Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Citel Technologies *CITEL*link SIP Handset Gateway with Avaya Communication Manager and Avaya SIP Enablement Services - Issue 1.0

Abstract

These Application Notes describe the steps for configuring the Citel Technologies *CITEL*link SIP Handset Gateway to communicate with Avaya Communication Manager and Avaya SIP Enablement Services (SES). The *CITEL*link SIP Handset Gateway is network appliance that allows Nortel Meridian digital telephones to register with Avaya Communication Manager and Avaya SIP Enablement Services as SIP telephones. Emphasis of testing was placed on verifying the ability of the *CITEL*link SIP Handset Gateway to interoperate with Avaya 4620 SW IP Telephones and Avaya SIP Enablement Services. Note that beginning with release 3.0 software, Avaya Converged Communication Server (CCS) has been renamed to Avaya SIP Enablement Services (SES). Information in these Application Notes has been obtained through Developer*Connection* compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

Avaya Communication Manager and Avaya SIP Enablement Services (SES) has the capability to extend advanced telephony features to SIP stations. This feature set can be extended to non-Avaya SIP telephones.

These Application Notes describe a solution for configuring the Citel Technologies *CITEL*link SIP Handset Gateway to enable Nortel Meridian digital telephones to register with Avaya Communication Manager via Avaya SIP Enablement Services. The *CITEL*link SIP Handset Gateway is a network appliance capable of registering directly connected Nortel Meridian digital telephones with Avaya SIP Enablement Services through standard SIP messaging. From the point of view of Avaya Communication Manager, each Nortel Meridian telephone is a SIP telephone. Quality of Service (QoS) is achieved through the use of user configurable Layer-2 (802.1p) and Layer-3 (DiffServ) parameters in the *CITEL*link SIP Handset Gateway. This ensures that voice packets from the *CITEL*link SIP Handset Gateway are given proper priority in the network.

1.1. Configuration

Figure 1 illustrates the configuration used in these Application Notes. The extension numbers used by the Nortel Meridian telephones are registered to Avaya Communication Manager via Avaya SIP Enablement Services and are administered as Off-PBX-Telephones stations in Avaya Communication Manager. As a result, Off-PBX-Station (OPS) features from Avaya Communication Manager are available to each Nortel Meridian telephone.

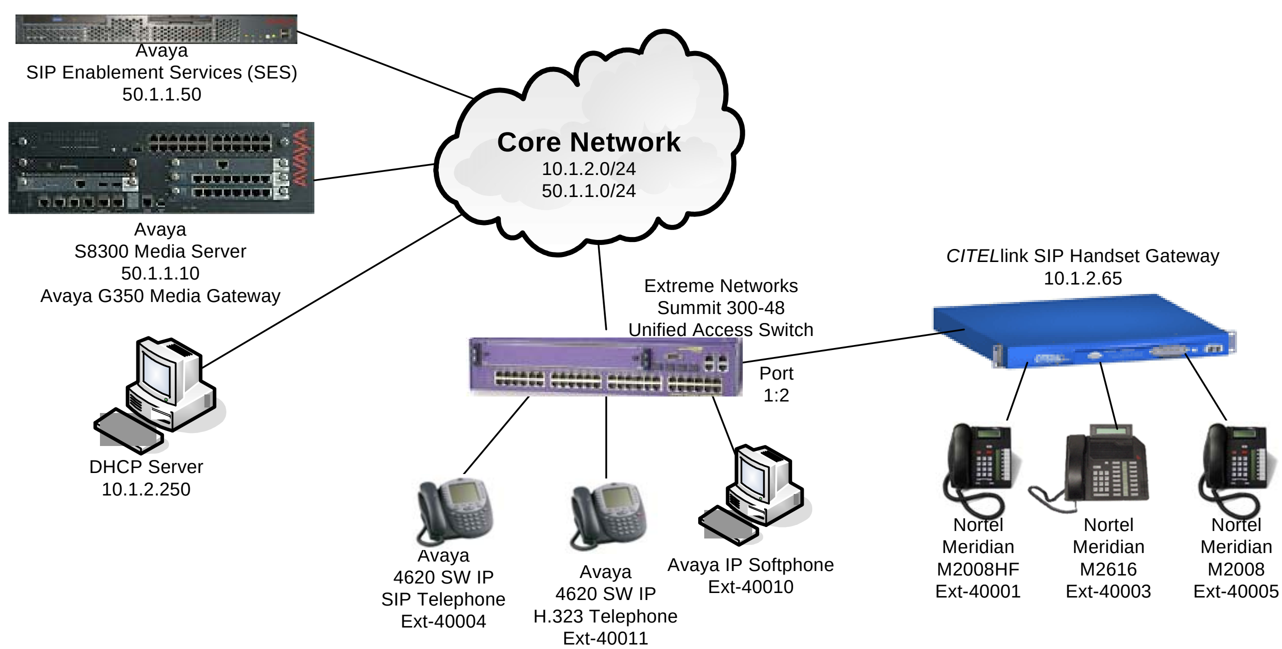


Figure 1: Sample Network Configuration

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8300 Media Server with a G350 Media Gateway	Communication Manager 3.0 (R0.13x.00.0.346.0)
Avaya SIP Enablement Services (SES) Server	3.0 (build 31)
Avaya 4620SW IP Telephone (H.323)	2.2.3
Avaya 4620SW IP Telephone (SIP)	2.2
Avaya IP Softphone	5.2.3.6
Extreme Networks Summit 300-48 Unified Access Switch	ExtremeWare 7.4e.1.5
Citel Technologies *CITEL*link SIP Handset Gateway	2066-SIP-Meridian_V_2_1_2_1
Nortel Meridian M2008, M2008HF, M2616 telephones	N/A

3. Configure Avaya Communication Manager

This section highlights the important commands for defining SIP telephones on Avaya Communication Manager. For complete documentation, see references [1] and [2]. Use the System Access Terminal (SAT) interface to perform these steps. Log in with the appropriate credentials.

3.1. Add New Stations to Avaya Communication Manager

Using the **add station** command, add a station for each SIP phone to be supported. The sample configuration uses ***6408D+*** for the station type. Include the coverage path for voice mail if it is available and use the appropriate COS value. Ensure that the station has three (3) "***call-appr***" buttons for **Button Assignment**. Set the **IP SoftPhone** field to ***n***. Repeat the following steps to add additional SIP telephone extensions.

```
add station 40003                                            Page   1 of   4
                                     STATION

Extension: 40003                        Lock Messages? n         BCC: 0
     Type: 6408D+                       Security Code:            TN: 1
     Port: X                          Coverage Path 1: 1         COR: 1
     Name: SIP40003                   Coverage Path 2:           COS: 1

STATION OPTIONS
              Loss Group: 2           Personalized Ringing Pattern: 1
             Data Module? n                       Message Lamp Ext: 40003
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english
                                                  Media Complex Ext:
                                                       IP SoftPhone? n
```

```
add station 40003                                              Page   3 of   4
                                     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:
 2: call-appr                          6:
 3: call-appr                          7:
```

Use the **change off-pbx-telephone station-mapping** command to map Avaya Communication Manager extensions to the Avaya SIP Enablement Service (SES) Server extensions. Select the trunk-group number for the **trunk-group** configured between Avaya Communication Manager and the Avaya SIP Enablement Services server. Select the **Configuration Set** number applicable for this configuration. The sample configuration uses **Configuration Set *1***. For additional information related to Avaya Communication Manager and OFF-PBX-EXTENSION support, refer to references [1], [2], [5], [6], and [7].

```
change off-pbx-telephone station-mapping 40003                    Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                   Selection   Set
 40001         OPS              - 40001              1          1
 40003         OPS              - 40003              1          1
 40005         OPS              - 40005              1          1

change off-pbx-telephone station-mapping 40000                    Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged
 Extension     Limit      Mode         Allowed     Calls
 40001        3         both          all         both
 40003        3         both          all         both
 40005        3         both          all         both
```

The screenshot below shows the settings for **trunk-group 1**.

```
display trunk-group 1                                             Page   1 of  20
                                TRUNK GROUP
Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: To CCS                     COR: 1       TN: 1       TAC: 101
   Direction: two-way       Outgoing Display? n
 Dial Access? n               Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                                      Signaling Group: 1
                                                    Number of Members: 24
TRUNK PARAMETERS
     Unicode Name? y
                                           Redirect On OPTIM Failure: 5000
            SCCAN? n                              Digital Loss Group: 18
```

The screenshot below shows the settings for **configuration-set 1**.

```
change off-pbx-telephone configuration-set 1                    Page   1 of   1
                               CONFIGURATION SET: 1
      Configuration Set Description:
               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
            Identity When Bridging: principal
```

3.2. Verify OPS Capacity

Use the display **system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones – OPS** has been set to a value that will accommodate the number of phones to be supported. Avaya Services has provisioned this during installation according to the system configuration purchased.

```
change system-parameters customer-options                       Page   1 of  10
                                OPTIONAL FEATURES
     G3 Version: V13
       Location: 1                          RFA System ID (SID): 1
       Platform: 13                         RFA Module ID (MID): 1
                                                            USED
                                Platform Maximum Ports: 900   48
                                      Maximum Stations: 40    20
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 50    0
                    Maximum Off-PBX Telephones -   OPS: 50    10
                    Maximum Off-PBX Telephones - SCCAN: 0     0
        (NOTE: You must logoff & login to effect the permission changes.)
```

3.3. IP Network Region

Note the value of the DiffServ/TOS parameters for Audio. This QoS value will be needed later when configuring the *CITEL*Link SIP Handset Gateway.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION

  Region: 1
Location: 1       Authoritative Domain: devcon.com
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? y
   UDP Port Max: 3028
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                  RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

3.4. Configure Audio Codec

In order for calls to be established successfully, during initial call setup the two end points must agree upon a mutually supported codec. Verify that the codec is set to G.711MU. *CITEL*link SIP Handset Gateway does not support G.729 codec in the version tested.

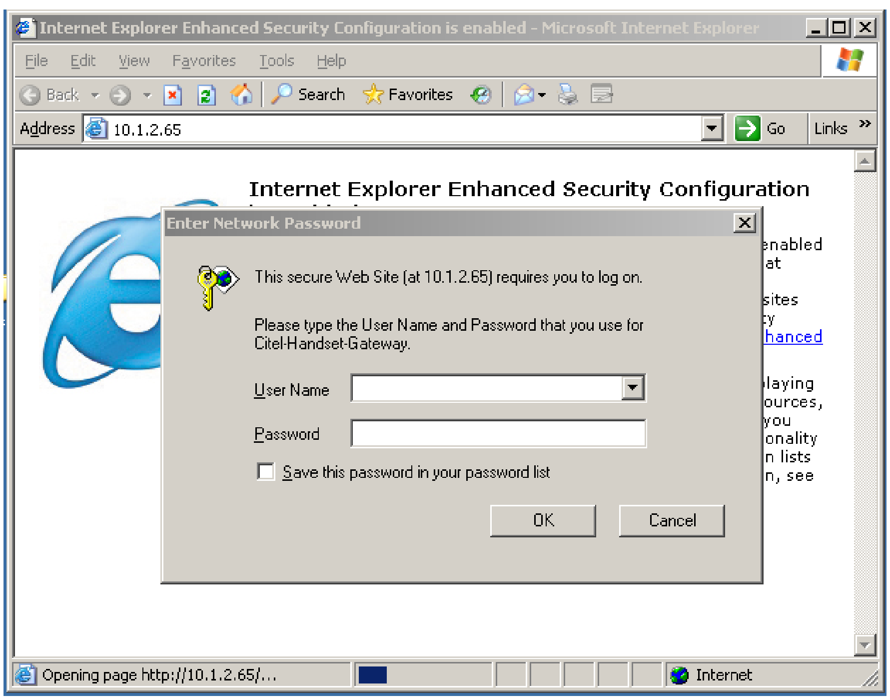
```
change ip-codec-set 1                                              Page   1 of   2
                          IP Codec Set
    Codec Set: 1
    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
```

4. Configure the Citel Technologies *CITEL*link SIP Handset Gateway

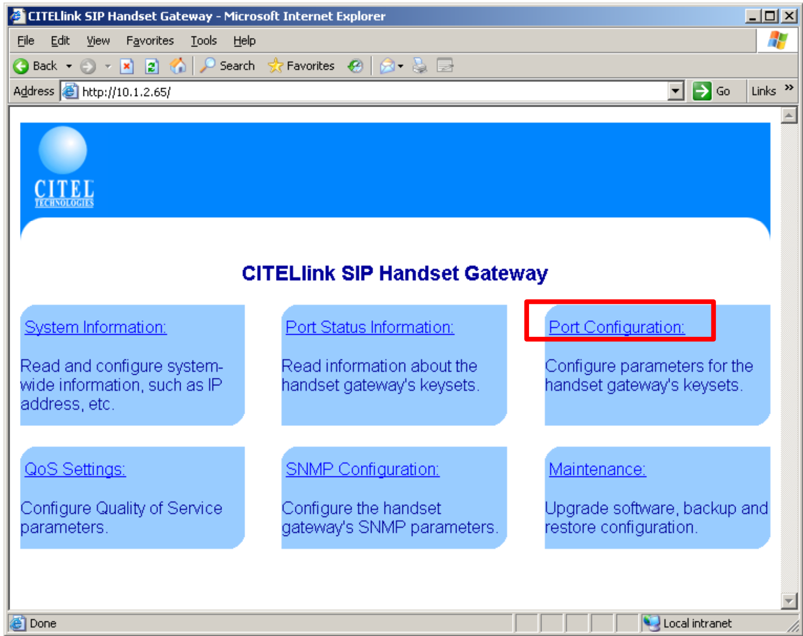
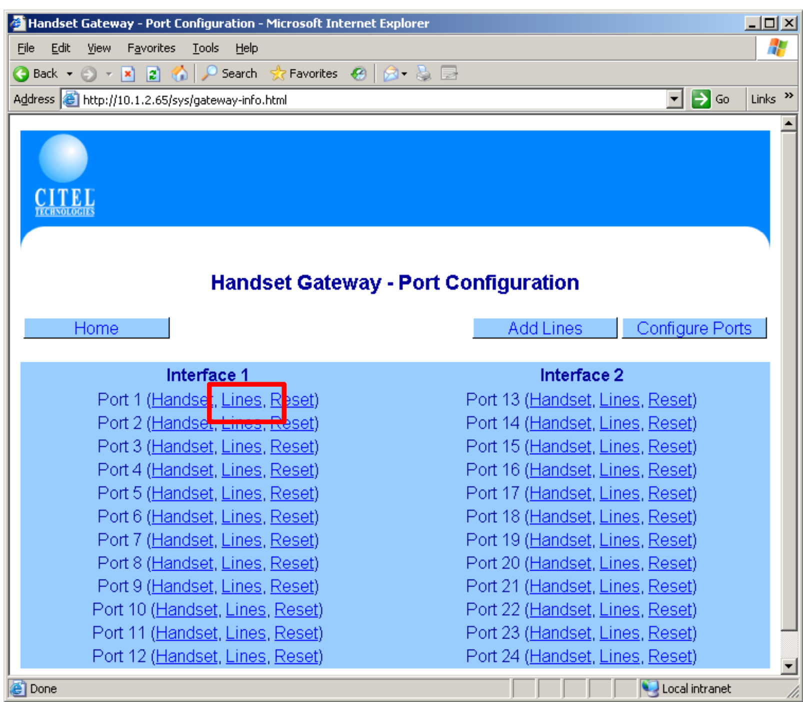
Most of the configuration is accomplished through a Web browser. Layer-2 (802.1p) and VLAN configuration are conducted through the serial interface on the *CITEL*link SIP Handset Gateway.

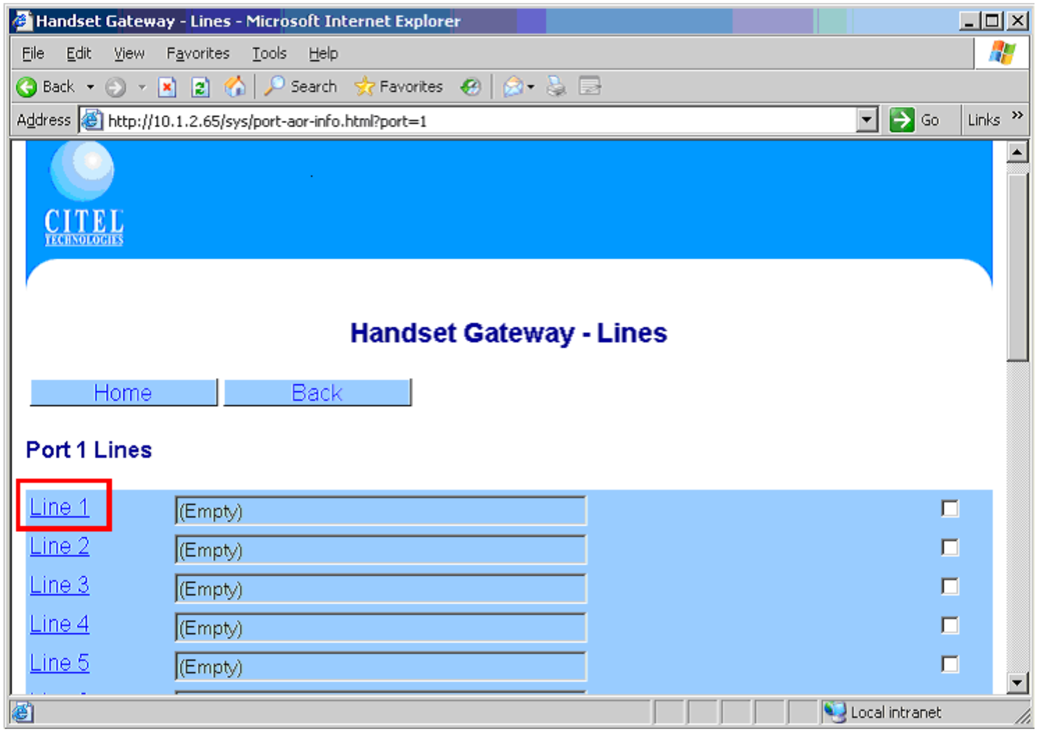
4.1. Web Configuration of the CITELlink SIP Handset Gateway

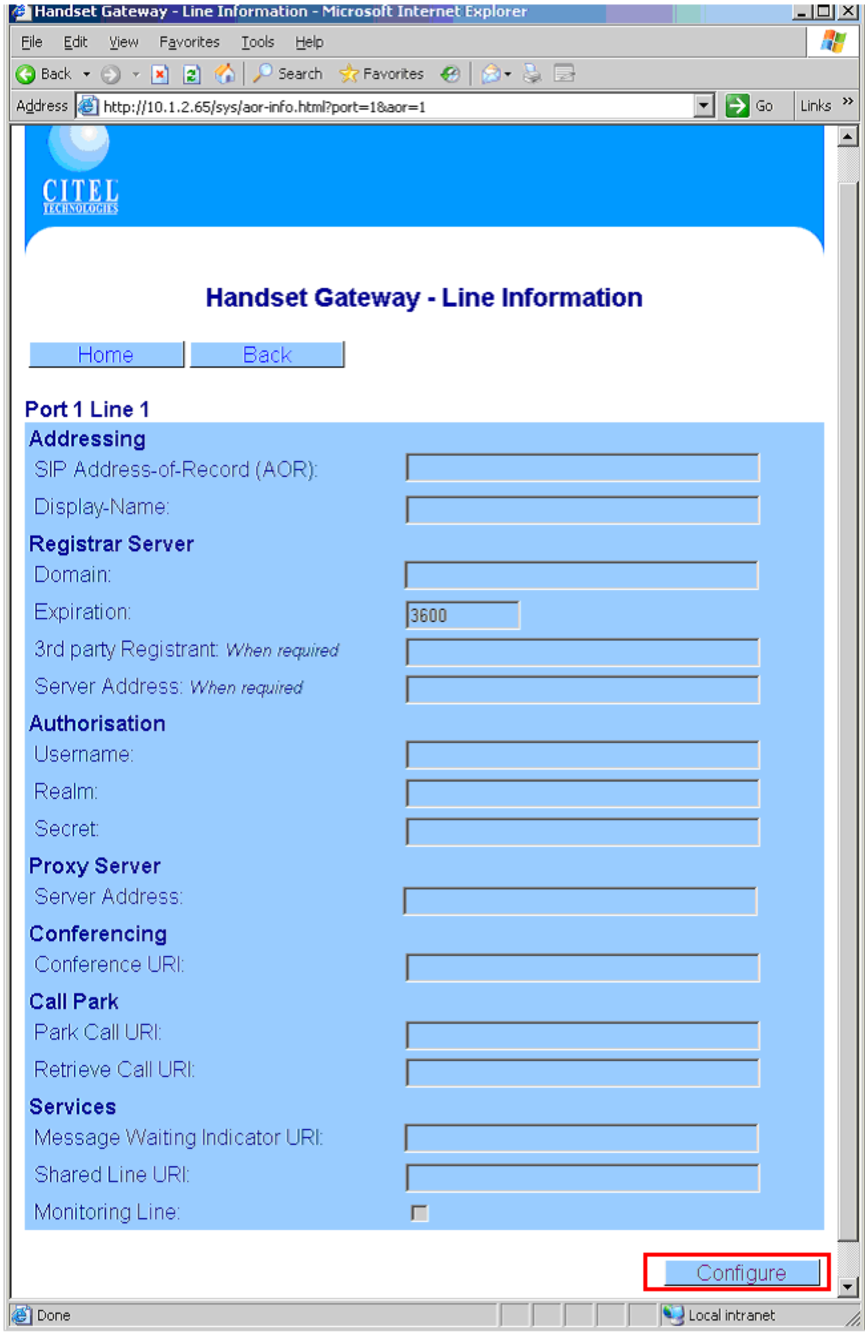
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>In the sample network, the CITELlink SIP Handset Gateway was set to acquire an IP address from the DHCP server (default). Begin configuration by entering the IP address of the CITELlink SIP Handset Gateway into the Web browser. The CITELlink SIP Handset Gateway in the sample network has IP address 10.1.2.65. Enter the appropriate User Name and Password to log into the system.



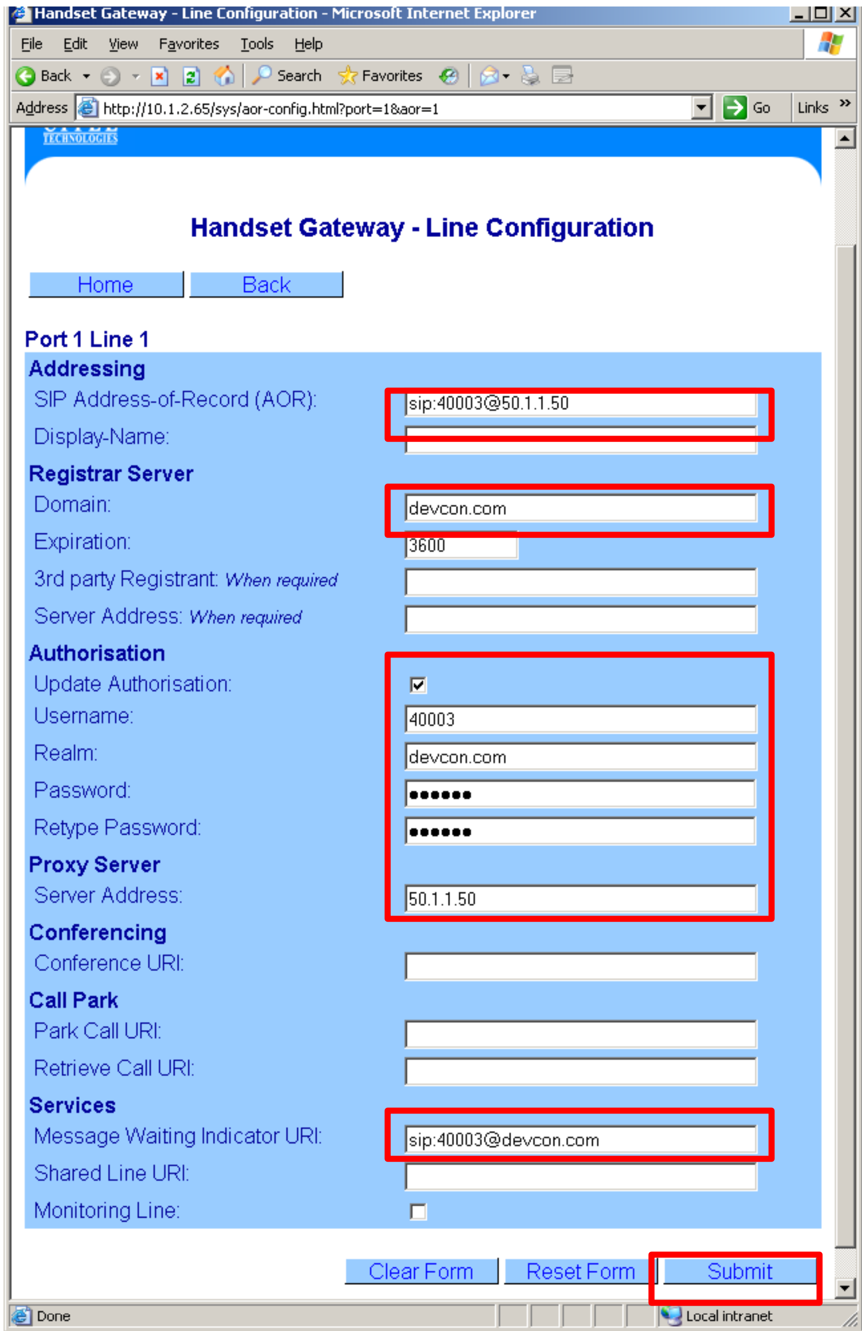
</td></tr>
</table>

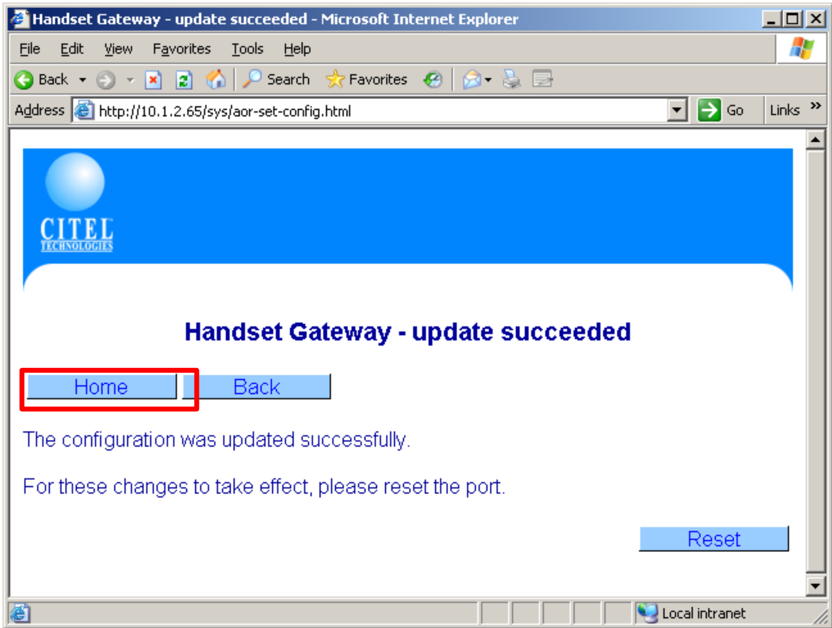
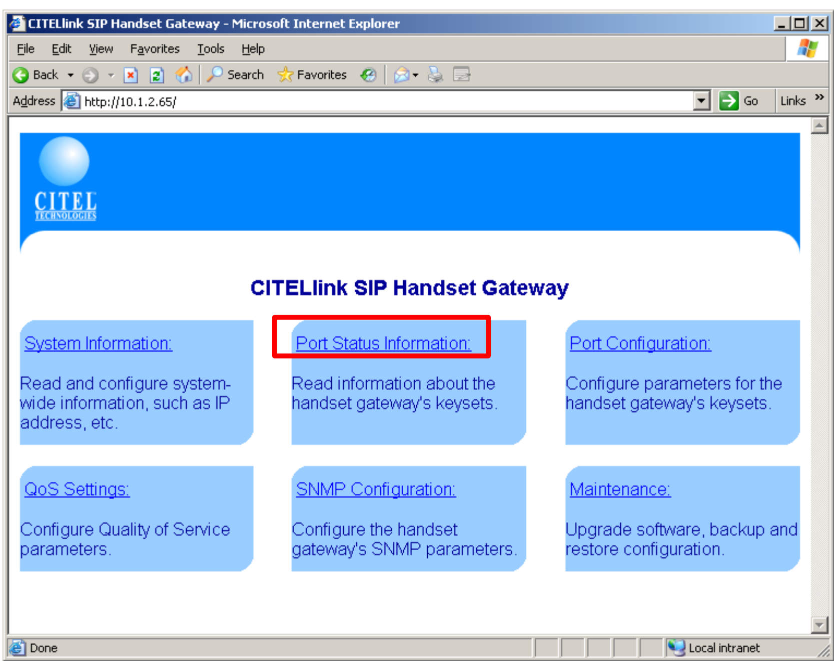
Step	Description
2.	After logging in, the following **CITELlink SIP Handset Gateway** screen will be displayed. Click on **Port Configuration** to begin configuration. 
3.	From the port that needs to be configured, select the **Lines** option. Port 1 in the sample network is connected to the Nortel Meridian M2616 telephone. 

Step	Description
4.	Select **Line 1** from the **Handset Gateway – Lines** menu. 

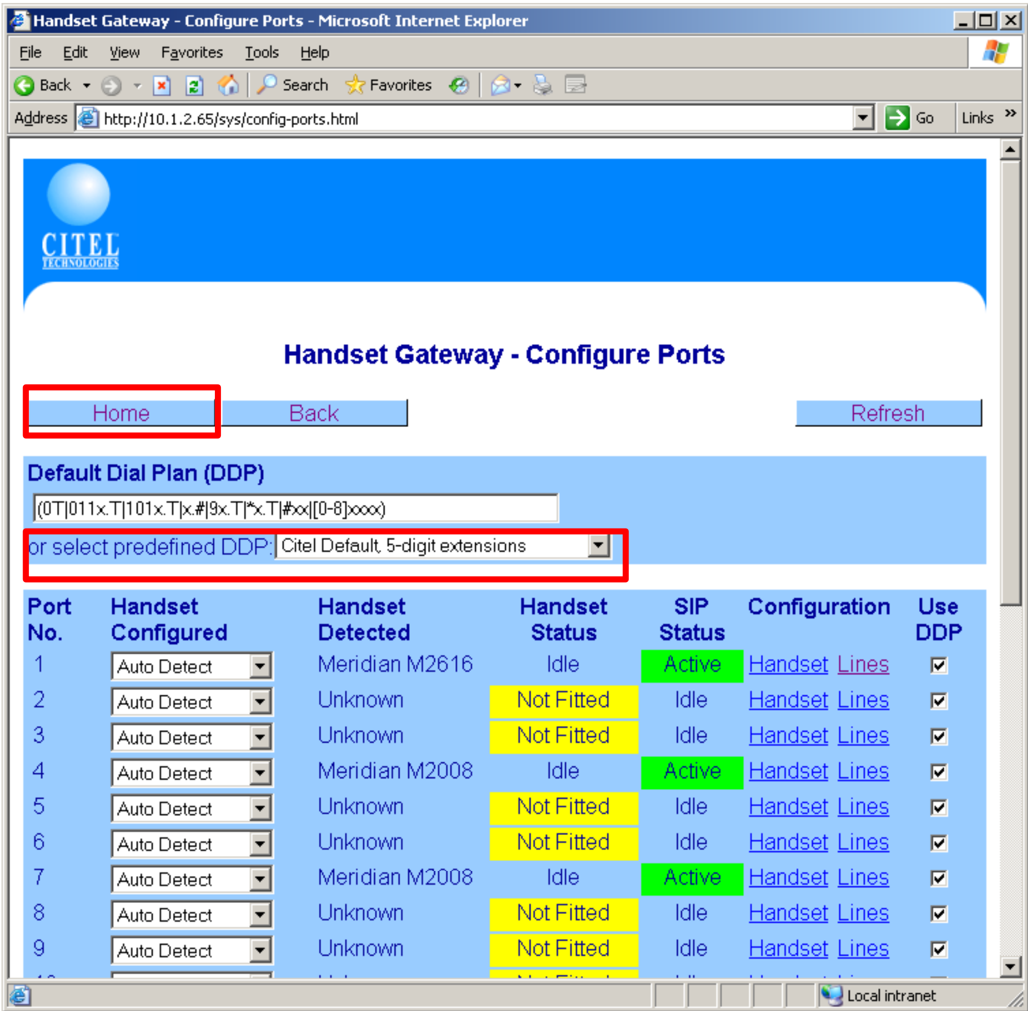
Step	Description
5.	Click **Configure** to begin configuration. 

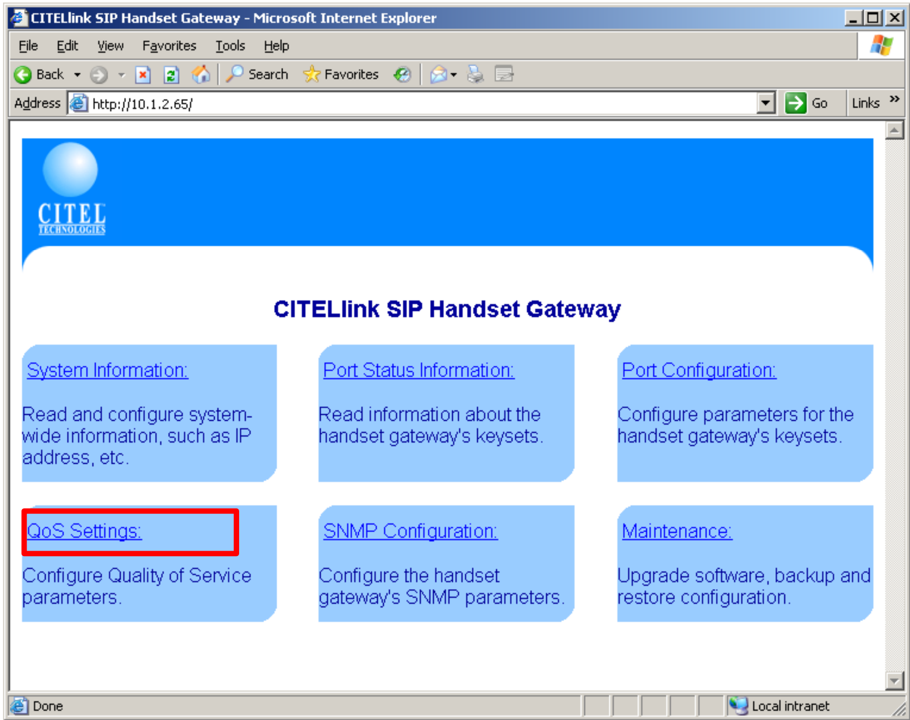
Step	Description
6.	Enter the appropriate information in the highlighted fields. The Avaya SIP Enablement Services (SES) server in the sample configuration is configured with an IP address ***50.1.1.50*** and a domain of ***devcon.com***. ***40003*** is the extension number that is assigned to the Nortel Meridian telephone connected to this port. Click **Submit** to update the changes.

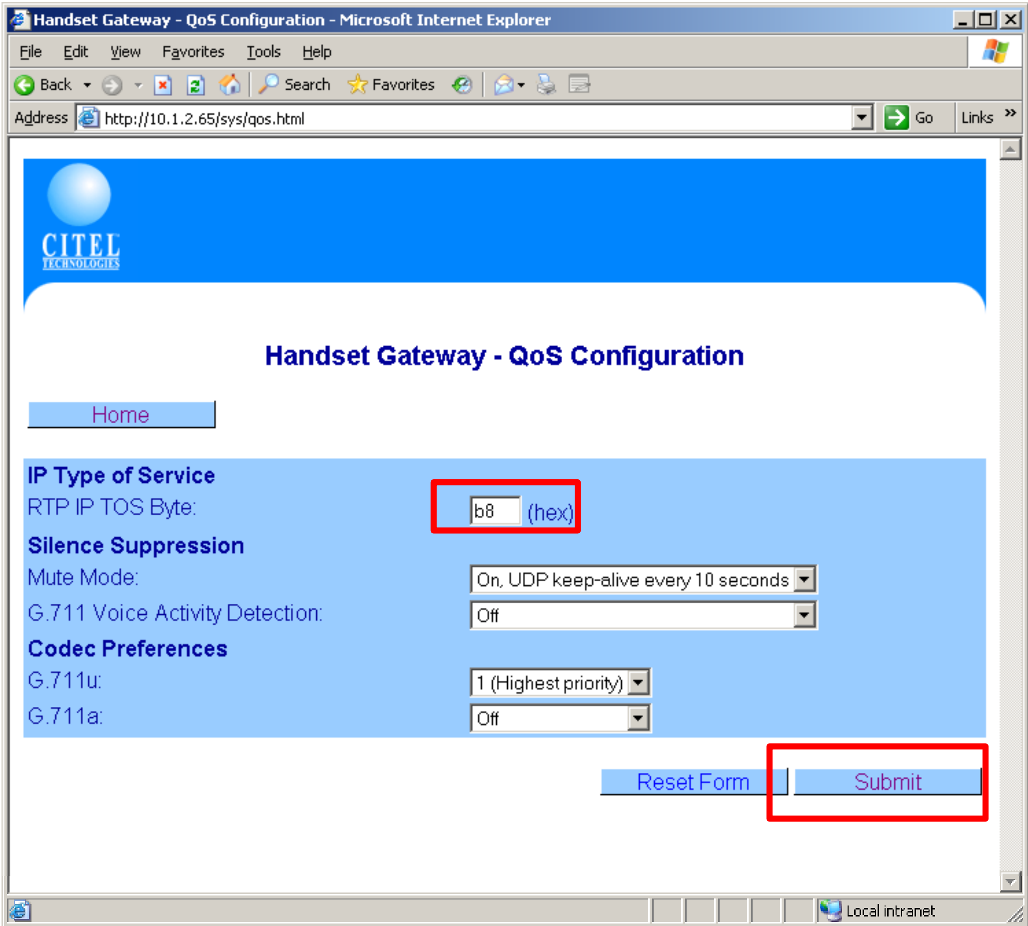


Step	Description
7.	Click **Home** to return to the main **CITELlink SIP Handset Gateway** menu. 
8.	Repeat Steps 2-6 for each telephone that is connected to the *CITEL*link SIP Handset Gateway.
9.	From the **CITELlink SIP Handset Gateway** menu, click on **Port Status Information** to configure the dial plan. 

Step	Description
10.	The sample configuration uses 5-digit dialing. From the **select predefined DDP** drop-down dialog box, select ***Citel Default 5-digit extensions***. Click on **Submit** (not shown) at the lower right hand corner to save the changes, then click **Home** to return to the main **CITELlink SIP Handset Gateway** menu.



Step	Description
11.	From the main **CITELlink SIP Handset Gateway** menu, click on **QoS Settings** to configure Layer-3 (DiffServ) parameters. 

Step	Description
12.	In the **Handset Gateway – QoS Configuration** screen, enter the appropriate **RTP IP TOS Byte value**. The sample network uses hex value ***b8***, which equates to the ASCII value 46 configured in Section 3.3. Click **Submit** to update the changes. 
13.	Close the Web browser to exit.

4.2. Layer-2 QoS and VLAN Configuration of the CITELlink SIP Handset Gateway

Layer-2 (802.1p) and VLAN configuration is conducted through the serial interface via a straight through cable.

Step	Description
1.	Use a terminal emulation program to access the serial interface on the *CITEL*link SIP Handset Gateway with the following settings. Speed 115200 Data bits 8 Parity N Stop bits 1 Flow control none
2.	Once connected via the serial interface, the **DCP [support]** prompt will appears. Type in ***su*** and enter the appropriate password to log into the system. DCP [support]$ su Password:*****
3.	Enter ***vlan/*** at the **DCP*[sys]** prompt. This will take the system into the vlan setting prompt **DCP*[vlan]**. In the sample configuration, the Layer-2 (802.1p) priority is set to 6 and the *CITEL*link SIP Handset Gateway is on VLAN "default" which has a VLAN tag of 1. Therefore, enter "***set 6 1***" at the prompt. The system will display back the information entered. DCP*[sys]$ vlan/ DCP*[vlan]$ set 6 1 info : vlanEnabled : TRUE info : vlanPriority : 6 info : vlanTag : 1
4.	Closed the terminal emulation program to exit.

5. Interoperability Compliance Testing

The interoperability compliance testing focused on assessing the ability of the *CITEL*link SIP Handset Gateway to interoperate with Avaya SIP Enablement Services in supporting the Nortel Meridian telephones.

5.1. General Test Approach

The general test approach was to verify that the Citel Technologies *CITEL*link SIP Handset Gateway could successfully interoperate with Avaya Communication Manager and Avaya SIP Enablement Services in supporting Nortel Meridian telephones. Basic features supported by the Nortel Meridian telephone such as hold, conference and transfer were exercised as well as many of the Avaya Feature Name Extensions such as Call-Pickup, Conferencing, Find-me, and Call Forwarding.

5.2. Test Results

Through the use of the *CITEL*link SIP Handset Gateway, the Nortel Meridian telephones were successful in providing all basic features supported by the telephone and were able to access the Avaya Feature Name Extensions. Both Layer-2 (802.1p) and Layer-3 (DiffServ) tagging for the *CITEL*link SIP Handset Gateway were verified through sniffer capture. There is however an issue with native conferencing when executed between Avaya and Nortel telephones. Some telephones may not receive two-way audio within the conference depending on the type of telephone, who initiated the conference and if shuffling is enabled or disabled.

6. Verification Steps

The following steps may be used to verify the configuration:

- Place and receive calls from the Nortel Meridian and Avaya telephones.
- Log in to the Avaya SIP Enablement Service (SES) Server via the Web browser. The registered users field under Users will also show all registered SIP users.
- Verify that all ports on the *CITEL*link SIP Handset Gateway that have a Nortel Meridian telephone connected to it are active under the SIP Status column shown as in Section 4.1 Step 10.

7. Support

For technical support on the *CITEL*link SIP Handset Gateway, contact Citel Technologies, Inc. at http://www.citel.com/service_support/.

8. Conclusion

These Application Notes described the administration steps required to configure the Citel Technologies *CITEL*link SIP Handset Gateway to interoperate with Avaya SIP Enablement Services (SES) and Avaya Communication Manager for Nortel Meridian telephones. With the exception of native conferencing, the *CITEL*link SIP Handset Gateway provided all tested basic and extended features to the Nortel Meridian telephones and can interoperate successfully with Avaya Communication Manager and Avaya SIP Enablement Services.

9. Additional References

[1] *Administrator Guide for Avaya Communication Manager*, Doc # 03-300509, Issue 1, June 2005
[2] *Avaya Communication Manager Advanced Administration Quick Reference*, Doc # 03-300364, Issue 2, June 2005 Release 3.0
[3] *Expanded Meet-me Conference (EMMC) version 1.0 Installation and Troubleshooting Guide for the S8500*, Doc # 04-300527, Issue 1, June 2005
[4] *Avaya IA 770 INTUITY AUDIX Messaging Application*, Doc # 585-313-159, Issue 4, December 2003
[5] *Converged Communications Server R3.0 Installation and Administration Guide (SIP Enablement Services R3.0)*, Doc # 555-245-705, Issue 5.1, July 2005
[6] *SIP Support in Release 3.0 of Avaya Communication Manager*, Doc # 555-245-206, Issue 5.1, July 2005
[7] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, Doc # 210-100-500, Issue 9, June 2005
[8] *CITELlink SIP Handset Gateway Installation & Configuration Guide version 2.2*, Document number 1764

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

Product documentation for Citel Technologies products may be found at
http://www.citel.com

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