



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	<ul style="list-style-type: none">4.0.4-017289 2009-11-19T05
Avaya 1210 SIP client Avaya 1230 SIP client Avaya SMC3456	<ul style="list-style-type: none">Model NTYS18, series 01.02.02.00Model NTYS20, series 01.02.02.00Version 2.6, build 56076
G-Tek MT-102H SIP Telephones	<ul style="list-style-type: none">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.

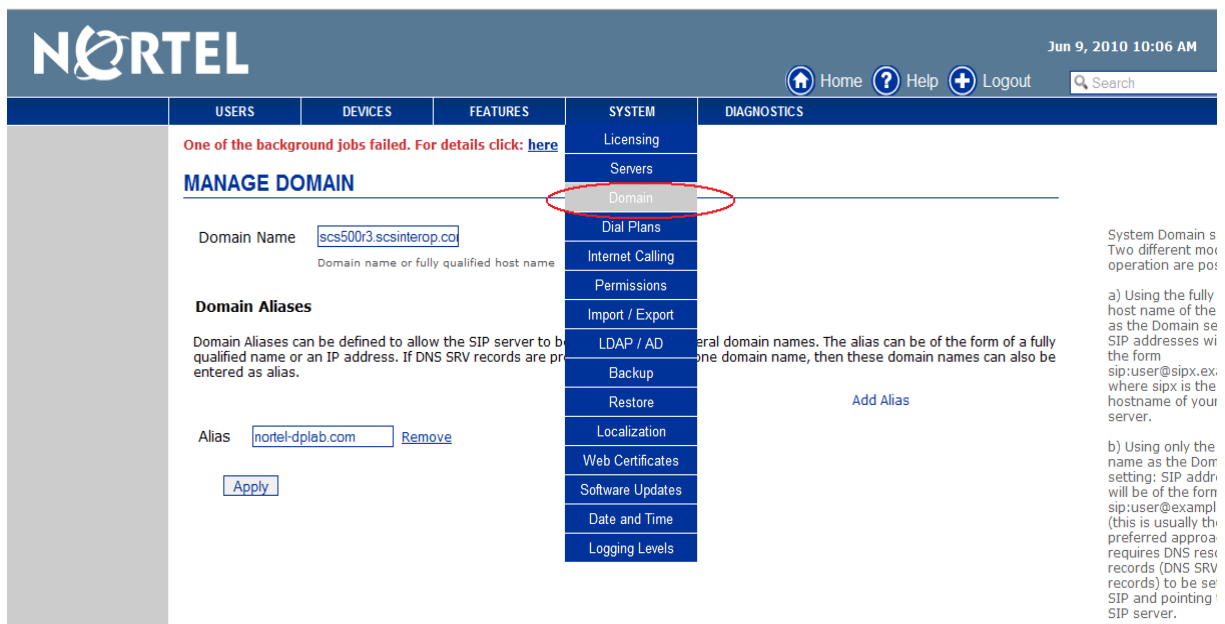


Figure 2: Domain management

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

The screenshot shows the 'MANAGE DOMAIN' page in the Nortel Software Communication System. The page has a blue header with the Nortel logo and navigation links: Home, Help, Logout, and a search bar. Below the header is a menu bar with tabs: USERS, DEVICES, FEATURES, SYSTEM, and DIAGNOSTICS. The main content area is titled 'MANAGE DOMAIN' and contains a 'Domain Name' field with the value 'scs500r3.scsinterop.com'. Below this is a section for 'Domain Aliases' with a description: 'Domain Aliases can be defined to allow the SIP server to be responsible for several domain names. The alias can be of the form of a fully qualified name or an IP address. If DNS SRV records are present for more than one domain name, then these domain names can also be entered as alias.' There is an 'Add Alias' button. Below the button are two rows of alias entries: 'Alias: nortel-dplab.com' with a 'Remove' link, and 'Alias: 47.248.100.216' with a 'Remove' link. An 'Apply' button is at the bottom of the alias section. On the right side of the page, there is a 'System Domain setting' section with two modes of operation: a) Using the fully qualified host name of the server as the Domain setting, and b) Using only the domain name as the Domain setting. A note at the bottom right states: 'Note: If the Domain Name setting is changed in a production system, all the phone profiles have to be regenerated. Use with caution. Using IP address as a domain alias will prevent high availability configuration from working properly.'

25-Sep-2009 10:06 AM

Home Help Logout Search

USERS DEVICES FEATURES SYSTEM DIAGNOSTICS

MANAGE DOMAIN

Domain Name
Domain name or fully qualified host name

Domain Aliases

Domain Aliases can be defined to allow the SIP server to be responsible for several domain names. The alias can be of the form of a fully qualified name or an IP address. If DNS SRV records are present for more than one domain name, then these domain names can also be entered as alias.

Add Alias

Alias Remove

Alias Remove

Apply

System Domain setting. Two different modes of operation are possible:

a) Using the fully qualified host name of the server as the Domain setting: SIP addresses will be of the form sip:user@sipx.example.com, where sipx is the hostname of your SIP server.

b) Using only the domain name as the Domain setting: SIP addresses will be of the form sip:user@example.com (this is usually the preferred approach). This requires DNS resource records (DNS SRV records) to be setup for SIP and pointing to the SIP server.

Note: If the Domain Name setting is changed in a production system, all the phone profiles have to be regenerated. Use with caution. Using IP address as a domain alias will prevent high availability configuration from working properly.

Software Communication System (4.0.1-015823 2009-06-19T00:09:44)

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**

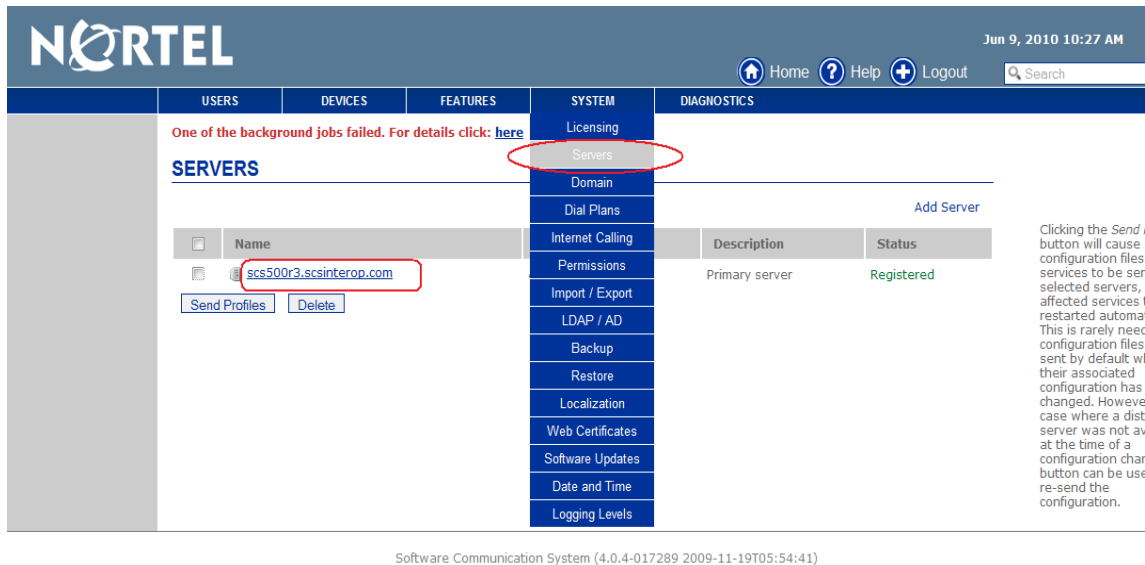


Figure 4: Domain Server Configuration Overview

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.

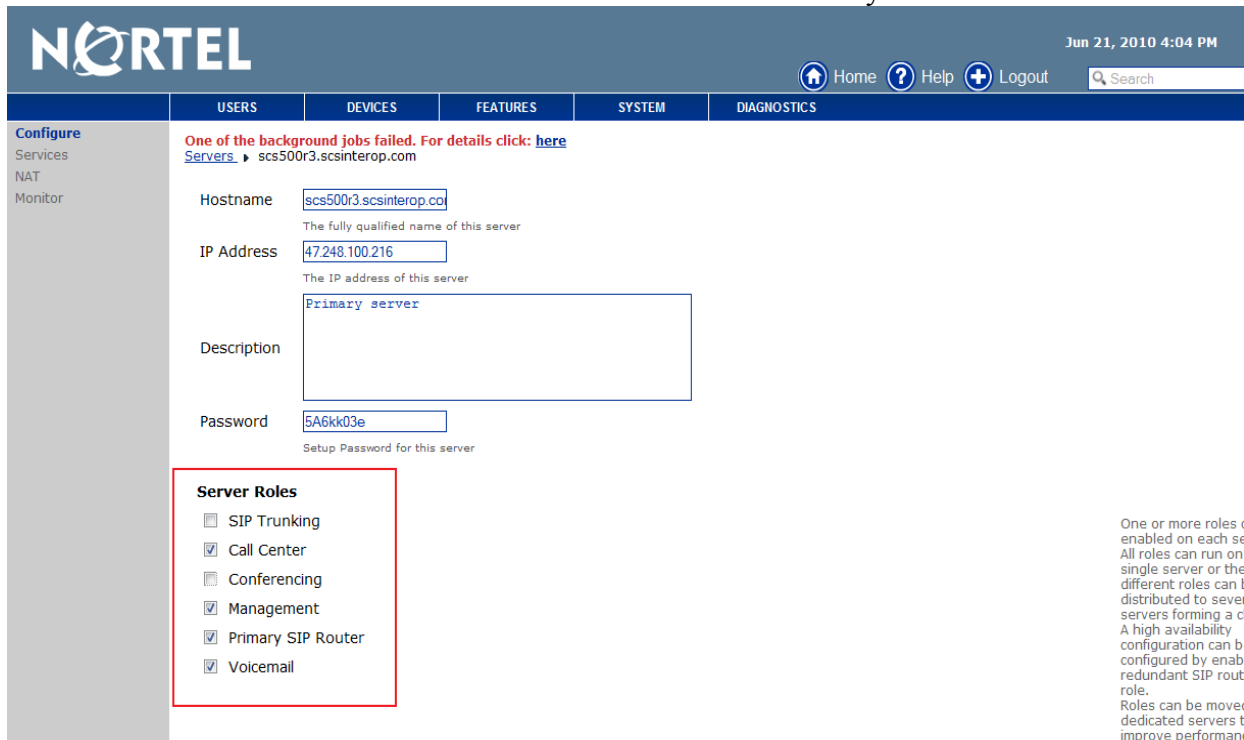


Figure 5: Domain Server Configuration with default Server Role

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**.

NORTEL Jun 9, 2010 10:30 AM

Home Help Logout Search

Configure **USERS** **DEVICES** **FEATURES** **SYSTEM** **DIAGNOSTICS**

One of the background jobs failed. For details click: [here](#)

Servers ▸ scs500r3.scsinterop.com

Hostname:
The fully qualified name of this server

IP Address:
The IP address of this server

Description:

Password:
Setup Password for this server

Server Roles

- ☒ SIP Trunking
- ☒ Call Center
- ☒ Conferencing
- ☒ Management
- ☒ Primary SIP Router
- ☒ Voicemail

One or more roles can be enabled on each server. All roles can run on a single server or the different roles can be distributed to several servers forming a cluster. A high availability configuration can be configured by enabling redundant SIP routing roles.

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**.

NORTEL Jun 9, 2010 10:32 AM

Home Help Logout Search

USERS **DEVICES** **FEATURES** **SYSTEM** **DIAGNOSTICS**

Users (selected) One of the background jobs failed. For details click: [here](#)

User Groups

Extension Pool

[Add New User](#)

Filter by: [v]

	User ID	First Name	Last Name	Aliases
<input type="checkbox"/>	2000	Nhan	2000	
<input type="checkbox"/>	2001	Nhan	2001	
<input type="checkbox"/>	20200	LAB_00	DP	
<input type="checkbox"/>	20201	LAB_01	DP	
<input type="checkbox"/>	20202	LAB_02	DP	
<input type="checkbox"/>	20203	LAB_03	DP	
<input type="checkbox"/>	20204	dat04	nguyen	
<input type="checkbox"/>	20205	dat05	nguyen	
<input type="checkbox"/>	20206	dat06	nguyen	

Select the Add New User link and create a new user.

After user is created can associate it with one or more managed phones

Figure 7: User Configuration

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

Jun 21, 2010 7:19 PM

Home
Help
Logout

USERS
DEVICES
FEATURES
SYSTEM
DIAGNOSTICS

One of the background jobs failed. For details click: [here](#)

NEW USER

User ID

20206

The User ID can be a numeric extension like "123" or a name like "jsmith". The User ID is displayed by the phone and it is therefore recommended to use the internal extension or the name of the user. If using Direct Inward Dialing (DID), then it is recommended to define the DID number (or its DNIS portion) as an alias.

Last name

First name

Active greeting

default system greeting ▾

Voicemail prompt callers will hear before leaving a message.

E-mail address

Used for sending notification about new voicemail left for this user. Leave empty to disable e-mail notification.

Attach voicemail

☐

If checked, the voicemail message will be attached to the notification e-mail. Otherwise, the e-mail will contain a link to retrieve voicemail message.

Additional E-mail address

Used for sending voicemail message notification to the additional e-mail address.

Attach voicemail

☐

If checked, the voicemail message will be attached to the notification email sent to the additional e-mail address.

PIN

Confirm PIN

SIP password

L7R17BD0

The SIP password is used by the user's phone to register with the SIP proxy. For phones managed by this system, the SIP password entered here will be configured automatically on the phone. For unmanaged phones, the SIP password is needed when manually configuring lines on the phone. The security of this password is very important and that is why a secure password is auto-generated.

Groups

List all groups for this user. If a group does not exist, it will be created. When entering multiple groups, separate them with spaces.

Quick Links

[Extension Pool](#)

Existing Groups:
administrators, dplab, 20201, 20405, 2200, 20200, 20206, administrators,, 2000

New Groups: You can create new groups simply by adding the group name to the Groups form value.

Done

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**.

NORTEL 24-Sep-2009 2:50 PM

Home ? Help + Logout Search

	USERS	DEVICES	FEATURES	SYSTEM	DIAGNOSTICS
Identification	<p>User: 20206</p> <p>Hide Advanced Settings</p> <p>Existing Groups: administrators, dplab, 20201, 20405, 2200, 20200, 20206</p> <p>New Groups: You can create new groups simply by adding the new group name to the Groups form value.</p> <p>Select <i>Phones</i> to add this user to one or more phones.</p>				
Phones	<p>User ID: <input type="text" value="20206"/></p> <p>The User ID can be a numeric extension like "123" or a name like "jsmith". The User ID is displayed by the phone and it is therefore recommended to use the internal extension or the name of the user. If using Direct Inward Dialing (DID), then it is recommended to define the DID number (or its DNIS portion) as an alias.</p>				
Call Forwarding	<p>Last name: <input type="text" value="nguyen"/></p>				
Schedules	<p>First name: <input type="text" value="dat06"/></p>				
Speed Dial	<p>Active greeting: <input type="text" value="default system greeting"/></p> <p>Voicemail prompt callers will hear before leaving a message.</p>				
ACD Agent Supervisor	<p>E-mail address: <input type="text"/></p> <p>Used for sending notification about new voicemail left for this user. Leave empty to disable e-mail notification.</p>				
Personal Auto-Attendant	<p>Attach voicemail: <input type="checkbox"/></p> <p>If checked, the voicemail message will be attached to the notification e-mail. Otherwise, the e-mail will contain a link to retrieve voicemail message.</p>				
Conferences	<p>Additional E-mail address: <input type="text"/></p> <p>Used for sending voicemail message notification to the additional e-mail address.</p>				
Registrations	<p>Attach voicemail: <input type="checkbox"/></p> <p>If checked, the voicemail message will be attached to the notification email sent to the additional e-mail address.</p>				
Permissions	<p>PIN: <input type="password" value="••••••••"/></p>				
Caller ID	<p>Confirm PIN: <input type="password" value="••••••••"/></p> <p>The PIN is a password used to log in to voicemail or to the user portal. Numeric PINs are recommended, since only numbers can be dialed.</p>				
	<p>SIP password: <input type="password" value="1234"/></p> <p>The SIP password is used by the user's phone to register with the SIP proxy. For phones managed by this system, the SIP password entered here will be configured automatically on the phone. For unmanaged phones, the SIP password is needed when manually configuring lines on the phone. The security of this password is very important and that is why a secure password is auto-generated.</p>				
	<p>Groups: <input type="text" value="administrators dplab"/></p> <p>List all groups for this user. If a group does not exist, it will be created. When entering multiple groups, separate them with spaces.</p>				
	<p>Aliases: <input type="text"/></p> <p>Aliases are additional names for the user. Like the user ID, an alias can be either a numeric extension or a name. When entering multiple aliases, separate them with spaces.</p>				
	<p><input type="button" value="OK"/> <input type="button" value="Apply"/> <input type="button" value="Cancel"/></p>				

Software Communication System (4.0.1-015823 2009-06-19T00:09:44)

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.

The screenshot displays the Nortel User Configuration interface for user 20206. The top navigation bar includes the Nortel logo, a search bar, and links for Home, Help, and Logout. The left sidebar lists various configuration categories, with 'Permissions' selected. The main content area is titled 'User: 20206' and 'Permissions'. It is divided into three sections: General Permission, Call Permission, and Voicemail Server. Each section contains a list of permissions with checkboxes and default status indicators.

Category	Permission	Default	
General Permission	Superadmin Access	checked	
	Change PIN from IVR	unchecked	
	Configure Personal Auto Attendant	checked	
Call Permission	900 Dialing	checked	
	Attendant Directory	checked	
	International Dialing	checked	
	Local Dialing	checked	
	Long Distance Dialing	checked	
	Mobile Dialing	checked	
	Toll Free	checked	
	Voice Mail	checked	
	Record System Prompts	checked	
	ToSPS	checked	
	Voicemail Server	Internal Voicemail Server	checked
		Microsoft Exchange UM Voicemail Server	checked

Buttons: OK, Apply, Cancel

Software Communication System (4.0.1-015823 2009-06-19T00:09:44)

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**.

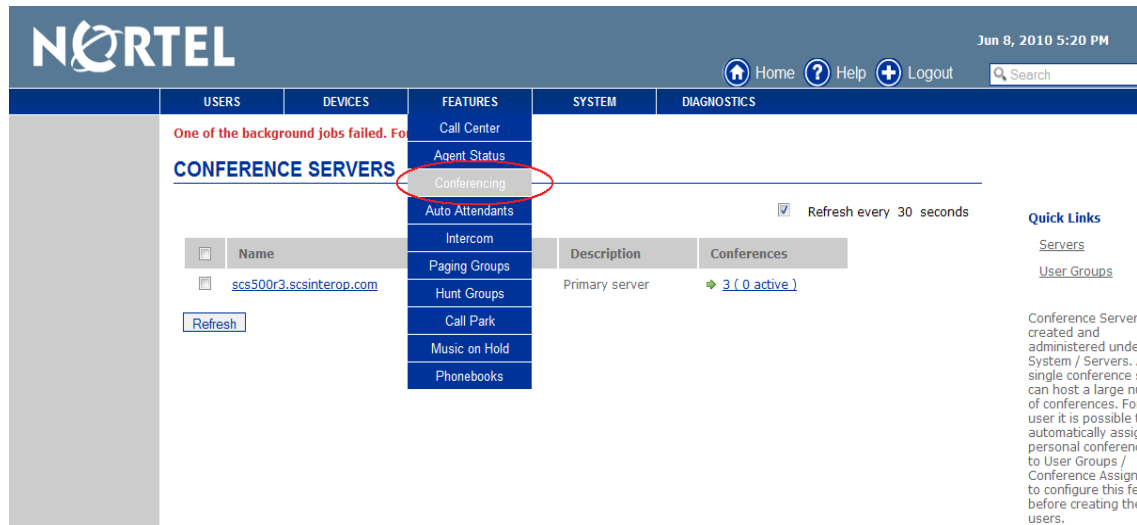


Figure 11: Conference Bridge overview page

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**.

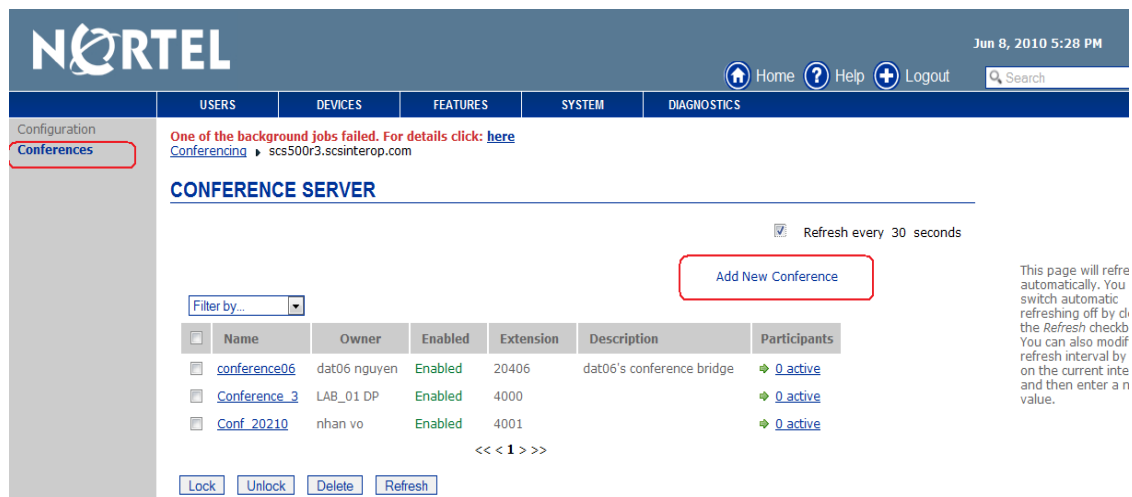


Figure 12: Adding Conference Bridge

- 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.

The screenshot shows the Nortel Configuration page for Conference Bridge configuration. The page has a header with the Nortel logo, navigation links (Home, Help, Logout), and a search bar. The main content area is divided into tabs: USERS, DEVICES, FEATURES, SYSTEM, and DIAGNOSTICS. The 'CONFIGURATION' tab is selected, and the 'Participants' sub-tab is active. The configuration form includes the following fields and options:

- Enabled:** A checkbox that is checked.
- Name:** A text field containing 'conference06'.
- Extension:** A text field containing '20406'.
- Description:** A text area containing 'dat06's conference bridge'.
- Conference owner:** A dropdown menu showing 'dat06 nguyen (20206)' with 'Change owner...' and 'Unassign' buttons. Below this, a note states: 'The user that should have permission to administer and control this conference. Unassigned conferences may only be controlled by administrators.'
- Participant PIN:** A text field containing '1234'. Below this, a note states: 'DTMF digits for participant PIN. Can be empty.'
- Maximum legs:** A text field containing '0'. To the right, it says '(Default: 0)'. Below this, a note states: 'The maximum number of call legs to be allowed by this bridge. 0 means unlimited.'

At the bottom of the form are three buttons: 'OK', 'Apply', and 'Cancel'. The footer of the page reads: 'Software Communication System (4.0.1-015823 2009-06-19T00:09:44)'.

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**.

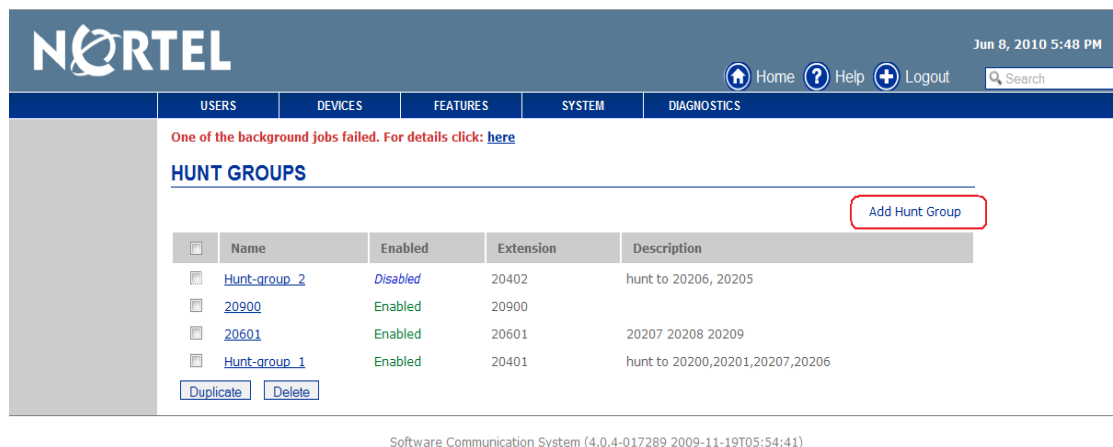


Figure 15: Hunt Group adding

- 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.

NORTEL 25-Sep-2009 10:12 AM

Home ? Help + Logout Search

USERS DEVICES FEATURES SYSTEM DIAGNOSTICS

HUNT GROUP

Click the Add User link to add users to this hunt group. You can search for users that match specified criteria. Change the calling sequence by moving users up and down. Specify expiration time (in seconds) to determine for how long the user's phone rings before a call is transferred to the next user on the list.

Enabled ☒

Name

Extension

Description

Call Sequence

Add User

<input type="checkbox"/>	Sequence	User	Aliases	Expiration [s]
<input type="checkbox"/>	Initially call	20206		<input type="text" value="10"/>
<input type="checkbox"/>	If no response	20200		<input type="text" value="30"/>

Move Up Move Down Delete

Use Voicemail ☒

If checked the call is sent to voicemail of the last user if no user picks up the phone. The last user has to have voicemail enabled. If unchecked the alternative fallback destination can be specified below.

Allow Call Forwarding ☒

If checked calls directed to hunt group follow user call forwarding rules. Clear this checkbox to force huntgroup to ignore call forwarding configured by hunt group members.

OK Apply Cancel

Software Communication System (4.0.1-015823 2009-06-19T00:09:44)

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.

NORTEL Jun 8, 2010 6:04 PM

Home ? Help + Logout Search

USERS **DEVICES** **FEATURES** **SYSTEM** **DIAGNOSTICS**

One of the background jobs failed. For details click: [here](#)

DIAL PLANS

Add New Rule... Reset

<input type="checkbox"/>	Name	Enabled	Type	Description	Schedule
<input type="checkbox"/>	Emergency	Disabled	Emergency	Emergency dialing plan	Always
<input type="checkbox"/>	International	Disabled	Long Distance	International dialing	Always
<input type="checkbox"/>	Local	Disabled	Long Distance	Local dialing	Always
<input type="checkbox"/>	Long Distance	Disabled	Long Distance	Long distance dialing plan	Always
<input type="checkbox"/>	Restricted	Disabled	Long Distance	Restricted dialing	Always
<input type="checkbox"/>	Toll free	Disabled	Long Distance	Toll free dialing	Always
<input type="checkbox"/>	AutoAttendant	Enabled	Attendant	Default autoattendant dialing plan	Always
<input type="checkbox"/>	Voicemail	Enabled	Voicemail	Default voicemail dialing plan	Always
<input type="checkbox"/>	ToSPS60	Enabled	Site To Site	Route to Rls6.0 system A, NMC, ICP	Always

Duplicate Delete Move Up Move Down

Quick Links

[Gateways](#)
[Permissions](#)

Dial plans consist of various types of dial rules. You can configure dial plans by adding, removing, editing, or reordering rules. It is possible to have more than one rule of each kind.

Rule order matters: Make sure that more specific rules precede more general rules. For example, move Long Distance rules for specified area codes above the default Long Distance rule.

Software Communication System (4.0.4-017289 2009-11-19T05:54:41)

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.

NORTEL 24-Sep-2009 2:50 PM

Home ? Help + Logout Search

USERS **DEVICES** **FEATURES** **SYSTEM** **DIAGNOSTICS**

DIAL PLANS

Add New Rule... Reset

<input type="checkbox"/>	Name	Enabled	Type	Description	Schedule
<input type="checkbox"/>	Emergency	Disabled	Emergency	Emergency dialing plan	Always
<input type="checkbox"/>	International	Disabled	Long Distance	International dialing	Always
<input type="checkbox"/>	Local	Disabled	Long Distance	Local dialing	Always
<input type="checkbox"/>	Long Distance	Disabled	Long Distance	Long distance dialing plan	Always
<input type="checkbox"/>	Restricted	Disabled	Long Distance	Restricted dialing	Always
<input type="checkbox"/>	Toll free	Disabled	Long Distance	Toll free dialing	Always
<input type="checkbox"/>	AutoAttendant	Enabled	Attendant	Default autoattendant dialing plan	Always
<input type="checkbox"/>	Voicemail	Enabled	Voicemail	Default voicemail dialing plan	Always
<input type="checkbox"/>	ToSPS60	Enabled	Site To Site	Route to Rls6.0 system A, NMC, ICP	Always

Duplicate Delete Move Up Move Down

Quick Links

[Gateways](#)
[Permissions](#)


Dial plans consist of various types of dial rules. You can configure dial plans by adding, removing, editing, or reordering rules. It is possible to have more than one rule of each kind.

Rule order matters: Make sure that more specific rules precede more general rules. For example, move Long Distance rules for specified area codes above the default Long Distance rule.

Software Communication System (4.0.1-015823 2009-06-19T00:09:44)

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.


Jun 21, 2010 5:43 PM

[Home](#)
[Help](#)
[Logout](#)

[USERS](#)
[DEVICES](#)
[FEATURES](#)
[SYSTEM](#)
[DIAGNOSTICS](#)

One of the background jobs failed. For details click: [here](#)

DIAL RULE

Enabled ☒

Name

Description

Internal station extension length
Number of digits used for internal extensions in your installation. Add another voicemail rule if you have phones with different length extensions.

Voicemail extension
Extension to dial to call voicemail. Leave empty to disable voicemail.

Voicemail inbox prefix
Dial voicemail prefix followed by an internal extension to directly call voicemail for that extension. Leave empty to disable.

Voicemail type
Only one voicemail type can be configured on a system.

Voicemail host
IP address or name of the voicemail server. Leave empty if the voicemail server runs on the same computer as the call server.

Schedule

Microsoft Exchange

Microsoft Exchange 2 can be used as an alternative to the internal Voicemail Server. Select "Exchange Voicemail Server" and enter its name or IP address into the field provided.

For every user or group of users the desired voicemail server needs to be selected. Select the "Permissions" tab in the "Users" menu to do this.

Choose only internal voicemail server or Exchange 2007 as they do not work when configured on the system.

Note: Due to a Microsoft problem Message Waiting Indication (MWI) does not work with Exchange 2007.

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.

NORTEL Jun 21, 2010 5:50 PM

Home ? Help + Logout Search

USERS DEVICES FEATURES SYSTEM DIAGNOSTICS

One of the background jobs failed. For details click: [here](#)

DIAL RULE

Enabled ☒

Name

Description

Extension

Attendant aliases

Attendant will be reachable through its extension and any of the above aliases. When entering multiple aliases, separate them with spaces.

Default attendant

Default attendant is used if Working time or Holiday attendant are not specified or if current time is neither holiday, nor working time.

Working time attendant

Select attendant to be used during working hours. Working hours can be specified once attendant is selected.

Holiday attendant

Select attendant to be used during holidays. If attendant is selected you can add and remove holiday dates.

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.

The screenshot shows the Nortel SCS Server web interface. The top navigation bar includes the Nortel logo, a date/time stamp (Jun 8, 2010 6:13 PM), and links for Home, Help, Logout, and a search bar. Below the navigation bar is a menu with tabs: USERS, DEVICES, FEATURES, SYSTEM, and DIAGNOSTICS. The FEATURES tab is selected, and a dropdown menu is open showing options: Call Center (highlighted with a red circle), Agent Status, Conferencing, Auto Attendants, Intercom, Paging Groups, Hunt Groups, Call Park, Music on Hold, and Phonebooks. The main content area displays the ACD SERVERS configuration. It includes a table with columns for Server Location and Configuration Port. The table contains one entry: Server Location is 'scs500r3.scsinterop.com' and Configuration Port is '8110'. Below the table is an 'Activate' button. A message at the top of the main content area states: 'One of the background jobs failed. For details click: here'. On the right side, there is a 'Quick Links' section with links for Presence Server, Servers, and Job Status. Below the links is a paragraph of text explaining the ACD configuration process.

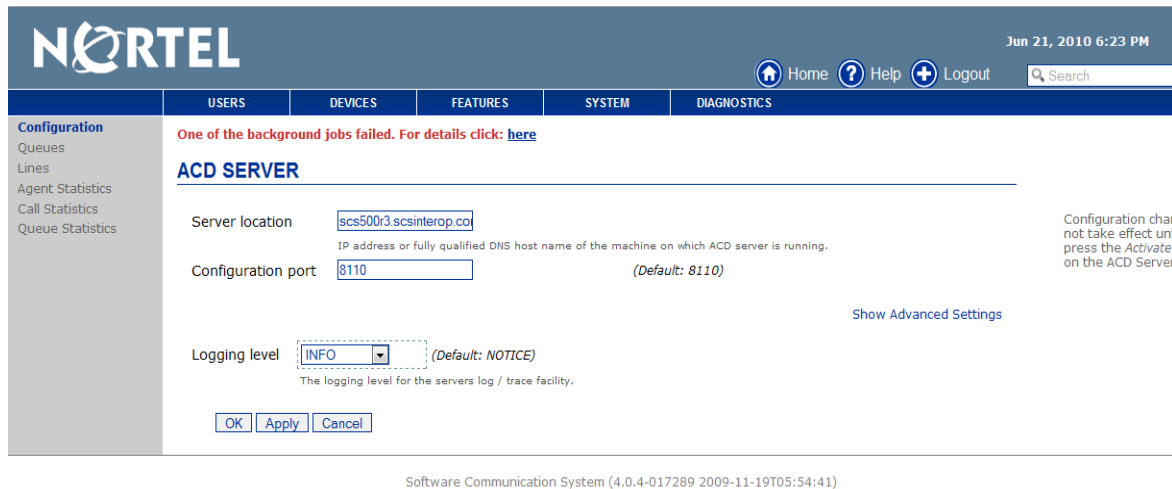
Figure 21: ACD Server list

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

This screenshot is similar to Figure 21, showing the ACD SERVERS configuration page. The 'Call Center' option is still selected in the FEATURES dropdown. The table shows the same server entry: 'scs500r3.scsinterop.com' with configuration port '8110'. In this image, the server name 'scs500r3.scsinterop.com' is highlighted with a red circle. The 'Activate' button remains below the table. The message at the top and the 'Quick Links' section on the right are also present.

Figure 22: ACD Server Name

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



NORTEL Jun 21, 2010 6:23 PM

Home ? Help + Logout Search

USERS DEVICES FEATURES SYSTEM DIAGNOSTICS

Configuration
Queues
Lines
Agent Statistics
Call Statistics
Queue Statistics

One of the background jobs failed. For details click: [here](#)

ACD SERVER

Server location
IP address or fully qualified DNS host name of the machine on which ACD server is running.

Configuration port (Default: 8110)

Logging level (Default: NOTICE)
The logging level for the servers log / trace facility.

[Show Advanced Settings](#)

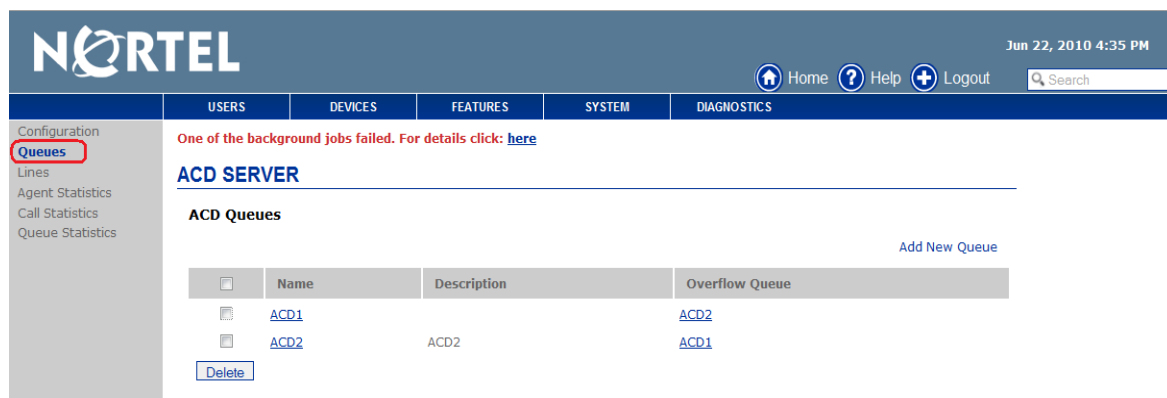
OK Apply Cancel

Configuration chan not take effect unt press the Activate I on the ACD Server:

Software Communication System (4.0.4-017289 2009-11-19T05:54:41)

Figure 23: ACD Server Configuration

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



NORTEL Jun 22, 2010 4:35 PM

Home ? Help + Logout Search

USERS DEVICES FEATURES SYSTEM DIAGNOSTICS

Configuration
Queues
Lines
Agent Statistics
Call Statistics
Queue Statistics

One of the background jobs failed. For details click: [here](#)

ACD SERVER

ACD Queues

[Add New Queue](#)

<input type="checkbox"/>	Name	Description	Overflow Queue
<input type="checkbox"/>	ACD1		ACD2
<input type="checkbox"/>	ACD2	ACD2	ACD1

[Delete](#)

Software Communication System (4.0.4-017289 2009-11-19T05:54:41)

Figure 24: ACD Server Queue overview

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

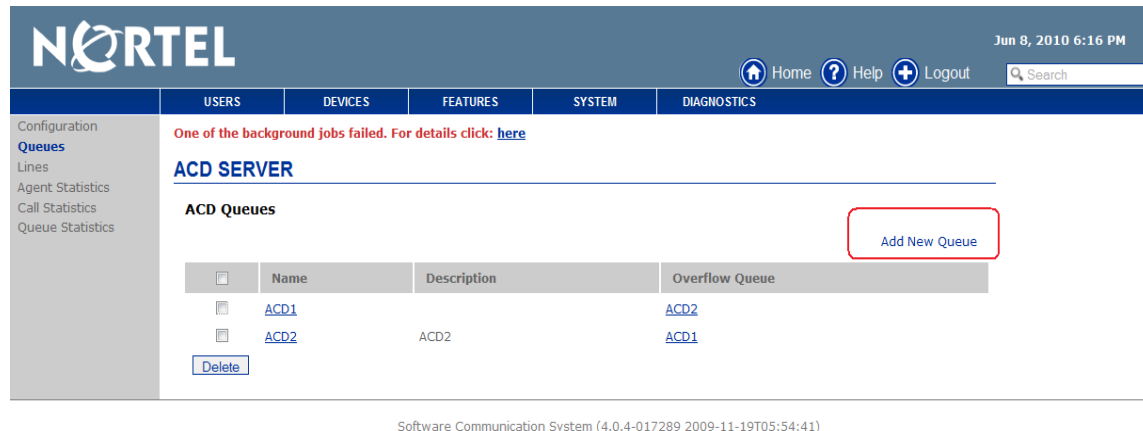


Figure 25: ACD Server Add New Queue

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.

24-Sep-2009 3:05 PM

Home
Help
Logout
Search

Configuration
Agents
Agent Statistics
Call Statistics

USERS
DEVICES
FEATURES
SYSTEM
DIAGNOSTICS

ACD QUEUE

Name

Description

Hide Advanced Settings

Overflow Type

select...

Define an overflow type

Overflow destination

select...

Overflow destination value depending on the selected overflow type.

Overflow entry

Call can be transferred to an internal extension or a SIP URI upon an overflow condition if no overflow destination is defined for this queue.

Call routing scheme

Circular

(Default: Circular)

The ACD call routing scheme that will be employed on this queue.

Maximum ring delay

15

(Default: 15)

The maximum time in seconds that the queue will allow an agent station to ring before a ring-no-answer condition is declared and the call is rerouted to a different agent.

Maximum queue length

10

(Default: 10)

The maximum number of calls that are allowed to wait in this queue. If a call arrives at this queue and the resulting call count exceeds this number, then an overflow condition for this queue will be triggered. A value of -1 disables this limit check.

Maximum wait time

60

(Default: 60)

The maximum time in seconds that a call can reside in a queue. When a waiting call exceeds this time limit, an overflow condition for this queue will be triggered. A value of zero disables timeouts.

FIFO overflow
☒

(Default: checked)

If set, then upon an overflow condition, a FIFO scheme will be employed in order to determine which call will be moved to the configured overflow-queue. If not set, then a LIFO scheme will be employed.

Answer mode

Immediate

(Default: Immediate)

If set to Immediate, the call will be answered immediately upon arriving at this queue and the configured welcome-audio file will be played to the caller. Once the audio has completed, the queue will then attempt to route the call. If set to Deferred, the queue will first attempt to route the call. If it is unable to immediately route the call, it will then be answered. If set to Never, the call will not be answered while on this queue other than when actually connecting to an agent.

Barge in
☐

(Default: unchecked)

If set, the welcome audio will be terminated early, should an agent become available while it is being played.

Welcome audio

XPStart.wav
Listen
Delete

Browse...

The welcome audio played to callers. A If no file is specified, then silence will be played. Several files can be uploaded and selected.

Queue audio

hpny.wav
Listen
Delete

Browse...

The queue audio played repeatedly to the caller until the queue either routes the call to an agent or to another queue. Several files can be uploaded and selected.

Audio interval

15

(Default: 15)

The interval, in seconds, to wait before repeating play of the specified Queue audio.

Call termination audio

select...

Browse...

The message played to the caller when it has been determined that the call must be terminated. Once the audio has completed, the call will be dropped. If no audio is specified, then a busy tone will be played prior to terminating the call. The duration of the busy tone is specified by the termination-tone-duration attribute.

Termination tone duration

2

(Default: 2)

The duration in seconds that the termination tone (busy tone) is to be played if no call-termination-audio is specified and the call is to be dropped by the queue. A value of zero indicates that no tone is to be played prior to dropping the call.

Agent wrap-up time

15

(Default: 15)

The period of time, in seconds, that has to pass before the ACD transfers a new call to an agent after a previous call has been completed. If set to 0, it will be disabled.

Agent Non-Responsive time

30

(Default: 30)

The period of time, in seconds, that has to pass before the ACD transfers a new call to an agent after a previous call was not answered.

Maximum Bounce Count

3

(Default: 3)

The number of rejected or non-answered calls an agent may have before being "bounced" (automatically signed out). If set to 0, it will be disabled.

OK
Apply
Cancel

Software Communication System (4.0.1-015823 2009-06-19T00:09:44)

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.

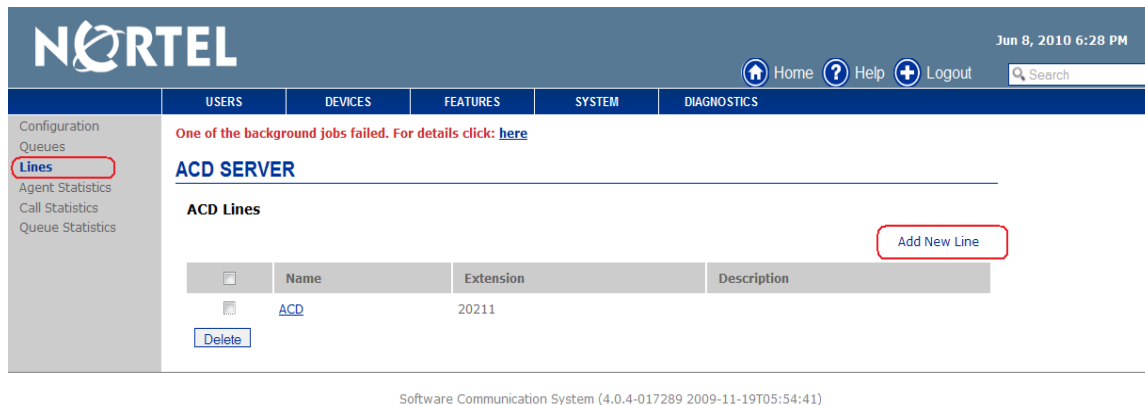


Figure 27: ACD Line overview

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.

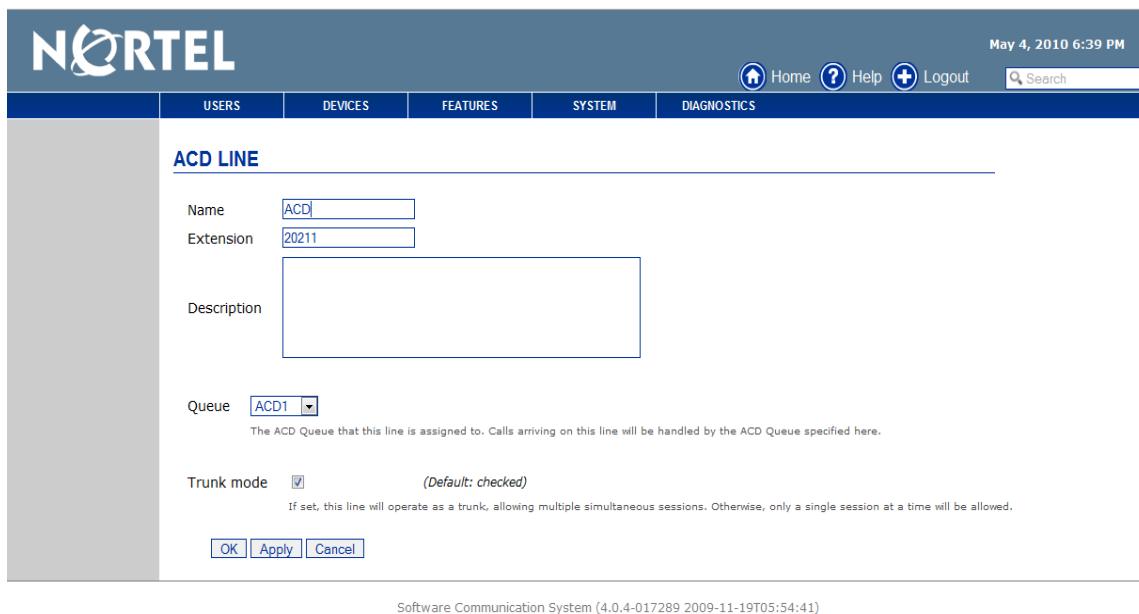


Figure 28: ACD Line configuration

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.

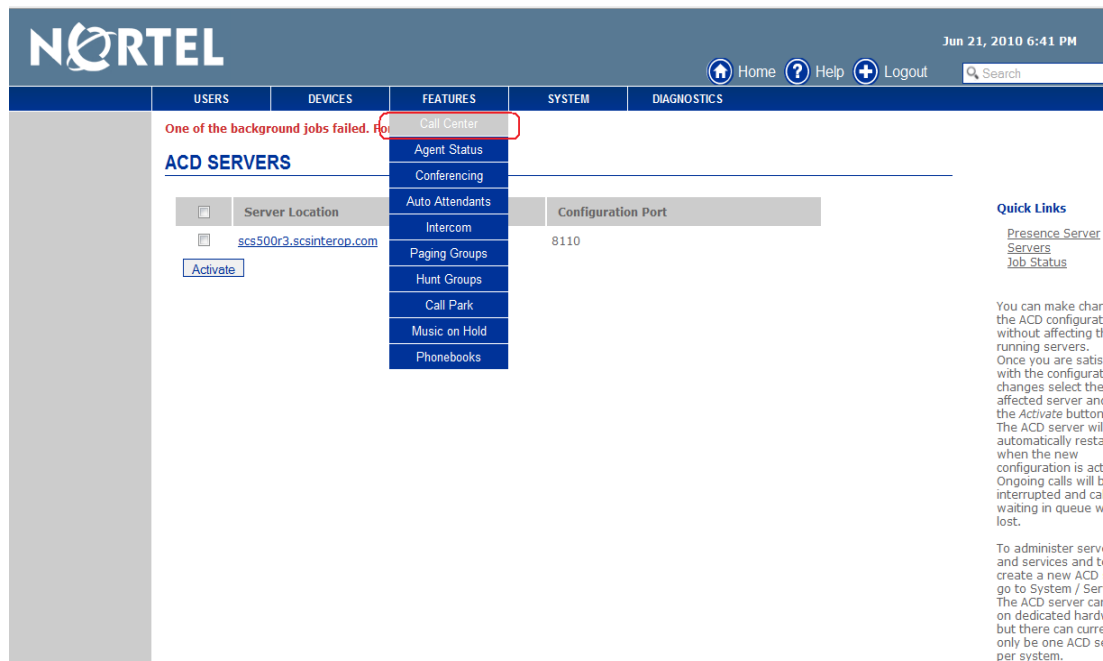


Figure 29: ACD Queue Agent overview

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

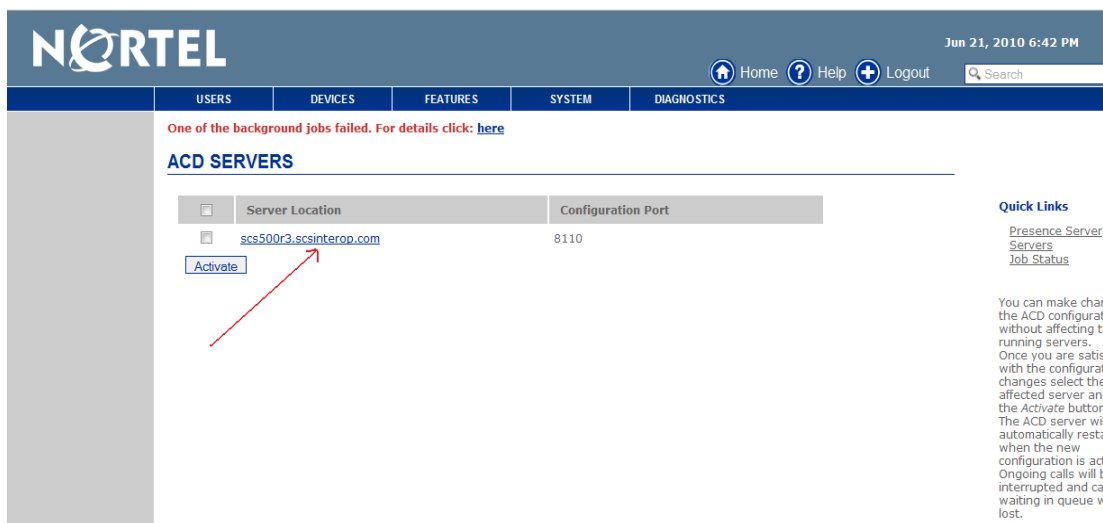


Figure 30: ACD Queue Agent overview

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.

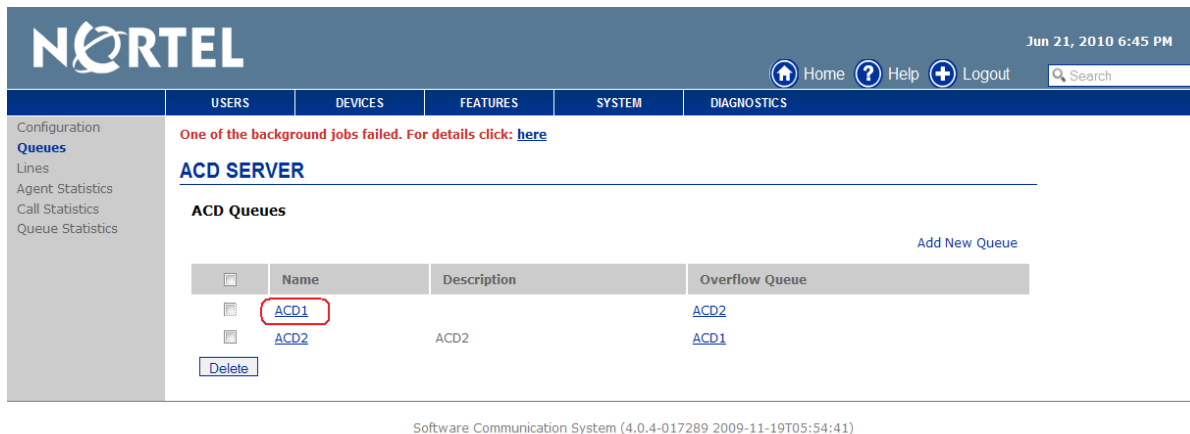


Figure 31: ACD Queue Server

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.

NORTEL Jun 21, 2010 6:50 PM

Home ? Help + Logout Search

USERS DEVICES FEATURES SYSTEM DIAGNOSTICS

One of the background jobs failed. For details click: [here](#)

ADD ACD AGENTS

Search for the user to be added to the ACD Queue.

User

Enter user ID, name, alias or description and press Search button. Leave empty and press Search to display all users.

Search Cancel

<input type="checkbox"/>	User ID	Last Name	First Name	Aliases
<input type="checkbox"/>				

Select

Software Communication System (4.0.4-017289 2009-11-19T05:54:41)

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.

NORTEL Jun 22, 2010 5:22 PM

Home ? Help + Logout Search

USERS DEVICES FEATURES SYSTEM DIAGNOSTICS

One of the background jobs failed. For details click: [here](#)

ADD ACD AGENTS

21 users found.

Search for the user to be added to the ACD Queue.

User

Enter user ID, name, alias or description and press Search button. Leave empty and press Search to display all users.

Search Cancel

<input type="checkbox"/>	User ID	Last Name	First Name	Aliases
<input type="checkbox"/>	20210	vo	nhan	nhan
<input type="checkbox"/>	20206	nguyen	dat06	
<input type="checkbox"/>	20207	nguyen	dat07	
<input type="checkbox"/>	20209	nguyen	dat09	
<input type="checkbox"/>	20204	nguyen	dat04	
<input type="checkbox"/>	20208	nguyen	dat08	
<input type="checkbox"/>	20205	nguyen	dat05	
<input type="checkbox"/>	20214	Nhan	Vo	
<input type="checkbox"/>	20213	Nhan	Vo	
<input checked="" type="checkbox"/>	20200	DP	LAB_00	
<input checked="" type="checkbox"/>	20201	DP	LAB_01	
<input checked="" type="checkbox"/>	20202	DP	LAB_02	
<input checked="" type="checkbox"/>	20203	DP	LAB_03	
<input type="checkbox"/>	20212	DP	LAB_12	
<input type="checkbox"/>	20215	DP	LAB_15	
<input type="checkbox"/>	20216	DP	LAB_16	

Figure 34: ACD Agents searching result page

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

NORTEL Jun 21, 2010 6:55 PM

Home ? Help + Logout Search

Configuration
Agents
Agent Statistics
Call Statistics

USERS DEVICES FEATURES SYSTEM DIAGNOSTICS

One of the background jobs failed. For details click: [here](#)

ACD QUEUE

ACD Agents [Add New Agent](#)

<input type="checkbox"/>	User ID	Aliases	Description
<input type="checkbox"/>	20205	nhan	
<input type="checkbox"/>	20210		
<input type="checkbox"/>	20212		
<input type="checkbox"/>	20200		
<input type="checkbox"/>	20215		
<input type="checkbox"/>	20201		
<input type="checkbox"/>	20202		
<input type="checkbox"/>	20203		

[Delete](#) [Move Up](#) [Move Down](#)

Software Communication System (4.0.4-017289 2009-11-19T05:54:41)

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.

NORTEL Jun 21, 2010 7:02 PM

Home ? Help + Logout Search

USERS DEVICES FEATURES SYSTEM DIAGNOSTICS

One of the background jobs failed. For details click: [here](#)

CALL PARK

[Defaults](#) [Add Call Park Extension](#)

<input type="checkbox"/>	Name	Enable	Extension	Background Music
<input type="checkbox"/>	park1	Enable	20301	default.wav
<input type="checkbox"/>	park2	Enable	20302	default.wav

[Delete](#)

Software Communication System (4.0.4-017289 2009-11-19T05:54:41)

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.

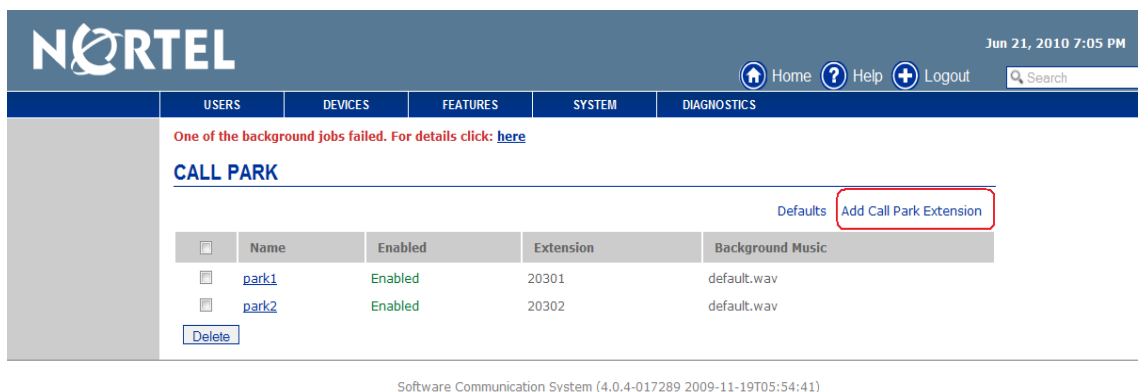


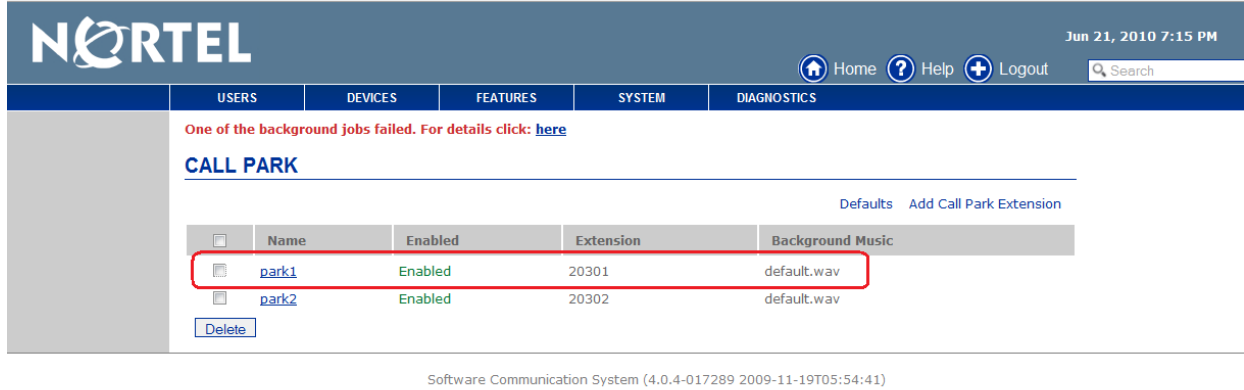
Figure 37: Call Park adding

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**.



The screenshot shows the Nortel web interface for Call Park configuration. The top navigation bar includes the Nortel logo, a date/time stamp (Jun 21, 2010 7:15 PM), and links for Home, Help, and Logout. Below this is a menu with tabs for USERS, DEVICES, FEATURES, SYSTEM, and DIAGNOSTICS. A message states: "One of the background jobs failed. For details click: [here](#)". The main section is titled "CALL PARK" and includes links for "Defaults" and "Add Call Park Extension". A table lists the configured extensions:

<input type="checkbox"/>	Name	Enabled	Extension	Background Music
<input type="checkbox"/>	park1	Enabled	20301	default.wav
<input type="checkbox"/>	park2	Enabled	20302	default.wav

A "Delete" button is located below the table. The footer text reads: "Software Communication System (4.0.4-017289 2009-11-19T05:54:41)".

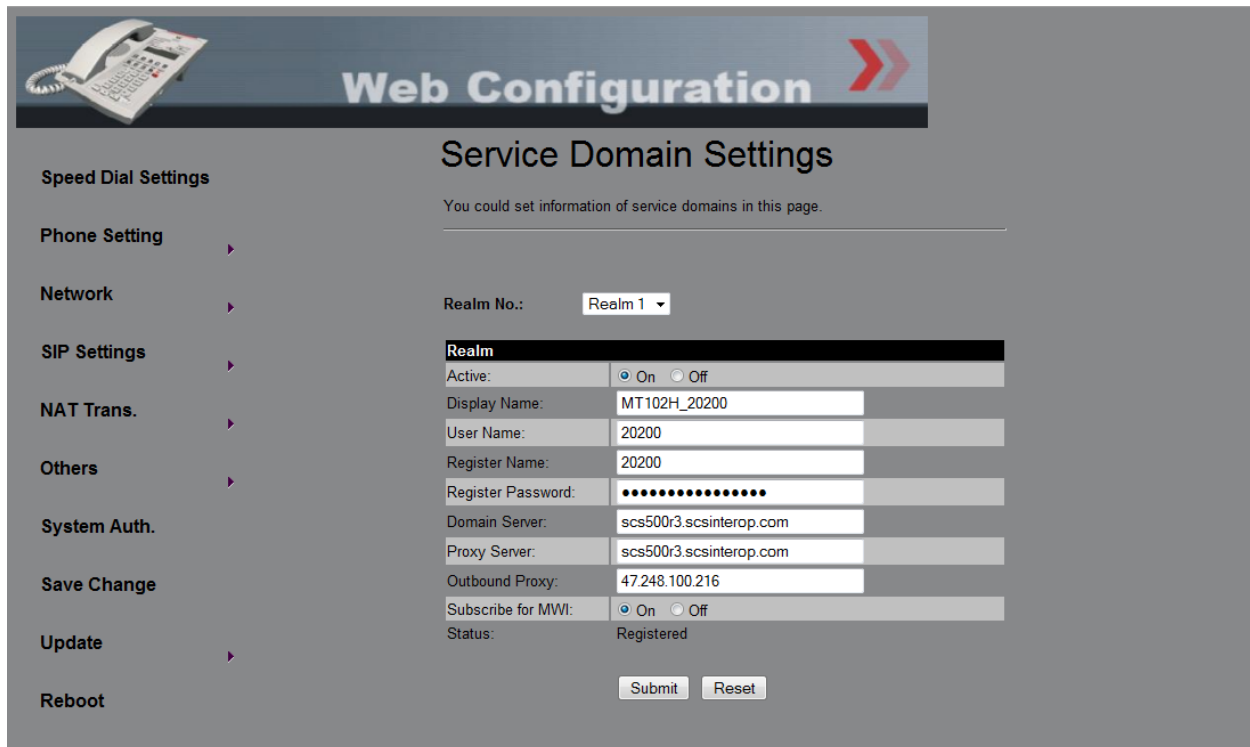
Figure 39: Call Park Extension list

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

The following sections below explain 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**.



The screenshot displays the 'Web Configuration' interface for a G-Tek MT-102H SIP telephone. The left sidebar contains navigation links: Speed Dial Settings, Phone Setting, Network, SIP Settings, NAT Trans., Others, System Auth., Save Change, Update, and Reboot. The main content area is titled 'Service Domain Settings' and includes a sub-header 'You could set information of service domains in this page.' Below this, the 'Realm No.' is set to 'Realm 1'. A table lists the configuration details for the selected realm.

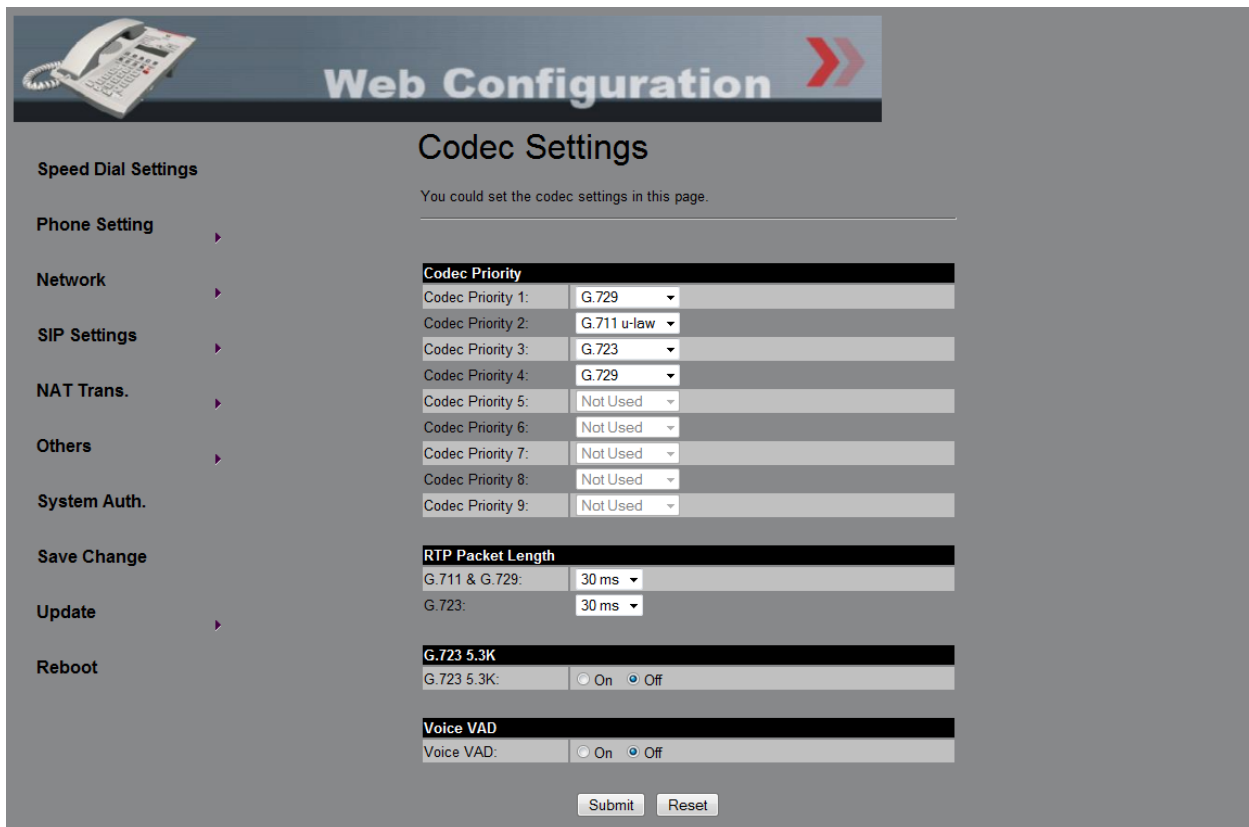
Realm	
Active:	<input checked="" type="radio"/> On <input type="radio"/> Off
Display Name:	MT102H_20200
User Name:	20200
Register Name:	20200
Register Password:	••••••••••
Domain Server:	scs500r3.scsinterop.com
Proxy Server:	scs500r3.scsinterop.com
Outbound Proxy:	47.248.100.216
Subscribe for MWI:	<input checked="" type="radio"/> On <input type="radio"/> Off
Status:	Registered

At the bottom of the configuration area are 'Submit' and 'Reset' buttons.

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

5.2. Codec settings

Configure Codec Settings as displayed in the below screenshot.



The screenshot displays the 'Web Configuration' interface for the G-Tek MT-102H. The left sidebar contains navigation links: Speed Dial Settings, Phone Setting, Network, SIP Settings, NAT Trans., Others, System Auth., Save Change, Update, and Reboot. The main content area is titled 'Codec Settings' and includes a sub-header 'You could set the codec settings in this page.' Below this, there are three sections: 'Codec Priority' with a table of 9 priorities, 'RTP Packet Length' with two rows, and 'G.723 5.3K' and 'Voice VAD' each with a single row. At the bottom are 'Submit' and 'Reset' buttons.

Codec Priority	
Codec Priority 1:	G.729
Codec Priority 2:	G.711 u-law
Codec Priority 3:	G.723
Codec Priority 4:	G.729
Codec Priority 5:	Not Used
Codec Priority 6:	Not Used
Codec Priority 7:	Not Used
Codec Priority 8:	Not Used
Codec Priority 9:	Not Used

RTP Packet Length	
G.711 & G.729:	30 ms
G.723:	30 ms

G.723 5.3K	
G.723 5.3K:	<input type="radio"/> On <input checked="" type="radio"/> Off

Voice VAD	
Voice VAD:	<input type="radio"/> On <input checked="" type="radio"/> Off

Submit Reset

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**.

The screenshot displays the 'Web Configuration' interface for the G-Tek MT-102H. The 'Speed Dial Settings' section is active, showing a 'Speed Dial Phone List'. The list contains 9 entries, each with a 'MKey', 'Name', 'Number or URL', and a 'Select' checkbox. The 5th entry is highlighted, showing 'MKey: 5', 'Name: 20206', and 'Number or URL: 20206'. Below the list are buttons for 'Delete Selected', 'Delete All', and 'Reset'. An 'Add New Phone' section is also visible, with fields for 'Position' (set to 5), 'Name' (set to 20206), and 'Number or URL' (set to 20206). Buttons for 'Add SpeedDial' and 'Reset' are at the bottom.

MKey	Name	Number or URL	Select
1	20201	20201	<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5	20206	20206	<input checked="" type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Buttons: Delete Selected, Delete All, Reset

Add New Phone

Position: 5 (1~9)

Name: 20206

Number or URL: 20206

Buttons: Add SpeedDial, Reset

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

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