Application Notes for the Citel Gateway 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 digital telephones connected to a Citel Gateway. Citel Gateway is a SIP-based VoIP appliance. During compliance testing, digital telephones connected to Citel Gateway 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 three-way conference, transfers, hold, etc. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services (SES) 3.1.1, and digital telephones connected to a Citel Gateway. Citel Gateway is a SIP-based VoIP appliance. During compliance testing, digital telephones connected to the Citel Gateway successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features such as three-way conference, transfers, hold, etc. During compliance testing, Nortel Meridian, Nortel Norstar, NEC and Panasonic model TDM telephones were connected to the Citel Gateway and these telephones appeared as SIP endpoints to Avaya SES. TDM telephones connected to Citel Gateway will be referred to as Citel Gateway Handsets. These application notes do not cover configuration of Citel Gateway Handsets.

Figure 1 illustrates a sample configuration consisting of a pair of Avaya S8710 Media Servers, an Avaya G650 Media Gateway, an Avaya SIP Enablement Services (SES) server, and the Citel Gateways. Avaya Communication Manager is installed on the Avaya S8710 Media Servers. The solution described herein is also extensible to other Avaya Media Servers and Media Gateways. For completeness, Avaya 4600 Series SIP IP Telephones, Avaya 4600 Series H.323 IP Telephones, and Avaya 6400 and 8400 Series Digital Telephones, are included in **Figure 1** to demonstrate calls between the SIP-based Citel Gateway and Avaya SIP, H.323, and digital telephones. The analog PSTN telephone is also included to demonstrate calls routed by Avaya Communication Manager between the Citel Gateway and the PSTN.

Citel Gateway originates a call by sending a call request (SIP INVITE message) to the Avaya SES server. The Avaya SES server 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 server for delivery to destination SIP telephone. Otherwise, Avaya Communication Manager routes the call to the PSTN, a local Avaya H.323, digital, or analog telephone, an adjunct, a vector, a hunt group, etc., depending on the destination number.

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

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 **[3**] and **[4]**.

Notes:

- During compliance testing, only the functionality of the Citel Gateway was verified.
- Citel Gateway only supports one phone range at a time. Following configurations were verified:
  - **Test 1** in **Figure 1** used Meridian on Citel Gateway 1 and Norstar on Citel Gateway 2.
  - **Test 2** in **Figure 1** used NEC on Citel Gateway 1 and Panasonic on Citel Gateway 2.

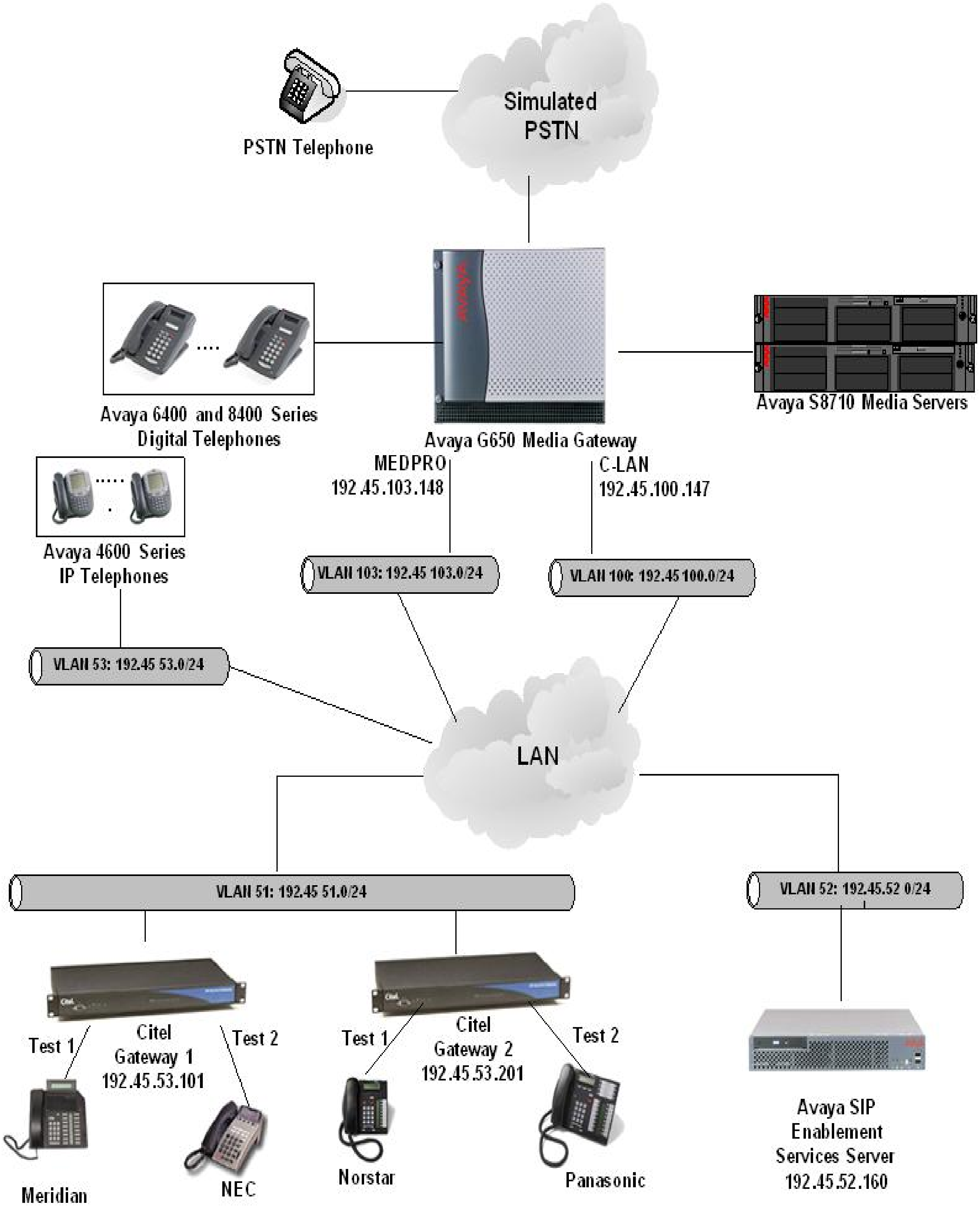


Figure 1: Sample configuration

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8710 Media Server	Avaya Communication Manager 3.1.2 (R013x.01.2.632.1)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	HW12 FW 31
TN799DP C-LAN Interface	HW01 FW 17
TN2302AP IP Media Processor	HW20 FW 112
Avaya SIP Enablement Services Server	SES 3.1.1(R03.1.1-03.1.114.0)
Avaya 4600 Series IP Telephones	2.3 (4602SW H.323) 2.5 (4625SW H.323) 2.2.2 (4610SW SIP)
Avaya 6400 and 8400 Series Digital Telephones	-
Avaya Analog Telephone	-
Citel Gateway	3.61.4d07 HSGW

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 IP codec sets, 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. Citel and other SIP telephones are configured as off-PBX telephones in Avaya Communication Manager. Avaya Communication Manager does not directly control an off-PBX telephone but its features and calling privileges can be applied to it by associating a local, on-PBX telephone with the off-PBX telephone. 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. Throughout the rest of this document, on-PBX telephones associated with SIP telephones in such a manner will be referred to as Outboard Proxy SIP (OPS) stations.

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  10
                                OPTIONAL FEATURES

     G3 Version: V13
       Location: 1                            RFA System ID (SID): 1
       Platform: 8                            RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 44000 908
                                      Maximum Stations: 36000 410
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 5     0
                    Maximum Off-PBX Telephones -   OPS: 200   50
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

Step	Description
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  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 200   148
           Maximum Concurrently Registered IP Stations: 1000  2
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 200   153

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 1     1
                    Maximum G250/G350/G700 VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 2     0
          Maximum TN2602 Boards with 320 VoIP Channels: 2     1
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
        (NOTE: You must logoff & login to effect the permission changes.)
```

3.2. IP Codec Set

This section describes the steps for administering a 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 <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.729AB** were used and **Media Encryption** was set to **none** as encryption is currently not supported for SIP telephony.

```
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.729AB            n          2         20
 3:
 4:
 5:
 6:
 7:

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

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.

Step	Description
1.	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 the **devconnect.com**. 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.

```
change ip-network-region 2                                      Page  1 of  19
                               IP NETWORK REGION
  Region: 2
Location:         Authoritative Domain: devconnect.com
    Name:
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? y
   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
```

Step	Description
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**.

```
Page 3 of 19
                     Inter Network Region Connection Management

 src dst  codec  direct   Total           Video                         Dyn
 rgn rgn   set    WAN  WAN-BW-limits  WAN-BW-limits Intervening-regions CAC IGAR
 2   1     2      y        :NoLimit                                         n
 2   2     2
 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. IP Node Names

This section describes the steps for setting an IP node name for Avaya SES in Avaya Communication Manager.

Step	Description
1.	Enter the **change node-names ip** command and add a node name for Avaya SES along with its IP address.

```
change node-names ip                                                   Page 1 of 1
                                  IP NODE NAMES
    Name              IP Address
CLAN-1A06           192.45 .100.147
MEDPRO-1A13         192.45 .103.148
SES                 192.45 .52 .160
```

3.5. 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.	Issue 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**. • **Near-end Node Name** - Set to CLAN name as displayed in **Section 3.4**. • **Far-end Node Name** - Set to Avaya SES name configured in **Section 3.4**. • **Far-end Network Region** - Set to the region configured in **Section 3.3**. • **Far-end Domain** - Set to the **devconnect.com**. This should match the **SIP Domain** value in **Section 4**, **Step 2**. • **DTMF over IP –** Set to **rtp-payload**. • **Direct IP-IP Audio Connections –** Set to **y** for shuffling.

```
add signaling-group 10                                          Page    1  of  5
                                SIGNALING GROUP

 Group Number: 10            Group Type: sip
                       Transport Method: tls





   Near-end Node Name: CLAN-1A06            Far-end Node Name: SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 2
       Far-end Domain:devconnect.com

                                            Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```

3.6. 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.

Step	Description
1.	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 3.5**. • **TAC** (Trunk Access Code) – Set to any available trunk access code. • **Signaling Group** – Set to the **Group Number** field value configured in **Section 3.5**. • **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 trunks for the duration of the call. The license file installed on the system controls the maximum permitted.*

```
add trunk-group 10                                             Page    1  of  21
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: SIP-SES-DevCon1             COR: 1        TN: 1        TAC: 110
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                          Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 10
                                                     Number of Members: 150
```

3.7. 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 Citel Gateway Handsets.

Step	Description
1.	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: • **Type –** Set to **6408D+**. • **Port –** Set to **X** for Administration Without Hardware (AWOH) as SIP stations are not directly connected to Avaya Communication Manager. • **Name –** Enter any descriptive name. • **IP Softphone** – Set to **y**.

```
add station 54020                                               Page   1 of   3
                                     STATION

Extension: 54020                              Lock Messages? n          BCC: 0
     Type: 6408D+                             Security Code:             TN: 1
     Port: X                                Coverage Path 1:            COR: 1
     Name: CGH54020                         Coverage Path 2:            COS: 1
                                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2                 Personalized Ringing Pattern: 1
             Data Module? n                             Message Lamp Ext: 54008
            Speakerphone: 2-way                      Mute Button Enabled? y
        Display Language: english

                                                           Media Complex Ext:
                                                                IP SoftPhone?y
```

2.	Proceed to **Page 3** of the **STATION** form and add the required number of **call-appr** entries in **BUTTON ASSIGNMENT** field. The number of call appearances should match the **Call Limit** field value in **Step 4**.

```
add station 54020                                               Page   3 of   3
                                     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:                                      7:
 4:                                      8:
```

Step	Description
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 in **Step 1**. • **Application –** Set to **OPS**. • **Phone Number –** Enter the number that the Citel Gateway Handset 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 3.6**.

```
add off-pbx-telephone station-mapping 54008                    Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial  Phone Number      Trunk       Configuration
 Extension                 Prefix                  Selection   Set
 54008         OPS              - 54008              10        1
```

Step	Description
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**.

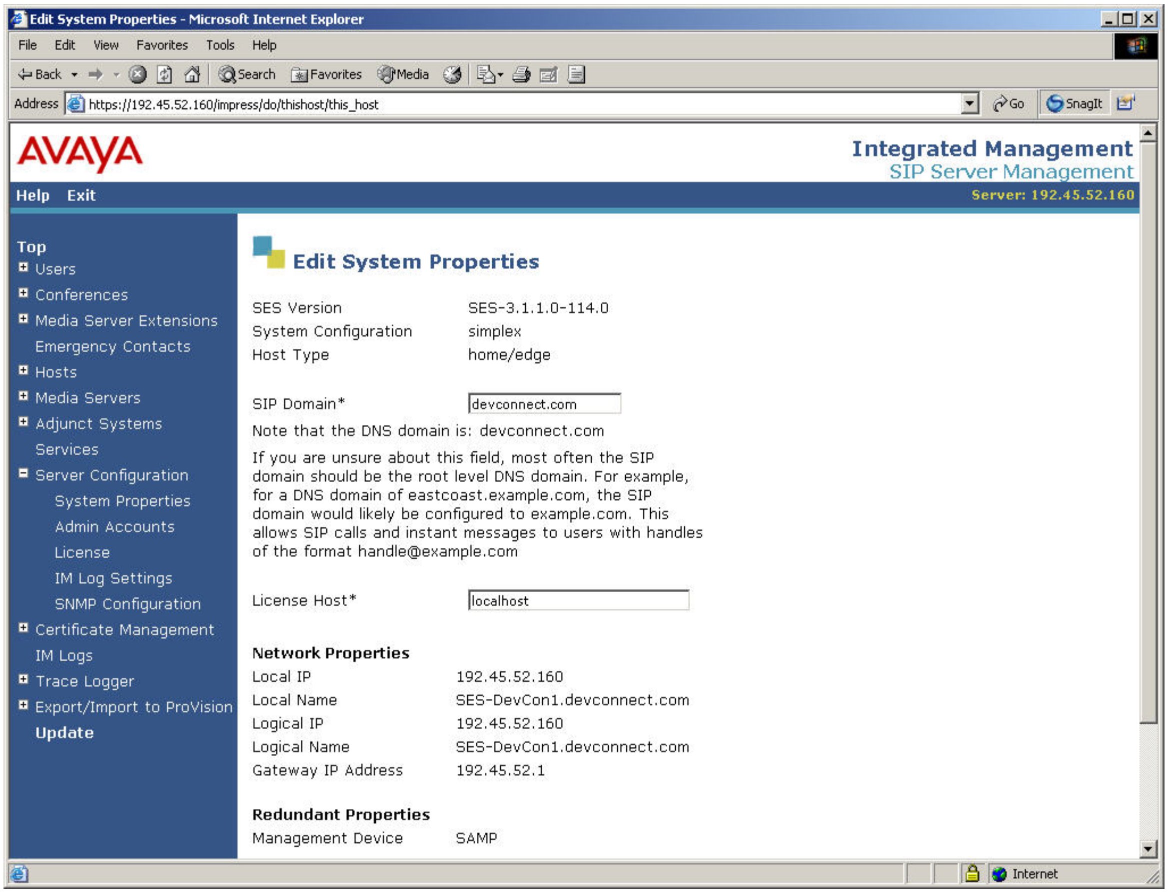
```
add off-pbx-telephone station-mapping 54008                    Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping    Calls Allowed       Bridged
 Extension     Limit      Mode       Allowed             Calls

 54008         2          both       all                 both         1
```

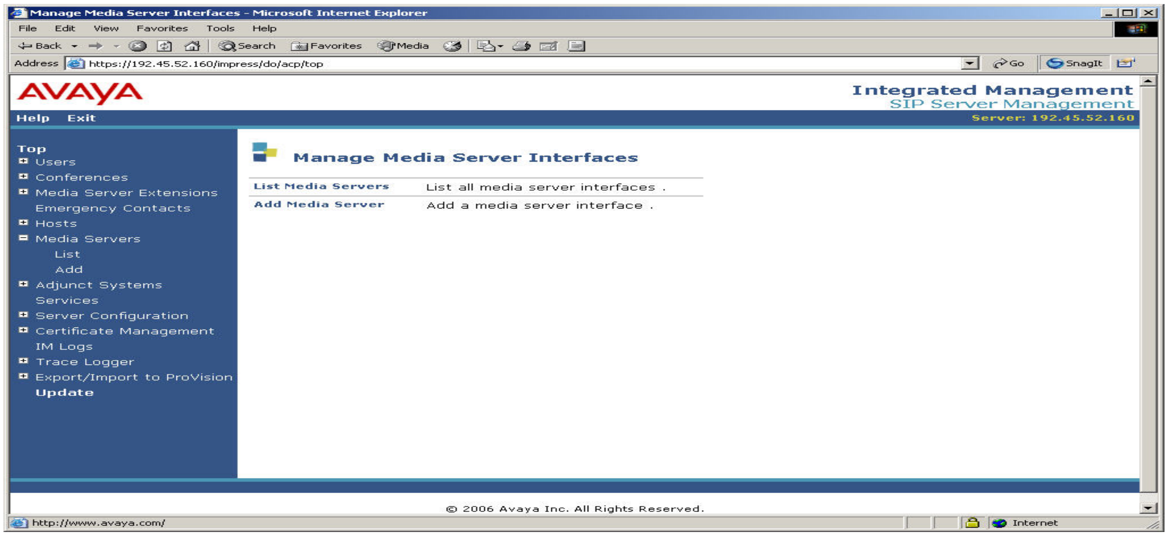
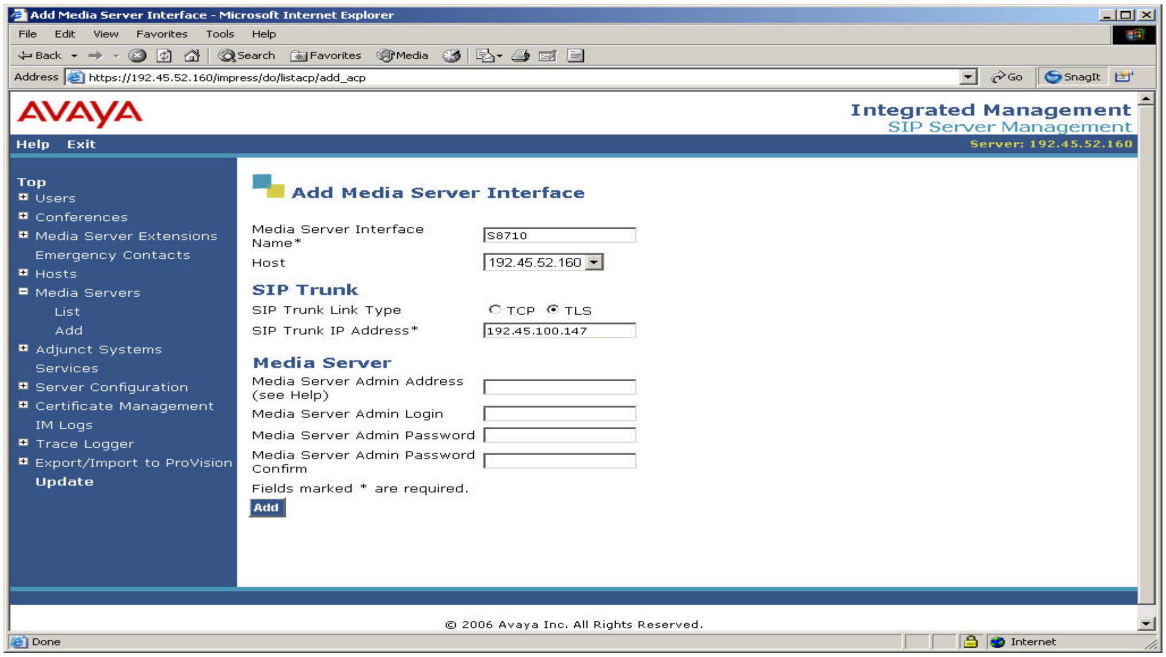
Step	Description
5.	Repeat **Steps 1 - 4** as necessary to administer additional OPS stations and associations for Citel Gateway Handsets.

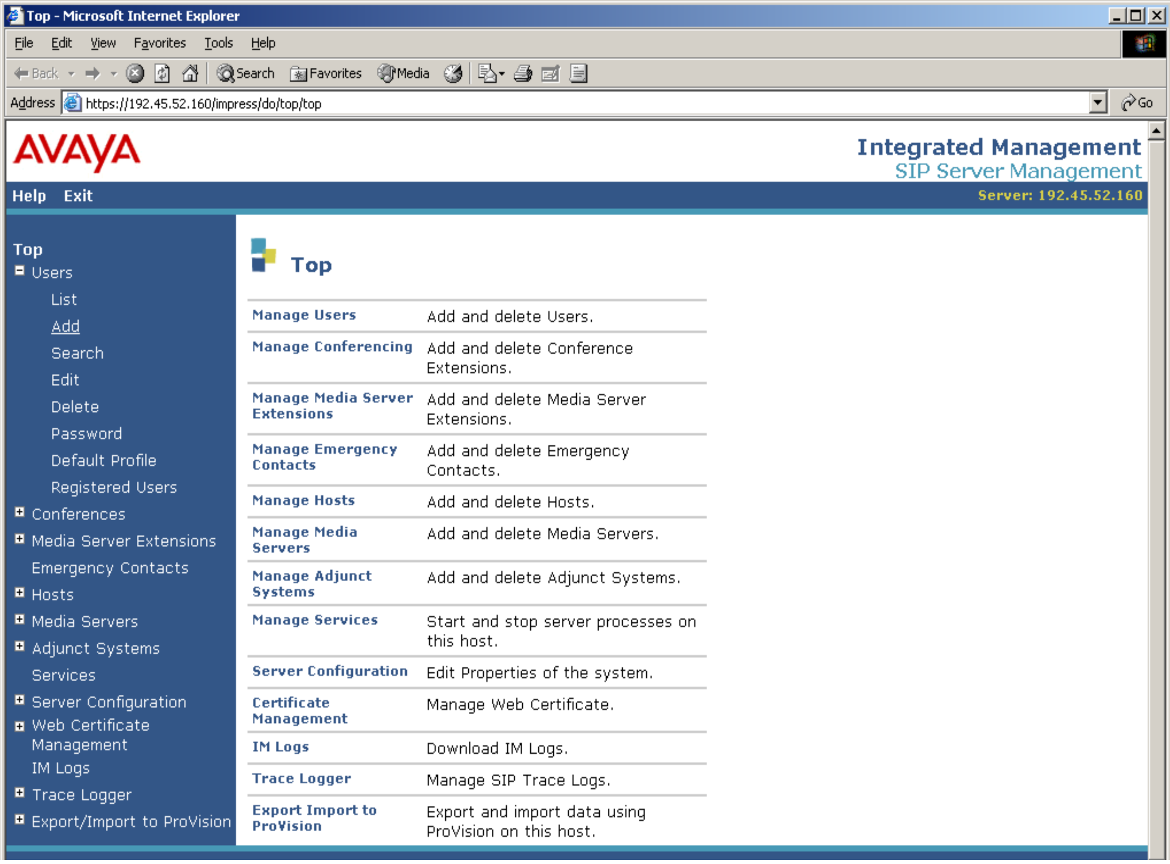
4. Configure Avaya SIP Enablement Services

This section describes the steps for creating 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 Citel Gateway will register with Avaya SES using the SIP user accounts. A separate SIP account will be created for each TDM telephone connected to Citel Gateway.

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 SES server>/admin for the URL, and log in with the appropriate credentials. Click on the **Launch Administration Web Interface** link upon successful login.
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 **Sections 3.3** and **3.5**.  https://192.45.52.160/impress/do/thishost/this_host Server: 192.45.52.160 **Edit System Properties** SES Version SES-3.1.1.0-114.0 System Configuration simplex Host Type home/edge SIP Domain* devconnect.com Note that the DNS domain is: devconnect.com If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com License Host* localhost **Network Properties** Local IP 192.45.52.160 Local Name SES-DevCon1.devconnect.com Logical IP 192.45.52.160 Logical Name SES-DevCon1.devconnect.com Gateway IP Address 192.45.52.1 **Redundant Properties** Management Device SAMP

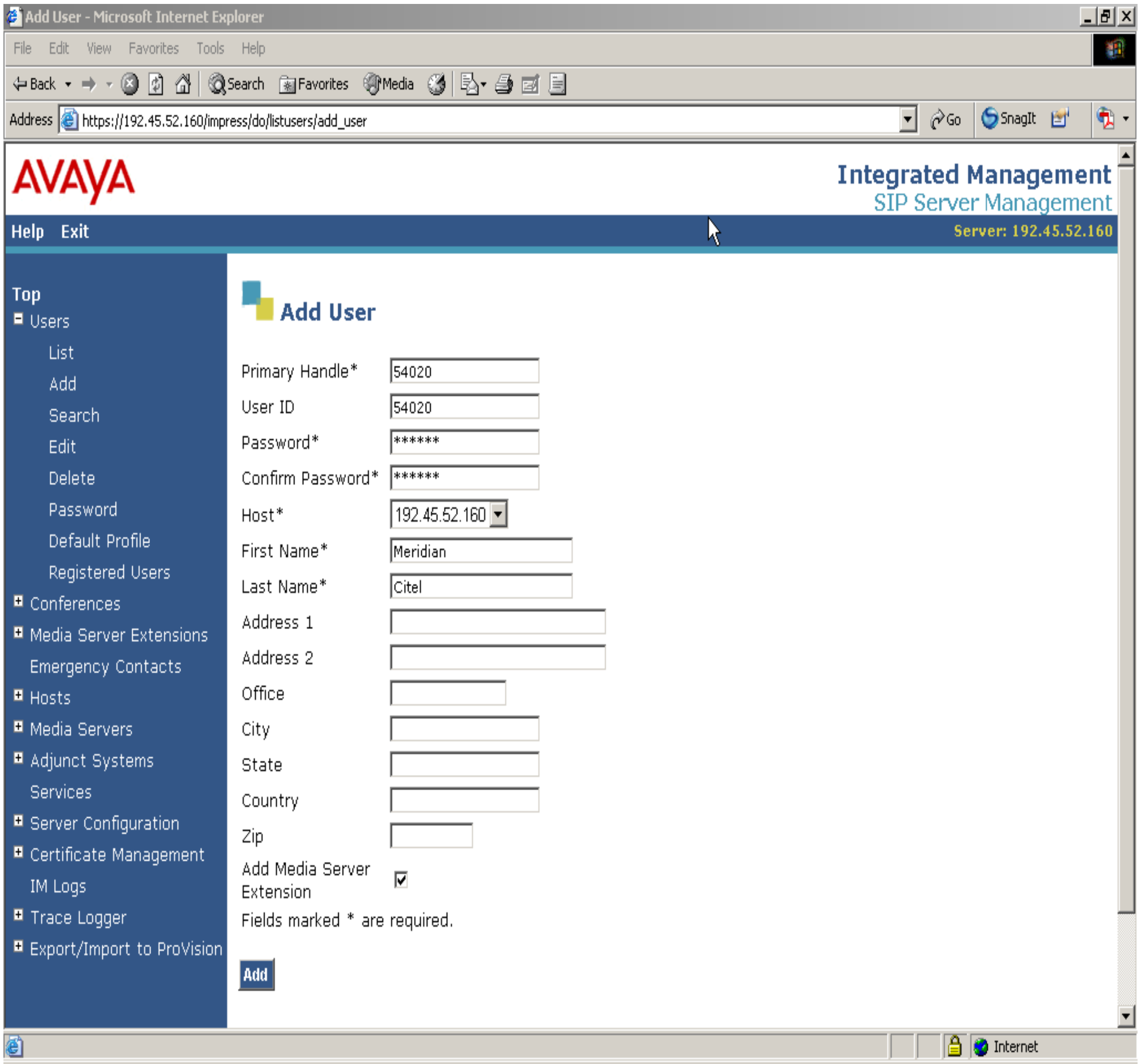
Step	Description
3.	To enable secure SIP trunking between Avaya SES and Avaya Communication Manager, add a media server corresponding to Avaya Communication Manager from the **SIP Server Management** page: • Click the + sign to expand the options under **Media Servers**. • Click **Add** in left pane under **Media Servers**.  Manage Media Server Interfaces List Media Servers — List all media server interfaces . Add Media Server — Add a media server interface .
4.	At the **Add Media Server Interface** page, provision **SIP Trunk** parameters as follows for connectivity to Avaya Communications Manager: • **SIP Trunk Link Type** - Set to the **Transport Method** field value in **Section 3.5**. • **SIP Trunk IP Address** - Set to the CLAN IP address as displayed in **Section 3.4**. • Click **Add** when finished and then click **Continue** on the confirmation page [not shown].  Add Media Server Interface Media Server Interface Name* S8710 Host 192.45.52.160 SIP Trunk SIP Trunk Link Type ○ TCP ● TLS SIP Trunk IP Address* 192.45.100.147 Media Server Media Server Admin Address (see Help) Media Server Admin Login Media Server Admin Password Media Server Admin Password Confirm Fields marked * are required. Add

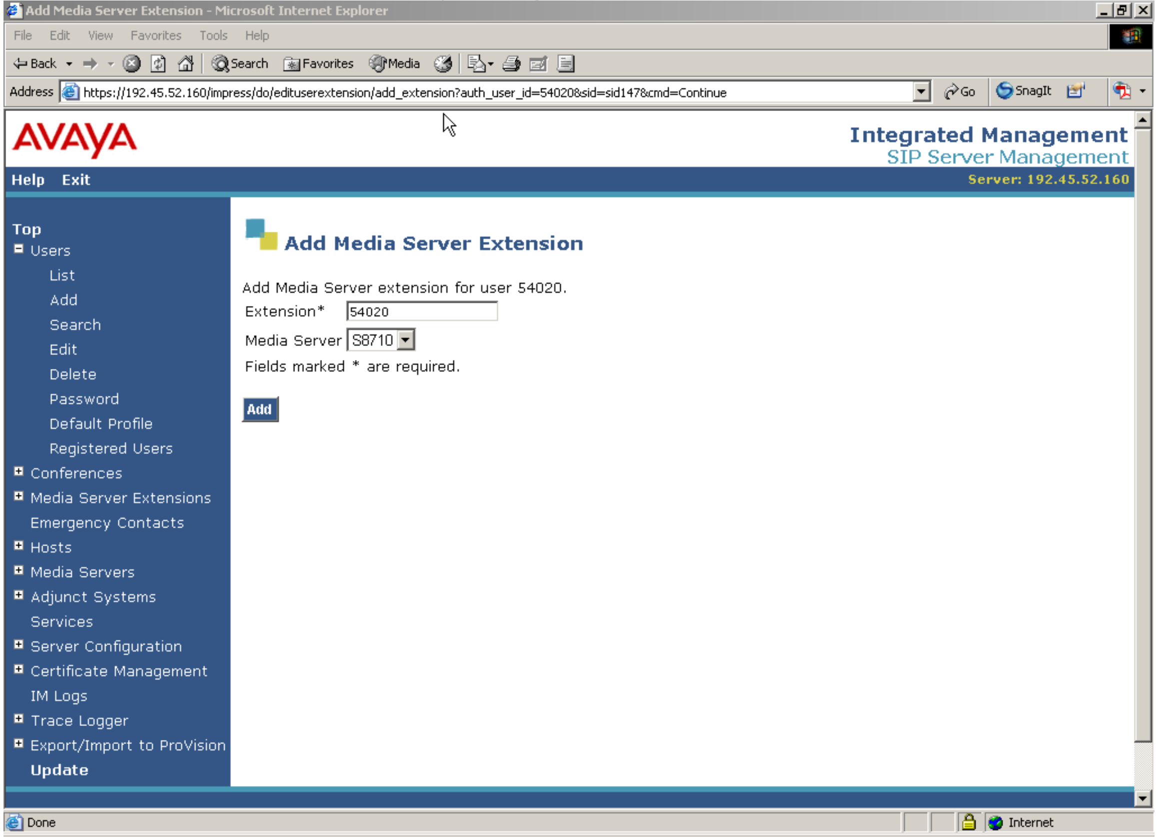
Step	Description
5.	In the left pane of the **SIP Server Management** page, expand **Users** and click **Add**. 

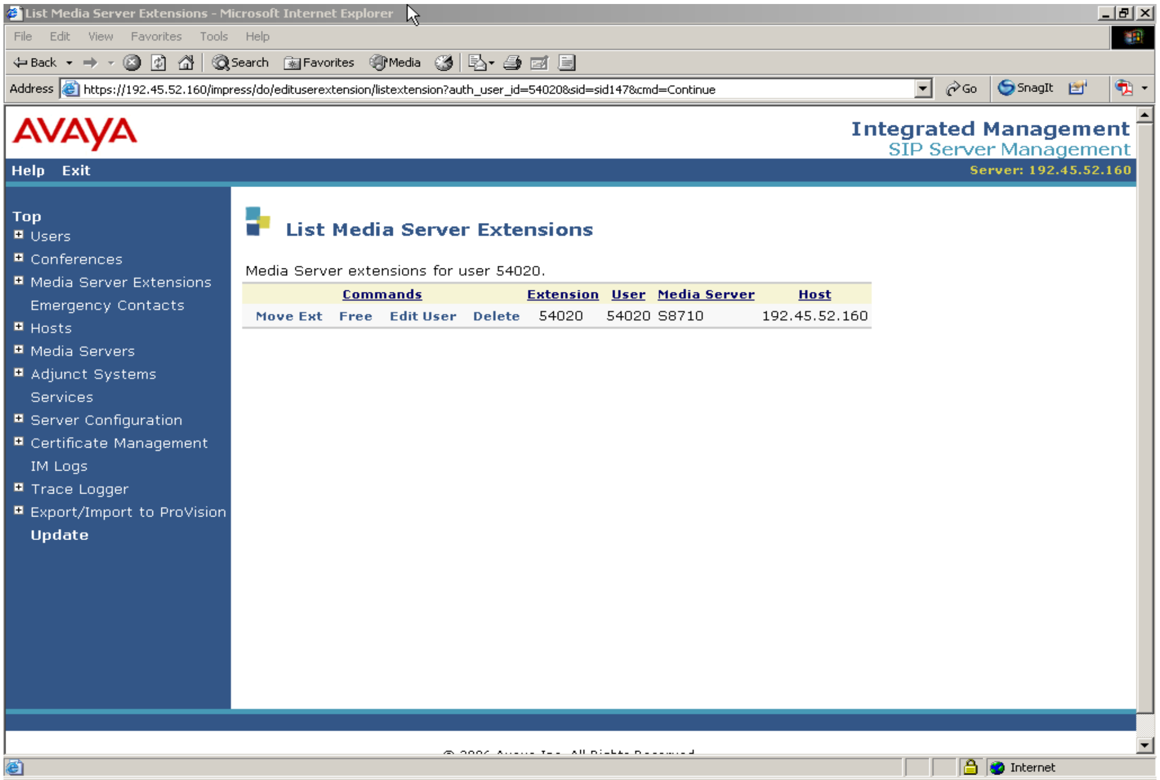
Top
- Users
  - List
  - Add
  - Search
  - Edit
  - Delete
  - Password
  - Default Profile
  - Registered Users
- Conferences
- Media Server Extensions
- Emergency Contacts
- Hosts
- Media Servers
- Adjunct Systems
- Services
- Server Configuration
- Web Certificate Management
- IM Logs
- Trace Logger
- Export/Import to ProVision

Top

Manage Users	Add and delete Users.
Manage Conferencing	Add and delete Conference Extensions.
Manage Media Server Extensions	Add and delete Media Server Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Manage Hosts	Add and delete Hosts.
Manage Media Servers	Add and delete Media Servers.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Services	Start and stop server processes on this host.
Server Configuration	Edit Properties of the system.
Certificate Management	Manage Web Certificate.
IM Logs	Download IM Logs.
Trace Logger	Manage SIP Trace Logs.
Export Import to ProVision	Export and import data using ProVision on this host.

Step	Description
6.	At the **Add User** page, configure the following: • **Primary Handle –** Enter the phone number of the Citel Gateway Handset. This number was configured in **Section 3.7**, **Step 1**. • **User ID** – Set to any descriptive name. • **Password** and **Confirm Password –** Specify a password that the Citel Gateway Handset will use to register with Avaya SES. • **Host –** Select the IP address or Fully Qualified Domain Name (FQDN) of the 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].  Add User Primary Handle* 54020 User ID 54020 Password* ****** Confirm Password* ****** Host* 192.45.52.160 First Name* Meridian Last Name* Citel Address 1 Address 2 Office City State Country Zip Add Media Server Extension ☑ Fields marked * are required.

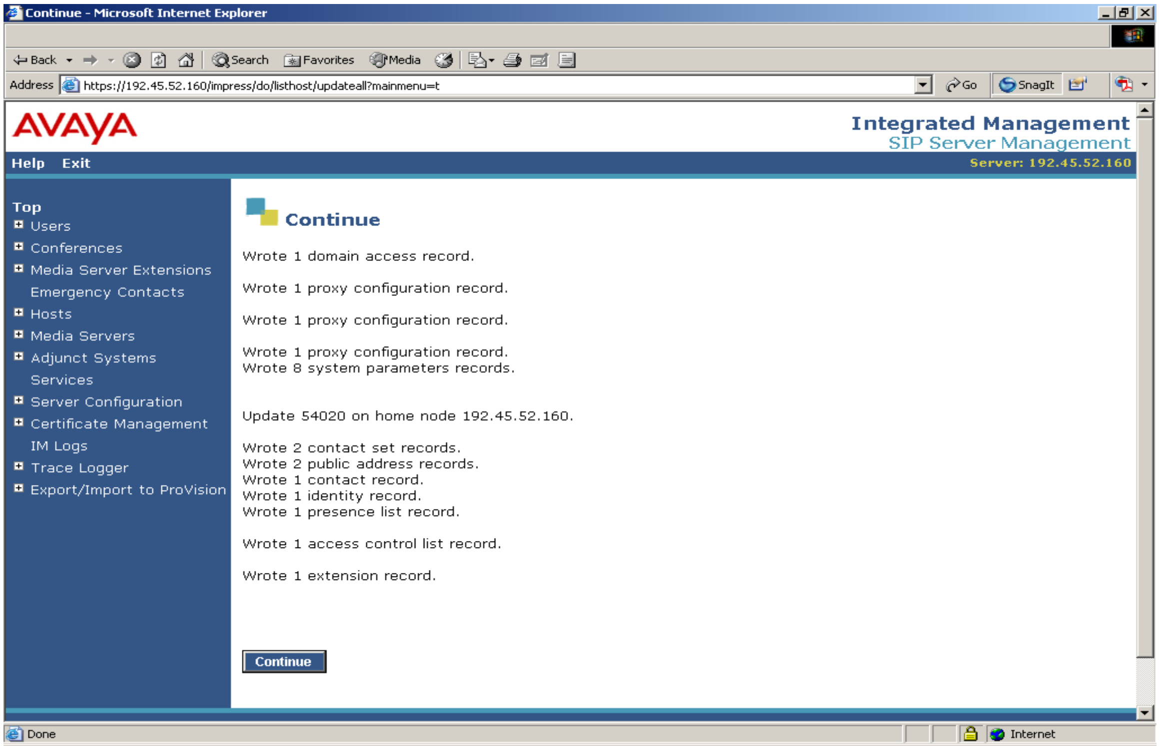
Step	Description
7.	At the **Add Media Server Extension** page, configure the following: • **Extension –** Set to **Phone Number** field value configured in **Section 3.7**, **Step 2**. • **Media Server –** Set to the media server where this OPS station is configured. • Click **Add** and then click **Continue** on the next page [not shown]. **Note:** Media Server was previously configured on SES.  Add Media Server Extension Add Media Server extension for user 54020. Extension* 54020 Media Server S8710 Fields marked * are required.
8.	Repeat **Steps 5 – 7** as necessary to configure additional Citel Gateway Handsets on Avaya SES.

Step	Description
9.	Click **Update** at the bottom of the left panel to save the configuration completed in the above steps. 

List Media Server Extensions

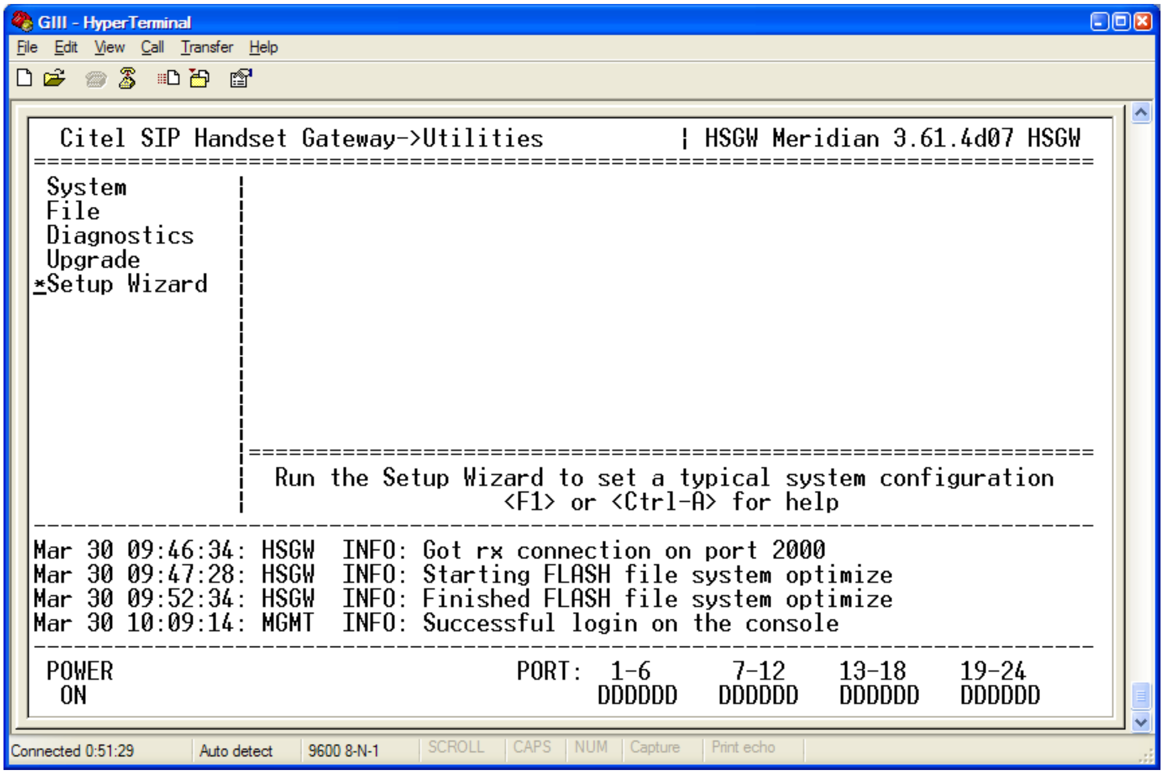
Media Server extensions for user 54020.

Commands	Extension	User	Media Server	Host
Move Ext Free Edit User Delete	54020	54020	S8710	192.45.52.160

Step	Description
10.	Click **Continue** at the bottom of the right panel.  Continue Wrote 1 domain access record. Wrote 1 proxy configuration record. Wrote 1 proxy configuration record. Wrote 1 proxy configuration record. Wrote 8 system parameters records. Update 54020 on home node 192.45.52.160. Wrote 2 contact set records. Wrote 2 public address records. Wrote 1 contact record. Wrote 1 identity record. Wrote 1 presence list record. Wrote 1 access control list record. Wrote 1 extension record.

5. Configure Citel Gateway

This section describes the steps for configuring the Citel Gateway. Citel Gateway supports a variety of TDM telephones including Nortel Meridian, Nortel Norstar, NEC and Panasonic. The configuration in this section is only for the Citel Gateway. This section assumes that the Citel Gateway's IP address is already configured. Configuration steps described in this section apply only to the fields where a value needs to be entered or modified. Default values are used for all other fields.

Step	Description
1.	Access the serial interface on the Citel Gateway using following settings for a terminal emulation program: • Speed 9600 • Data bits 8 • Parity N • Stop bits 1 • Flow control Xon/Xoff The Setup Wizard runs automatically the first time the unit is started; or the user can select **Setup Wizard** on the **Citel SIP Handset Gateway->Utilities** screen and hit the **Enter** key to setup the Citel Gateway. 

```
 Citel SIP Handset Gateway->Utilities          | HSGW Meridian 3.61.4d07 HSGW
================================================================================
 System         |
 File           |
 Diagnostics    |
 Upgrade        |
*Setup Wizard   |
                |
                |
                |
                |
                |
                |
                |=============================================================
                | Run the Setup Wizard to set a typical system configuration
                |                <F1> or <Ctrl-A> for help
--------------------------------------------------------------------------------
Mar 30 09:46:34: HSGW  INFO: Got rx connection on port 2000
Mar 30 09:47:28: HSGW  INFO: Starting FLASH file system optimize
Mar 30 09:52:34: HSGW  INFO: Finished FLASH file system optimize
Mar 30 10:09:14: MGMT  INFO: Successful login on the console
--------------------------------------------------------------------------------
 POWER                              PORT:  1-6      7-12     13-18     19-24
  ON                                     DDDDDD    DDDDDD    DDDDDD    DDDDDD
```

Step	Description
2.	Only relevant setup steps are shown here. PBX type can be changed here depending upon what type of telephones will be connected to the Citel Gateway. In this example, Nortel Meridian telephone sets will be used.

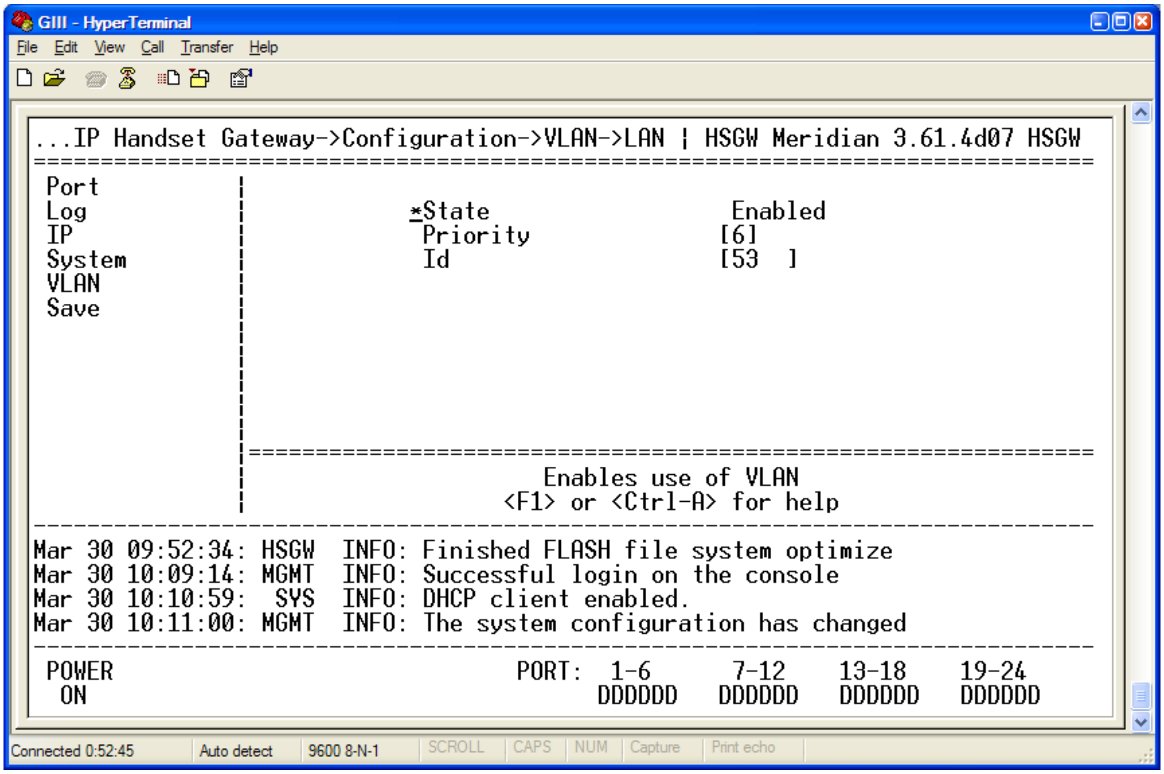
```
************************************************************
************  We need to specify the PBX type  *************
************************************************************
Currently configured PBX type is 'Meridian'
Do you want to select a new PBX type? [y/n] n
Enable all 24 ports? [y/n] y

************************************************************
*  Finally, let's configure this unit's IP configuration  **
************************************************************
Set System IP Configuration? [y/n] y
Use DHCP to acquire unit's IP address? [y/n] n
Enter the IP Address: 192.45.53.101
Enter the Subnet Mask: 255.255.255.0
Enter the Default Router: 192.45.53.1
```

Step | **Description**

3. At the **SIP Handset Gateway** screen, select **Configuration →VLAN →LAN** to configure the following:

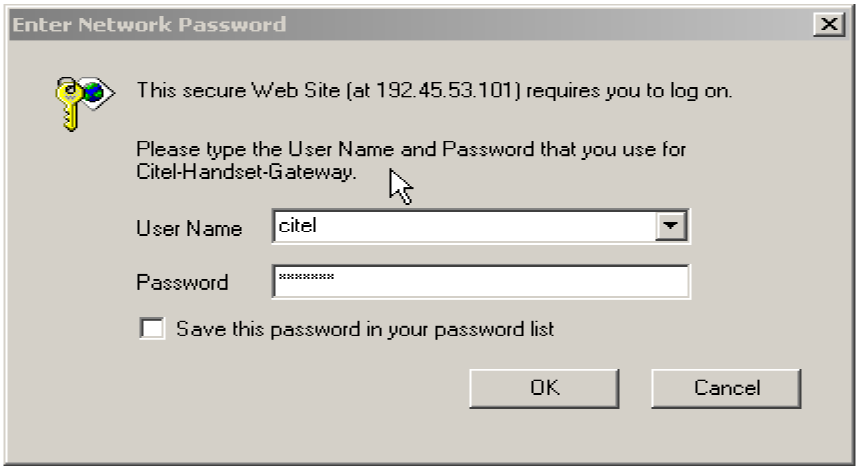
- **Priority –** This is layer-2 priority setting and can assume any value between **1** and **7**.
- **Id –** Set to **53**.

Save the configuration by selecting **Save** in the left pane of the screen.



```
...IP Handset Gateway->Configuration->VLAN->LAN | HSGW Meridian 3.61.4d07 HSGW
=============================================================================
 Port            |
 Log             |            *State                    Enabled
 IP              |             Priority                [6]
 System          |             Id                      [53   ]
 VLAN            |
 Save            |
                 |
                 |
                 |
                 |
                 |==============================================================
                 |                Enables use of VLAN
                 |           <F1> or <Ctrl-A> for help
-----------------------------------------------------------------------------
Mar 30 09:52:34: HSGW  INFO: Finished FLASH file system optimize
Mar 30 10:09:14: MGMT  INFO: Successful login on the console
Mar 30 10:10:59:  SYS  INFO: DHCP client enabled.
Mar 30 10:11:00: MGMT  INFO: The system configuration has changed
-----------------------------------------------------------------------------
 POWER                        PORT:  1-6      7-12     13-18     19-24
  ON                                DDDDDD   DDDDDD   DDDDDD    DDDDDD
```

4. 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 Citel Gateway. Enter the **User Name** and **Password**. Click **OK** to proceed to the next screen.



This secure Web Site (at 192.45.53.101) requires you to log on.

Please type the User Name and Password that you use for Citel-Handset-Gateway.

User Name: citel

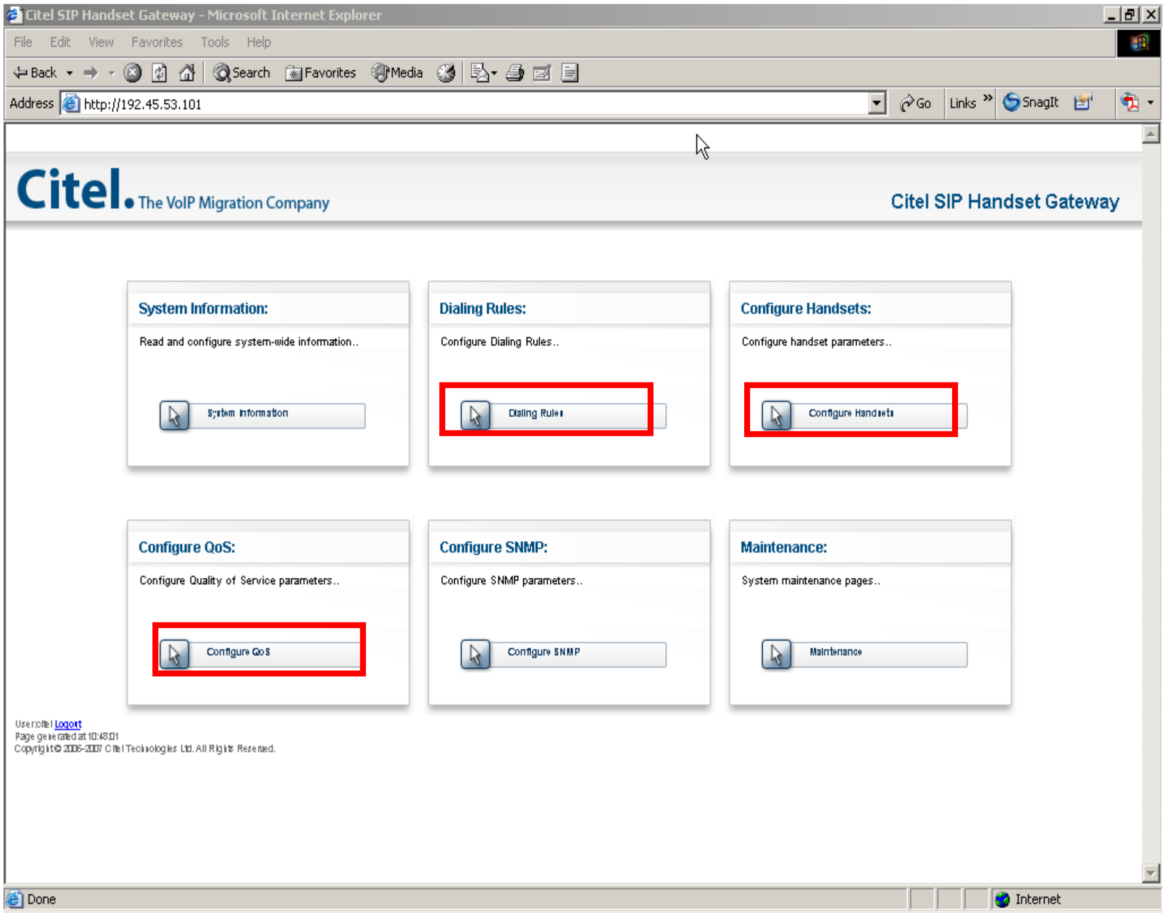
Password: ********

☐ Save this password in your password list

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>

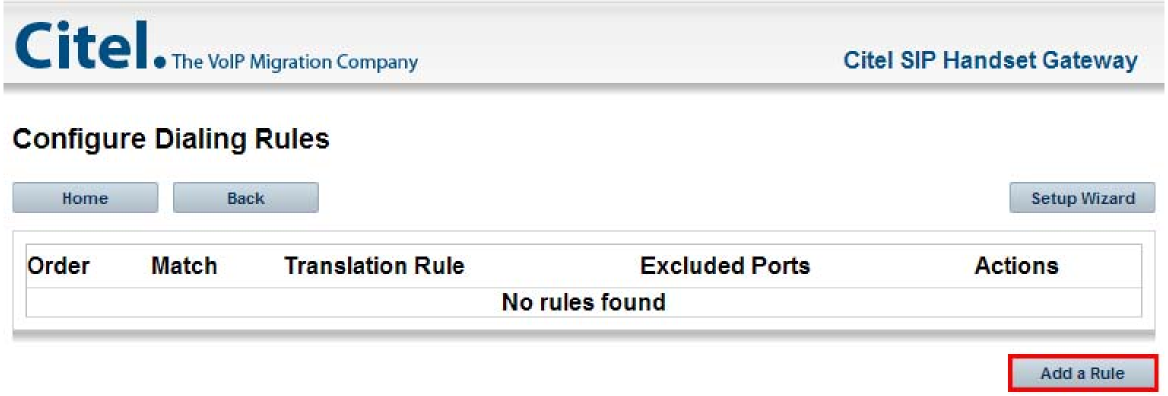
This is the main screen on the Citel Gateway to perform various configuration steps. Click on any of the following buttons to configure the Citel Gateway:

- **Dialing Rule** – To configure the dial plan for the Citel Gateway.
- **Configure Handsets –** Configure the telephones connected to the Citel Gateway.
- **Configure QoS –** Configure the QoS values and the codecs.

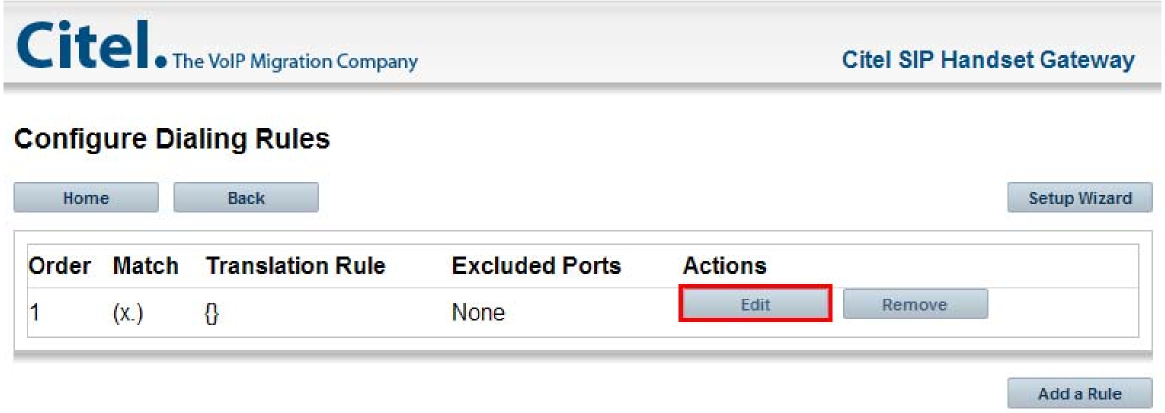


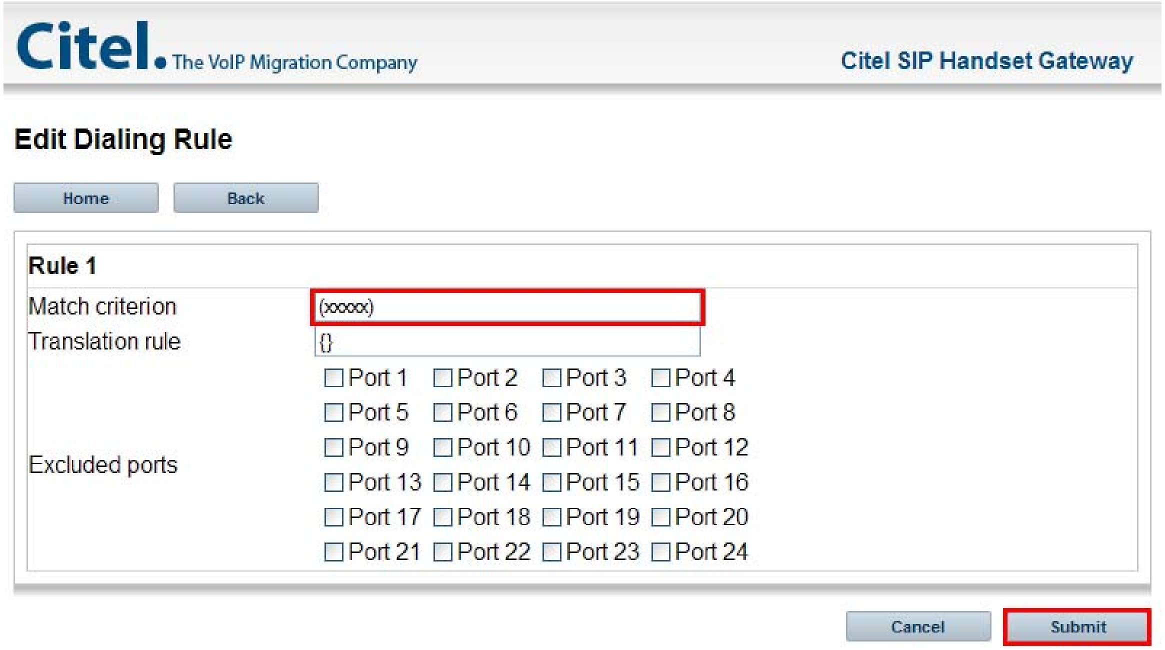
</td></tr>
<tr><td>6.</td><td>

Select the **Dialing Rule** button from the screen at **Step 5** and click on **Add a Rule** to generate a default rule for the dial plan.

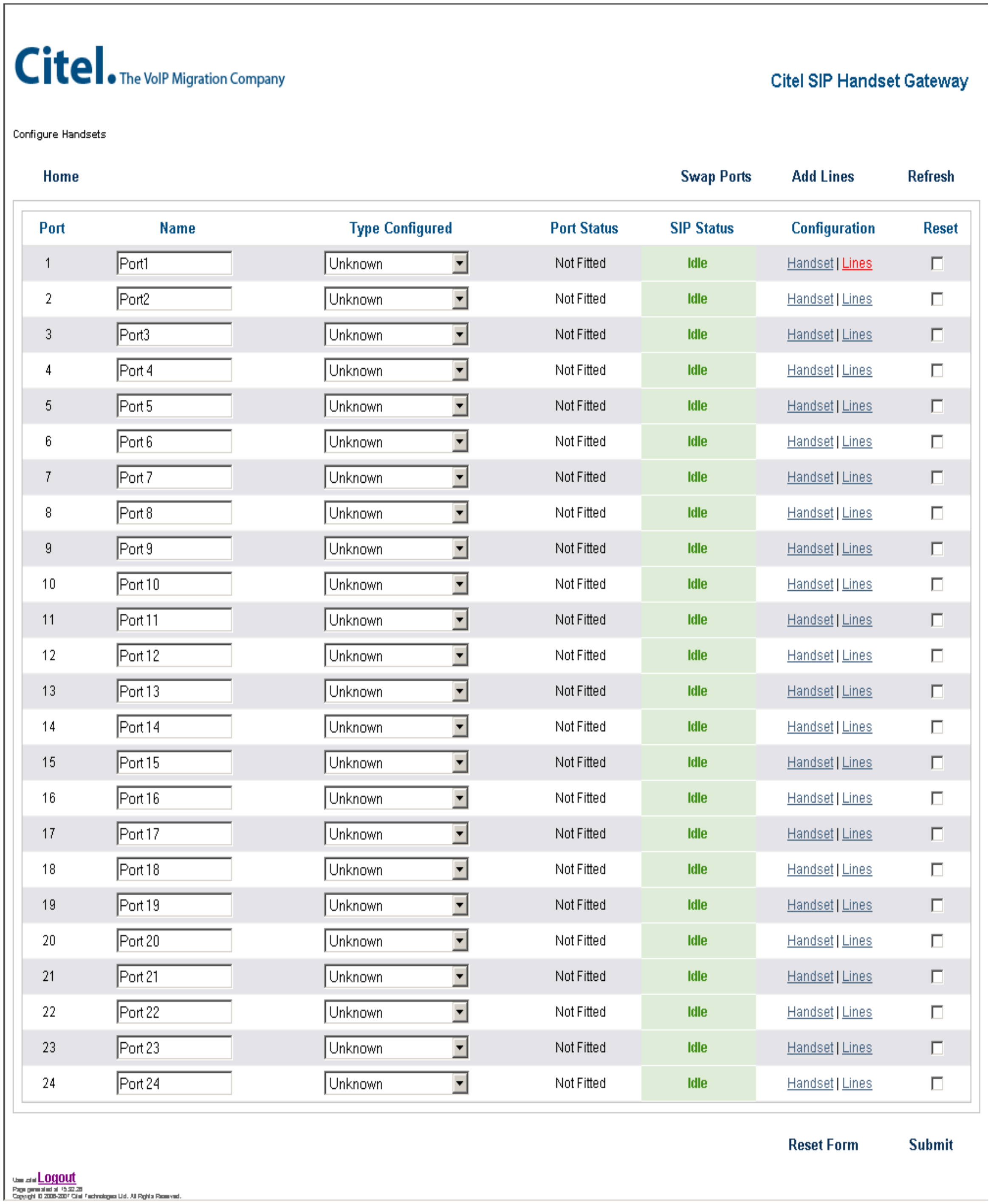


</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>At the Configure Dialing Rules screen, click Edit to edit the default rule.


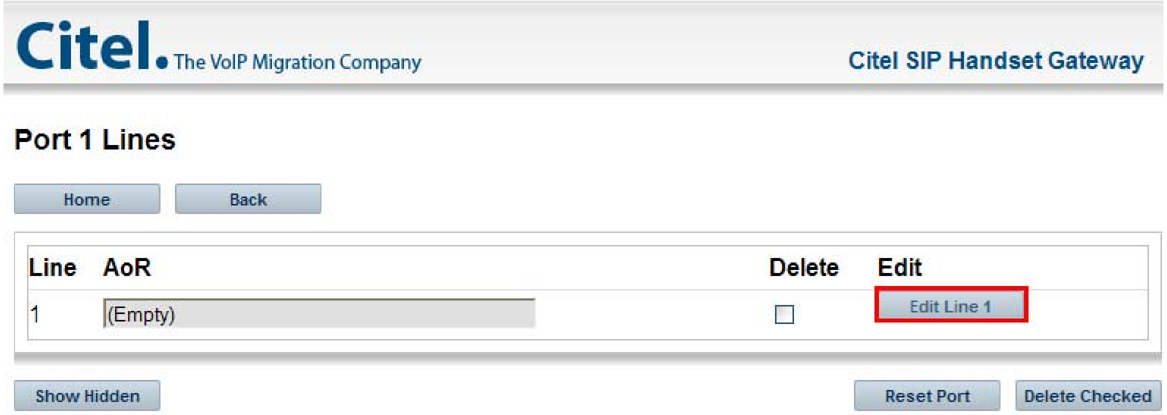
Configure Dialing Rules
<table>
<tr><th>Order</th><th>Match</th><th>Translation Rule</th><th>Excluded Ports</th><th>Actions</th></tr>
<tr><td>1</td><td>(x.)</td><td>{}</td><td>None</td><td>Edit
Remove</td></tr>
</table></td></tr>
<tr><td>8.</td><td>At Edit Dialing Rule screen, set Match criterion field to any value. In this example, xxxxx is used for dialing a five-digit extension. Click Submit to complete this configuration and then click Home on the subsequent screen [not shown] to go back to the screen in Step 5.


Edit Dialing Rule
Rule 1
Match criterion: (xxxxx)
Translation rule: {}
Excluded ports:
☐ Port 1 ☐ Port 2 ☐ Port 3 ☐ Port 4
☐ Port 5 ☐ Port 6 ☐ Port 7 ☐ Port 8
☐ Port 9 ☐ Port 10 ☐ Port 11 ☐ Port 12
☐ Port 13 ☐ Port 14 ☐ Port 15 ☐ Port 16
☐ Port 17 ☐ Port 18 ☐ Port 19 ☐ Port 20
☐ Port 21 ☐ Port 22 ☐ Port 23 ☐ Port 24</td></tr>
</table>

Step	Description
9.	Select the **Configure Handsets** button from the screen at **Step 5**. To configure a SIP telephone line on Citel Gateway, click **Lines** for the port used by the Citel Gateway Handset. 

Configure Handsets

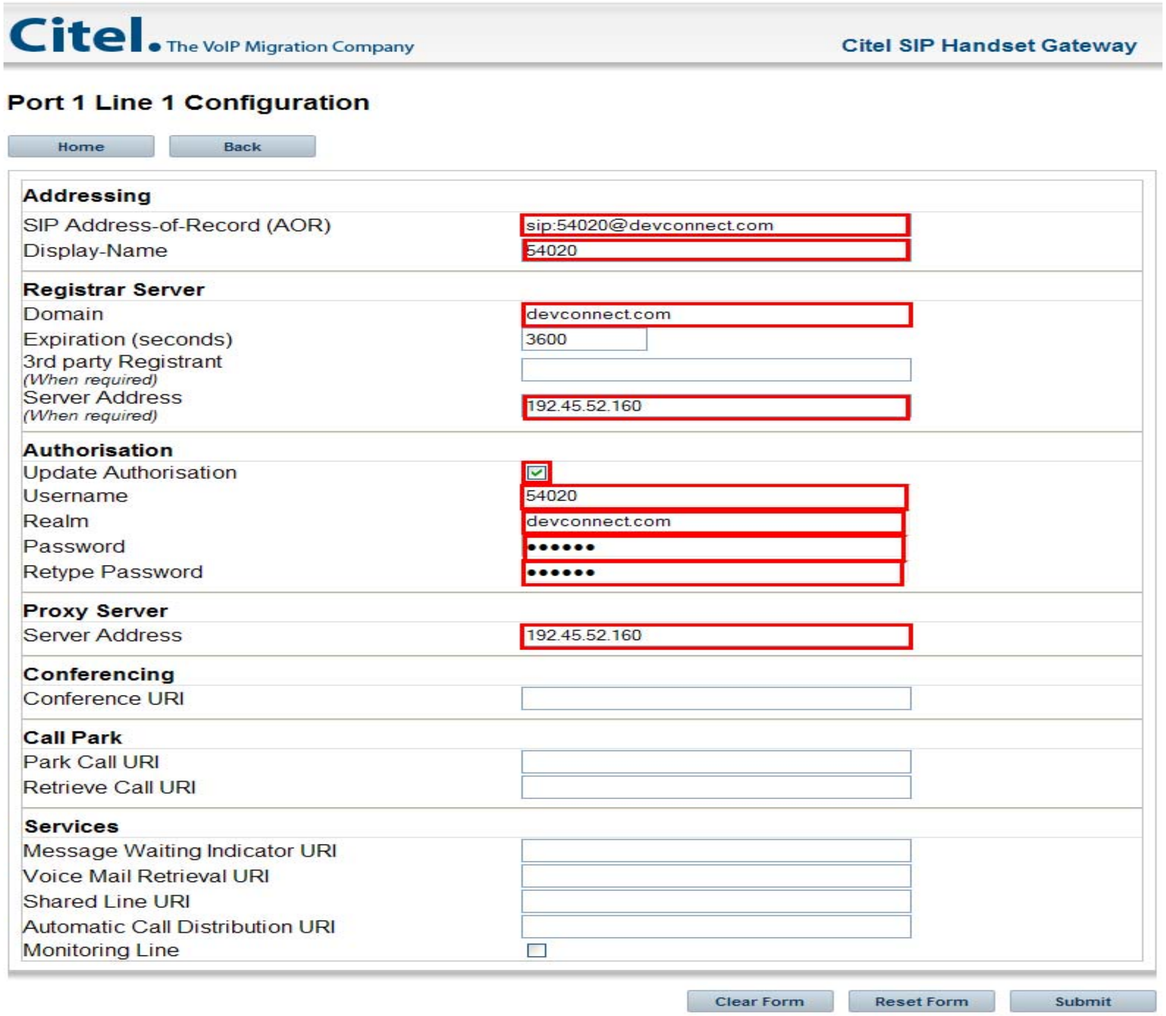
Port	Name	Type Configured	Port Status	SIP Status	Configuration	Reset
1	Port1	Unknown	Not Fitted	Idle	Handset | Lines	☐
2	Port2	Unknown	Not Fitted	Idle	Handset | Lines	☐
3	Port3	Unknown	Not Fitted	Idle	Handset | Lines	☐
4	Port 4	Unknown	Not Fitted	Idle	Handset | Lines	☐
5	Port 5	Unknown	Not Fitted	Idle	Handset | Lines	☐
6	Port 6	Unknown	Not Fitted	Idle	Handset | Lines	☐
7	Port 7	Unknown	Not Fitted	Idle	Handset | Lines	☐
8	Port 8	Unknown	Not Fitted	Idle	Handset | Lines	☐
9	Port 9	Unknown	Not Fitted	Idle	Handset | Lines	☐
10	Port 10	Unknown	Not Fitted	Idle	Handset | Lines	☐
11	Port 11	Unknown	Not Fitted	Idle	Handset | Lines	☐
12	Port 12	Unknown	Not Fitted	Idle	Handset | Lines	☐
13	Port 13	Unknown	Not Fitted	Idle	Handset | Lines	☐
14	Port 14	Unknown	Not Fitted	Idle	Handset | Lines	☐
15	Port 15	Unknown	Not Fitted	Idle	Handset | Lines	☐
16	Port 16	Unknown	Not Fitted	Idle	Handset | Lines	☐
17	Port 17	Unknown	Not Fitted	Idle	Handset | Lines	☐
18	Port 18	Unknown	Not Fitted	Idle	Handset | Lines	☐
19	Port 19	Unknown	Not Fitted	Idle	Handset | Lines	☐
20	Port 20	Unknown	Not Fitted	Idle	Handset | Lines	☐
21	Port 21	Unknown	Not Fitted	Idle	Handset | Lines	☐
22	Port 22	Unknown	Not Fitted	Idle	Handset | Lines	☐
23	Port 23	Unknown	Not Fitted	Idle	Handset | Lines	☐
24	Port 24	Unknown	Not Fitted	Idle	Handset | Lines	☐

Step	Description
10.	At the **Port 1 Lines** screen, click **Edit Line 1**.  Port 1 Lines Line 1: AoR (Empty), Delete ☐, Edit Line 1

Step | **Description**

11. At **Port 1 Line 1 Configuration** screen, configure the following:

- **SIP Address-of-Record (AOR) –** Set to SIP URI. In this example, it is set to **sip:54020@devconnect.com** where **54020** is **User ID** configured in **Section 4**, **Step 6** and **devconnect.com** is the domain name of the Avaya SES server configured in **Section 4**, **Step 2**.
- **Display-Name –** Enter any descriptive name.
- **Domain –** Set to the **SIP Domain** field value configured in **Section 4**, **Step 2**.
- **Server Address –** Set to the IP address of the Avaya SES server configured in **Section 4**, **Step 2**.
- Check the **Update Authorisation** box.
- **Username –** Set to the **Primary Handle** field value configured in **Section 4**, **Step 6**.
- **Realm –** Set to the domain name configured in **Section 4**, **Step 2**.
- **Password** and **Retype Password –** Set to the Password field value configured in **Section 4**, **Step 6**.
- **Proxy Server** - Set to the IP address of the Avaya SES server configured in **Section 4**, **Step 2**.
- Click **Submit**.



Port 1 Line 1 Configuration

Addressing
SIP Address-of-Record (AOR): sip:54020@devconnect.com
Display-Name: 54020

Registrar Server
Domain: devconnect.com
Expiration (seconds): 3600
3rd party Registrant (When required):
Server Address (When required): 192.45.52.160

Authorisation
Update Authorisation: ☑
Username: 54020
Realm: devconnect.com
Password: ••••••
Retype Password: ••••••

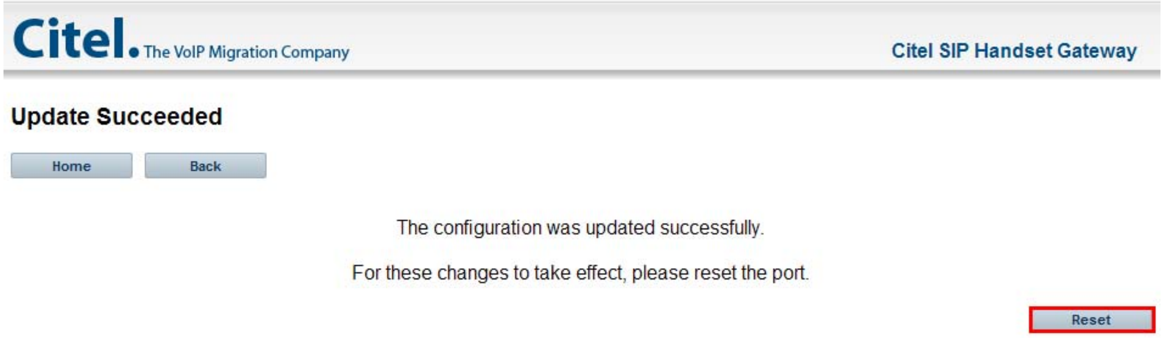
Proxy Server
Server Address: 192.45.52.160

Conferencing
Conference URI:

Call Park
Park Call URI:
Retrieve Call URI:

Services
Message Waiting Indicator URI:
Voice Mail Retrieval URI:
Shared Line URI:
Automatic Call Distribution URI:
Monitoring Line: ☐

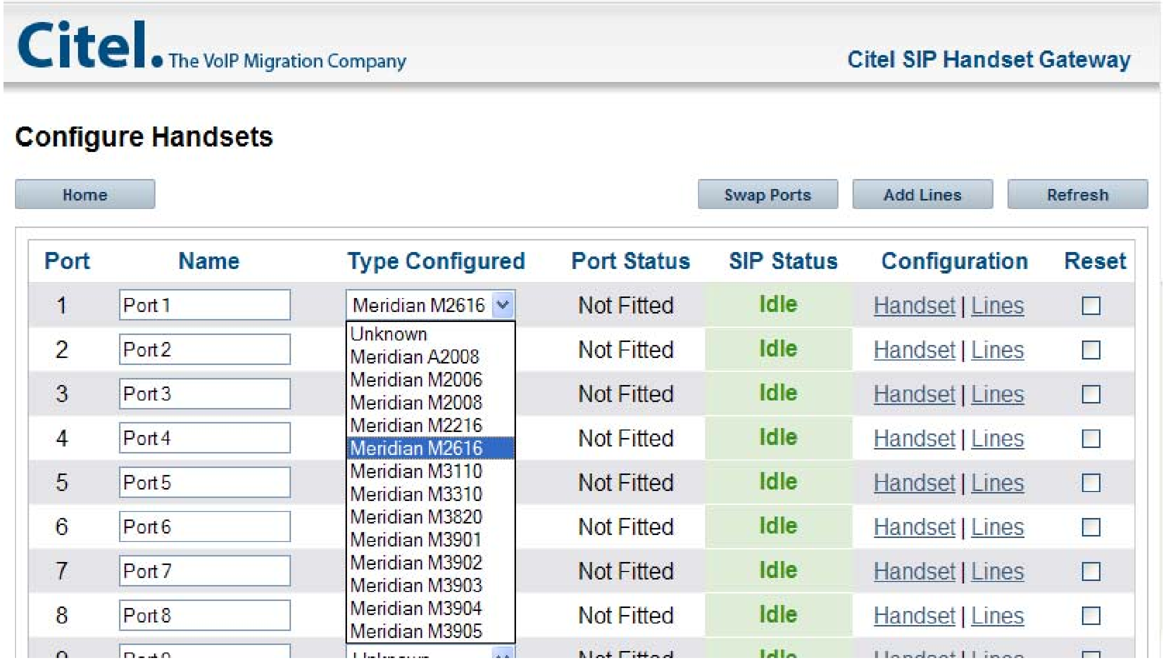
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>12.</td><td>
At the Update Succeeded screen, click Reset and click Home on the subsequent screen [not shown] to go back to the screen on Step 5.



Update Succeeded

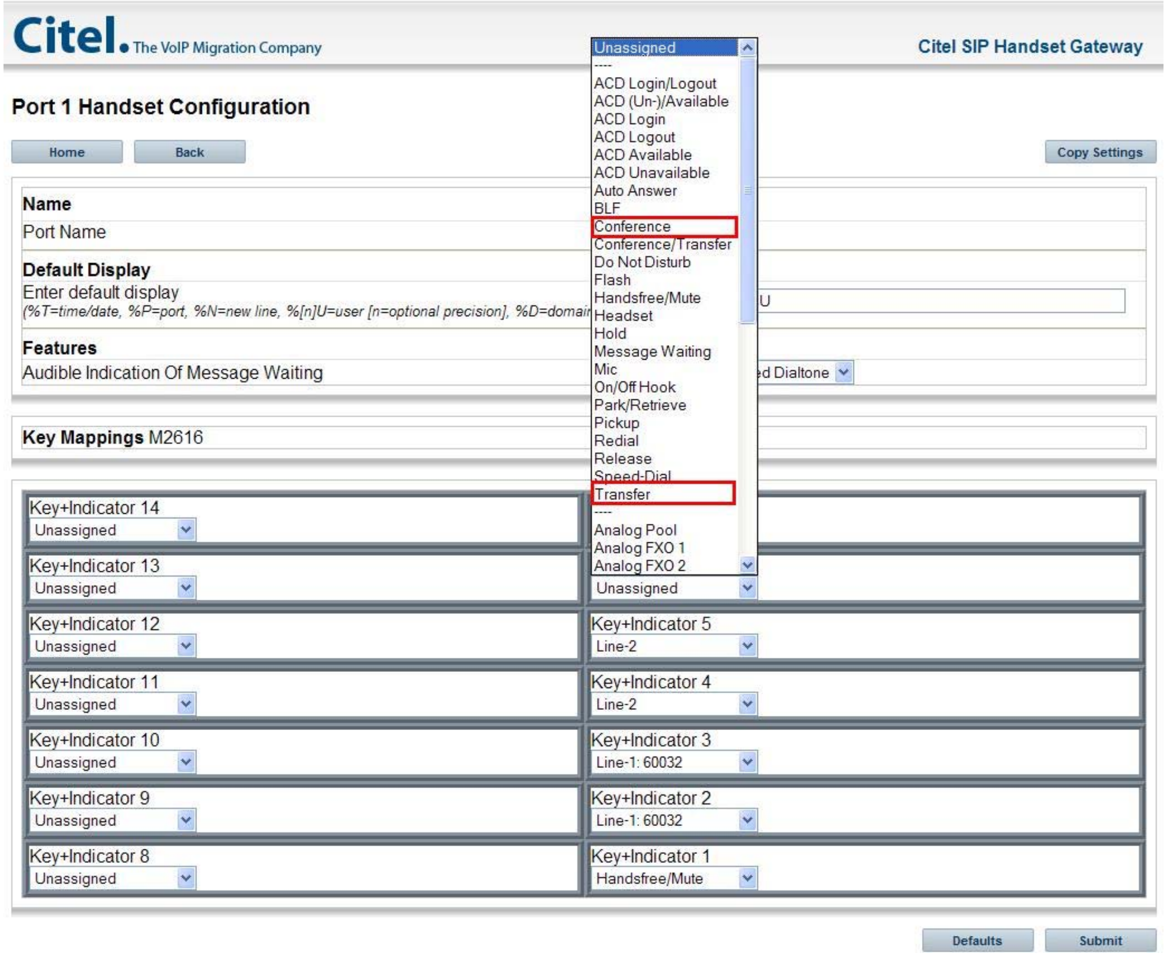
The configuration was updated successfully.

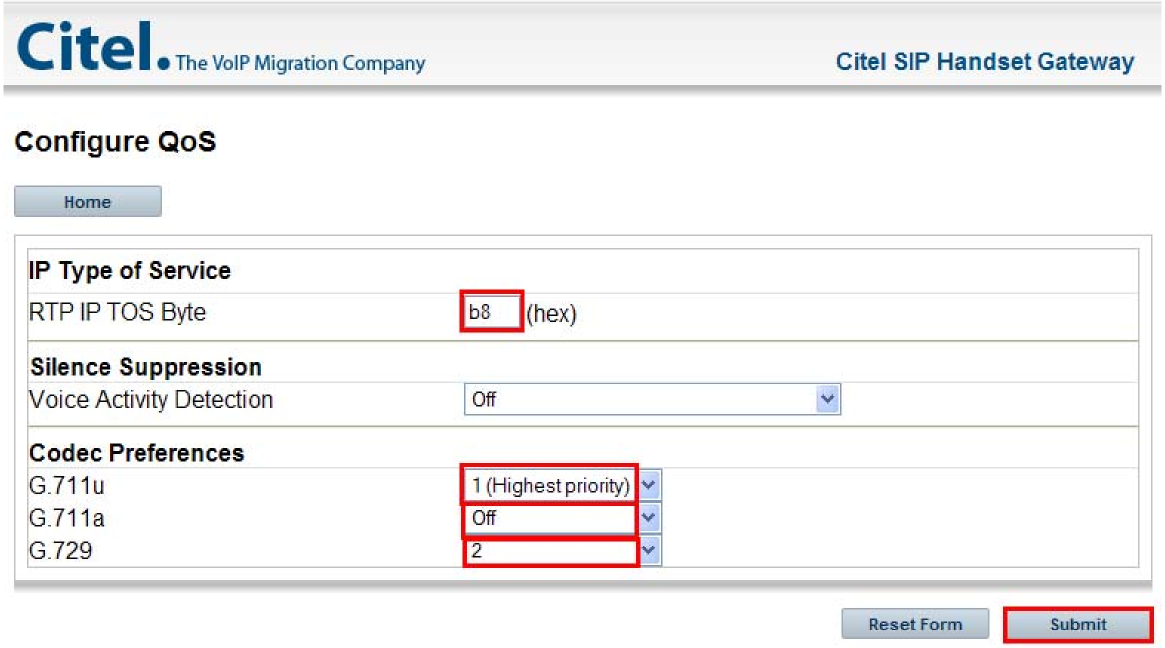
For these changes to take effect, please reset the port.
</td></tr>
<tr><td>13.</td><td>
Repeat Steps 6 - 12 to add additional lines on the same Citel Gateway port or to add another handset on a different Citel Gateway port.
</td></tr>
<tr><td>14.</td><td>
Select the Configure Handsets button from the screen at Step 5 and click on Type Configured to configure the appropriate telephone model on Citel Gateway.
Then click Submit [not shown].



Configure Handsets

<table>
<tr><th>Port</th><th>Name</th><th>Type Configured</th><th>Port Status</th><th>SIP Status</th><th>Configuration</th><th>Reset</th></tr>
<tr><td>1</td><td>Port 1</td><td>Meridian M2616</td><td>Not Fitted</td><td>Idle</td><td>Handset | Lines</td><td>☐</td></tr>
<tr><td>2</td><td>Port 2</td><td></td><td>Not Fitted</td><td>Idle</td><td>Handset | Lines</td><td>☐</td></tr>
<tr><td>3</td><td>Port 3</td><td></td><td>Not Fitted</td><td>Idle</td><td>Handset | Lines</td><td>☐</td></tr>
<tr><td>4</td><td>Port 4</td><td></td><td>Not Fitted</td><td>Idle</td><td>Handset | Lines</td><td>☐</td></tr>
<tr><td>5</td><td>Port 5</td><td></td><td>Not Fitted</td><td>Idle</td><td>Handset | Lines</td><td>☐</td></tr>
<tr><td>6</td><td>Port 6</td><td></td><td>Not Fitted</td><td>Idle</td><td>Handset | Lines</td><td>☐</td></tr>
<tr><td>7</td><td>Port 7</td><td></td><td>Not Fitted</td><td>Idle</td><td>Handset | Lines</td><td>☐</td></tr>
<tr><td>8</td><td>Port 8</td><td></td><td>Not Fitted</td><td>Idle</td><td>Handset | Lines</td><td>☐</td></tr>
</table>
</td></tr>
</table>

Step	Description
15.	Select the **Configure Handsets** button from the screen at **Step 5** and click on **Handset** [not shown] to configure telephone button mappings on Citel Gateway. Use the drop-down menu to designate the choice of buttons for Conference, Transfer, and any additional features required and click **Submit**.  **Port 1 Handset Configuration** **Name** Port Name **Default Display** Enter default display *(%T=time/date, %P=port, %N=new line, %[n]U=user [n=optional precision], %D=domai* U **Features** Audible Indication Of Message Waiting d Dialtone **Key Mappings** M2616 Key+Indicator 14 Unassigned Key+Indicator 13 Unassigned Unassigned Key+Indicator 12 Unassigned Key+Indicator 5 Line-2 Key+Indicator 11 Unassigned Key+Indicator 4 Line-2 Key+Indicator 10 Unassigned Key+Indicator 3 Line-1: 60032 Key+Indicator 9 Unassigned Key+Indicator 2 Line-1: 60032 Key+Indicator 8 Unassigned Key+Indicator 1 Handsfree/Mute
16.	At the **Update Succeeded** screen [not shown], click **Reset** and click **Home** on the subsequent screen [not shown] to go back to the screen on **Step 5**.

Step	Description
17.	Select the **Configure QoS** button from the screen at **Step 5** and configure the following fields: - **RTP IP TOS Byte –** Set this field for the Layer-3 Diffserv value to be used. In this example, a Hex value of **b8** was used, which calculates to **ASCII 46** after low order bit shifting. - **Codec Preferences –** Set the priority for each Codec value to be used. - Click **Submit**.  **IP Type of Service** RTP IP TOS Byte: b8 (hex) **Silence Suppression** Voice Activity Detection: Off **Codec Preferences** G.711u: 1 (Highest priority) G.711a: Off G.729: 2

6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment using the Citel Gateway and operations such as dialing methods (manual, re-dial, and phone book), hold, mute, transfer and conference. Verified Citel Gateway interactions with SES, Avaya Communication Manager, and Avaya SIP, H.323, and digital telephones.

6.1. General Test Approach

The general test approach was to place calls to and from the Citel Gateway Handsets and exercise basic telephone operations. The main objectives were as follows:

- The Citel Gateway successfully registers with Avaya SES.
- Successfully establish calls between the Citel Gateway and Avaya SIP, H.323, and digital telephones attached to Avaya SES or Avaya Communication Manager.
- Successfully establish calls between the Citel Gateway and PSTN telephone through Avaya Communication Manager.
- The Citel Gateway successfully handles concurrent calls on its two lines.
- The Citel Gateway successfully negotiates the right codec.
- The Citel Gateway successfully shuffles for VoIP calls.
- The Citel Gateway successfully transmits DTMF during a call.
- The Citel Gateway successfully holds a call, transfers a call, establishes a three party conference call, and displays calling party number.
- The Citel Gateway successfully handles layer-3 (DiffServ) QoS for Audio.

For serviceability testing, failures such as cable pulls and hardware resets were applied. For performance testing, a conference call involving two Citel Gateways and two Avaya telephones was formed as follows. A call was established between an Avaya telephone and a Citel Gateway. The Citel Gateway then established a call with another Citel Gateway, and bridged the two calls together, forming a 3-party conference. The second Citel Gateway then established a call with another Avaya telephone, and bridged the two calls together, effectively forming a 4-party conference.

6.2. Test Results

The test objectives of **Section 6.1** were verified. For serviceability testing, the Citel Gateway operated properly after recovering from failures such as cable disconnects, and resets of the Citel Gateway, the Avaya SES server, and Avaya Communication Manager. For performance testing, the conference call was successfully maintained for approximately two hours. Citel Gateway successfully shuffles to communicate directly with the other endpoints.

The following observations were made during testing:

- Citel Gateway cannot negotiate with Avaya Communication Manager for the correct codec when establishing a conference call if the codec used during negotiation is not the top priority codec for the Citel Gateway. After the conference setup, the other two parties cannot hear audio from the last party added.
- Citel Gateway cannot negotiate with Avaya Communication Manager for the correct codec when putting a held call off hold. This happens if the codec used during negotiation is not the top priority codec for the Citel Gateway and hence no audio for the call when the call is put off hold.

Citel will address and resolve all the above observations with future firmware releases. Contact Citel (www.citel.com) for further updates.

7. Verification Steps

The following steps may be used to verify the configuration:

- Verify that the Citel Gateway Handsets successfully register with the Avaya SES server by following the **Users -> Registered Users** links on the SES Administration Web Interface.
- Place calls to and from the Citel Gateway Handset 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 3.6**. Note the **Member** with **Service State** set to **in-service/active**. In this example, **0010/002** and **0010/006** are active and either member can be used to verify whether calls shuffled and which codec was used.

```
Status trunk 10

                          TRUNK GROUP STATUS

Member   Port    Service State      Mtce Connected Ports
                                    Busy

0010/001 T00046  in-service/idle    no
0010/002 T00047  in-service/active  no   T0051
0010/003 T00048  in-service/idle    no
0010/004 T00049  in-service/idle    no
0010/005 T00050  in-service/idle    no
0010/006 T00051  in-service/active  no   T0047
0010/007 T00052  in-service/idle    no
0010/008 T00053  in-service/idle    no
0010/009 T00054  in-service/idle    no
0010/010 T00055  in-service/idle    no
```

Step	Description
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** are using the same port and the **Audio Connection Type** is **ip-direct**, it signifies that shuffling was successful. In this example, shuffling was successful.

```
status trunk 10/2                                           Page 1 of 2

                            TRUNK STATUS

 Trunk Group/Member: 0010/002           Service State: in-service/active
               Port: T00047          Maintenance Busy? No
Signalling Group ID:



   Connected Ports: T0051

           Port     Near-end IP Addr  : Port    Far-end IP Addr  : Port
Signaling: 01A0617  192. 45.100.147   : 5061    192. 45. 52.160  : 5061

G.711MU   Audio:    192. 45. 53.101   : 34008   192. 45. 53.102  : 34008
          Video:
    Video Codec:
                                    Authentication Type: None
  Audio Connection Type: ip-direct
```

8. Support

For technical support on Citel Gateway and how to configure TDM telephones connected to it, consult the support pages at http://www.citel.com/Support/ or contact Citel technical support at:

- Telephone: 1 (877)248-3587
- E-mail: support@citel.com

9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services 3.1.1, and digital telephones connected to Citel Gateway. Citel Gateway is a SIP-based VoIP appliance. During compliance testing, digital telephones connected to Citel Gateway 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 three-way conference, transfers, hold, etc.

10. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, Issue 2.1, May 2006, Document Number 03-300509
[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 11, February 2006, Document Number 555-233-504
[3] *SIP Support in Release 3.1 of Avaya Communication Manager*, Issue 6, February 2006, Document Number 555-245-206
[4] *Installing and Administering SIP Enablement Services R3.1.1*, Issue 2.0, August 2006, Document Number 03-600768

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

[5] Citel SIP Gateway install manual G-SIP3D-RUC

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 Developer*Connection* Program at devconnect@avaya.com.