Avaya Solution & Interoperability Test Lab

Application Notes for the UniData Communication Systems WPU-7700 Wireless IP Phones with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager, Avaya SIP Enablement Services, and UniData Communication Systems WPU-7700 Wireless IP Phones. During compliance testing, UniData WPU-7700 Wireless IP Phones successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features such as conference, transfer, and hold.

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 a solution comprised of Avaya Communication Manager 5.1.1, Avaya SIP Enablement Services (SES) 5.1.1, and UniData Communication Systems WPU-7700 Wireless IP Phones. Avaya Communication Manager and Avaya SIP Enablement Services has the capability to extend advanced telephony features to SIP stations. These features can be extended to non-Avaya SIP telephones such as the UniData WPU-7700 Wireless IP Phones.

Figure 1 illustrates a sample configuration consisting of an Avaya Communication Manager running on an Avaya S8300C Server with the Avaya G250 Media Gateway, the Avaya SIP Enablement Services (SES) that is co-resident on the S8300C, and the UniData WPU-7700 Wireless IP Phones. For completeness, an Avaya 4610SW SIP IP Telephone, an Avaya 9630 H.323 IP Telephone and an Avaya 6221 Analog Telephone were included to demonstrate calls between the SIP-based UniData telephones and Avaya SIP, H.323 and Analog telephones. The Fast Ethernet ports on the G250 Media Gateway provide LAN connectivity and power to the Avaya IP Telephones through Power-over-Ethernet (PoE). Wireless LAN connectivity for the UniData telephones is provided by the Corega Wireless Access Point. Avaya IA 770 INTUITY AUDIX Messaging (IA 770) is used to support voice messaging. An audio wav file is used as the music-on hold (MOH) through the virtual Voice Announcement with LAN (VAL) feature in the G250 Media Gateway. The ISDN-BRI trunk is also included to demonstrate calls routed by Avaya Communication Manager between the UniData telephones and the PSTN.

The UniData telephone originates a call by sending a call request (SIP INVITE message) to the Avaya SES. The Avaya SES routes the call over a SIP trunk to Avaya Communication Manager for origination services. If the call is destined for another local SIP telephone, then Avaya Communication Manager routes the call back over the SIP trunk to Avaya SES for delivery to the destination SIP telephone. Otherwise, Avaya Communication Manager routes the call to the PSTN, a local Avaya H.323, digital, or analog telephone, as appropriate depending on the destination number.

For a call arriving at Avaya Communication Manager that is destined for the UniData telephone, Avaya Communication Manager routes the call over the SIP trunk to the Avaya SES for delivery to the UniData telephone.

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 [1] thru [4].

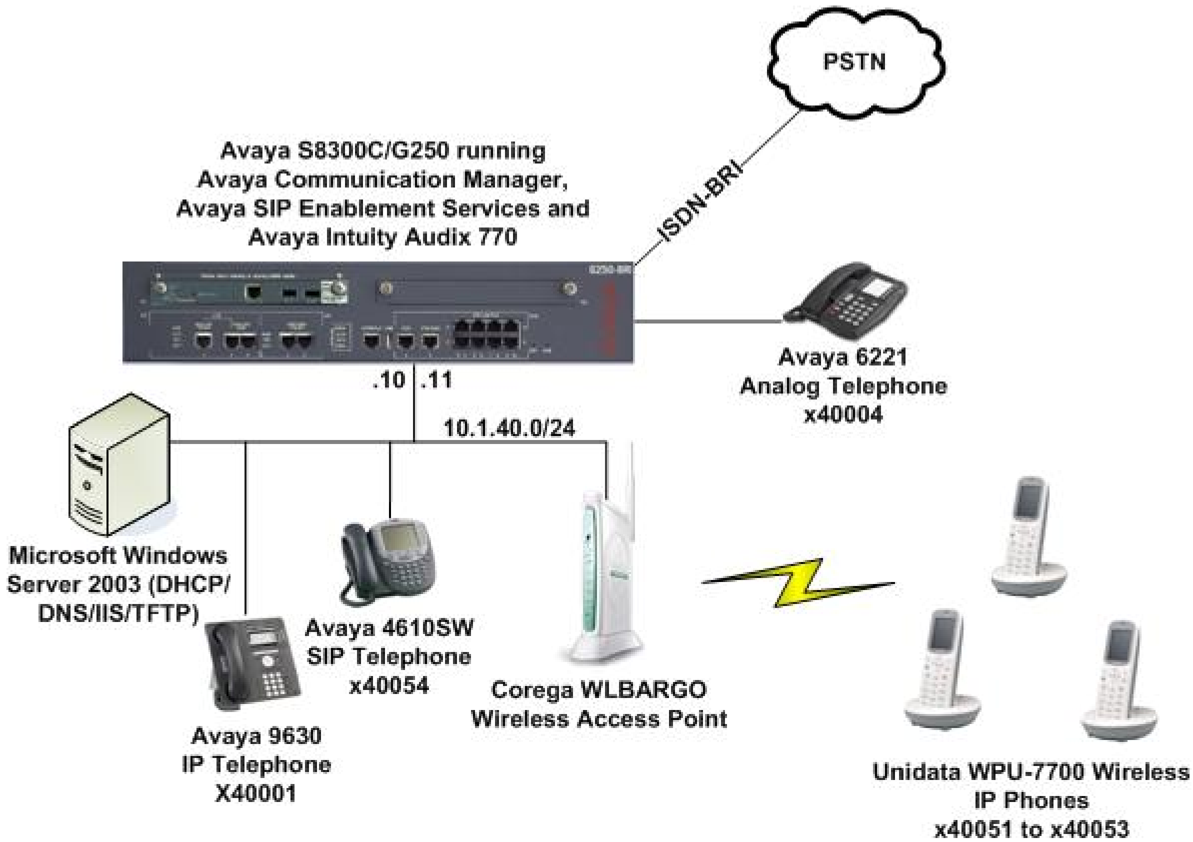


Figure 1: Sample Configuration

2. Equipment and Software Validated

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

Equipment	Software / Firmware
Avaya S8300C Server	Avaya Communication Manager, Avaya SIP Enablement Services and Avaya IA 770 5.1.1 (Service Pack 01.1.415.1-16402)
Avaya G250 Media Gateway	28.19.0
Avaya 4600 Series IP Telephones - 4610SW	2.2.2 (SIP)
Avaya 9600 Series IP Telephones - 9630	2.0 (H.323)
Avaya 6221 Analog Telephone	-
UniData WPU-7700 Wireless IP Phones	V2.1.1-a0
Corega WLBARGO Wireless Access Point	1.0.9

3. Configure Avaya Communication Manager

This section describes a procedure for setting up a SIP trunk between Avaya Communication Manager and Avaya SES which includes steps for setting up a list of IP codecs, an IP network region, a signaling group and its interface. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. Also, a procedure is described here to configure SIP telephones in Avaya Communication Manager. Configuration in the following sections is only for the fields where a value needs to be entered or modified. Default values are used for all other fields.

These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. UniData and other SIP telephones are configured as Off-PBX Stations (OPS) in Avaya Communication Manager. Avaya Communication Manager does not directly control an OPS endpoint, but its features and calling privileges can be applied to it by associating a local, on-PBX telephone with the OPS endpoint. Similarly, a SIP telephone in Avaya SES is associated with an on-PBX telephone on Avaya Communication Manager. SIP Telephones register with the Avaya SES and use Avaya Communication Manager for call origination and termination services.

3.1. Capacity Verification

Step	Description
1.	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: V15                           Software Package: Standard
       Location: 2                          RFA System ID (SID): 1
       Platform: 18                         RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 2850  22
                                      Maximum Stations: 14    9
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 2400  0
                    Maximum Off-PBX Telephones -   OPS: 2400  5
                    Maximum Off-PBX Telephones - PBFMC: 2400  0
                    Maximum Off-PBX Telephones - PVFMC: 2400  0
                    Maximum Off-PBX Telephones - SCCAN: 2400  0
```

2.	Proceed to **Page 2** of **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. **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.

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

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 10     3
           Maximum Concurrently Registered IP Stations: 12     1
             Maximum Administered Remote Office Trunks: 0      0
Maximum Concurrently Registered Remote Office Stations: 0      0
              Maximum Concurrently Registered IP eCons: 2      0
  Max Concur Registered Unauthenticated H.323 Stations: 0      0
                  Maximum Video Capable H.323 Stations: 10     0
                   Maximum Video Capable IP Softphones: 10     0
                       Maximum Administered SIP Trunks: 10     8
  Maximum Administered Ad-hoc Video Conferencing Ports: 0      0
   Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                             Maximum TN2501 VAL Boards: 10     0
                     Maximum Media Gateway VAL Sources: 1      1
           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 an IP 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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>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 an 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 and Media Encryption was set to none.</td></tr>
<tr><td><pre>
change ip-codec-set 2   Page   1 of   2

                          IP Codec Set

    Codec Set: 2

    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:
</pre></td></tr>
</table>

3.3. 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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>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 d.com in this example. This should match the SIP Domain value in Section 4 Step 2.
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.
Call Control PHB Value – Note the value to be configured on the UniData telephone in Section 5 Step 8.
Audio PHB Value – Note the value to be configured on the UniData telephone in Section 5 Step 8.
</td></tr>
<tr><td><pre>
change ip-network-region 2   Page   1 of 19
                               IP NETWORK REGION
  Region: 2
Location: Authoritative Domain: d.com
    Name: Local
MEDIA PARAMETERS   Intra-region IP-IP Direct Audio: yes
      Codec Set: 2   Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048   IP Audio Hairpinning? n
   UDP Port Max: 65535
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
</pre></td></tr>
</table>

2.	Proceed to **Page 3** of IP network region configuration and enable inter-region connectivity between regions as per below. For this compliance testing, codec set was set to the IP codec set configured in **Section 3.2**.

```
change ip-network-region 2                                      Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video      Intervening      Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC IGAR AGL
 2   1    2     y    NoLimit                                             n
 2   2    2                                                                  all
 2   3
 2   4
 2   5
 2   6
 2   7
 2   8
 2   9
 2   10
 2   11
 2   12
 2   13
 2   14
 2   15
```

3.4. 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.

Step	Description
1.	Enter the command **add signaling-group s**, where **s** is an available signaling group and configure the following: • **Group Type** – Set to **sip.** • **Transport Method** – Set to **tls**. • **Co-Resident SES** – Set to **y** to connect to the co-resident SES. • **Near-end Node Name** - Set to **procr**. • **Near-end Listen Port** - Set to **6001** for co-resident Avaya SES. • **Far-end Node Name** - Set to **procr** for co-resident Avaya SES. • **Far-end Network Region** - Set to the region configured in **Section 3.3**. • **Far-end Domain** - Set to **d.com** in this example. This should match the **SIP Domain** value in **Section 4 Step 2**.

```
add signaling-group 50                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 50             Group Type: sip
                        Transport Method: tls
                         Co-Resident SES? y




   Near-end Node Name: procr                    Far-end Node Name: procr
 Near-end Listen Port: 6001                   Far-end Listen Port: 5061
                                           Far-end Network Region: 2
       Far-end Domain: d.com

                                             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              Alternate Route Timer(sec): 6
```

3.5. SIP Trunking

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Issue the command add trunk-group t, where t is an unallocated trunk group and configure the following:

Group Type – Set to the Group Type field value configured in Section 3.4.
TAC (Trunk Access Code) – Set to any available trunk access code.
Signaling Group – Set to the Group Number field value configured in Section 3.4.
Number of Members – Allowed values are between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.
Group Name – Enter any descriptive name.

Note: Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunk members for the duration of the call. The license file installed on the system controls the maximum permitted.</td></tr>
<tr><td></td><td><pre>add trunk-group 50   Page   1 of 21
                                TRUNK GROUP

Group Number: 50   Group Type: sip   CDR Reports: n
  Group Name: SIP Local   COR: 1   TN: 1   TAC: 750
   Direction: two-way   Outgoing Display? n
 Dial Access? n   Night Service:
Queue Length: 0
Service Type: tie   Auth Code? n

                                                       Signaling Group: 50
                                                     Number of Members: 8</pre></td></tr>
<tr><td>2.</td><td>Proceed to Page 4 and set Telephone Event Payload Type to 101 to match the default value used by the UniData telephone. Leaving this value blank is also acceptable as the Avaya Communication Manager and the UniData telephone will negotiate the payload type.</td></tr>
<tr><td></td><td><pre>add trunk-group 50   Page   4 of 21
                              PROTOCOL VARIATIONS

                        Mark Users as Phone? n
              Prepend '+' to Calling Number? n
        Send Transferring Party Information? n

             Telephone Event Payload Type: 101</pre></td></tr>
</table>

3.6. SIP Stations

This section describes the steps for administering OPS stations in Avaya Communication Manager and associating the OPS station extensions with the telephone numbers of the UniData telephones.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter the add station s command, where s is an available extension in the dial plan, to administer an OPS station. On Page 1 of the station form configure the following fields:

Type – Set to 6408D+.
Port – Set to X.
Name – Enter any descriptive name.

<pre>
add station 40051   Page   1 of   5
                                     STATION

Extension: 40051   Lock Messages? n   BCC: 0
     Type: 6408D+   Security Code: TN: 1
     Port: X   Coverage Path 1:   COR: 1
     Name: Al   Coverage Path 2:   COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 2   Personalized Ringing Pattern: 1
             Data Module? n   Message Lamp Ext: 40051
            Speakerphone: 2-way   Mute Button Enabled? y
        Display Language: english

           Survivable COR: internal   Media Complex Ext:
   Survivable Trunk Dest? y   IP SoftPhone? n
</pre></td></tr>
<tr><td>2.</td><td>Proceed to Page 4 of the STATION form and add the required number of call-appr entries in the BUTTON ASSIGNMENTS section. The number of call appearances should match the Call Limit field value in Step 4. Configure additional feature buttons such as no-hld-cnf (required for Conference on Answer) and auto-cback (required for Automatic Call Back) as required.
<pre>
add station 20051   Page   4 of   5
                                     STATION
SITE DATA
      Room: Headset? n
      Jack: Speaker? n
     Cable: Mounting: d
     Floor: Cord Length: 0
  Building: Set Color:

ABBREVIATED DIALING
     LIST1:   List2:   List3:

BUTTON ASSIGNMENTS
 1: call-appr   5: no-hld-cnf
 2: call-appr   6: auto-cback
 3:   7:
 4:   8:
</pre></td></tr>
</table>

3.	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 the UniData telephone 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 3.5**.

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

 Station          Application Dial   CC  Phone Number      Trunk       Config
 Extension                    Prefix                       Selection   Set
 40051              OPS           -     40051               50          1
```

4.	Proceed to **Page 2** of station mapping form and verify that the **Call Limit** field value matches the number of call appearances configured in **Step 2**.

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

 Station        Call       Mapping      Calls       Bridged         Location
 Extension      Limit      Mode         Allowed     Calls
 40051          2        both           all         both
```

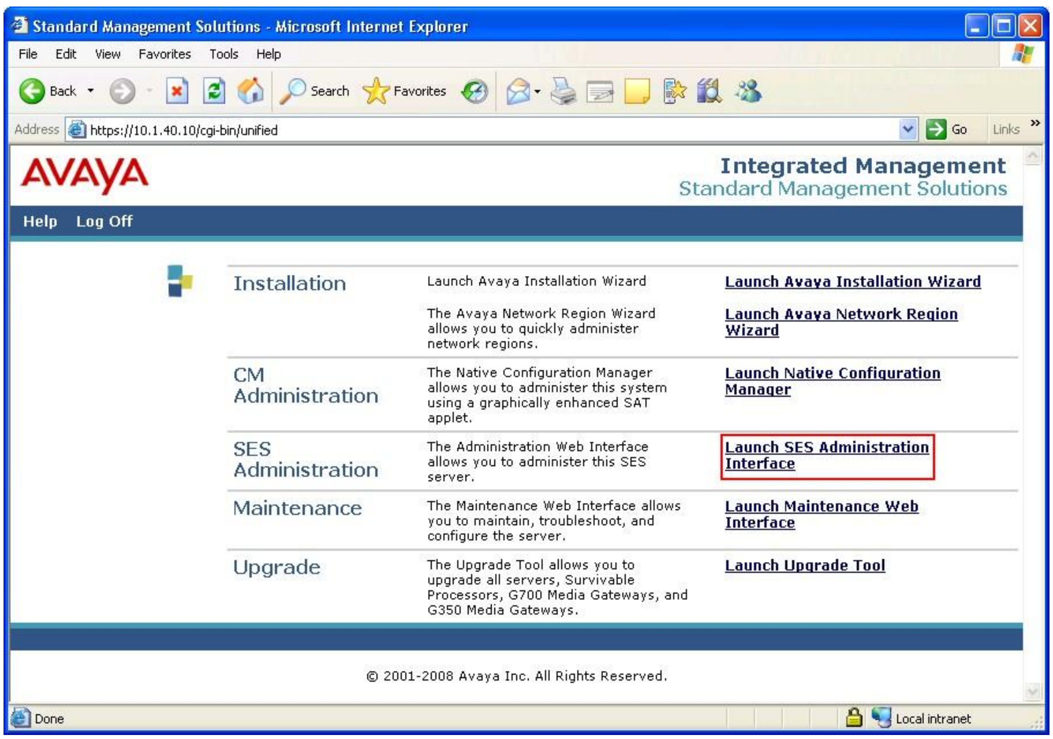
5.	Repeat **Steps 1 - 4** as necessary to administer additional OPS stations and associations for UniData telephones.

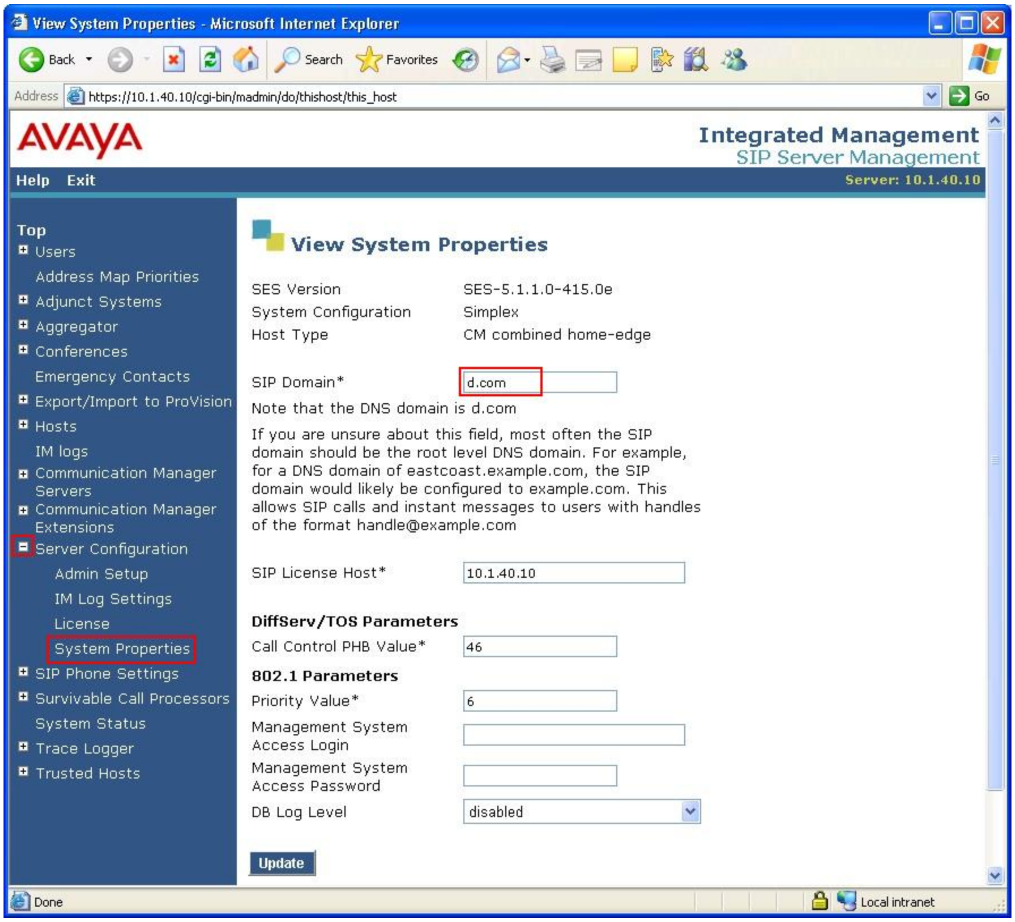
4. Configure Avaya SIP Enablement Services

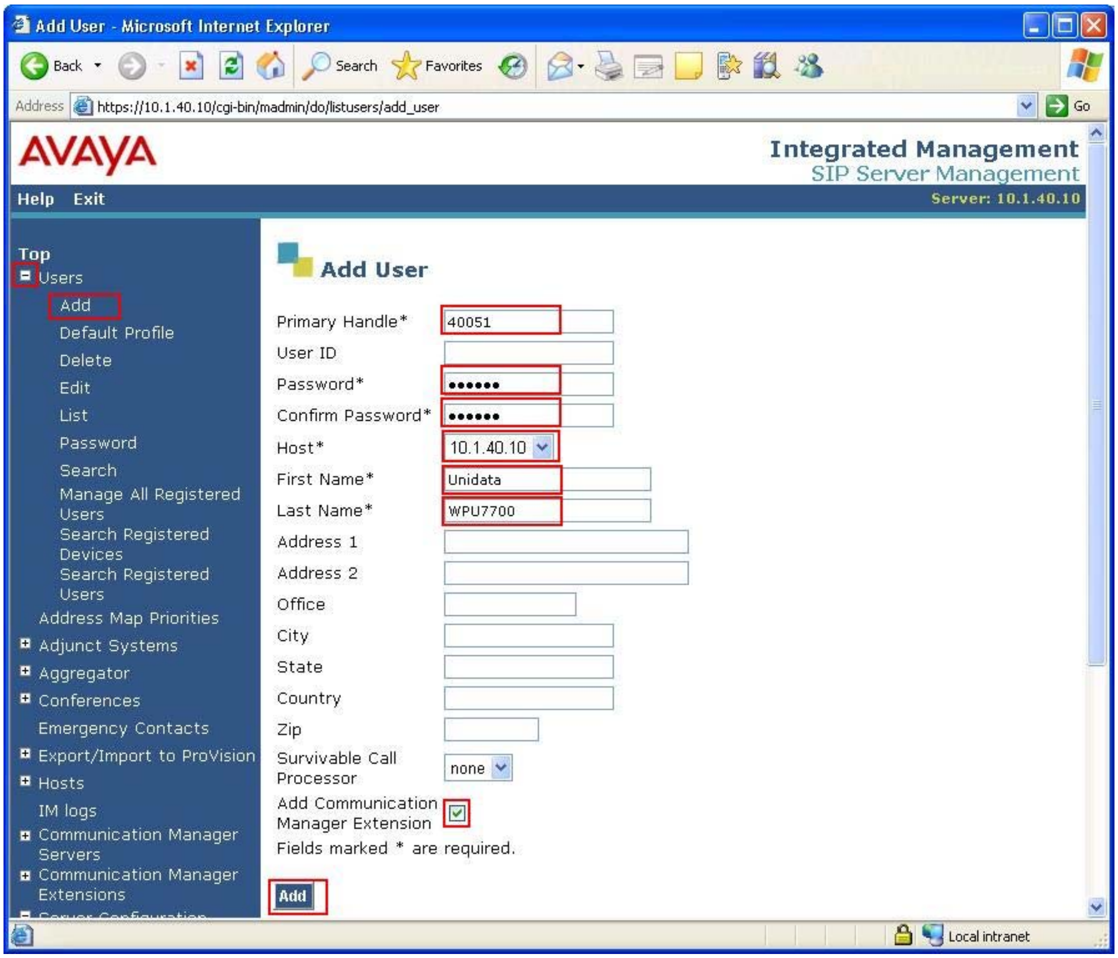
This section describes the steps for creating a SIP trunk between Avaya SES and Avaya Communication Manager. Also, SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. The UniData telephones will register with Avaya SES using the SIP user accounts.

Configuration in the following steps is only for the fields where a value needs to be entered or modified. Default values are used for all other fields.

Step	Description
1.	Open a web browser, enter **http://<IP address of Avaya S8300C server>** for the URL, and log in with the appropriate credentials. Click on the **Launch SES Administration Interface** link upon successful login.

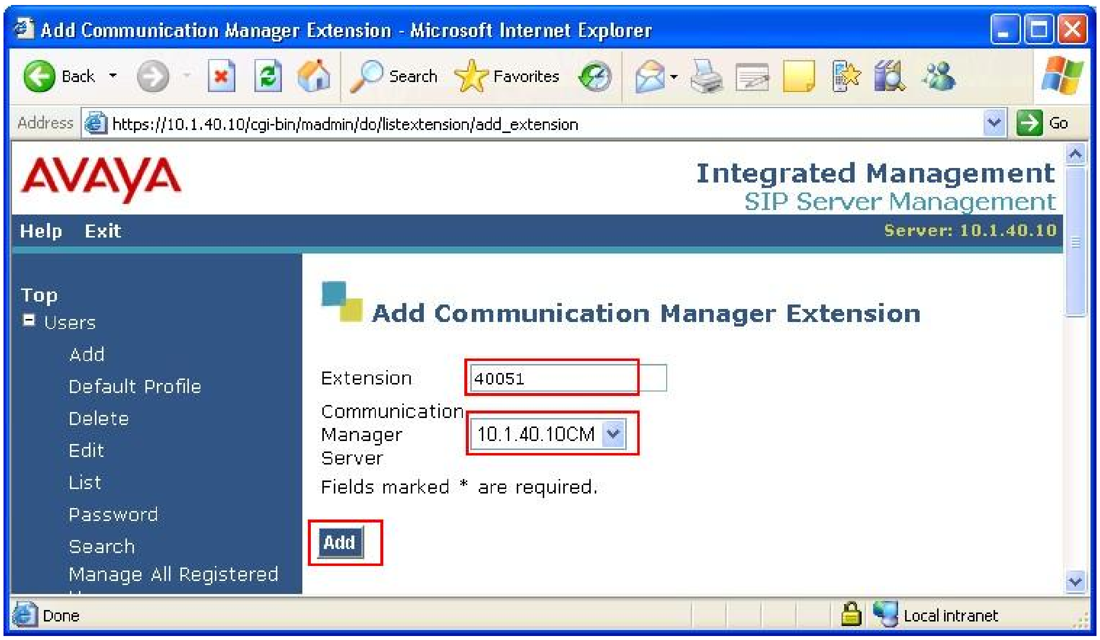


Step	Description
2.	On the **SIP Server Management** page: • Click the **+** sign to expand the options under **Server Configuration**. • Click **System Properties**. • Verify the **SIP Domain** matches the **Far-end Domain** field value configured for the signaling group on Avaya Communication Manager in **Section 3.4**. 

Step	Description
3.	In the left pane of the **SIP Server Management** page, expand **Users** and click **Add**. At the **Add User** page, configure the following: • **Primary Handle –** Enter the phone number of the UniData telephone. This number was configured in **Section 3.6 Step 1**. • **User ID** – Set to any descriptive name (optional). • **Password** and **Confirm Password –** Specify a password that the UniData telephone will use to register with Avaya SES. • **Host –** Select the IP address of the co-resident Avaya SES server. • **First Name** and **Last Name –** Enter descriptive names. • Check the **Add Media Server Extension** checkbox. Click **Add** when finished and then click **Continue** on the next page [not shown]. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>
At the Add Communication Manager Extension page, configure the following:

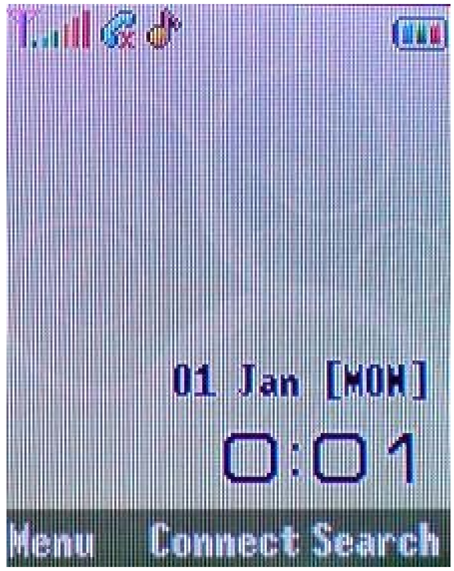
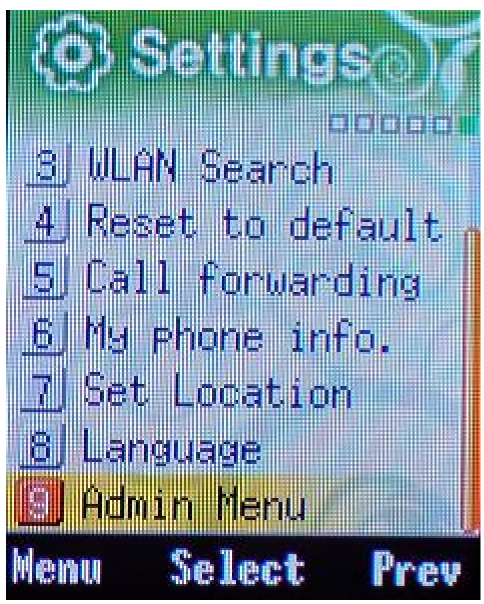
Extension – Set to Phone Number field value configured in Section 3.6 Step 1.
Communication Manager Server – Set to the co-resident Communication Manager where this OPS station is configured.
Click Add and then click Continue on the next page [not shown].

Note: Communication Manager Server was previously configured during the initial setup of SES.


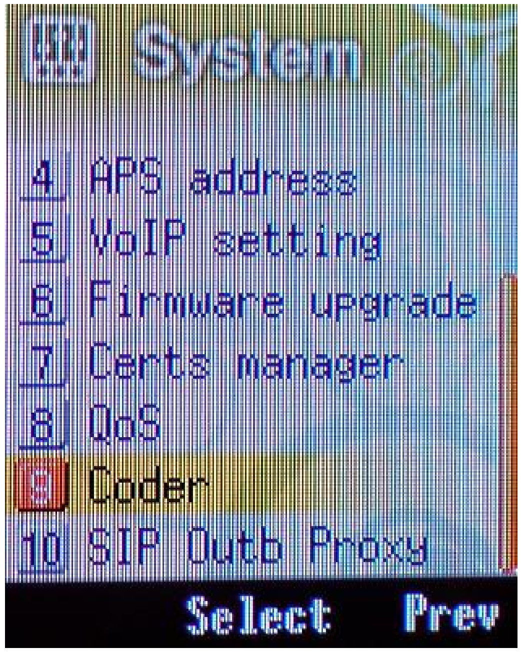
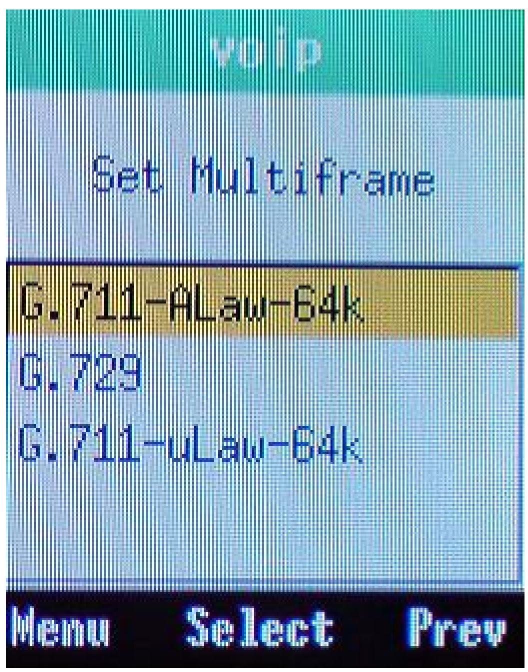
</td></tr>
<tr><td>5.</td><td>Repeat Steps 3 and 4 as necessary to configure additional UniData telephones.</td></tr>
</table>

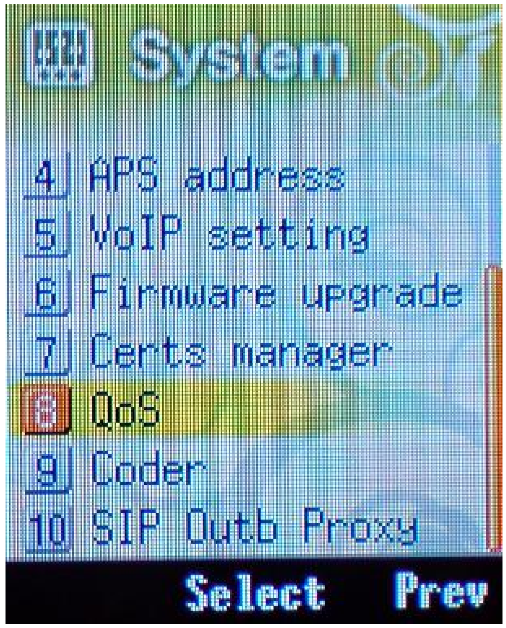
5. Configure UniData WPU-7700 Wireless IP Phones

This section describes the steps for configuring the UniData WPU-7700 Wireless IP Phones. This section assumes that the UniData telephone's connection to the wireless access point is already configured and the telephone is assigned an IP address via DHCP. The UniData telephones can be provisioned via the TFTP, HTTP or HTTPS protocol, or by configuring through the phone manually. The configuration steps in this section described the manual method.

Step	Description
1.	From the home screen as shown below, press **Menu**. 
2.	Use the direction keys to select **Admin Menu** on the Settings screen. Enter the Administrator password when prompted (not shown). 

Step	Description
3.	On the System screen, select **VoIP setting**. 
4.	On the VoIP setting screen, configure the following: • Display name: Enter the name to display on the phone. • User name: Enter the **Primary Handle** configured in **Section 4 Step 3**. • Auth. User name: Enter the **Primary Handle** configured in **Section 4 Step 3**. • Auth. password: Enter the **Password** configured in **Section 4 Step 3**. • Domain: Enter the IP address of the Avaya SES. • 1st_Proxy: Enter the IP address of the Avaya SES. • 2nd_Proxy: Enter the IP address of an alternative SIP proxy, not used in this case. 

Step	Description
5.	On the System screen, select **Coder** and then **voip** on the next screen (not shown) to configure the preferred codecs to use on the phone. 
6.	On the voip screen, set the priority of the codec to use. The order of the priority can be changed by pressing **Menu**. 

Step	Description
7.	On the System screen, select **QoS** and then **voip** on the next screen (not shown) to configure the DSCP values to use on the UniData telephone. 
8.	On the voip screen, set the Signal DSCP and Voice DSCP values (in hexadecimal notation) to correspond to the **Call Control PHB Value** and **Audio PHB Value** configured in **Section 3.3 Step 1**. 

For advance configuration of the UniData telephone features, configure the following parameters in the **e1_common.ini file** on the provisioning server. This completes the configuration of the UniData telephone.

Note: Set the value to “0” to disable the feature, or “1” to enable it.

Feature	Parameters
Call Waiting	[BASIC_CALL] Use_Call_Waiting = 1
Do Not Disturb	[DND] Use_DND = 1 Enable_DND = 0
Message Waiting Indicator	[MWI] Use_MWI = 1 Use_Subscribe = 0

6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment on the UniData WPU-7700 Wireless IP Phones and operations such as dialing methods (manual, re-dial, and phone book), hold, mute, transfer and conference. UniData WPU-7700 Wireless IP Phones’ interactions with SES, Avaya Communication Manager, and Avaya SIP, H.323, and Analog telephones were also verified.

6.1. General Test Approach

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

- UniData telephones successfully register with Avaya SES.
- UniData telephones successfully establish calls with Avaya SIP, H.323, and Analog telephones attached to Avaya SES or Avaya Communication Manager.
- UniData telephones successfully establish calls with PSTN telephones through Avaya Communication Manager.
- UniData telephones successfully handle concurrent calls.
- UniData telephones successfully negotiate the right codec.
- UniData telephones successfully shuffle for VoIP calls.
- UniData telephones successfully transmit DTMF during a call.
- UniData telephones successfully hold and transfer a call.
- UniData telephones establish a three party conference call, and display calling party number.

6.2. Test Results

The test objectives of **Section 6.1** were verified. UniData telephones successfully shuffled to communicate directly with the other telephones and negotiated the codec.

The following observations were made during testing:

- The UniData telephones do not have built-in support for 3-party conference. To setup a conference using the UniData telephones, use the Conference on Answer OPS feature on Avaya Communication Manager.
- Priority Call OPS feature is not supported.

UniData may address the above observations in future firmware releases. Contact UniData for further updates.

7. Verification Steps

The following steps may be used to verify the configuration:

- Verify that the UniData telephones successfully register with the Avaya SES server by using the **Users -> Search Registered Users** link on the SIP Server Management Web Interface.
- Place calls to and from the UniData telephones and verify that the calls are successfully established with two-way talk path.
- From the Avaya Communication Manager System Access Terminal (SAT) interface, perform the following steps to verify:
  - Audio codec used between two telephones
  - Shuffling between two telephones

Step	Description
1.	Enter **status trunk t** command, where **t** is the SIP trunk configured in **Section 3.5**. Note down the **Member** with **Service State** set to **in-service/active**. In this example, **0050/001** and **0050/005** are active and either member can be used to verify whether calls shuffled and which codec was used.

```
status trunk 50

                             TRUNK GROUP STATUS

Member    Port     Service State      Mtce Connected Ports
                                      Busy

0050/001 T00011   in-service/active   no   T00015
0050/002 T00012   in-service/idle     no
0050/003 T00013   in-service/idle     no
0050/004 T00014   in-service/idle     no
0050/005 T00015   in-service/active   no   T00011
0050/006 T00016   in-service/idle     no
0050/007 T00017   in-service/idle     no
0050/008 T00018   in-service/idle     no
0050/009 T00019   in-service/idle     no
0050/010 T00020   in-service/idle     no
0050/011 T00021   in-service/idle     no
0050/012 T00022   in-service/idle     no
```

2.	Enter **status trunk m,** where **m** is the member in active state as noted in the previous step for verification of codec used and shuffling status: • **Codec –** The codec used for Audio is **G.711MU** in this example. • **Shuffling** - If the **Near-end IP Addr** and **Far-end IP Addr** for **Audio** belongs to the UniData telephones and the **Audio Connection Type** is **ip-direct**, it signifies that shuffling was successful. In this example, shuffling was successful.

```
status trunk 50/1                                                 Page   1 of   2
                              TRUNK STATUS

 Trunk Group/Member: 0050/001              Service State: in-service/active
               Port: T00011             Maintenance Busy? no
 Signaling Group ID:

   IGAR Connection? no

    Connected Ports: T00015




                   Port      Near-end IP Addr : Port     Far-end IP Addr : Port
        Signaling: 01A0017    10.   1. 40. 10  : 6001     10.   1. 40. 10 : 5061

G.711MU      Audio:             10.   1. 40.202  : 20000    10.   1. 40.207  : 20000
             Video:
       Video Codec:
                                                Authentication Type: None
    Audio Connection Type: ip-direct
```

8. Support

For technical support on UniData WPU-7700 Wireless IP Phones, contact UniData technical support at:

- Telephone: +82-2-3443-3390
- E-mail: mini@udcsystems.com

9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 5.1.1, Avaya SIP Enablement Services 5.1.1 and UniData WPU-7700 Wireless IP Phones. During compliance testing, UniData telephones 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. The objectives of **Section 6.1** were met with some exceptions as noted in **Section 6.2**.

10. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com/.

[1] *Administrator Guide for Avaya Communication Manager*, Release 5.0, Issue 4.0, January 2008, Document Number 03-300509.

[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 13, January 2008, Document Number 555-233-504.

[3] *SIP Support in Avaya Communication Manager Running on Avaya S8xxx Servers*, Issue 8, January 2008, Document Number 555-245-206.

[4] *Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services*, Issue 6.0, June 2008, Document Number 03-600768.

Product information for UniData products may be found at http://www.udcsystems.com.

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.