Avaya Solution & Interoperability Test Lab

Application Notes for Bittel Electronics Kingstar BT-2008 SIP Telephones 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 Bittel Electronics Kingstar BT-2008 SIP Telephones. During compliance testing, Bittel Kingstar BT-2008 SIP Telephones successfully registered with Avaya SIP Enablement Services, 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 5.1.2, Avaya SIP Enablement Services (SES) 5.1.2, and Bittel Electronics Kingstar BT-2008 SIP Telephones. 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 Bittel Kingstar BT-2008 SIP Telephones.

1.1. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment on the Bittel Kingstar BT-2008 SIP Telephones and operations such as dialing methods (manual, re-dial, and phone book), hold, mute, transfer and conference. Bittel Kingstar BT-2008 SIP Telephones' interactions with Avaya SES, Avaya Communication Manager, and Avaya SIP, H.323, and Analog telephones were also verified.

1.2. Support

For technical support on Bittel Kingstar BT-2008 SIP Telephones, contact Bittel technical support at:

- Telephone: +86-633-2212125
- E-mail: tech@bittelcom.com

2. Reference Configuration

Figure 1 illustrates a sample configuration consisting of an Avaya Communication Manager running on an Avaya S8300C Server with the Avaya G350 Media Gateway, the Avaya SIP Enablement Services (SES) that is co-resident on the S8300C, and the Bittel Kingstar BT-2008 SIP Telephones. For completeness, an Avaya 9630 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 Bittel SIP Telephones and Avaya SIP, H.323 and Analog telephones. The Fast Ethernet ports on the Avaya G350 Media Gateway provide LAN connectivity and power to the Avaya and Bittel IP Telephones through Power-over-Ethernet (PoE). 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 Avaya G350 Media Gateway. The ISDN-BRI trunk is also included to demonstrate calls routed by Avaya Communication Manager between the Bittel SIP Telephones and the PSTN.

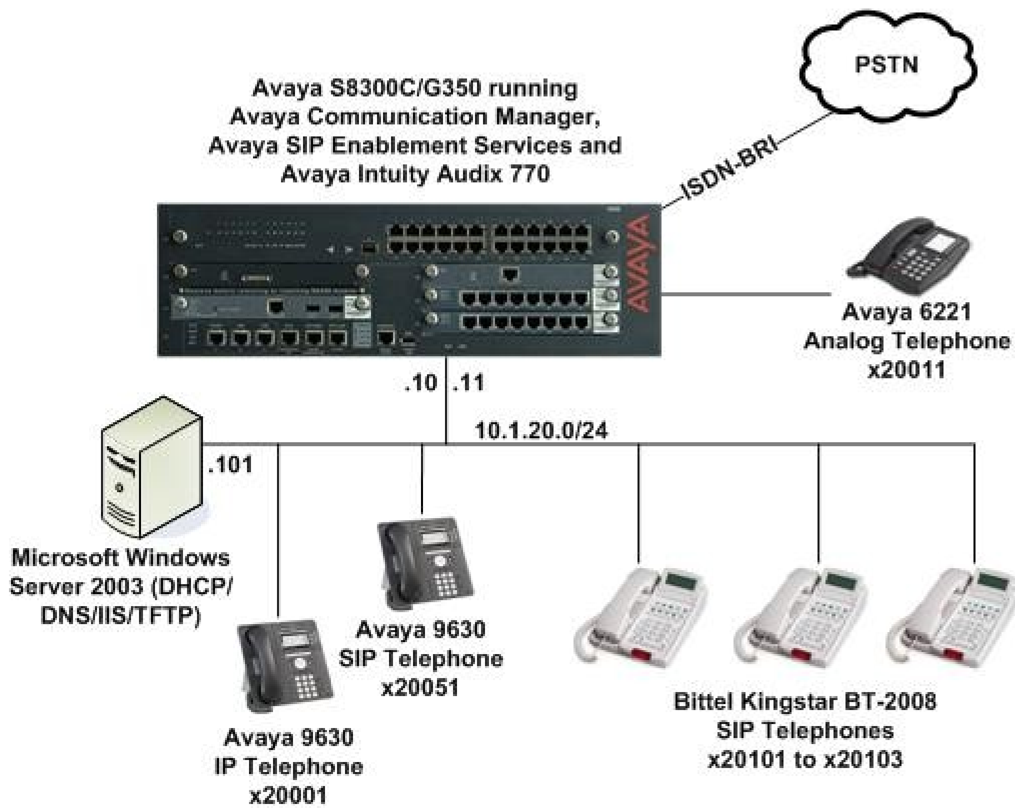


Figure 1: Sample Configuration

The Bittel SIP Phone 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 Bittel SIP Phone, Avaya Communication Manager routes the call over the SIP trunk to the Avaya SES for delivery to the Bittel SIP Phone.

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].

3. 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 5.1.2 (Service Packs 01.2.416.4-17067 and SES-01.2.416.4-SP1)
Avaya G350 Media Gateway	28.25.0
Avaya 9600 Series IP Telephones - 9630	3.0 (H.323), 2.0.5 (SIP)
Avaya 6221 Analog Telephone	-
Bittel Kingstar BT-2008 SIP Telephones	Ver 0.43.018

4. 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. Bittel and other SIP telephones are configured as Outboard-Proxy SIP (OPS) Stations 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 extension with the OPS endpoint. Similarly, a SIP telephone in Avaya SES is associated with an extension on Avaya Communication Manager. SIP Telephones register with the Avaya

SES and use Avaya Communication Manager for call origination and termination services. Enter the **save translation** command after completing this section.

4.1. 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: 13                             RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 900     244
                                      Maximum Stations: 450     156
                              Maximum XMOBILE Stations: 0       0
                    Maximum Off-PBX Telephones - EC500: 450     0
                    Maximum Off-PBX Telephones -   OPS: 450     5
                    Maximum Off-PBX Telephones - PBFMC: 0       0
                    Maximum Off-PBX Telephones - PVFMC: 0       0
                    Maximum Off-PBX Telephones - SCCAN: 0       0
```

2.	Proceed to **Page 2** of **OPTIONAL FEATURES** form. Verify that the number of **Maximum Administered 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: 200    66
           Maximum Concurrently Registered IP Stations: 450    1
             Maximum Administered Remote Office Trunks: 450    0
Maximum Concurrently Registered Remote Office Stations: 450    0
              Maximum Concurrently Registered IP eCons: 2      0
  Max Concur Registered Unauthenticated H.323 Stations: 200    0
                  Maximum Video Capable H.323 Stations: 200    0
                   Maximum Video Capable IP Softphones: 200    0
                       Maximum Administered SIP Trunks: 450    20
  Maximum Administered Ad-hoc Video Conferencing Ports: 0      0
   Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                             Maximum TN2501 VAL Boards: 0      0
                     Maximum Media Gateway VAL Sources: 2      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
```

4.2. 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.

Step	Description
1.	Enter the **change ip-codec-set n** command, where **n** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 4.3** 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**, **G.711A** and **G.729B** were used and **Media Encryption** was set to **none**. Note also the value for **Frames Per Pkt** for each codec which should match the values configured on the Bittel telephones in **Section 6 Step 4**.

```
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.711A             n          2        20
 3: G.729B             n          2        20
 4:
 5:
 6:
 7:


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

4.3. 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 b.com in this example. This should match the SIP Domain value in Section 5 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.
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.
Codec Set – Set the codec set number as provisioned in Section 4.2.
Audio PHB Value – Note the value to be configured on the Bittel telephone in Section 6 Step 2.
Audio 802.1p Priority – Note the value to be configured on the Bittel telephone in Section 6 Step 2.
</td></tr>
<tr><td><pre>
change ip-network-region 2   Page   1 of 19
                               IP NETWORK REGION
  Region: 2
Location: Authoritative Domain: b.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 4.2**.

```
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
```

4.4. 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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter the command add signaling-group n, where n 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 Listen Port - Set to 5061 for co-resident Avaya SES.
Far-end Network Region - Set to the Region configured in Section 4.3.
Far-end Domain - Set to b.com in this example. This should match the SIP Domain value in Section 5 Step 2.

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

4.5. 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 n, where n is an unallocated trunk group and configure the following:

Group Type – Set to the Group Type field value configured in Section 4.4.
Group Name – Enter any descriptive name.
TAC (Trunk Access Code) – Set to any available trunk access code.
Signaling Group – Set to the Group Number field value configured in Section 4.4. (i.e., 50)
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.

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.
<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: 20
</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 Bittel telephone. Leaving this value blank is also acceptable as the Avaya Communication Manager and the Bittel telephone will negotiate the payload type.
<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>

4.6. 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 Bittel telephones.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>Enter the add station n command, where n 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.</td></tr>
<tr><td><pre>
add station 20101   Page   1 of   5
                                     STATION

Extension: 20101   Lock Messages? n   BCC: 0
     Type: 6408D+   Security Code: TN: 1
     Port: X   Coverage Path 1:   COR: 1
     Name: John Doe   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: 20101
            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>
</table>

2.	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.

```
add station 20101                                               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:
```

3.	Enter the **change off-pbx-telephone configuration-set n** command, where **n** is an unused configuration set to be used for the Bittel telephones. On the **CONFIGURATION SET** form, configure the following fields: • **Configuration Set Description –** Set to a descriptive name. • **Calling Number Style –** Set to the recommended value of **network**. Use the default values for the remaining fields. For the detail explanation of each field, refer to **Chapter 19: Screen Reference** in **[1]**.

```
change off-pbx-telephone configuration-set 1                      Page   1 of   1



                                  CONFIGURATION SET: 1

                        Configuration Set Description: SIP Phones
                                 Calling Number Style: network
                                  CDR for Origination: phone-number
                    CDR for Calls to EC500 Destination? y
                         Fast Connect on Origination? n
                         Post Connect Dialing Options: dtmf
                        Cellular Voice Mail Detection: none
                                         Barge-in Tone? n
                         Calling Number Verification? y
          Call Appearance Selection for Origination: primary-first
                                     Confirmed Answer? n
```

4.	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 Bittel 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 4.5**. • **Config Set –** Set to the configuration set configured in **Step 3**.

	<pre>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
	 20101 OPS - 20101 50 1</pre>

5.	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**.

	<pre>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
	 20101 2 both all both</pre>

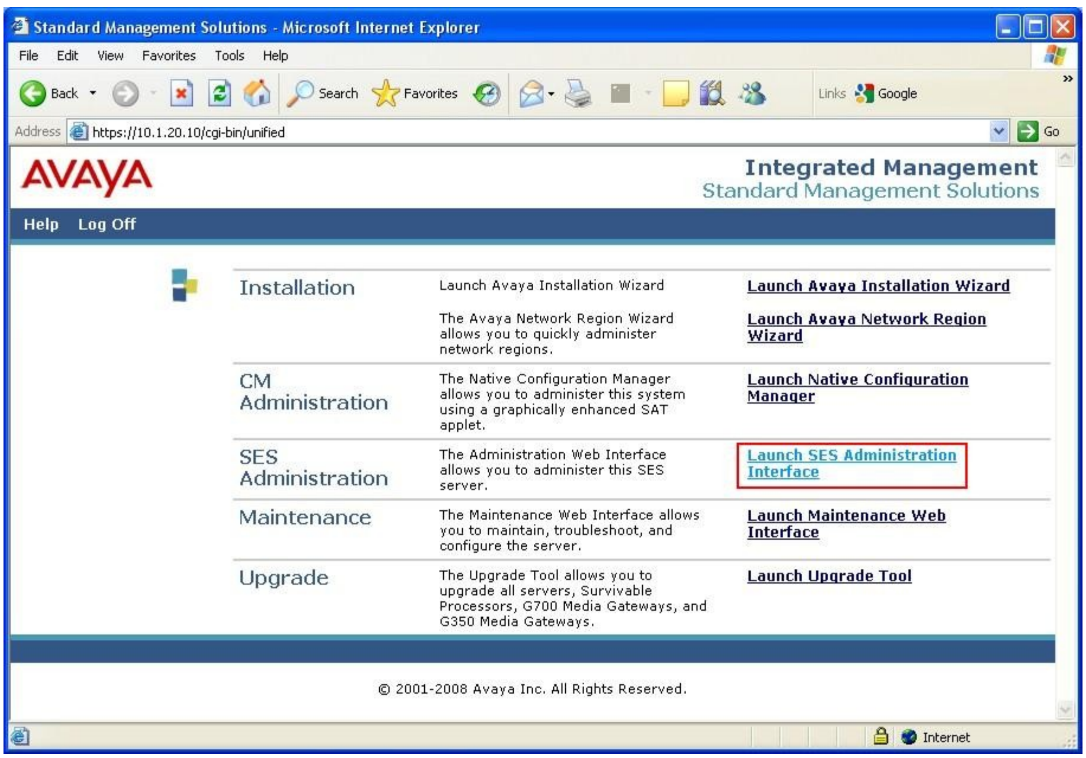
6.	Repeat **Steps 1 - 5** as necessary to administer additional OPS stations and associations for Bittel telephones.
7.	Enter the **save translation** command after completing **Section 4** to make the changes permanent.

5. Configure Avaya SIP Enablement Services

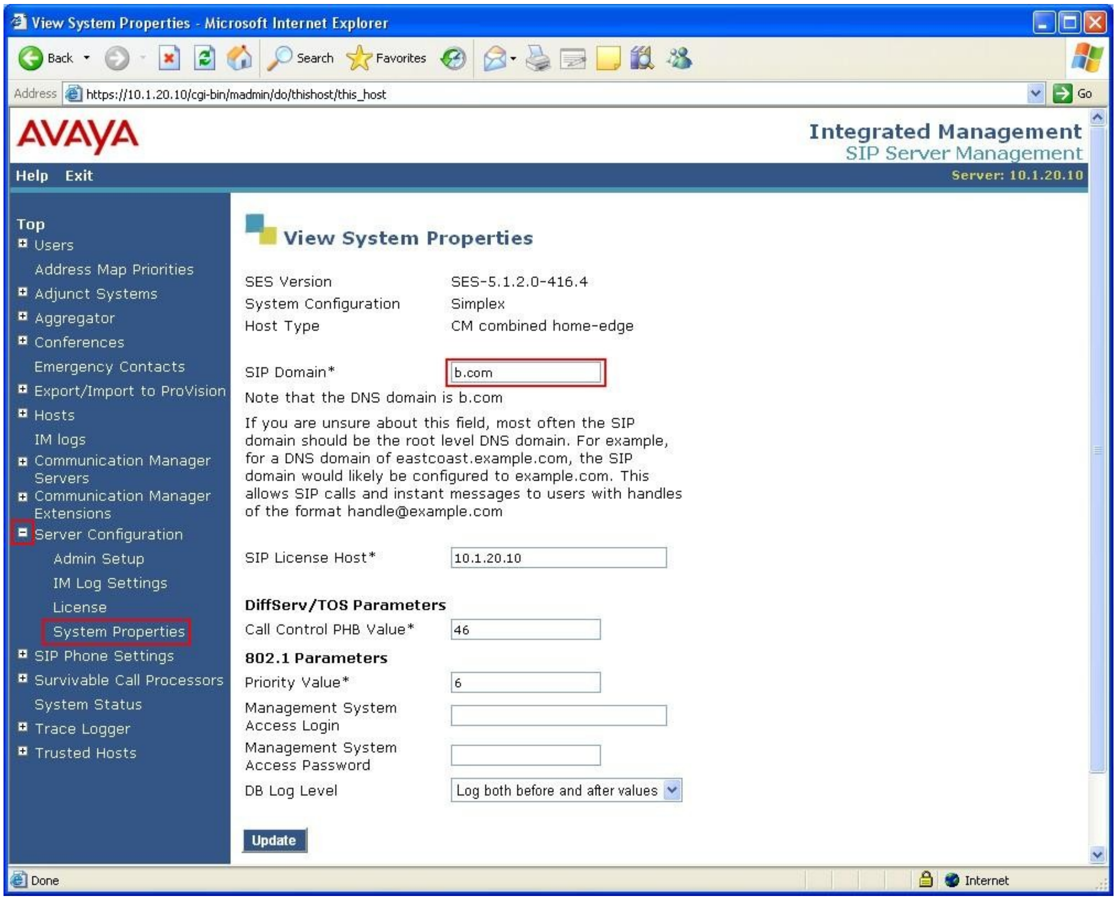
This section describes the steps for creating a SIP trunk between Avaya SES and Avaya Communication Manager. Also, SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. The Bittel telephones will register with Avaya SES using the SIP user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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Open a web browser, enter http://<IP address of co-resident Avaya SES> for the URL, and log in with the appropriate credentials (not shown). Click on the Launch SES Administration Interface link upon successful login.



</td></tr>
</table>

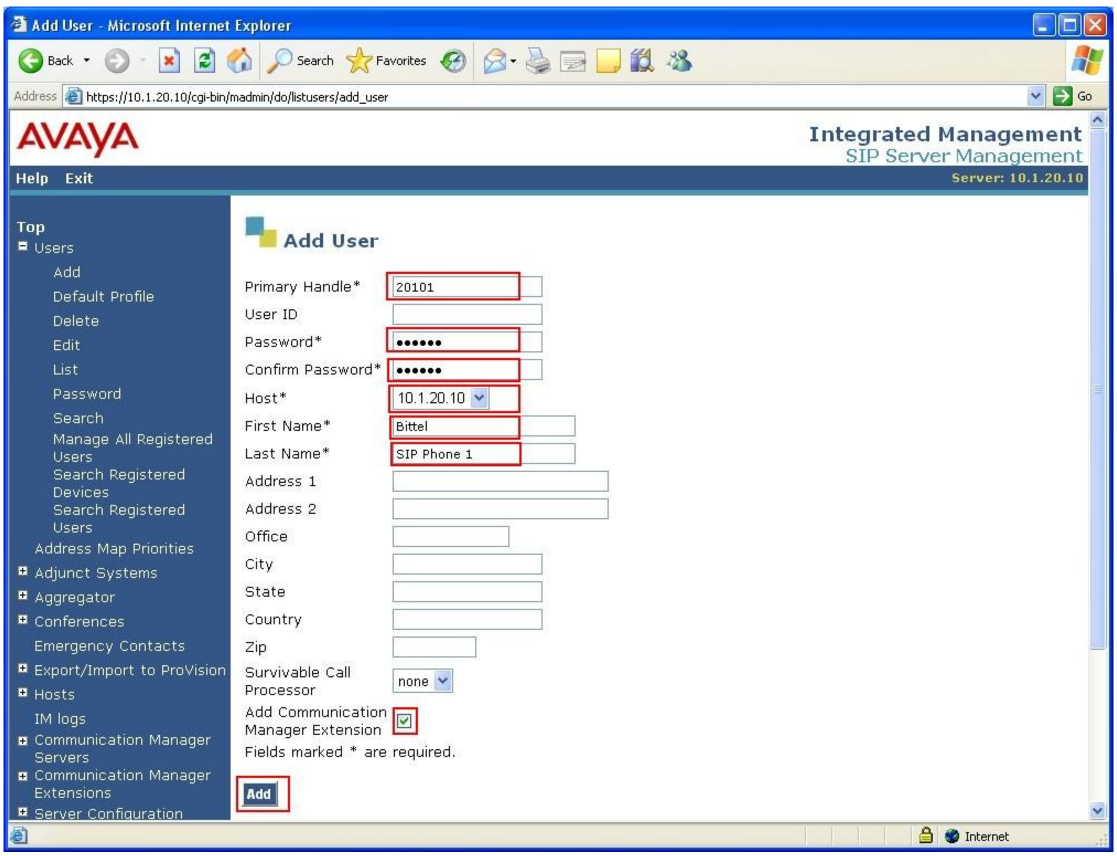
Step	Description
2.	On the **SIP Server Management** page: • Click the + sign to expand the options under **Server Configuration**. • Click **System Properties**. • Verify the **SIP Domain** matches the **Far-end Domain** field value configured for the signaling group on Avaya Communication Manager in **Section 4.4**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>

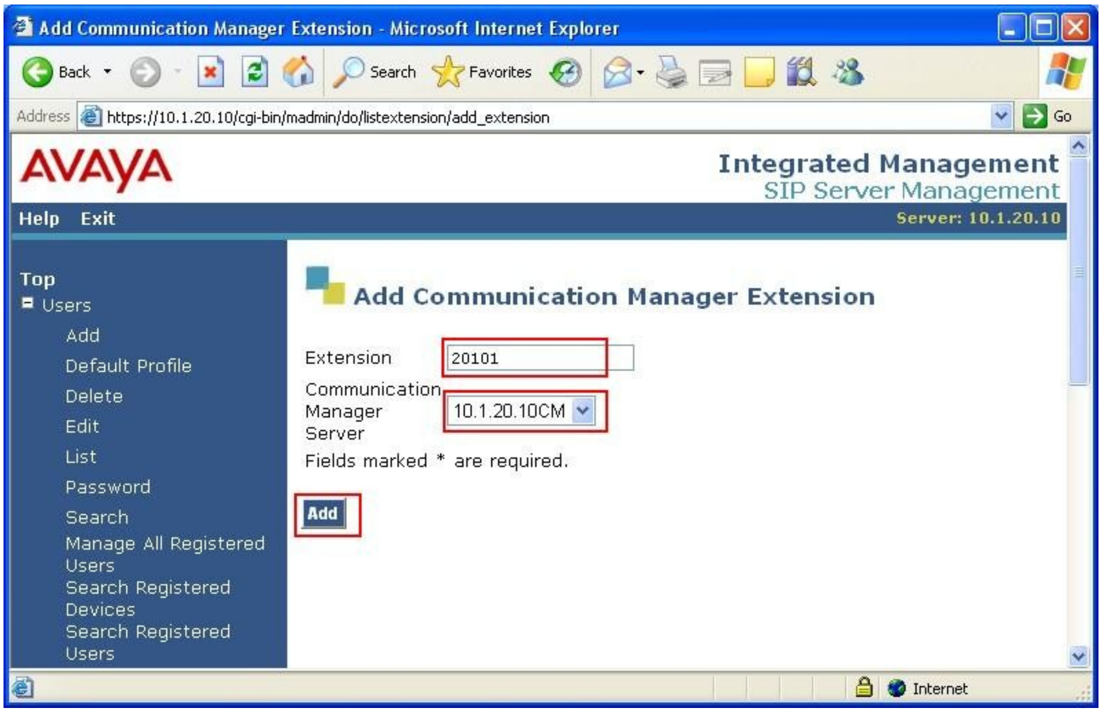
In the left pane of the **SIP Server Management** page, expand **Users** and click **Add**. At the **Add User** page, configure the following:

- **Primary Handle –** Enter the phone number of the Bittel telephone. This number was configured in **Section 4.6 Step 1**.
- **User ID** – Set to any descriptive name (optional).
- **Password** and **Confirm Password –** Specify a password that the Bittel 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 Communication Manager Extension** checkbox.

Click **Add** when finished and then click **Continue** on the next page [not shown].

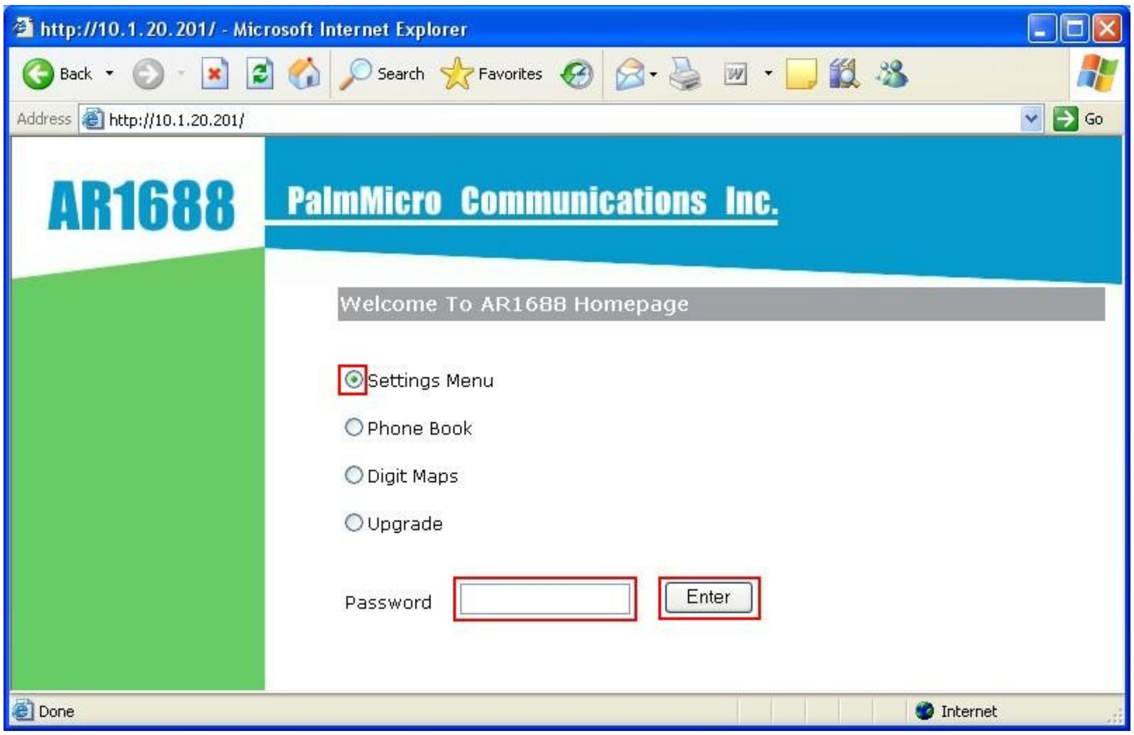


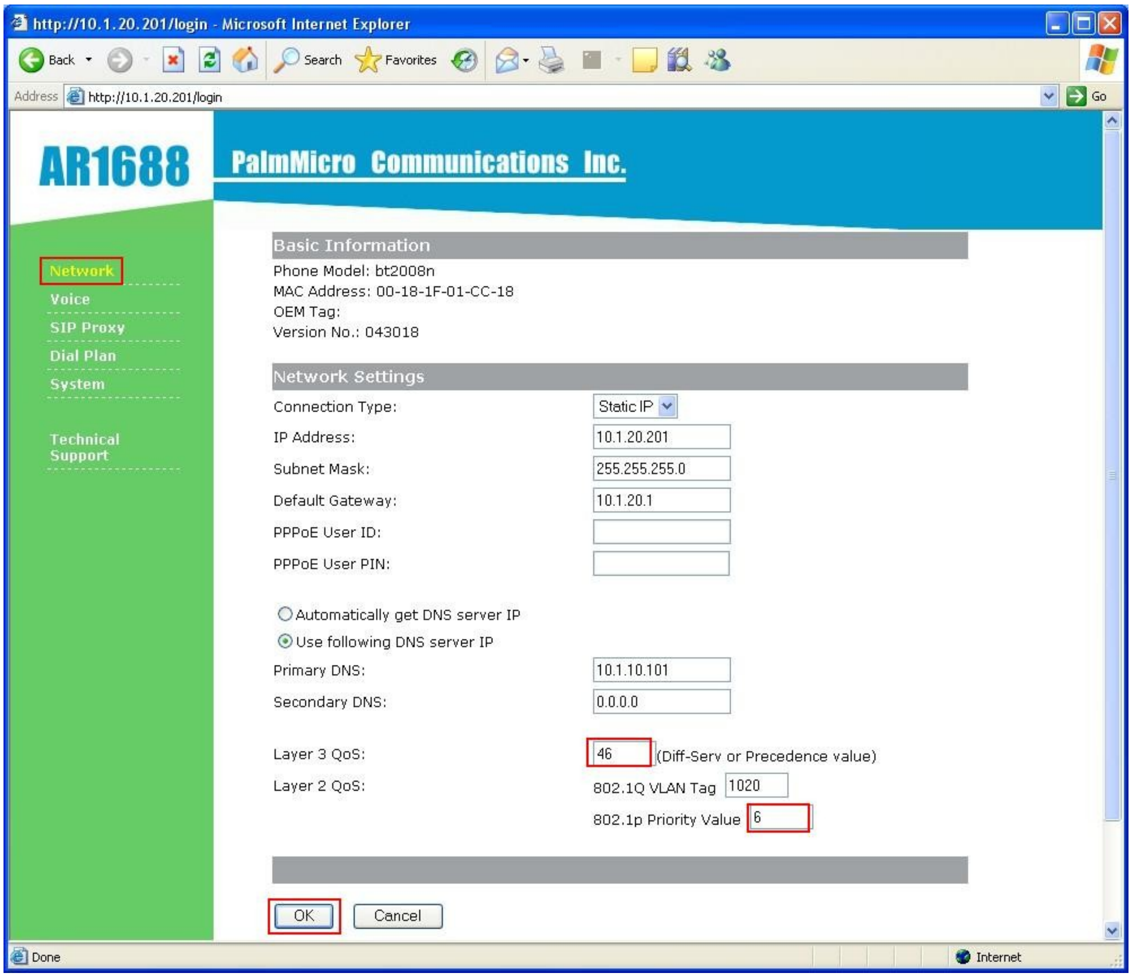
</td></tr>
</table>

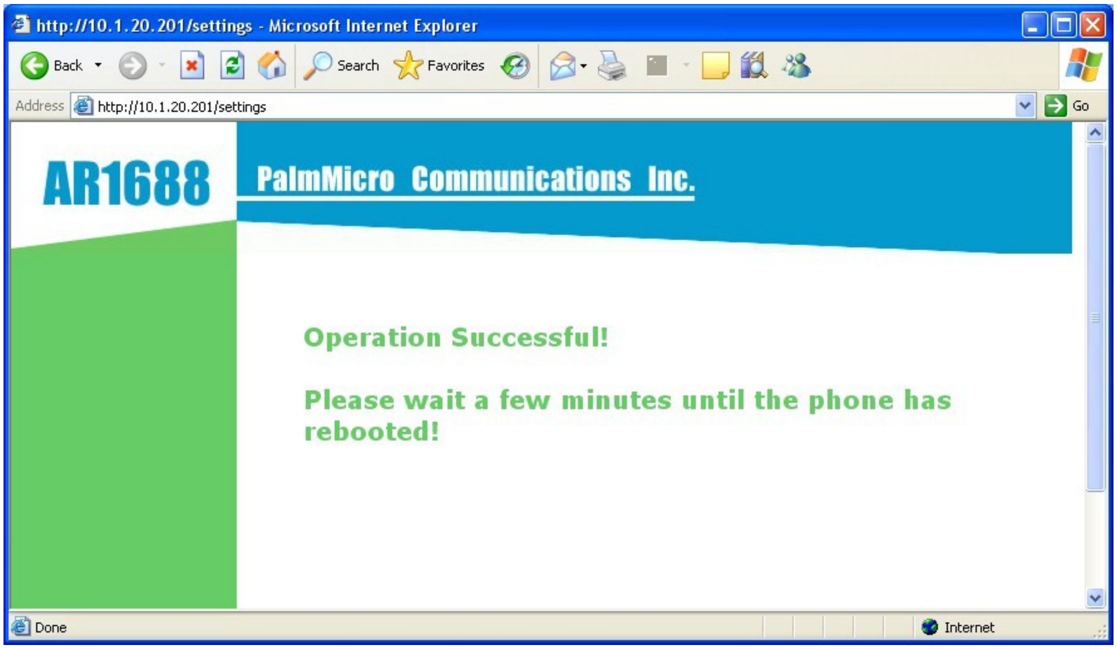
Step	Description
4.	At the **Add Communication Manager Extension** page, configure the following: • **Extension –** Set to **Phone Number** field value configured in **Section 4.6 Step 1**. • **Communication Manager Server** – Set to the co-resident Communication Manager where this OPS station is configured. • 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. 
5.	Repeat **Steps 3** and **4** as necessary to configure additional Bittel telephones.

6. Configure Bittel Kingstar BT-2008 SIP Telephones

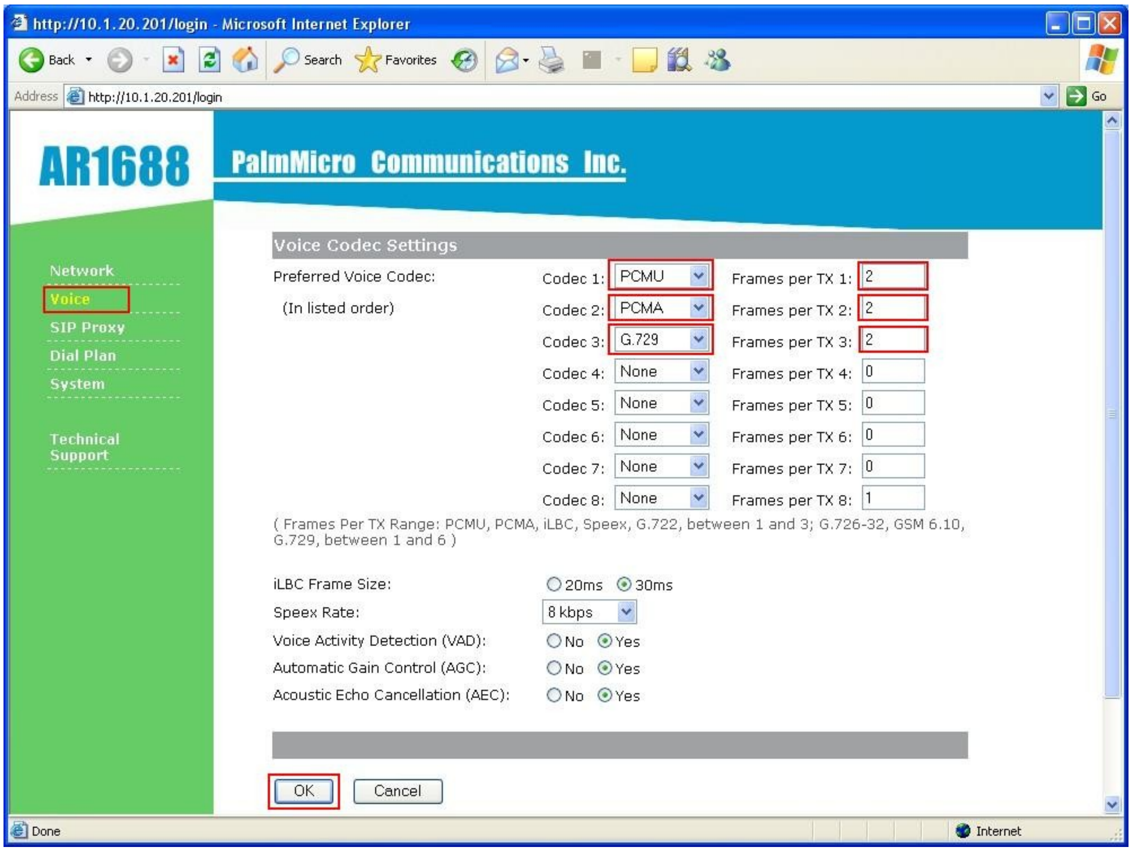
This section describes the steps for configuring the Bittel Kingstar BT-2008 SIP Telephones. The Bittel telephones support DHCP and automatic provisioning, but for the compliance testing, the Bittel telephones were assigned static IP addresses and manually configured via the Bittel telephones' web interface.

Step	Description
1.	Open a web browser and enter **http://a.b.c.d** for the URL, where **a.b.c.d** is the IP address of the Bittel telephone. Select **Settings Menu**, enter the password (if configured) and click **Enter**. 

Step	Description
2.	The Network page will be shown. In the **Network Settings** section, configure the following: • **Layer 3 QoS** – Set to the **Audio PHB Value** field value configured in **Section 4.3 Step 1**. • **Layer 2 QoS: 802.1p Priority Value –** Set to the **Audio 802.1p Priority** field value configured in **Section 4.3 Step 1**. Click **OK** to continue. 

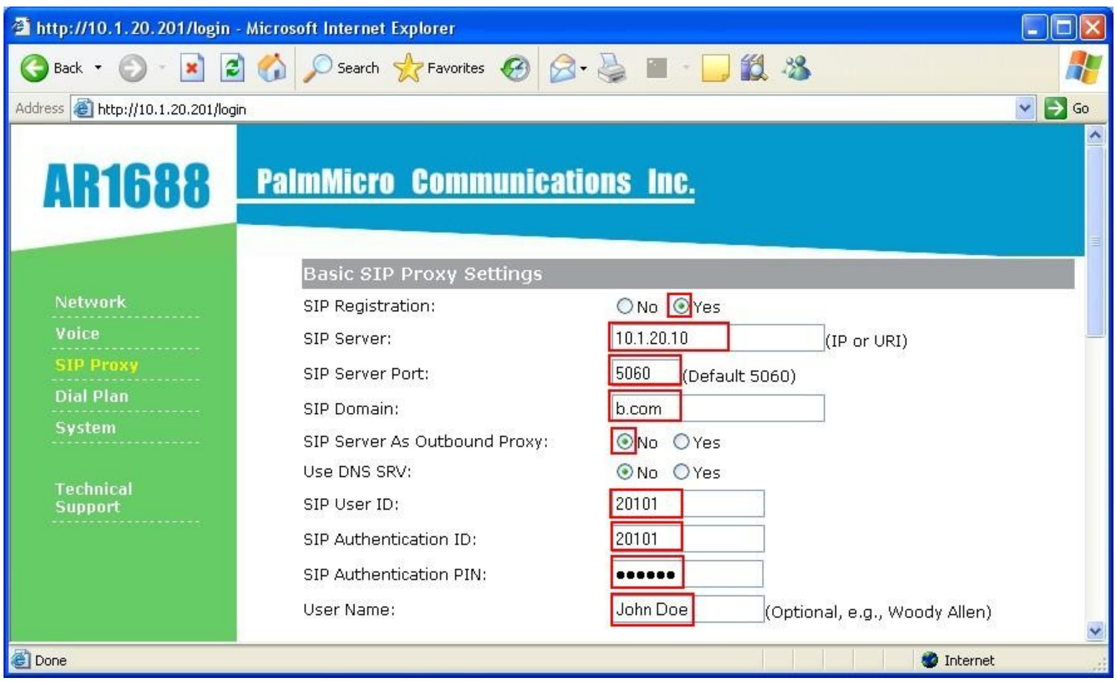
Step	Description
3.	The following web page is displayed. This occurs whenever the phone settings are changed. Repeat **Step 1** to return to the **Settings Menu**. 

Step	Description
4.	Click **Voice** on the left navigation menu. In the **Voice Codec Settings** section, for **Preferred Voice Codec**, select the codecs in the order of preference. For compliance testing, the codecs used were **PCMU**, **PCMA** and **G.729**. For the codecs **PCMU**, **PCMA** and **G.729**, set **Frames per TX** to **2** to match the values configured on Avaya Communication Manager in **Section 4.2 Step 1**. Click **OK**.

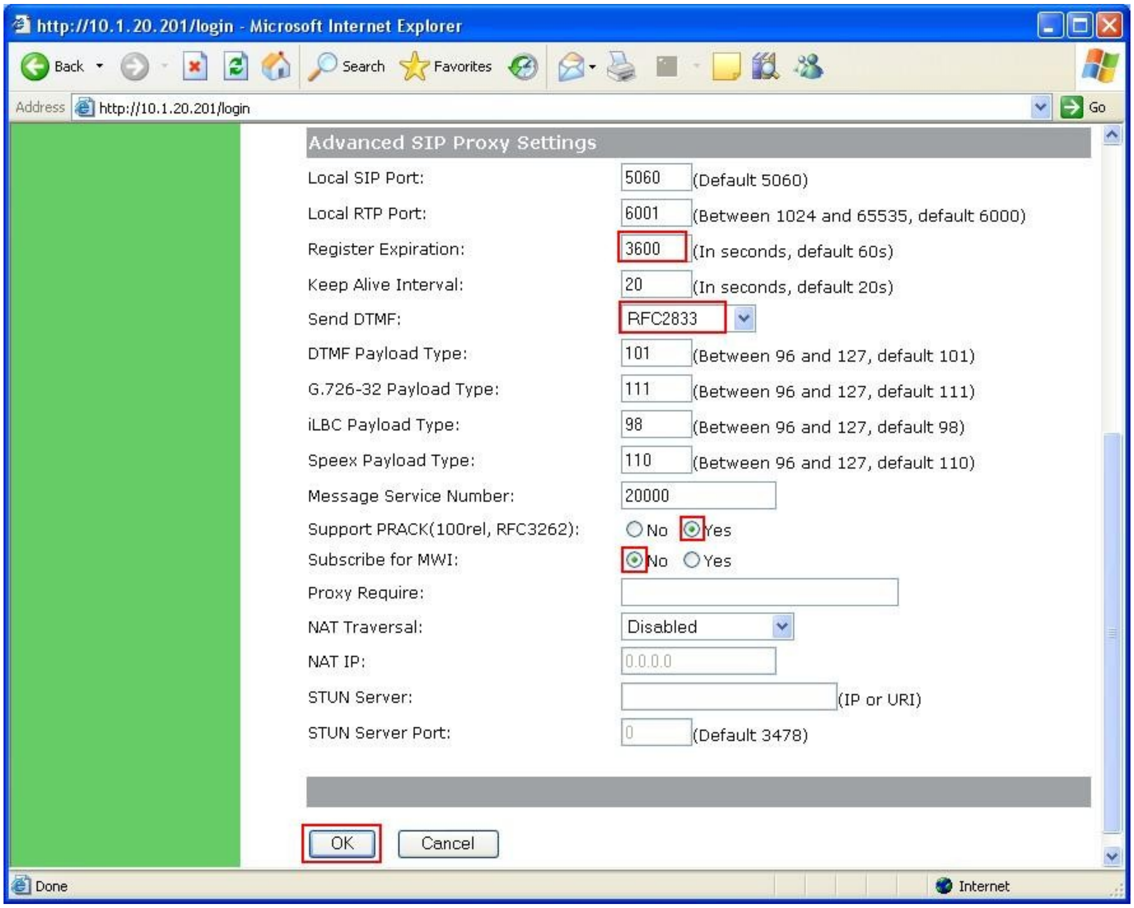


<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>Click SIP Proxy on the left navigation menu. In the Basic SIP Proxy Settings section, configure the following:

SIP Registration – Select Yes.
SIP Server – Set to the IP address of the co-resident Avaya SES.
SIP Server Port – Enter the default SIP port 5060.
SIP Domain – Set to the SIP Domain of the co-resident Avaya SES as shown in Section 5 Step 2.
SIP Server As Outbound Proxy – Select No.
SIP User ID – Set to the Primary Handle of a SIP User on Avaya SES as configured in Section 5 Step 3.
SIP Authentication ID – Set to the Primary Handle of a SIP User on Avaya SES as configured in Section 5 Step 3.
SIP Authentication PIN – Set to the Password of the SIP User as configured in Section 5 Step 3.
User Name – Enter an optional name for the user.



</td></tr>
</table>

Step	Description
6.	Scroll down to the **Advanced SIP Proxy Settings** section and configure the following: • **Register Expiration** – Set to **3600**, the recommended value for Avaya SES. • **Send DTMF –** Select **RFC2833**, which is supported by Avaya SES. • **Support PRACK (100rel, RFC3262)** – Select **Yes**. • **Subscribe for MWI –** Select **No**. Click **OK** to save the changes. This completes the configuration of the Bittel Kingstar BT-2008 SIP Telephones for basic operation. 

7. General Test Approach and Test Results

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

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

All test cases were successfully completed. The following observations were made during testing:

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

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

8. Verification Steps

The following steps may be used to verify the configuration:

- Verify that the Bittel 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 Bittel 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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter status trunk t command, where t is the SIP trunk configured in Section 4.5. 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.
<pre>
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
</pre></td></tr>
<tr><td>2.</td><td>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 Bittel telephones and the Audio Connection Type is ip-direct, it signifies that shuffling was successful. In this example, shuffling was successful.

<pre>
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. 20. 10 : 6001   10. 1. 20. 10 : 5061

G.711MU   Audio: 10. 1. 20.202 : 20000   10. 1. 20.201 : 20000
            Video:
      Video Codec:
                                              Authentication Type: None
    Audio Connection Type: ip-direct
</pre></td></tr>
</table>

9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 5.1.2, Avaya SIP Enablement Services 5.1.2 and Bittel Electronics Kingstar BT-2008 SIP Telephones. During compliance testing, Bittel 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-party conference, transfers, hold, etc. All test cases were successfully completed with some exceptions as noted in **Section 7**.

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 Bittel products may be found at http://www.chinabittel.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.