show	78000@avaya.com	SIP	78000	---	192.168.100.63	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/> > Show	---	H175	78401	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/> > Show	78400@avaya.com	SIP	78400	---	192.168.100.60	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/> > Show	---	H175	78402	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>

2. Verify basic telephony features by establishing calls between a MT-752 SIP Telephone with and another phone.

## 9. Conclusion

These Application Notes have described the administration steps required to integrate the AEi Communications MT-752 SIP Telephone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The AEi Communications MT-752 SIP Telephone successfully registered with Session Manager and basic telephony and hospitality features were verified. All test cases passed with observations noted in **Section 2.2**.

## 10. 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*, Release 7.0, Issue 1, August 2015, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, Release 7.0, Issue 1, August 2015.

---

**©2016 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](mailto:devconnect@avaya.com).