Avaya Solution & Interoperability Test Lab

Application Notes for G-Tek SIP Telephone MT-102H version 1510X.27.1.02i with Avaya Software Communication System Release 3.0 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Software Communication System Release 3.0 (SCS) and the G-Tek SIP telephones MT-102H firmware version 1510X.27.1.02i. During the compliance testing, the MT-102H was able to register, as a SIP Client endpoint, with the Software Communication System. The MT-102H was able to place and receive calls from the Software Communication System Release 3.0 SIP Line clients. Other telephony features such as transfer as the transferee, Forward in Parallel, Forward Sequentially, Forked Invite, Call Pick-up, Call Park and Retrieved were executed. This solution is currently supported only in Malaysia, Japan, and Brazil.

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.

Introduction

These application notes provide detailed configurations of Avaya Software Communication System Release 3.0 and G-Tek SIP telephone MT-102H rel. 1510X.27.1.02i during the compatibility testing session. The MT-102H was tested against the SIP clients of the Software Communication System Release 3.0. All the applicable telephony feature test cases of Avaya Software Communication System Release 3.0 were executed on the MT-102H, to ensure the interoperability with Avaya Software Communication System.

1.1. Interoperability Compliance Testing

The focus of this compliance testing is to verify that the MT-102H is able to interoperate with Avaya Software Communication System Release 3.0. The following interoperability areas are:

- Registration of MT-102H to the Avaya Software Communication System Release 3.0.
- Calls establishment of MT-102H with Avaya SIP phones on the Avaya Software Communication System.
- Telephony features: DTMF transmission, voicemail with MWI notification, speed dial, perform blind transfer as the transferee, Forward in Parallel, Forward Sequentially, Forked Invite, Call Pick-up, Call Park and Retrieved.
- Specific hospitality feature requirement such as handling of detected loops or too many hops, long “Via” path resulting in large SIP messages.
- Codec negotiation.

1.2. Support

For technical support on G-Tek SIP telephones, please contact G-Tek technical support at:

- Telephone: +886-2-26962665 ext. 221
- E-mail: support@GTek.com.tw

2. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya Software Communication System Release 3.0 and the MT-102H.

Figure 1: Test Configuration

3. Equipment and Software Validated

System	Software/Hardware Version
Avaya Software Communication System Server	• 4.0.4-017289 2009-11-19T05
Avaya 1210 SIP client Avaya 1230 SIP client Avaya SMC3456	• Model NTYS18, series 01.02.02.00 • Model NTYS20, series 01.02.02.00 • Version 2.6, build 56076
G-Tek MT-102H SIP Telephones	• 1510X.27.1.02i

4. Configure Software Communication System

This section describes the steps to configure Software Communication System (SCS Server).

4.1. SIP Domain and Domain Aliases

This section shows the steps to configure and manage the domain on the SCS server. The Domain settings can be used to assign domain name, create and edit domain aliases.

Login to the SCS server webpage. Under **SYSTEM** menu tab as shown in **Figure 2**, select **Domain** from the pull-down menu. In the Domain Name attribute, specify the target domain name to be used as shown and its alias.

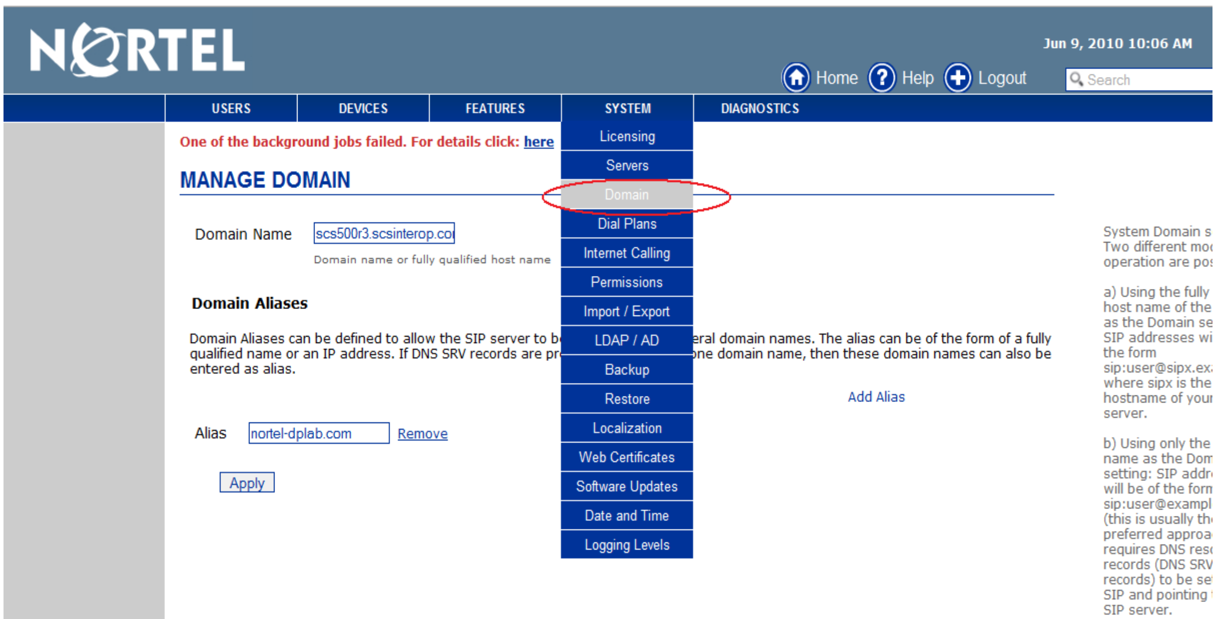

Figure 2: Domain management

On **Figure 2**, click “Add Alias” to enter the IP address of host server as shown in **Figure 3**.

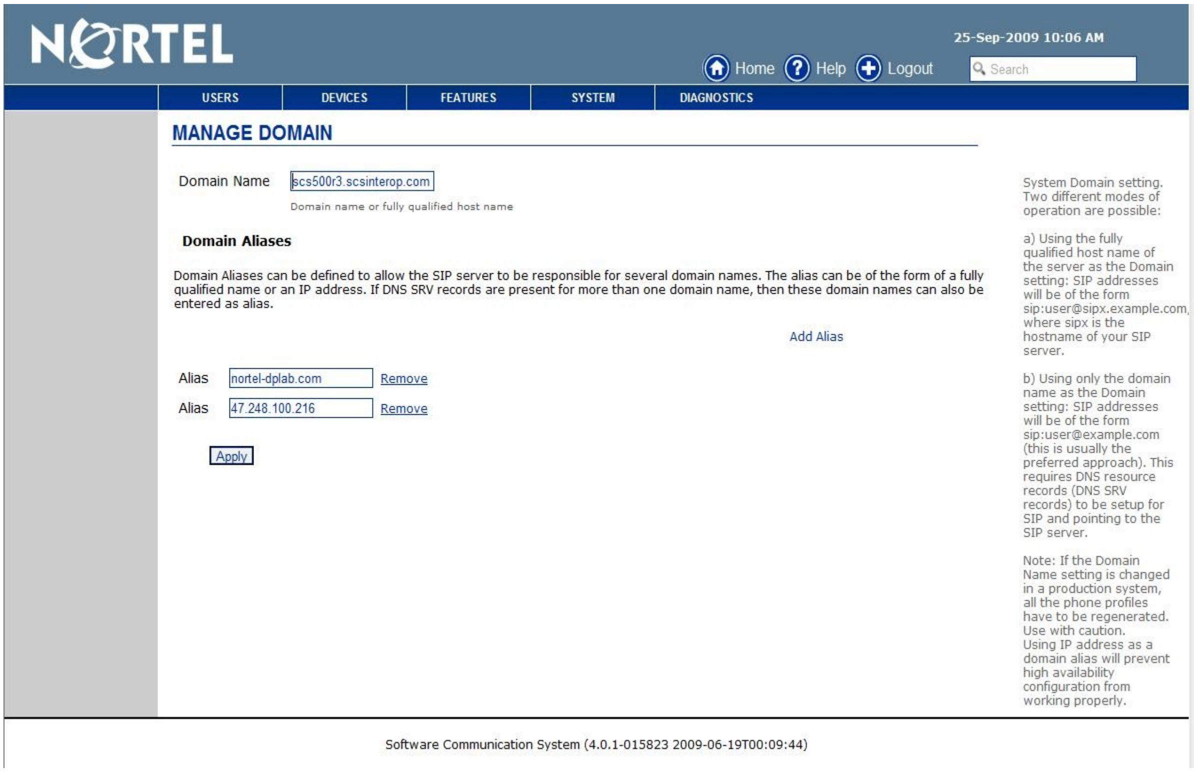

Figure 3: Alias management

4.2. Domain Server Configuration

On the SCS server webpage, navigate to **SYSTEM** menu tab and select **Server** from pull down list. The list of Host Server name will be shown in **Figure 4** below.

+ Hostname: **scs500r3.scsinterop.com**
+ IP Address**: 47.248.100.216**

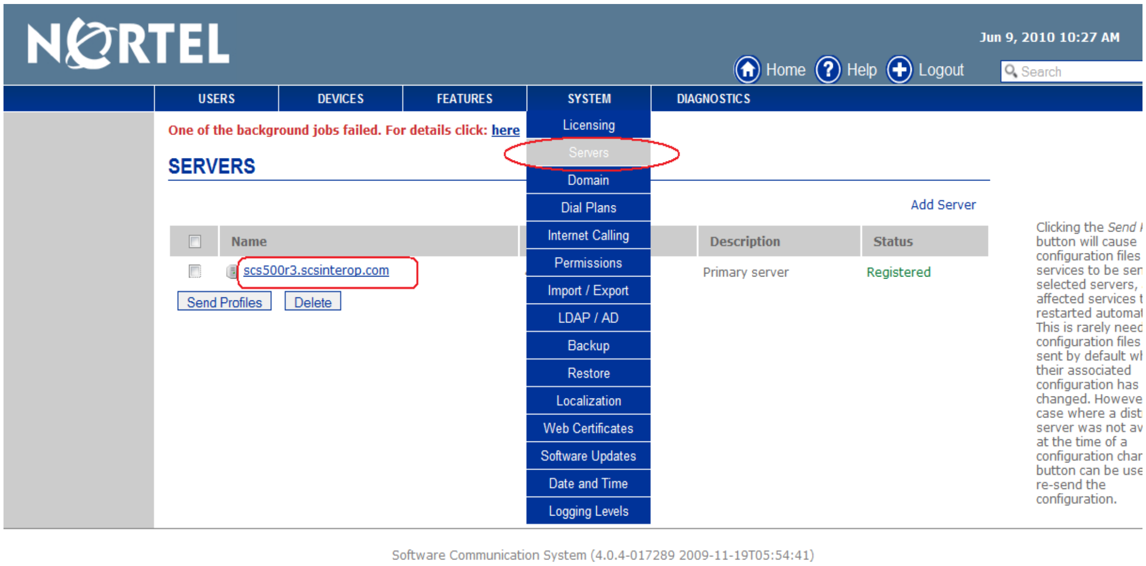

Figure 4: Domain Server Configuration Overview

From the servers list shown in **Figure 4**, choose the target server name. The details configuration of this server will be shown in **Figure 5**. The **Management**, **Primary SIP Router**, **Call Center** and **Voicemail** server roles are selected and enabled by default.

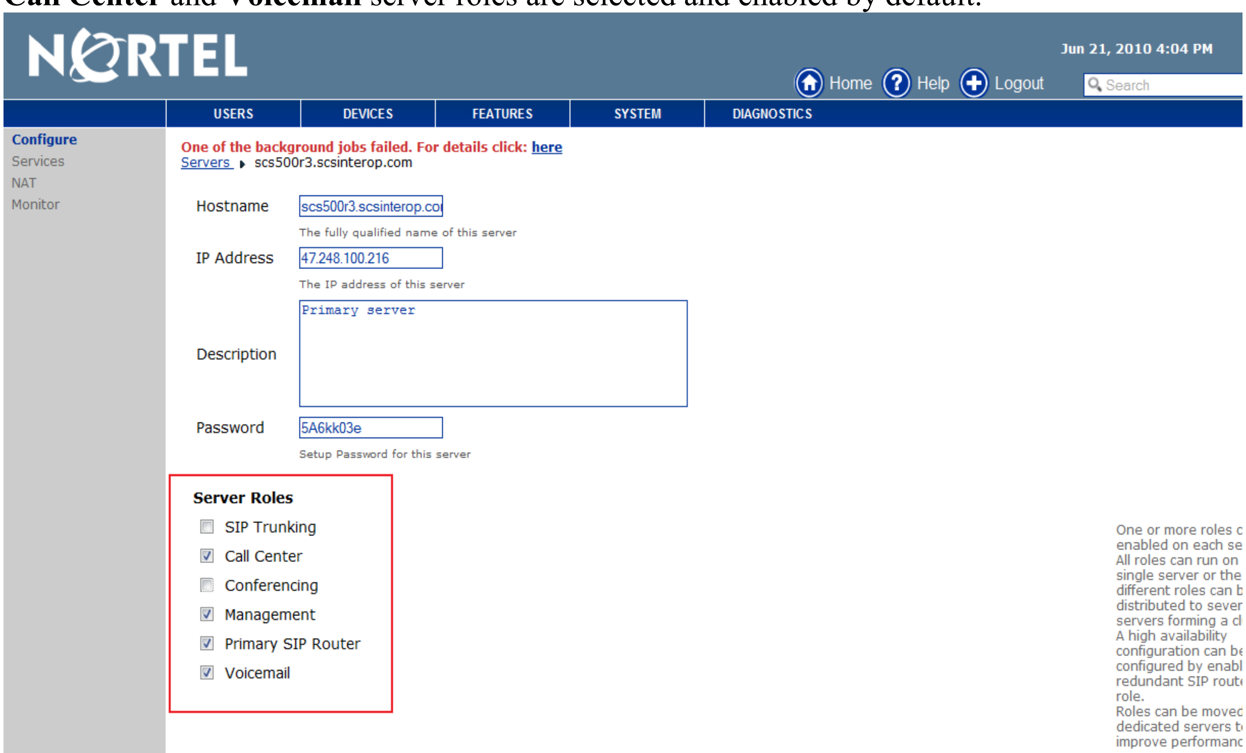

Figure 5: Domain Server Configuration with default Server Role

In **Figure 5**, under **Server Roles**, click on the check boxes of **SIP Trunking** and **Conferencing** to enable these roles on the target server as shown in **Figure 6.**

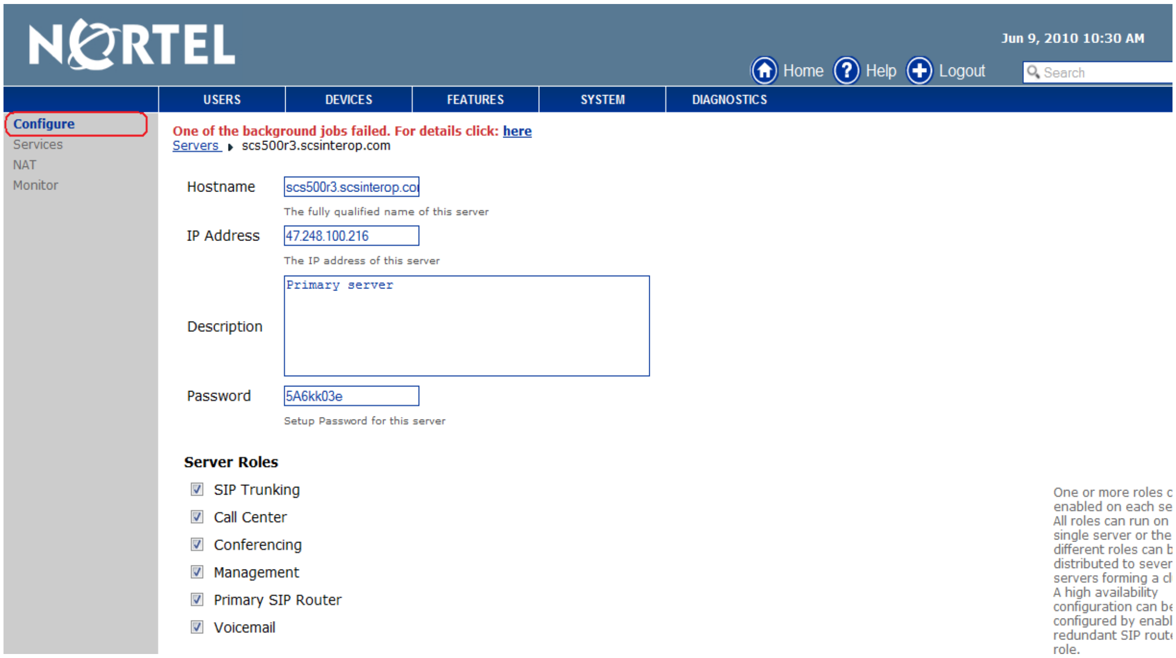

Figure 6: Server Roles Configuration

4.3. User Configuration – Identification

This section is to show how to create and configure user settings. Click on the **USERS** menu tab, select **Users** on the pull down list. The users' page will appear as shown in **Figure 7**.

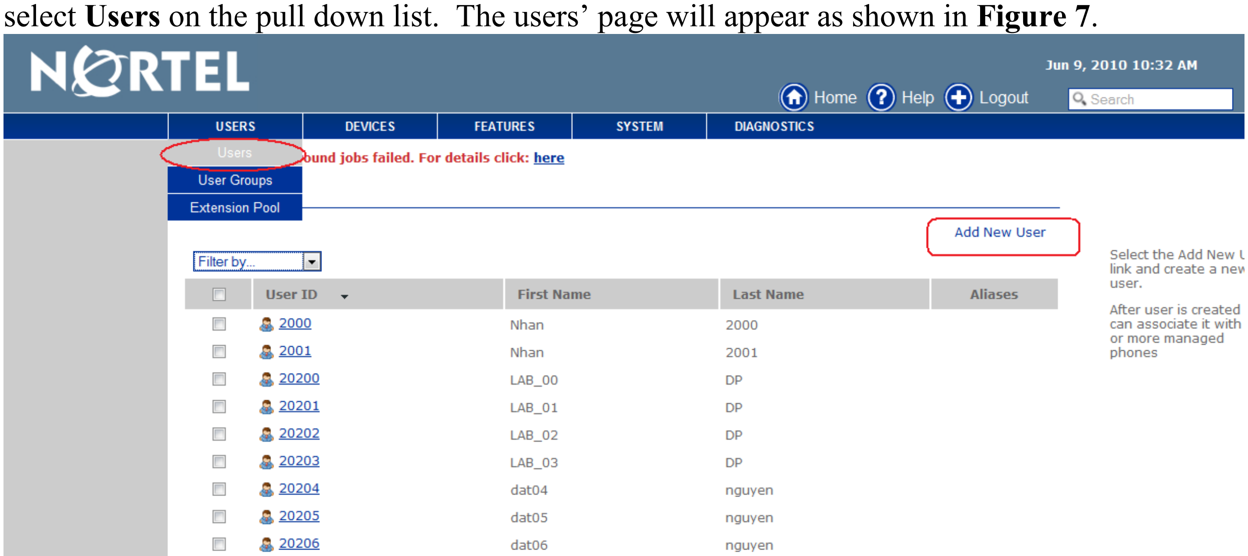

Figure 7: User Configuration

Click on **Add New User**, the user details configuration page will appear as shown in **Figure 8**.

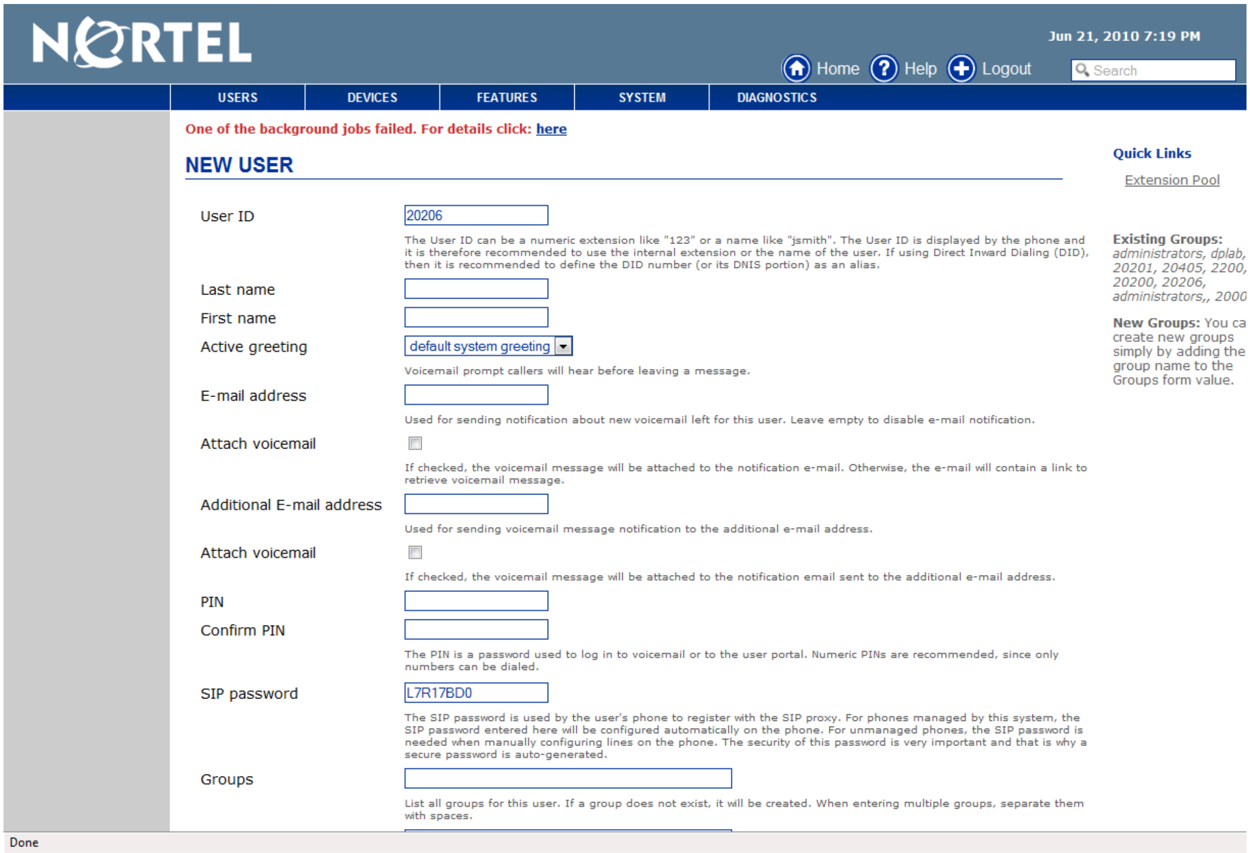

Figure 8: Adding New User

Enter the user information details as shown in **Figure 9.**

- The following fields are required: **User ID**, **Last name**, **First name**, **Active greeting**, **PIN**, **Confirm PIN**, **SIP password**, and Groups. The User ID and PIN will be used to configure the MB-102H in Section 5.1.
- Other fields are optional and can be left blank.
- Click **Apply** to save the user information and click **OK** to return to **Figure 7**.

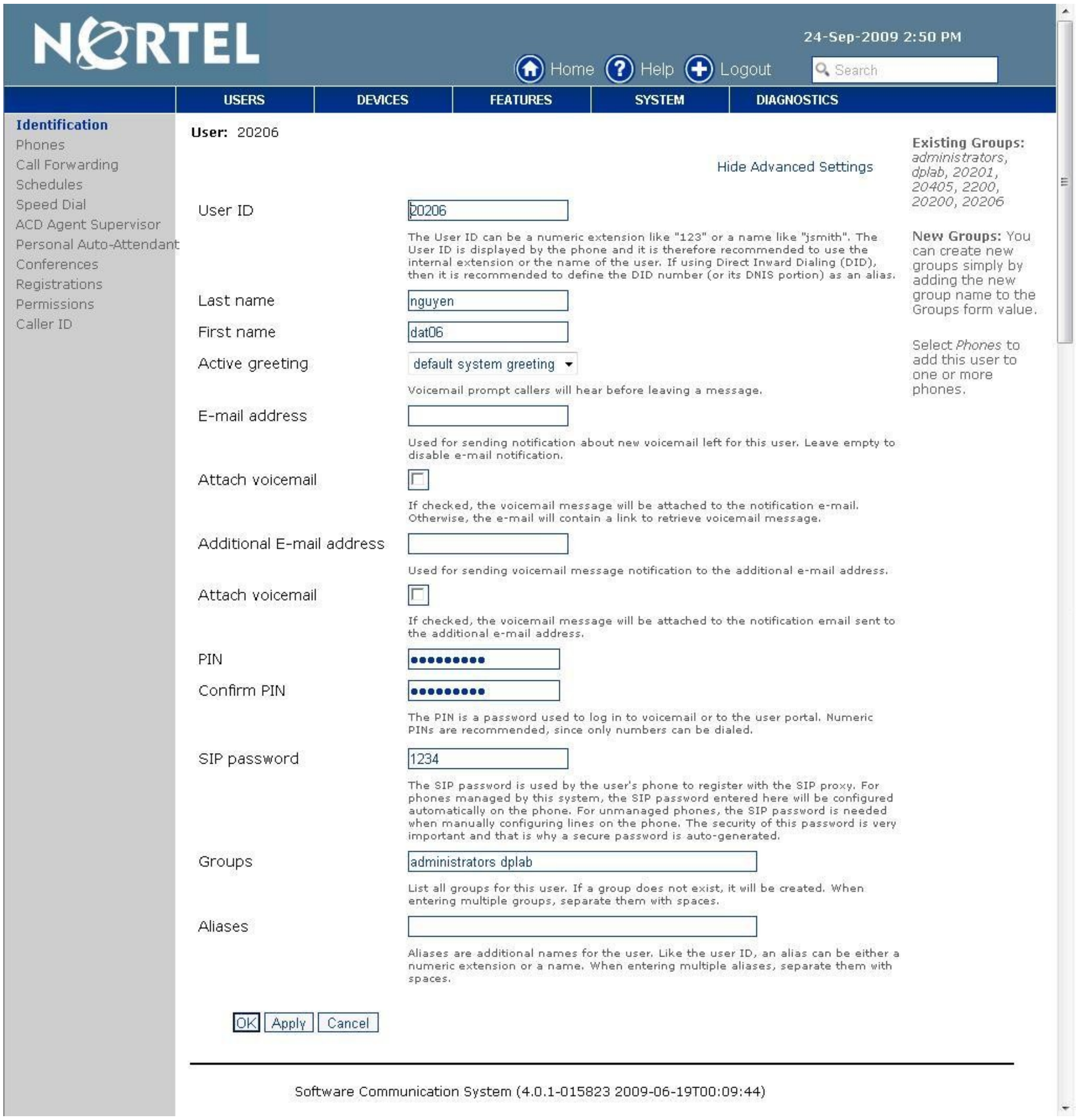

Figure 9: New User Details

4.4. User Configuration – Permission Settings

To assign permission settings for the new user click on **USERS** menu tab and select **Users** from the pull down menu. Select the new user created in Section 4.3 and click Permissions on the left menu column to display the page as shown in **Figure 10**. All the check boxes are checked by default except **Change PIN from IVR** attribute. Administrator can uncheck the box individually to turn specific function/feature off for the user. In this case, we leave everything at default values.

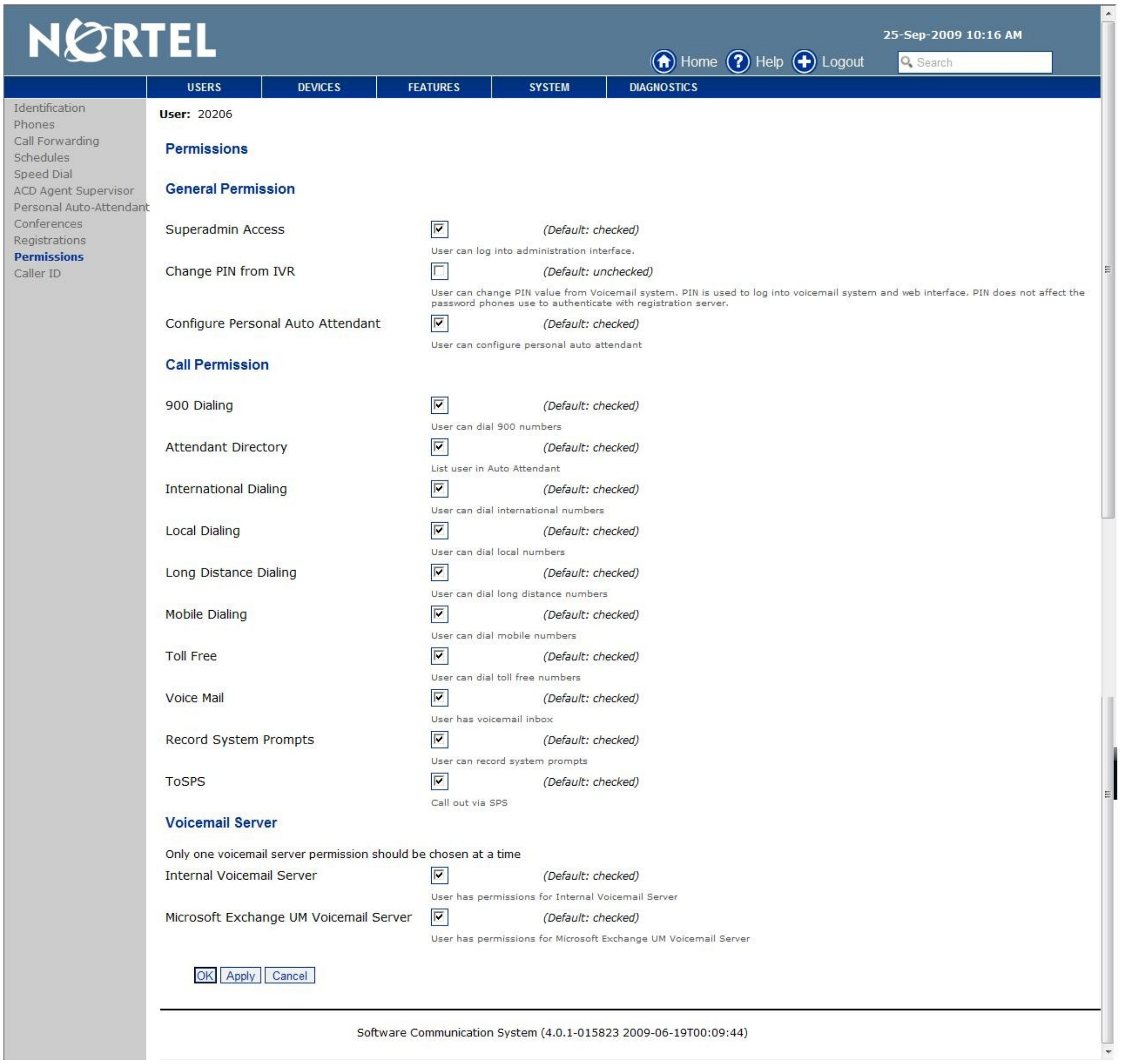

Figure 10: User permission setting

4.5. User Configuration – Assigning Conference Bridge

The following steps will show how to configure a Conference Bridge.

- On the SCS Server web page, navigate to **FEATURES** menu tab.
- Select **Conferencing** from the pull down menu as shown in **Figure 11**.

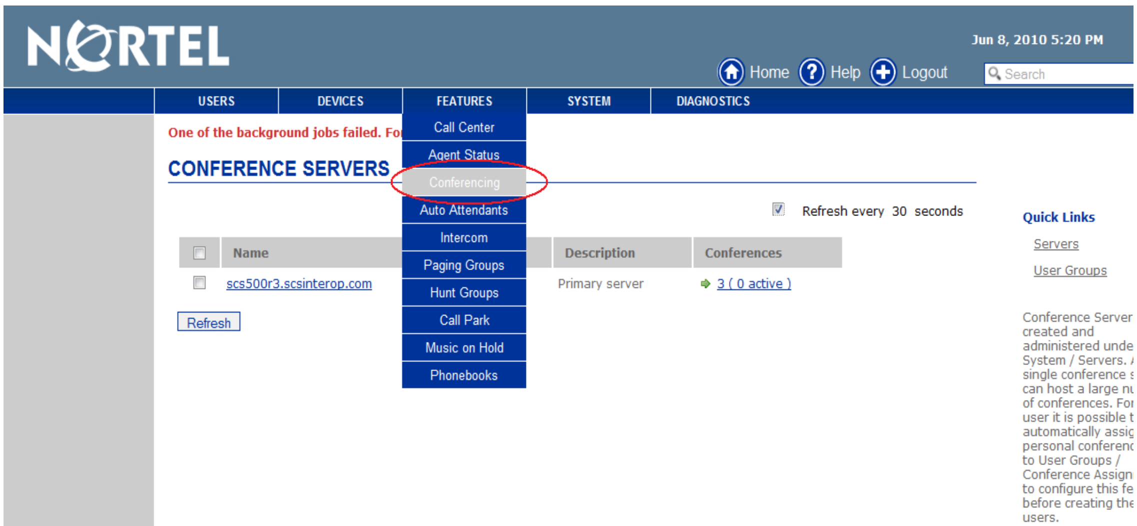

Figure 11: Conference Bridge overview page

Choose the target server from the server list (not shown) to display the page in **Figure 12**. On the left column menu, select **Conferences** and click on **Add New Conference**.

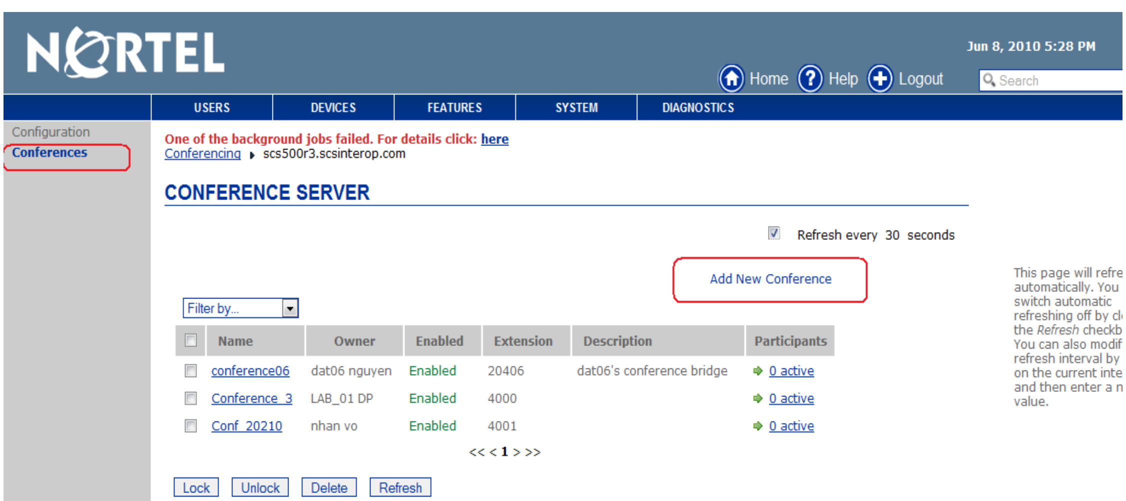

Figure 12: Adding Conference Bridge

- A page will appear as shown in **Figure 13** for the detail configuration of a conference bridge.
- Check **Enabled** box to enable this conference bridge, enter conference name and its associated **Extension**.
- Choose the owner of the conference.
- Enter participant pin.
- Other fields are left at default.
- Click on **Apply** button to save the setting information.
- Click **OK** to return to the Conference Server page.

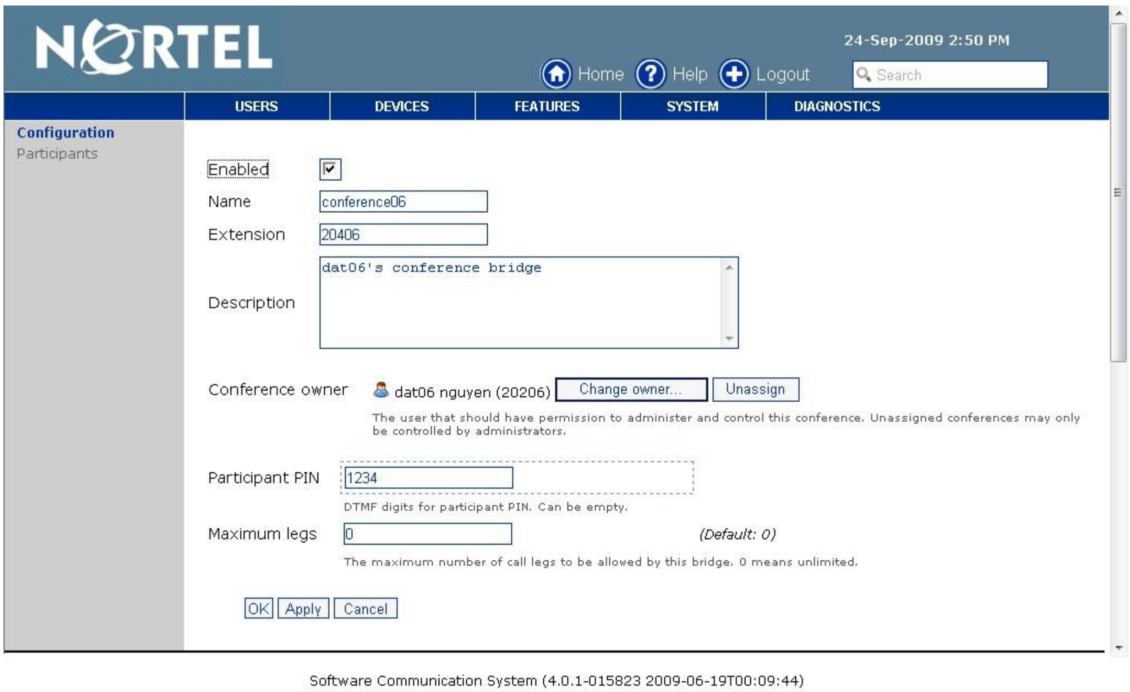

Figure 13: Conference Bridge configuration

4.6. Hunt Group Creation

The SCS system can be configured with a hunt group extension that when called, triggers a calling sequence to a group of member extensions. The calling sequence can be determined by the SCS administrator.

The following steps define how a Hunt Group can be configured:

- On the SCS Server web page, navigate to **FEATURES** menu tab.
- Select **Hunt Groups** from pull down menu as shown in **Figure 14**.

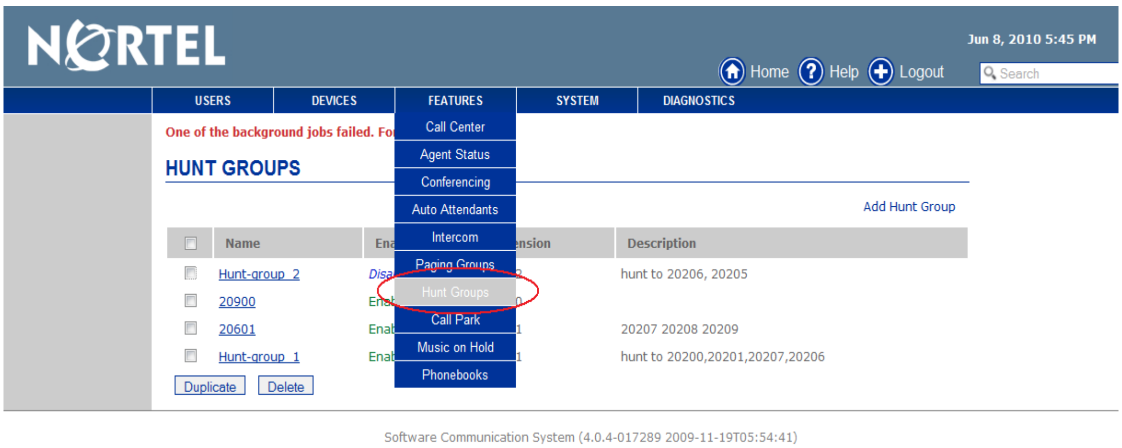

Figure 14: Hunt Group overview page

From the Hunt Groups page as shown in **Figure 15**, click **Add Hunt Group**.

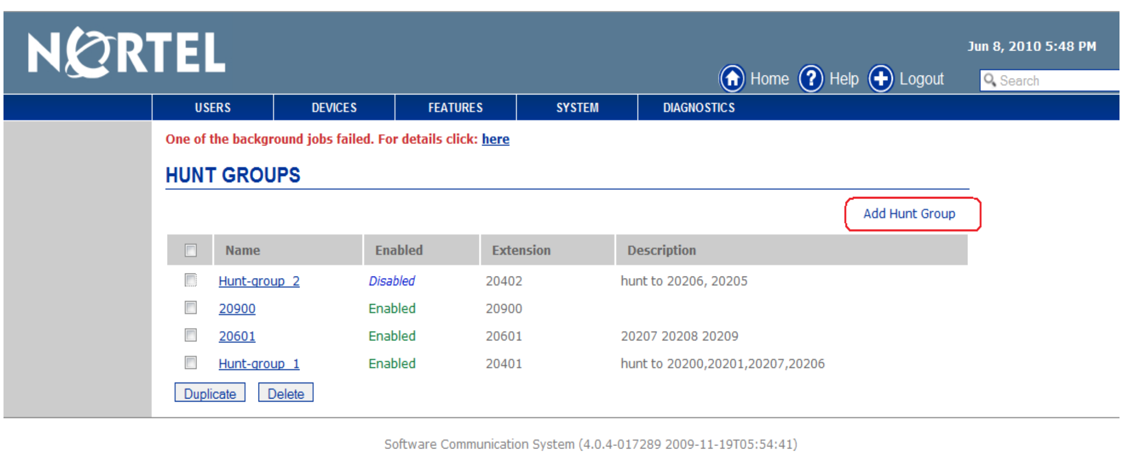

Figure 15: Hunt Group adding

- A page will appear as shown in **Figure 16** for the detail configuration of a hunt group.
- Check **Enabled** box to enable this hunt group, enter hunt group name and its associated **Extension**.
- Enter the **Description** for the Hunt Group.
- Click on **Apply** button to save the setting information.
- Click **OK** to return to the Hunt Group page.

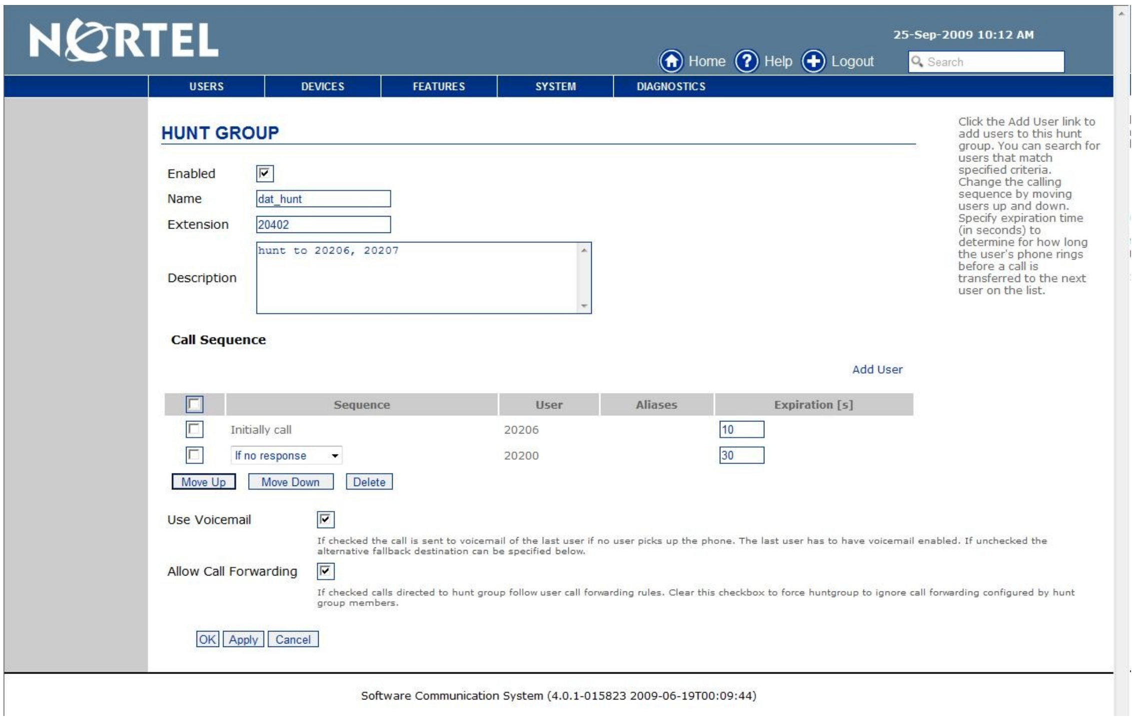

Figure 16: Hunt Group configuration

4.7. Dial Plan Configuration

Dial Plan Rules determine how calls are routed to and from the SCS system. The rules can be associated with configured SIP Trunk Gateways. The dial plan rules can be created to route calls internally and externally both locally and internationally. There are several types of rules that can be configured. Custom dial plan rules for specific call routing scenarios can be defined. From the main Dial Plans screen new rules can be created, existing rules can be modified, enabled or disabled, and the order of rules can be modified. Voicemail Dial Plan and Auto Attendant Dial Plan are available by default.

On the SCS server webpage, navigate to **SYSTEM** menu tab; select **Dial Plans** from pull down list. The list of dial plans will be shown in **Figure 17** below.

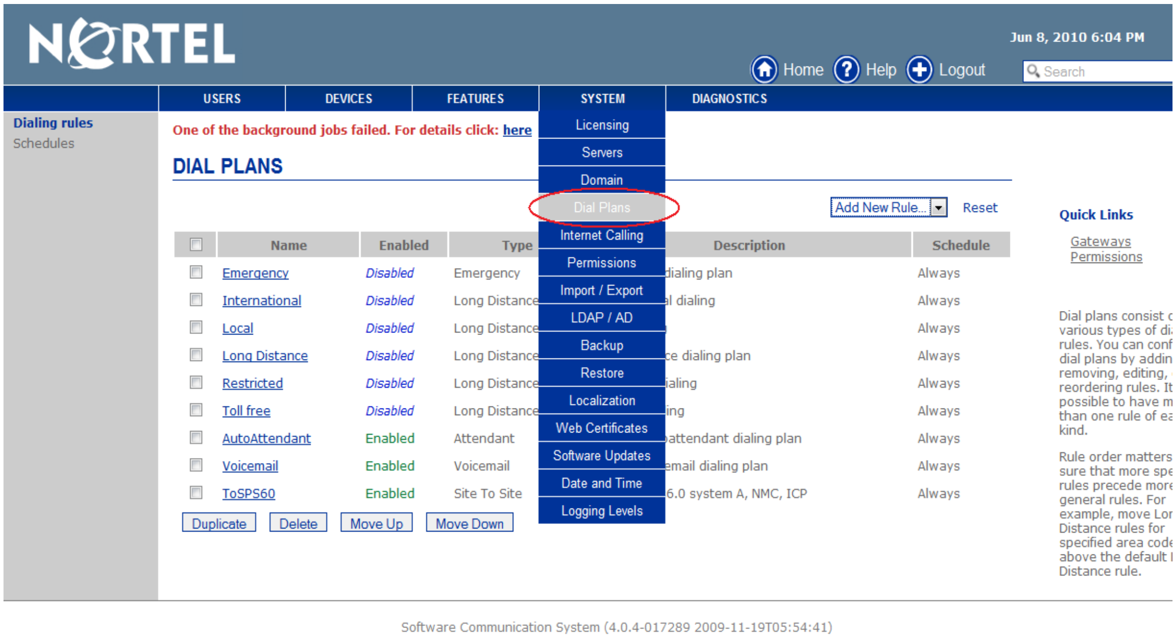

Figure 17: Dial Plan overview page

On the Dialing plan page, Auto Attendant and Voicemail are enabled by default as shown in **Figure 18** below.

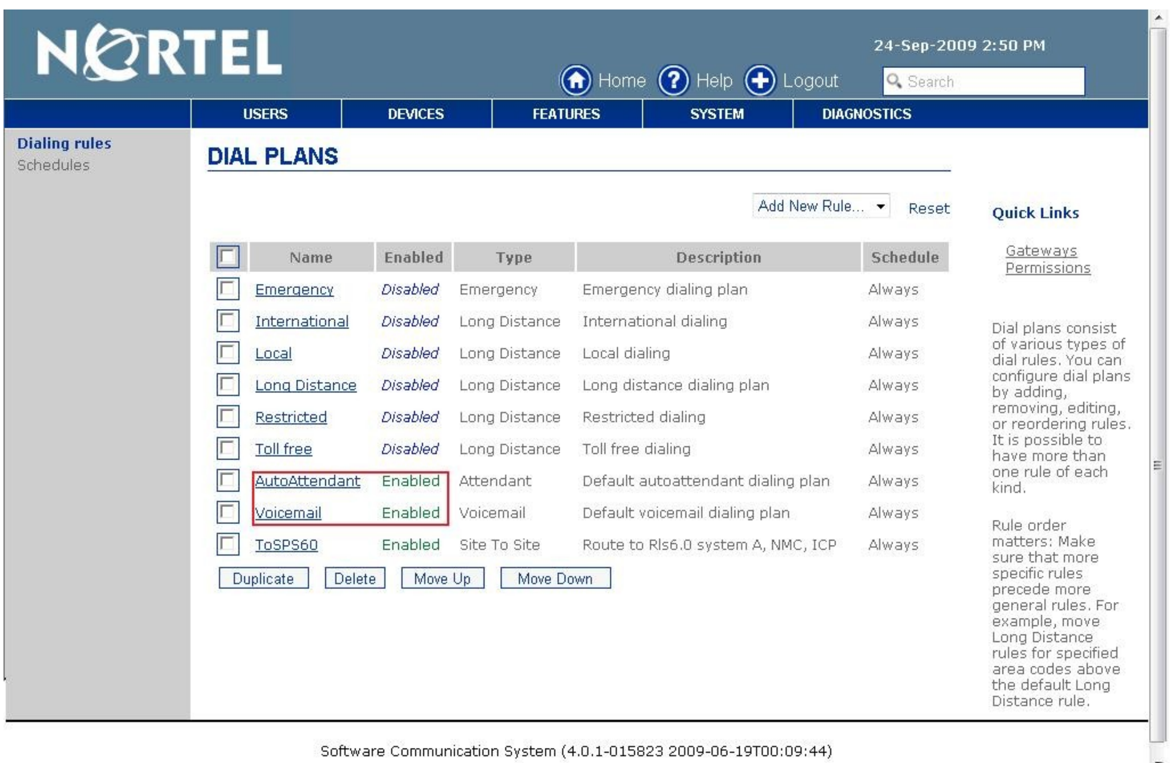

Figure 18: Dial Plan rules overview

On the Dial plan rules on **Figure 18**, click on **Voicemail** link. The Dial Rule for voice mail is displayed with extension **101** as default in **Figure 19** as shown below.

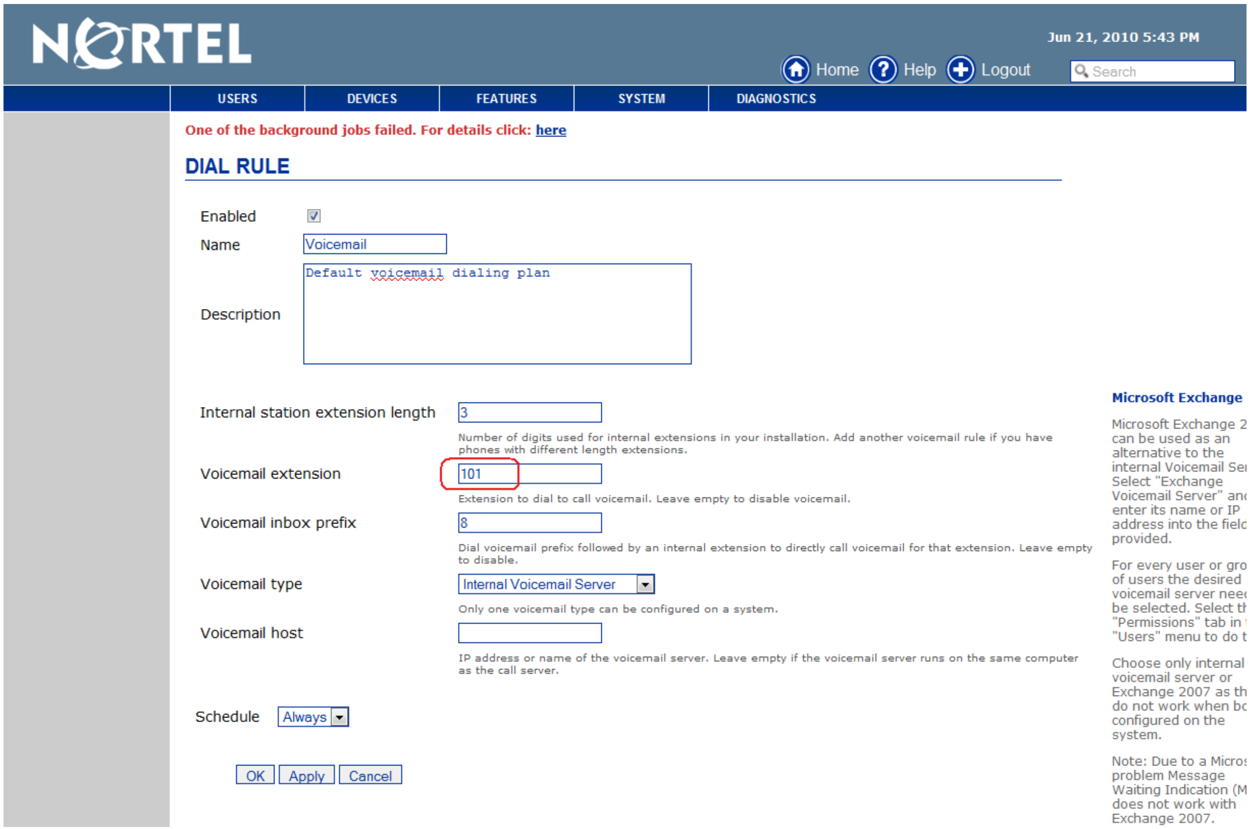

Figure 19: Dial Plan rules for Voice Mail

Return to the **Figure 18** above and click on **Auto Attendant** link. The Dial Plan Rule for auto attendant is displayed. The Auto Attendant Dial Plan allows user to select an auto attendant and its extension. By default, there is a single Auto Attendant Dialing rule that associates extension **100** with the operator as shown in **Figure 20** below.

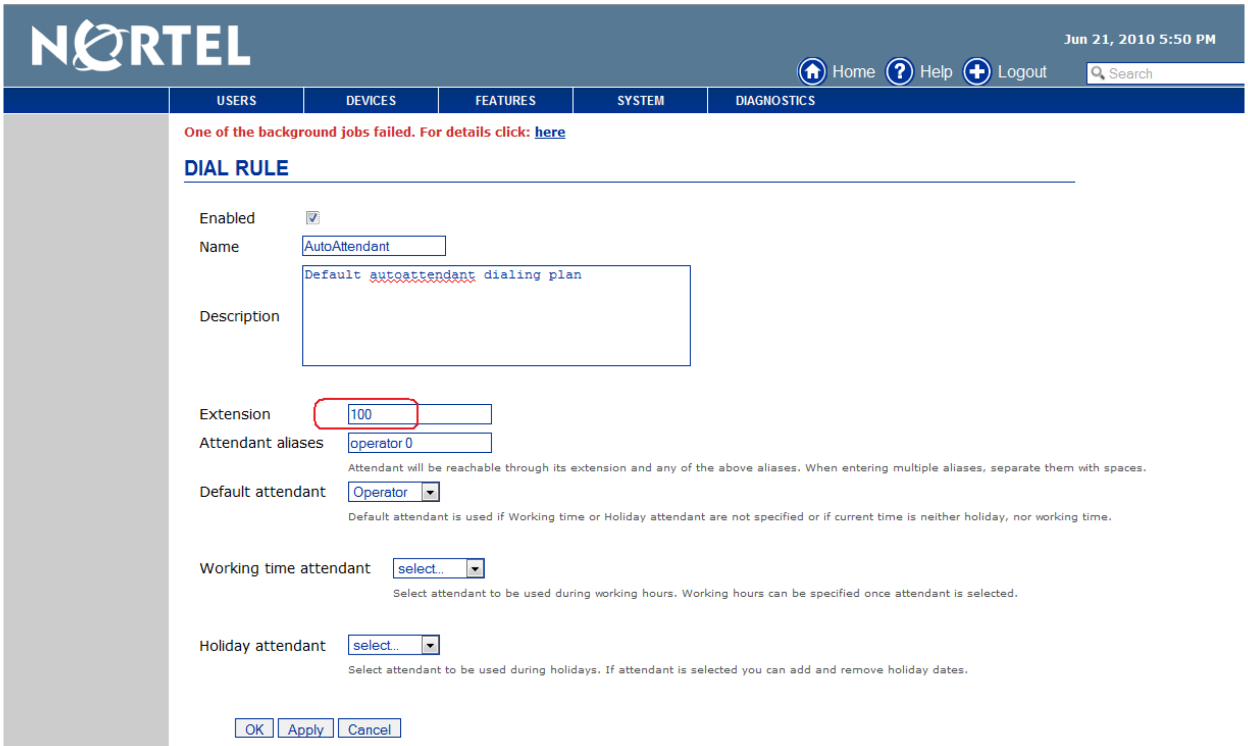

Figure 20: Dialing Plan rules for Auto Attendant

4.8. ACD Queue Configuration

An Automated Call Distribution (ACD) can be configured on the SCS system. This facilitates the management of incoming calls and their distribution to ACD agents. The agents are required to 'login' to their telephones in order to receive ACD calls.

On the SCS Server web page, click on the **FEATURES** menu tab; select **Call Center** on the pull down list. The list of Host Server name will be as shown in **Figure 21** below.

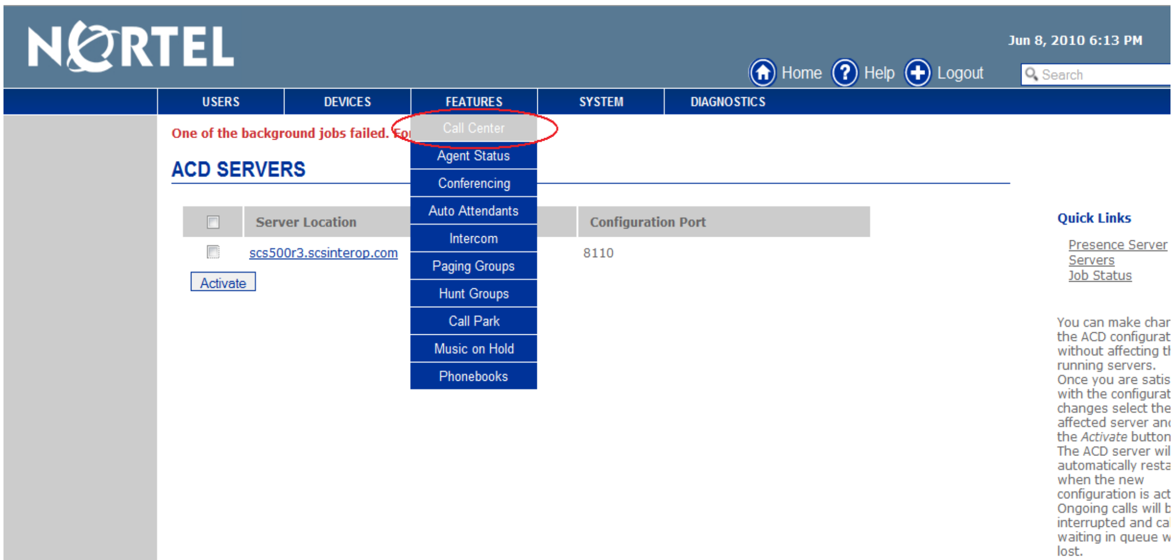

Figure 21: ACD Server list

Select the target server name as shown in **Figure 22**.

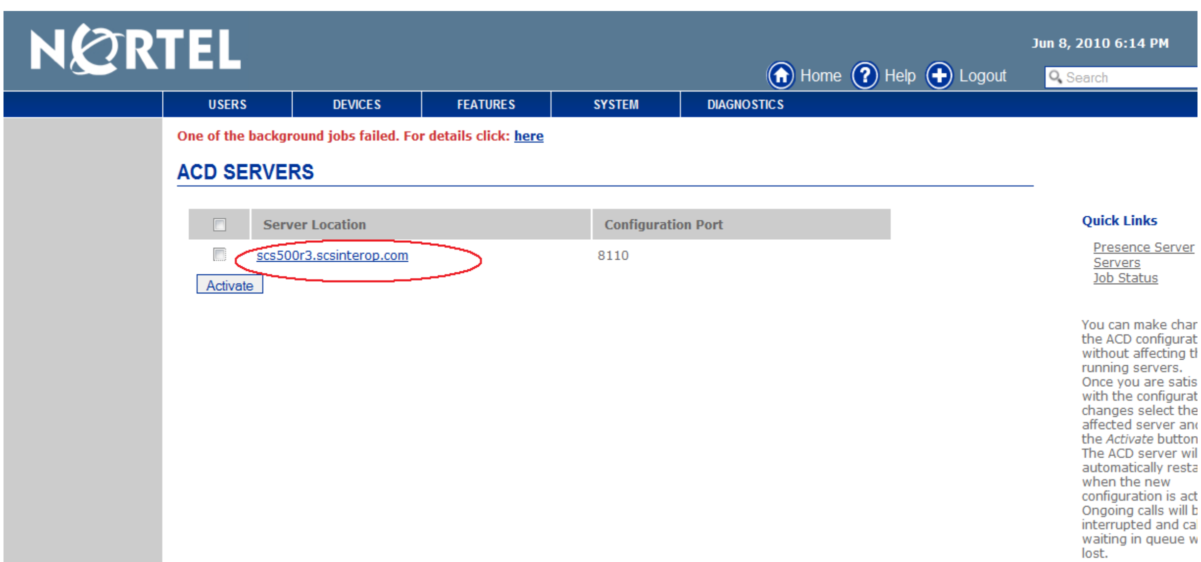

Figure 22: ACD Server Name

The ACD Server configuration will be shown as in **Figure 23**.

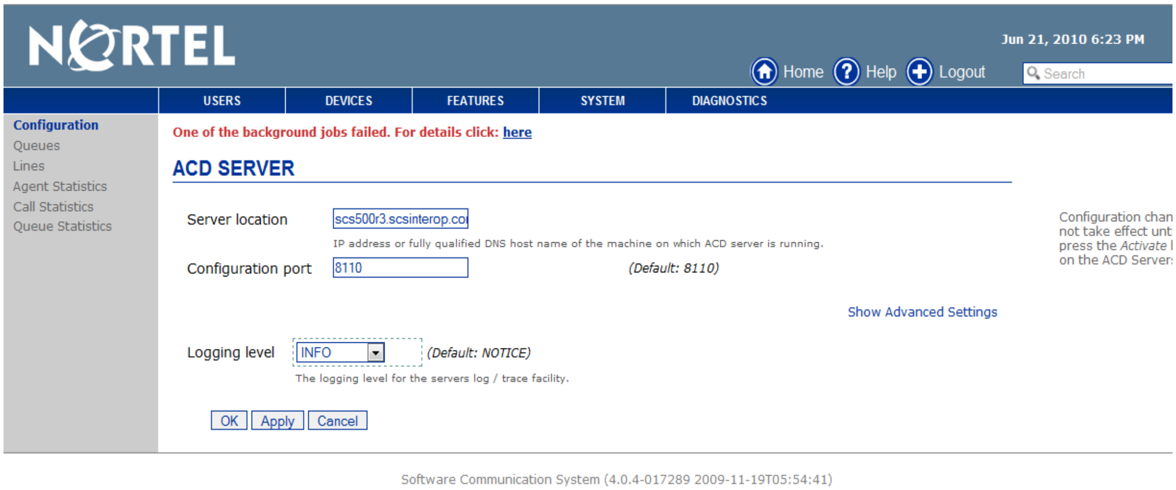

Figure 23: ACD Server Configuration

Click on **Queues** on the left panel to show the ACD Queues screen in **Figure 24**.

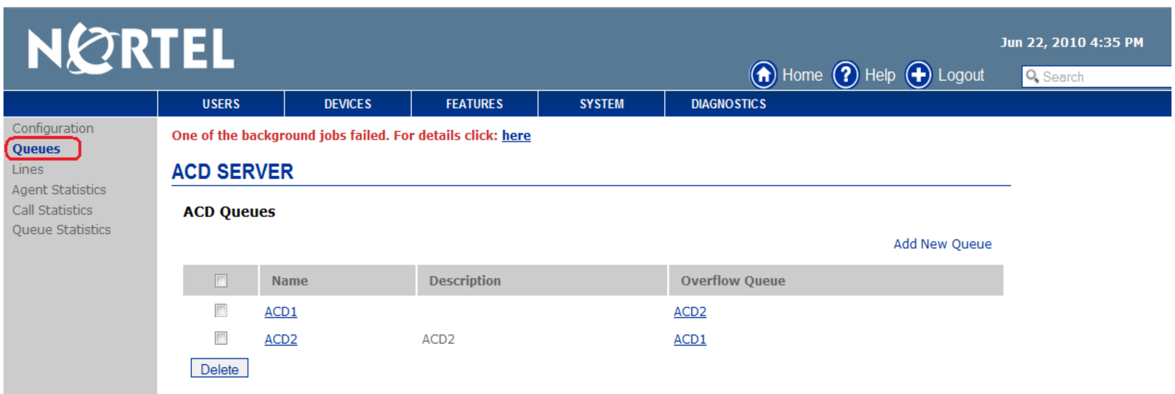

Figure 24: ACD Server Queue overview

Create a new ACD Queue by clicking **Add New Queue** as shown in **Figure 25**.

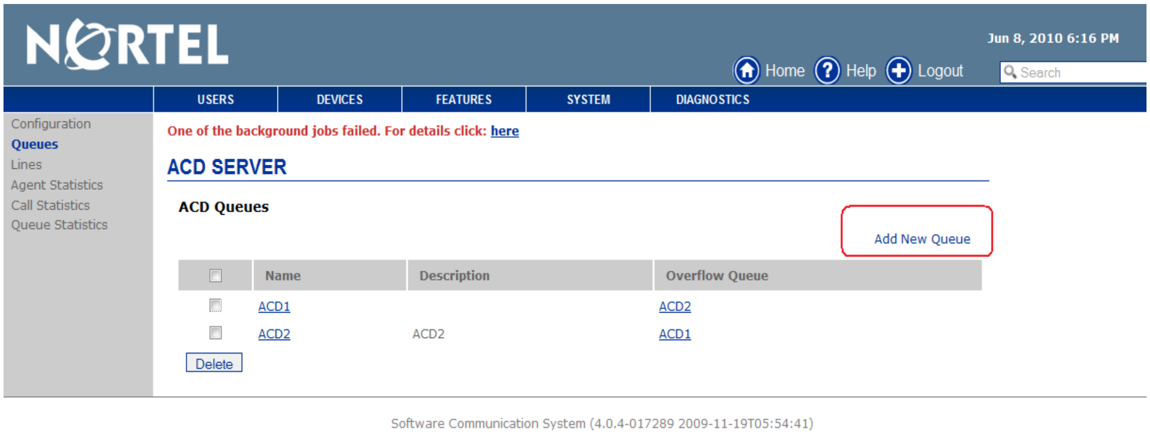

Figure 25: ACD Server Add New Queue

A page will appear as shown in **Figure 26** for detail configuration of an ACD Queue.
- Enter Name of ACD Queue.
- Others are at default.
- Click on **Apply** button to save the setting information.
- Click **OK** to return to the ACD Queue page.

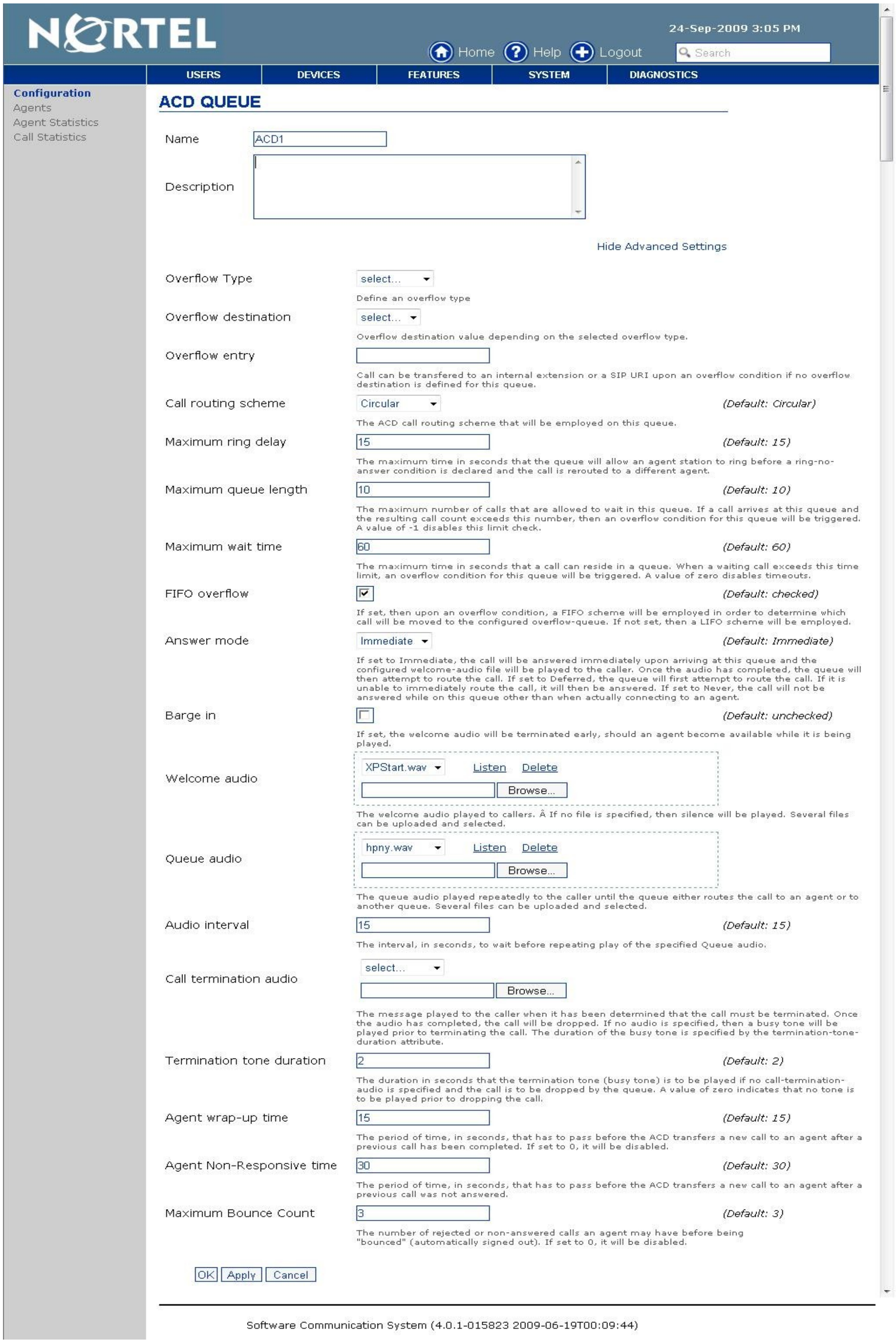

Figure 26: ACD Server Queue configuration

4.9. ACD Line Configuration

This section is to show how to create and configure ACD Line settings.

On **Figure 23**, on the left column menu, select **Lines** and click **Add New Line** as shown in **Figure 27** below.

Figure 27: ACD Line overview

A page will appear as shown in **Figure 28** for the detail configuration of an ACD Line.
- Enter ACD Line **Name** and its associated **Extension**.
- Choose **Queue** from the Queue list box.
- Others are at default.
- Click on **Apply** button to save the setting information.
- Click **OK** to return to the ACD Line page.

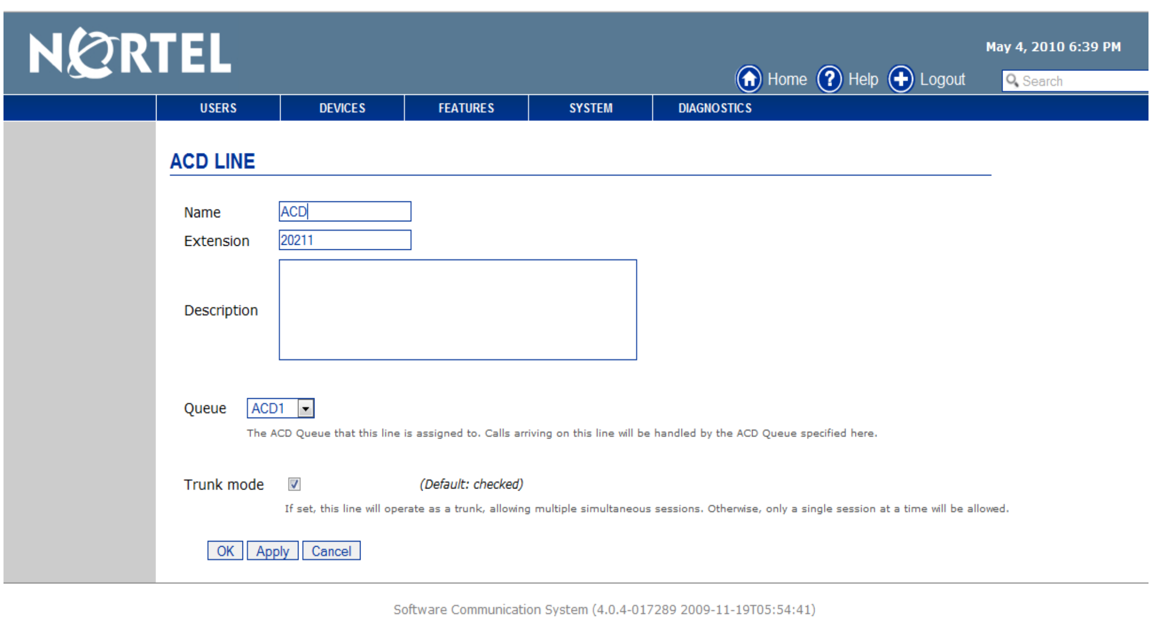

Figure 28: ACD Line configuration

4.10. Adding ACD Queue Agents

The agents who will answer the ACD calls can now be created and added to the ACD Queue.

On the SCS server webpage, navigate to **FEATURE** menu tab followed by **Call Center**. The list of Host Server name will be as shown in **Figure 29** below.

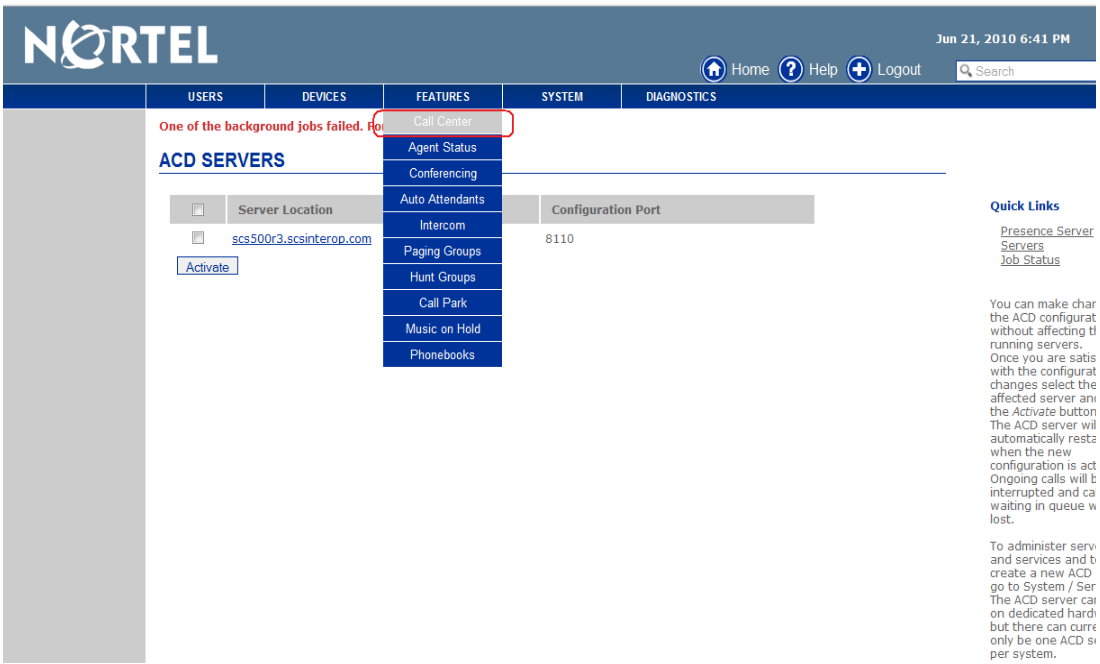

Figure 29: ACD Queue Agent overview

From the servers list shown in **Figure 30**, choose the target server name.

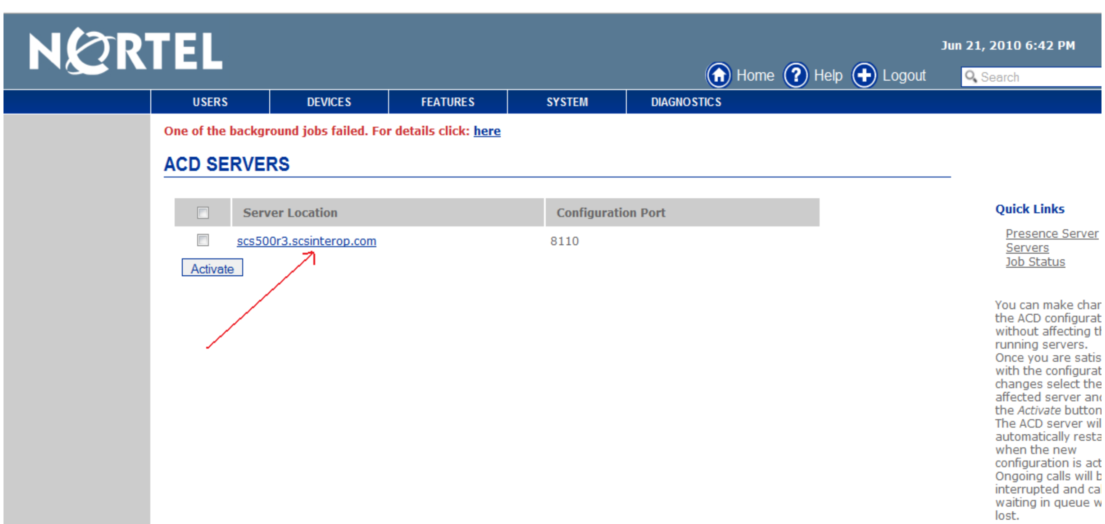

Figure 30: ACD Queue Agent overview

Then click on the ACD Queue name to which the agents will be added as shown in **Figure 31** below. In this example, a queue named **ACD1** has been selected.

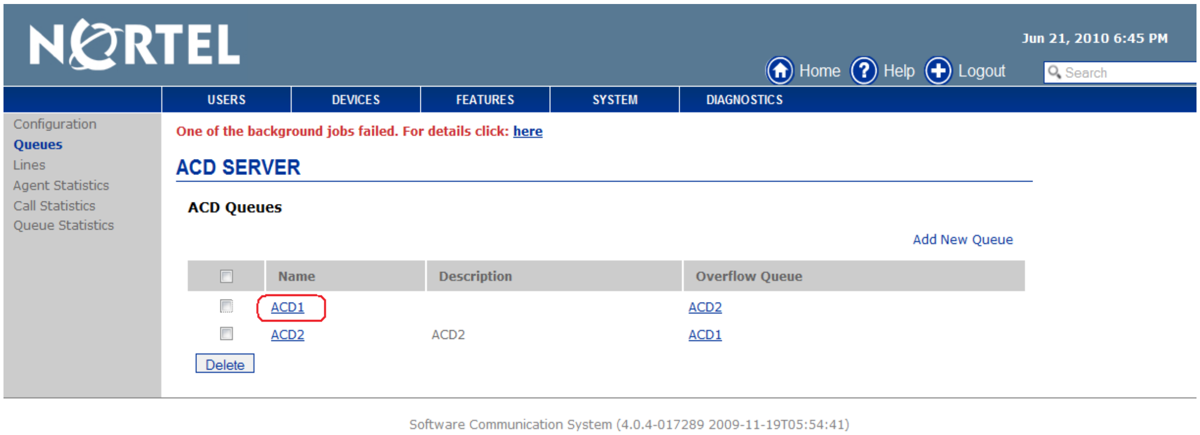

Figure 31: ACD Queue Server

A page will appear. User clicks the **Agents** on the left panel and click on **Add New Agent** to create a new Agent as shown in **Figure 32** below.

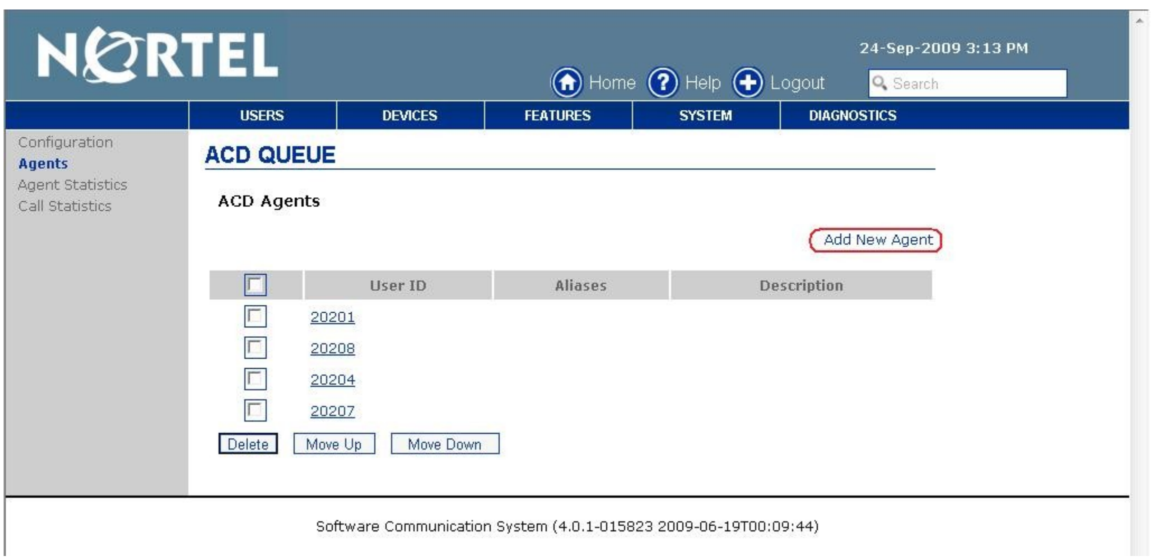

Figure 32: ACD Agents Adding

The Add ACD Agents screen will be displayed as shown in **Figure 33**. To view a list of administered agents, click on the **Search** button.

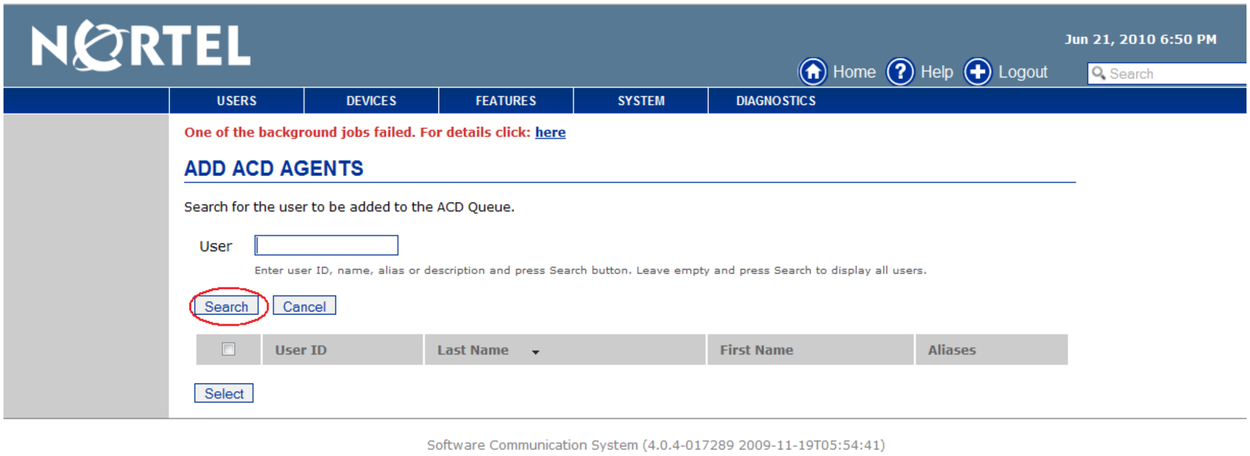

Figure 33: ACD Agents searching page

Select the check boxes next to the users to be added as agents to this ACD Queue. Then click the **Select** button.

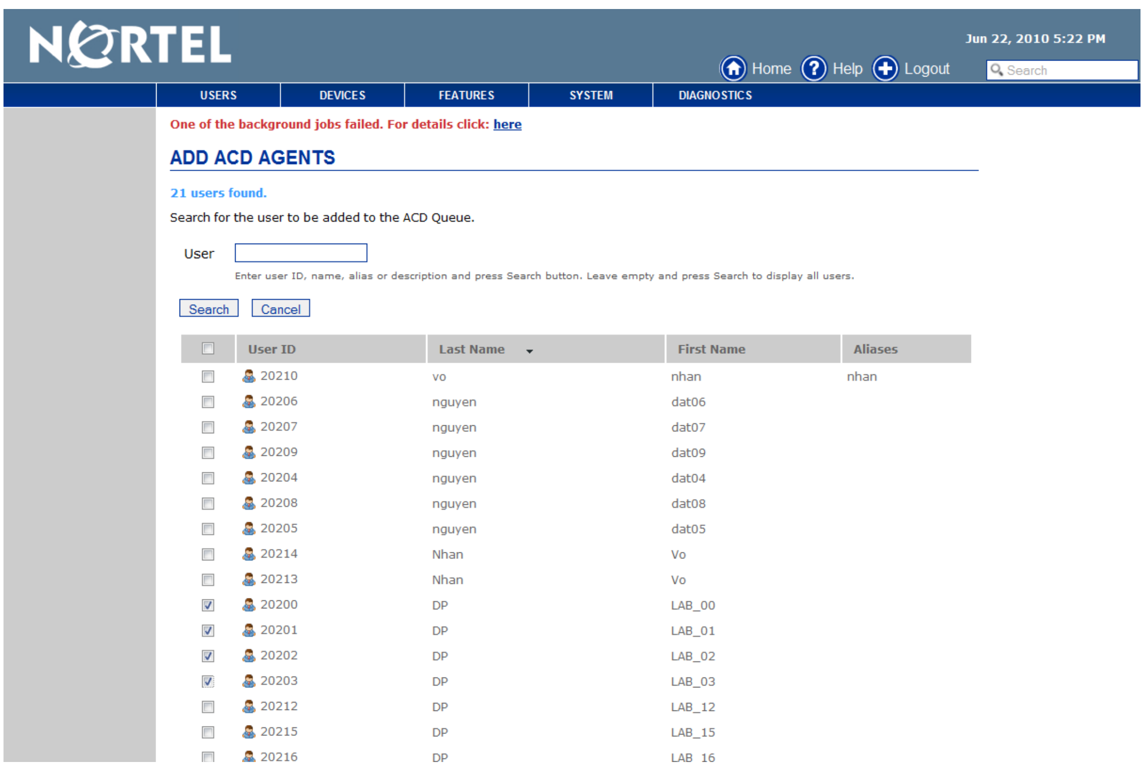

Figure 34: ACD Agents searching result page

The agents assigned to this ACD Queue will be displayed as shown in **Figure 35** below.

Figure 35: ACD Agents list on ACD Queue

4.11. Call Park Extension Creation

The call park feature enables calls to be transferred to a specified park 'extension'. When a call is parked, it can be retrieved by pressing *4 followed by the extension number. Music files can be uploaded to the SCS system to provide background music to parked callers. If there are several calls parked on the same 'extension', the first parked call is retrieved.

On the SCS server web page, navigate to **FEATURES** menu tab; select **Call Park** from pull down list as shown in **Figure 36** below.

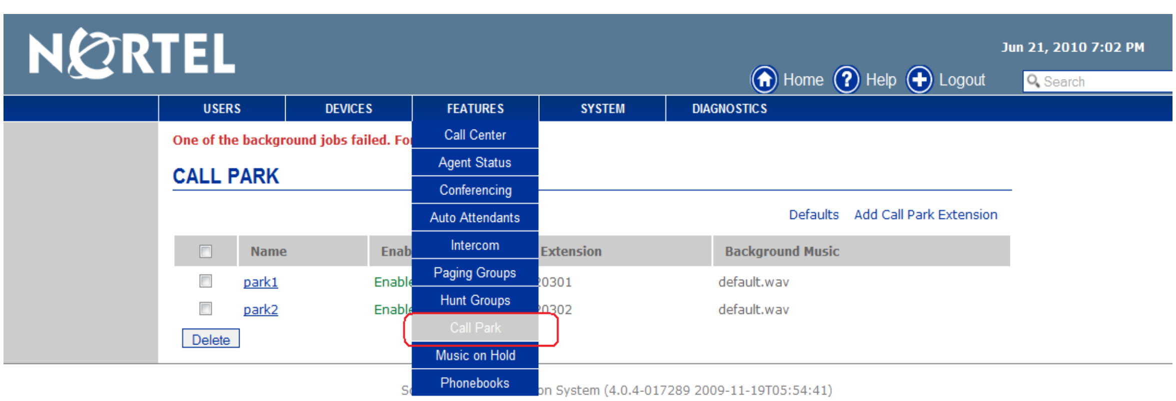

Figure 36: Call Park overview

The Call Park screen will be displayed. Click on the **Add Call Park Extension** link as shown in **Figure 37** below.

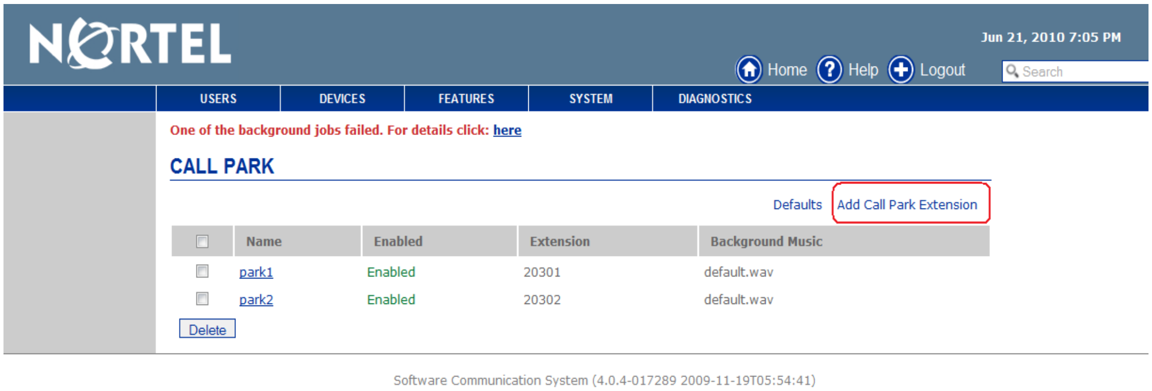

Figure 37: Call Park adding

A page will appear as shown in **Figure 38** for detail configuration of a Call Park Extension.
- Check **Enabled** box to enable this call park, enter call park name and its associated Extension.
- Enter **Description**.
- Select **Background music**.
- Others are at default.
- Click on **Apply** button to save the setting information.
- Click **OK** to return to the Call Park page.

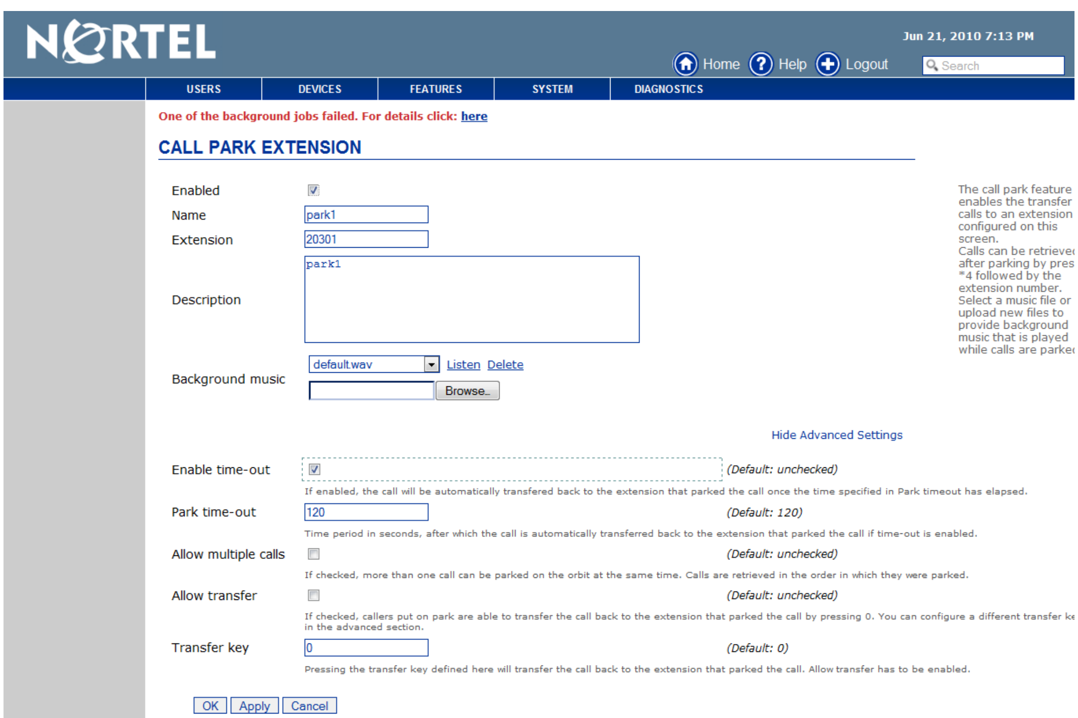

Figure 38: Call Park Extension configuration

The Call Park configuration details will be displayed on **Figure 39**.

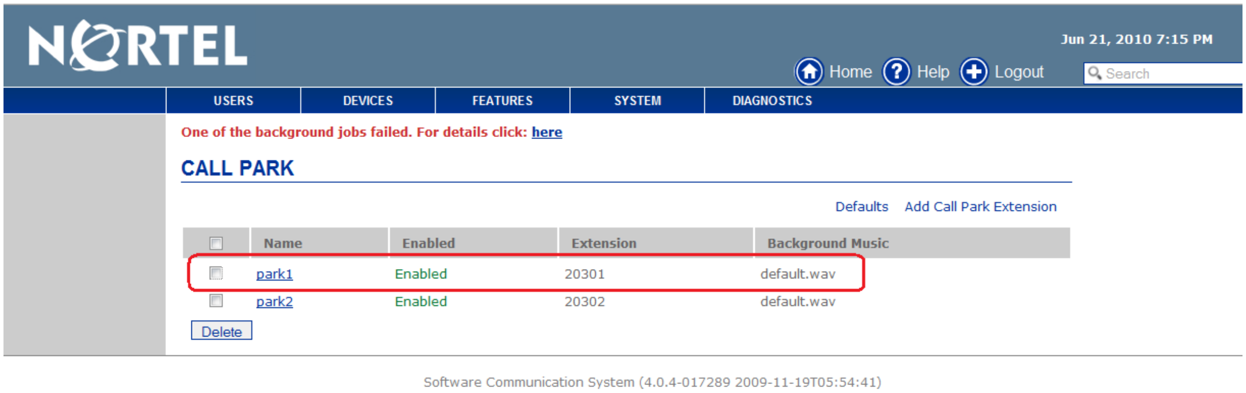

Figure 39: Call Park Extension list

5. Configure G-Tek MT-102H SIP Telephone

The following sections below explains the configuration for the G-Tek MT-102H SIP Telephones with respect to SIP account, Codec and Speed dial settings.

5.1. SIP Account Settings

Configure SIP Settings as displayed in the below screenshot. The Domain Server, Proxy Server and Outbound Proxy attribute values are as per the configurations explained in **Section 4.1** and **4.2**.

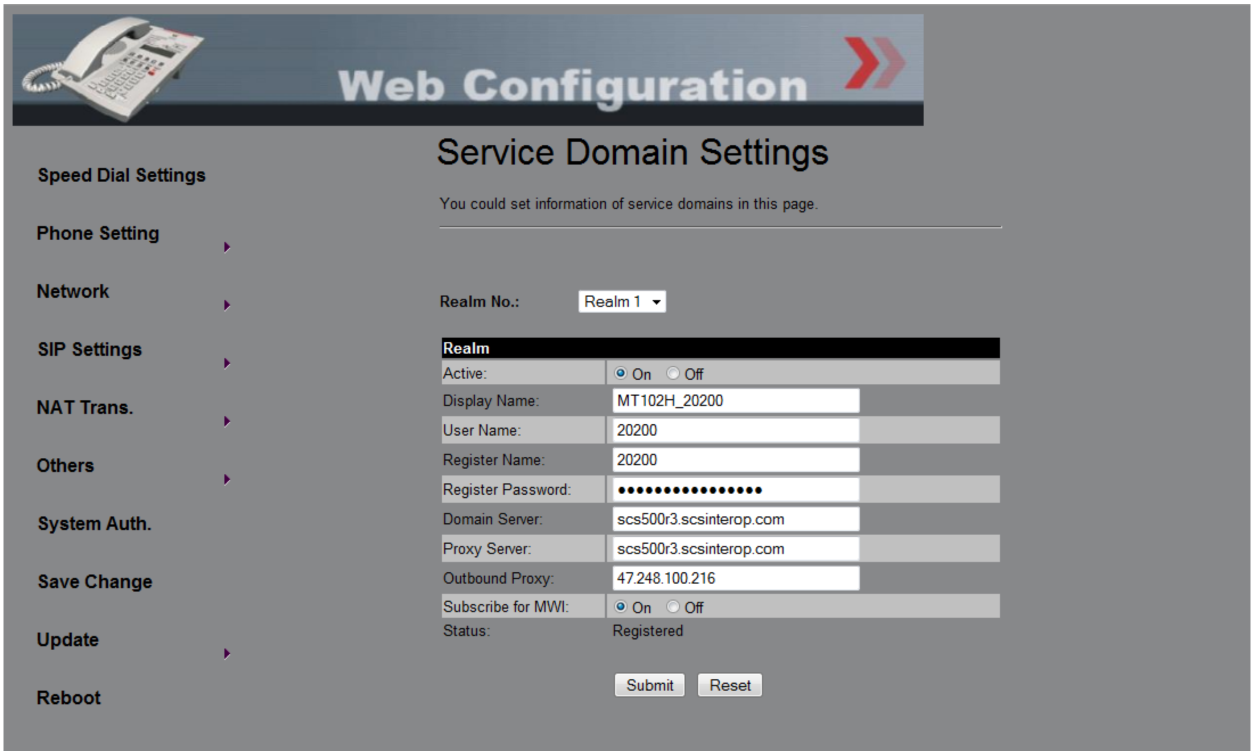

Figure 40 – G-Tek MT-102H SIP Account Settings

5.2. Codec settings

Configure Codec Settings as displayed in the below screenshot.

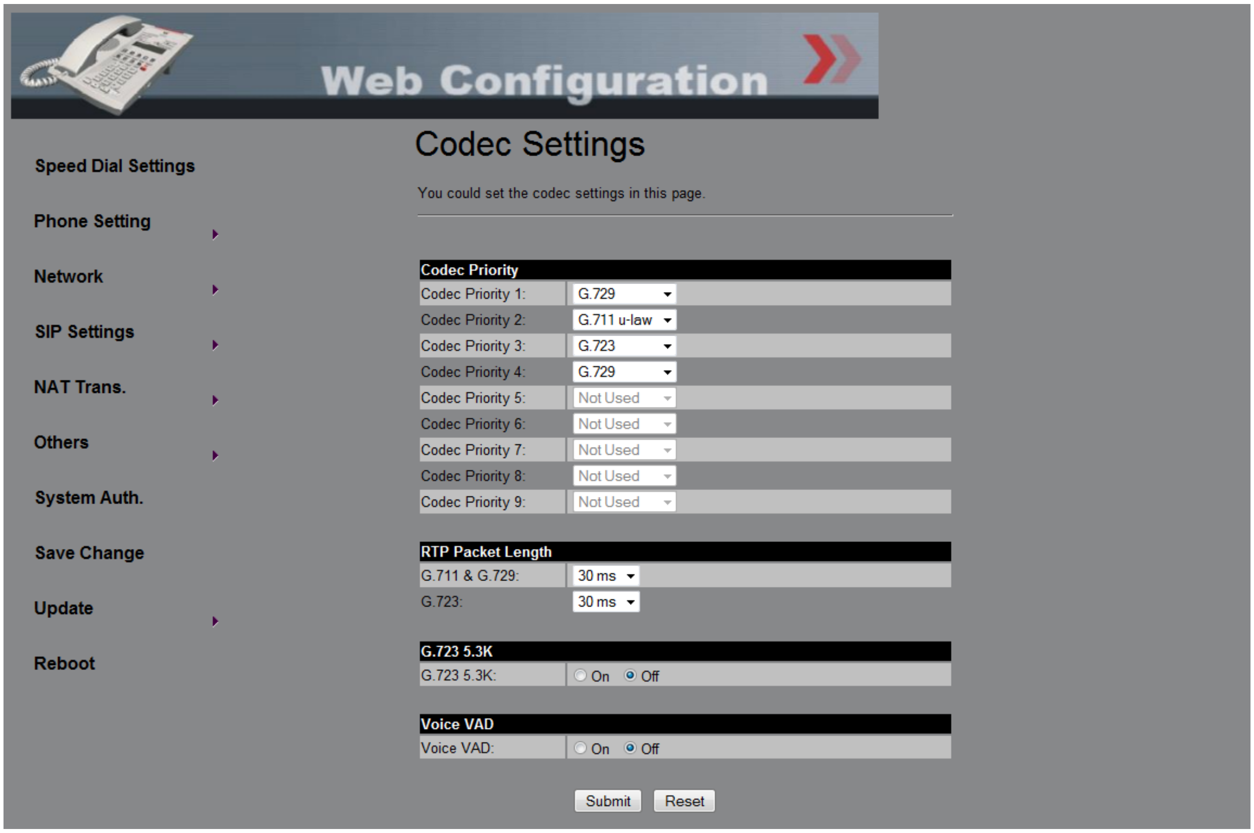

Figure 41 – G-Tek MT-102H Codec Settings

5.3. Speed Dial

Configure Speed Dial Settings as displayed in the below screenshot. The example 20206 shown below is configured in **Section 4.3**.

Figure 42 – G-Tek MT-102H Speed Dial Settings

6. General Test Approach and Test Results

The focus of this interoperability compliance testing was primarily to verify the call establishment on the G-Tek MT-102H SIP telephones and the feature operations such as: DTMF, MWI, codec negotiation, perform blind transfer as the transferee, Forward in Parallel, Forward Sequentially, Forked Invite, Call Pick-up, Call Park and Retrieved.

6.1. General Test Approach

The general test approach was to have one of the SCS clients/users to place a call to and from the G-Tek MT-102H and exercise the telephony features. The main objectives were to verify the MT-102H successfully perform the following:

- Register to Software Communication System Release 3.0.
- Call establishment with Avaya Software Communication System SIP clients
- Basic call operation: DTMF transmission, voicemail with MWI notification,
- Advance Software Communication System features: perform blind transfer as the transferee, Forward in Parallel, Forward Sequentially, Forked Invite, Call Pick-up, Call Park and Retrieved, ACD Server, Meet me conference.
- Handle of loop or too many hops, long "Via" path resulting in large SIP messages.
- Call redirection and conference: Avaya phones as a transferor for blind/consultative transfers and as a moderator for the 3 way conference call.
- Specific hospitality feature requirement, speed dial.
- Codec negotiations.

6.2. Test Results

The objectives outlined in **Section 6.1** were verified and met. The following observations were made during the compliance testing:

- On Blind Transfer scenario with Avaya Sip Phone 12xx as transferor, "Ringback – Remote Hold" shows on LCD of the transferee MT 102H instead of "DN Transferring".
- Media Service on the SCS supports only G711; therefore the MT/MB102H codec options should be set to either G711 only or G711/G729.
- The CLID is not displayed correctly sometimes when Auto Attendant transfers. It is intermittent.

7. Verification Steps

This section includes some steps that can be followed to verify the configuration.

- Verify that the MT-102H telephone registers successfully with the Software Communication System by using its Web GUI http: //<IPAddress>:8000 with username/password: admin/1234.
- On Web GUI of MT-102H phone, check for the status as "Registered" on *SIP Setting > Service Domain.*
- Place a call from and to the MT-102H phone and verify that the call is established with 2 way speech path.
- During the call, use pcap tool (ethereal/wireshark) at the SCS Server and clients to make sure that all SIP request/response messages are displayed correctly.

8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 6.1**, with some exceptions outlined in **Section 6.2**. The outstanding issues are being investigated by G-Tek and Avaya. Some of these issues are considered as exceptions. G-Tek MT-102H SIP telephone version 1510X.27.1.02i has passed compliance testing with Software Communication System Release 3.0.

9. Additional References

Product documentation for Avaya Software Communication System may be found at:
http://support.nortel.com/go/main.jsp

Product information for G-Tek products can be found at
http://www.gtek.com.tw/en/offering_products.php?mw=9

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.