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SIP Provisioning Client User Guide



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Welcome

The Session Initiation Protocol (SIP) Provisioning Client is a web-based Graphical User Interface (GUI) for service providers to complete administrative tasks, for example, provisioning administrators, domains, IP Client Managers (IPCMs) and IPCM clusters.

Overview

Service providers access the SIP Provisioning Client with any web browser. Being a web-based system, the SIP Provisioning Client provides the service providers with an highly efficient access from anywhere capability.



Note: The provisioning abilities visible to an administrator in the SIP Provisioning Client GUI depends on the level of authority assigned to their account.

Audience

This guide is intended for service providers and administrators (with varying levels of privileges) assigned to manage customer domains.

What is not covered

This document does not cover

- Configuration
- System Management Console

Text and graphic conventions

This guide uses the following text and graphic conventions:

bold text

Indicates a menu option, link, or command key you need to click. Example: Click **Ok**.

italic text

Indicates

- a domain name or subdomain name
- a document title
- a variable name

separator (>)

Shows menu paths.

Example: **Gateway > Add Gateway**



Indicates a top level menu item in the SIP Provisioning Client menu hierarchy.



Indicates a sub menu item in the SIP Provisioning Client menu hierarchy.



Indicates a form in the SIP Provisioning Client menu hierarchy.

Acronyms

This guide uses the following acronyms

| | |
|--------|-------------------------------|
| CAS | Channel Associated Signalling |
| CPL | Call Processing Language |
| COS | Class of Service |
| CS2000 | Communication Server 2000 |
| DSL | Digital Subscriber Line |
| GMT | Greenwich Mean Time |

| | |
|---------|-------------------------------------|
| DHCP | Dynamic Host Configuration Protocol |
| DIGMAN | Digit Manipulation |
| GUI | Graphical User Interface |
| IP | Internet Protocol |
| IPCM | Internet Protocol Client Manager |
| MAddr | Media Access Control Address |
| MCP | Multimedia Communications Portfolio |
| MSD | Most Significant Digit |
| MWI | Message Waiting Indicator |
| PDIL | Partial Dial |
| PRI | Primary Rate Interface |
| PSEIZ | Permanent Seizure |
| PSTN | Public Switched Telephone Network |
| SA | System Administrator |
| SIP | Session Initiation Protocol |
| SMDI | Simplified Message Desk Interface |
| TDM | Time Division Multiplex |
| UNIStim | Unified Network IP Stimulus |
| URI | Uniform Resource Identifier |
| URL | Uniform Resource Locator |
| VMS | Voice Mail Server |
| VPN | Virtual Private Network |

Related publications

- *SIP Personal Agent Getting User Guide (Part No. NN10039-112)*
- *SIP Multimedia Web Client User Guide (Part No. NN10040-112)*
- *SIP Multimedia PC Client User Guide (Part No. NN10041-112)*
- *i2004 Internet Telephone User Guide (Part No. NN10042-113)*

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Chapter 1

Getting Started

Topics in this section:

- [“SIP Provisioning Client overview” on page 1](#)
- [“SIP Provisioning Client features” on page 2](#)
- [“System requirements” on page 2](#)
- [“Using the online help” on page 5](#)
- [“Setting up time zones” on page 5](#)
- [“Changing the admin password” on page 9](#)
- [“Exiting the SIP Provisioning Client” on page 9](#)

SIP Provisioning Client overview

The SIP Provisioning Client is a web-based GUI that you use to perform the following administrative and provisioning tasks:

- define administration roles and privileges
- manage (list add, modify, or delete) users and devices
- define multiple service packages
- assign services to service packages
- assign service packages to domains
- assign translation information specific to a domain or subdomain
- provision voice mail servers
- provision 911 emergency notification

SIP Provisioning Client features

The SIP Provisioning Client gives service providers the capability to create and set security, access, and permissions for the following components and roles of the user framework:

- administration
- domains and subdomains (users, devices, telephony routes)
- gateways
- IPCM clusters
- voice mail servers
- services
- system



Note: The provisioning capabilities visible to an administrator in the SIP Provisioning Client menu depend on the level of authority (rights) assigned to their account.

System requirements

The following section describes the minimum and recommended hardware and software requirements for the SIP Provisioning Client.

Minimum hardware and software requirements

- 200MHz Pentium class or equivalent processor
- No additional free RAM is required. (The minimum memory requirements of your browser must be met. See your browser documentation for more information.)
- No additional harddrive space is required. (Local caching of web pages is performed by your web browser. See your browser documentation for information on the minimum disk space requirements.)
- Mouse (optional)
- 640x480 @8bpp (256 colors) VGA graphics card

- Microsoft* Windows* 98 Second Edition (SE), Microsoft Windows Millennium Edition (Me), Microsoft Windows 2000, Microsoft XP, or Microsoft Windows NT* 4.0 with Service Pack 5 (SP5)
- 28.8Kbps modem
- Netscape* Communicator 4.77 or Internet Explorer 5.0
- Cookies and javascript enabled

Recommended hardware and software requirements

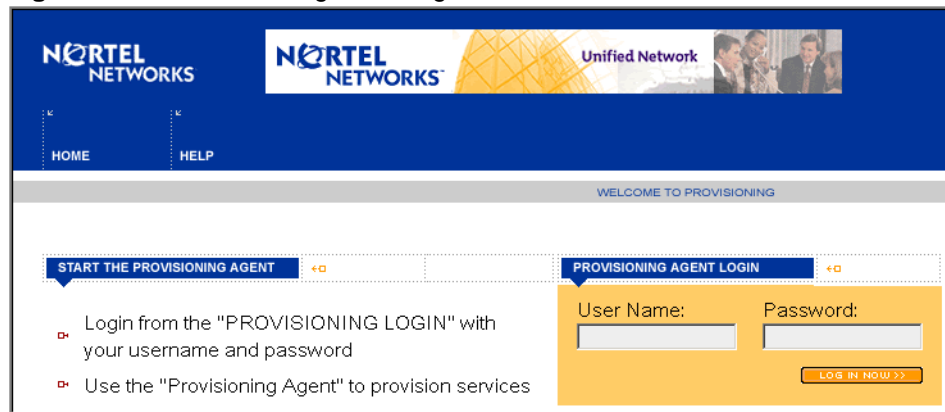
- 300MHz (or higher) Pentium class
- No additional free RAM is required. (The minimum memory requirements of your browser must be met. See your browser documentation for more information.)
- No additional harddrive space is required. (Local caching of web pages is performed by your web browser. See your browser documentation for more information on the minimum disk space requirements.)
- Mouse (required)
- 800x600 or higher @ 16bpp (65,536 colors) VGA or better video graphics card
- Microsoft Windows 98 Second Edition (SE), Microsoft Windows 2000, Microsoft XP, or Microsoft Windows NT 4.0 with Service Pack 5 (SP5)
- 56Kbps modem or other high speed connection (For example, cable modem, Digital Subscriber Line (DSL) modem, 10baseT ethernet, an so forth)
- Netscape Communicator 4.77 or greater, or Internet Explorer 5.0 or greater
- Cookies and javascript enabled

Starting the SIP Provisioning Client

To start the SIP Provisioning Client from your PC

- 1 Start any supported web browser.
- 2 Enter the Uniform Resource Locator (URL) for the SIP Provisioning Client. The SIP Provisioning Client Login window appears ([Figure 1](#)).

Figure 1 SIP Provisioning Client login window.



3 Enter your username and password in the respective fields.

4 Click **LOG IN NOW** to log in. Upon a successful login, the SIP Provisioning Client Welcome window appears.

Figure 2 SIP Provisioning Client welcome window



If the SIP Provisioning Client GUI is inactive for a period of time, your session will expire. The following message appears:

Figure 3 Session inactivity message

To restart the SIP Provisioning Client, click on the link to login again. Contact your service provider to request a change to the session expiration time limit.

Using the online help

There are several ways that you can access SIP Provisioning Client help:

- online help - from the SIP Provisioning Client screen, click the **Help** button to view a task-based help system. The online help provides:
 - help pages containing forward and backward navigation icons
 - procedures that help you use the SIP Provisioning Client
 - a Table of Contents with hypertext links
 - an Index with hypertext links
- question mark help. Click the question mark icon to display a description and possible values for parameter fields on a form.

Setting up time zones

The SIP Provisioning Client allows you to define and manage as many time zones as you require to meet your needs, and to create your own time zones for easy reference. Time zones define the time zone from which a subscriber normally accesses the system (that is, the subscribers's local time zone). Time zones are used in the SIP Personal Agent Call Manager function to set a screening rule that applies during a certain time. This time is dependent on the time zone selected.

The SIP Provisioning Client is initially set up with the standard 24 time zones: plus or minus 12 hours from the Greenwich Mean Time (GMT) format. However, there are more than 300+ time zones. What defines a time zone is not just the offset from the GMT. It is also whether or not the time zone uses Daylight Saving Time, when Daylight Saving Time occurs, and what the offset is.

Setting up times zones in the United States

As an example, the time zones you would set up across the United States from East to West are listed in the following table:

| Zone | Standard Time | Daylight Saving Time |
|----------|---------------|----------------------|
| Atlantic | AST | ADT |
| Eastern | EST | EDT |
| Central | CST | CDT |
| Mountain | MST | MDT |
| Pacific | PST | PDT |
| Alaska | AKST | AKDT |
| Hawaii | HST | Not Observed |

Selecting a time zone

To select a time zone

- 1 Click the **System** menu option.
- 2 Click the **Timezones** form. The Timezone details window appears.

Figure 4 Time Zone details window

Timezone details

Current timezone options

| Name | Timezone | Save | Delete |
|------------------------|---------------------------|------|--------|
| Pacific Standard Time | GMT-8.0 PST | Save | Delete |
| Mountain Standard Tim | GMT-7.0 MST | Save | Delete |
| Central Standard Time | GMT-6.0 CST | Save | Delete |
| Eastern Standard Time | GMT-5.0 EST | Save | Delete |
| GMT-11:00 | GMT-11.0 MIT | Save | Delete |
| Hawaii Standard Time | GMT-10.0 HST | Save | Delete |
| Alaska Standard Time | GMT-9.0 AST | Save | Delete |
| GMT-04:00 | GMT-4.0 PRT | Save | Delete |
| Newfoundland Standan | GMT-3.5 CNT | Save | Delete |
| GMT-03:00 | GMT-3.0 BET | Save | Delete |
| Greenwich Mean Time | GMT GMT | Save | Delete |
| Central European Stanc | GMT+1.0 ECT | Save | Delete |
| GMT+02:00 | GMT+2.0 EET | Save | Delete |
| GMT+03:00 | GMT+3.0 EAT | Save | Delete |
| GMT+03:30 | GMT+3.5 MET | Save | Delete |
| GMT+04:00 | GMT+4.0 NET | Save | Delete |
| GMT+05:00 | GMT+5.0 PLT | Save | Delete |
| GMT+05:30 | GMT+5.5 IST | Save | Delete |
| GMT+06:00 | GMT+6.0 BST | Save | Delete |
| GMT+07:00 | GMT+7.0 VST | Save | Delete |
| China Standard Time | GMT+8.0 CTT | Save | Delete |
| Japan Standard Time | GMT+9.0 JST | Save | Delete |
| GMT+09:30 | GMT+9.5 ACT | Save | Delete |
| GMT+10:00 | GMT+10.0 AET | Save | Delete |
| GMT+11:00 | GMT+11.0 SST | Save | Delete |
| GMT+12:00 | GMT+12.0 NST | Save | Delete |
| Saskatchewan | GMT-6.0 America/Guatemala | Save | Delete |

- 3** Select a zone from the list of time zones.
- 4** From the drop down list, select a time zone.

- 5 Click **Save**.

Creating a new time zone

To create a new time zone

- 1 Click the **System** menu option.
- 2 Click the **Timezones** form. The Timezone details window appears.
- 3 In the Add new timezone box, enter a meaningful name for the time zone in the Name field. (This field is visible to subscribers.) Both the name and the time zone need to be unique.

Figure 5 Add new time zone



| Name | Timezone | |
|-----------|--------------------------|------|
| Manhattan | GMT-5.0 America/New_York | Save |

- 4 Click **Save** to save the new time zone. Click **View details** to see the list of time zones.

Modifying or Deleting a time zone

The SIP Provisioning Client is preloaded with a number of default times zones. You may wish to modify or delete a time zone to prevent the time zone from showing up in the drop down list that is available when creating or modifying a user.

To modify or delete a time zone

- 1 Click the **Services** menu option.
- 2 Click the **Timezones** form. The Timezone details window appears.
- 3 Select the time zone that you wish to delete. Click the **Delete** link next to any time zone that you do not need. In the confirmation dialog box, Click **Ok** to permanently delete the time zone or **Cancel** to abort the action.



Tip: You cannot modify or delete a time zone that is currently in use.

Changing the admin password

To change the admin password

- 1 Click the **System** menu option.
- 2 Click the icon for the **Change Password** form. The Change Admin Password window appears.
- 3 Enter your new password.
- 4 Confirm your new password.
- 5 Enter your current password.
- 6 Click **Save**. Your new password is activated upon your next login.

Exiting the SIP Provisioning Client

To close the SIP Provisioning Client window and exit the SIP Provisioning Client, click the **Logout** menu option.

Chapter 2

Administration Overview

Topics in this section:

- [“Introduction” on page 11](#)
- [“SIP Provisioning roles, rights, and tasks” on page 12](#)
- [“Understanding provisioning roles and rights” on page 15](#)

Introduction

In the SIP Provisioning Client, a provisioning role is defined as a collection of access rights and privileges that allow an administrator to perform various provisioning tasks. As a System Administrator, you can create any number of administrator roles, depending on your particular needs. For example, you can add a provisioning role to allow an administrator to perform multiple tasks, however, giving them the rights to only a subset of everything available.

After logging in to the SIP Provisioning Client, you will see a menu list on the left side of your screen. This list varies based on the privileges available to you. The types of tasks you can perform as an administrator is based on the scope of your defined role.



Note: The provisioning abilities visible to an administrator in the SIP Provisioning Client GUI depends on the rights assigned to their provisioning role.

SIP Provisioning roles, rights, and tasks

The following table provides an overview of the SIP Provisioning tasks.

Table 1 SIP Provisioning tasks overview





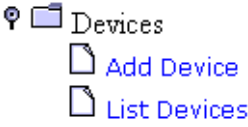
| If you have the right for | Use this Provisioning menu option... | To perform these tasks... |
|---------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Full domain access |  <ul style="list-style-type: none"> 📁 Domains <ul style="list-style-type: none"> 📄 List Foreign Domains 📄 Add Domain | Administer all current domains and future domains regardless of initial provisioning. This allows the administrator with this role to see all domains, versus other administrators that can see only a subset of domains. |
| Domain management |  <ul style="list-style-type: none"> 📁 Domains <ul style="list-style-type: none"> 📄 List Foreign Domains 📄 Add Domain | List, add, modify, or delete domains and subdomains. |
| IPCM cluster provisioning |  <ul style="list-style-type: none"> 📁 IPCM Clusters <ul style="list-style-type: none"> 📄 List IPCM Clusters 📄 Add IPCM Cluster 📄 List Physical IPCMs | Add, list, modify, or delete an IPCM or IPCM cluster. If you add an IPCM through the SIP Provisioning Client, the Management Console GUI will not know that the IPCM exists. It is recommended that you deploy an IPCM through the System Management Console. |
| Service Package Creation |  <ul style="list-style-type: none"> 📁 Services <ul style="list-style-type: none"> 📄 Define Service Parameters 📄 Assign Services | Add, list, modify, or delete services and service packages to a particular domain or all domains. If you have the administrator privileges to add a domain or multiple domains, then you can add or create service packages within that domain. |
| Device Management |  <ul style="list-style-type: none"> 📁 Devices <ul style="list-style-type: none"> 📄 Add Device 📄 List Devices | Add, list, modify, or delete i2004 devices. |

Table 1 SIP Provisioning tasks overview
























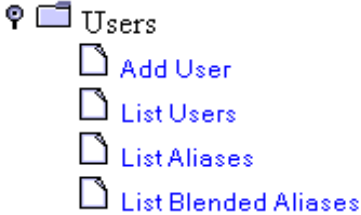
| If you have the right for | Use this Provisioning menu option... | To perform these tasks... |
|------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Telephony Route/Digit Manipulation |   Telephony Routes <ul style="list-style-type: none">  Routing COS  Add Telephony Route  List Telephony Routes  Add RouteList  List RouteLists  Translation Tool | Add, list, modify, or delete telephony routes |
| Admin |   Admins <ul style="list-style-type: none">  List Admins  Add Admin  List Roles  Add Roles | Add, list, modify, or delete administrators and roles. |
| Gateway Management |   Gateways <ul style="list-style-type: none">  Add Gateway  List Gateways  Add Route  List Routes  Add TrunkGroup  List TrunkGroups | Add, list, modify, remove, reorder, new or existing gateways, routes, and trunk groups. When you deploy and provision a gateway through the management console, you must provision the gateway with the similar trunk groups and routes through the SIP Provisioning Client to ensure that the information is saved into the database that the SIP Application Module uses. |

Table 1 SIP Provisioning tasks overview

| If you have the right for | Use this Provisioning menu option... | To perform these tasks... |
|---------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Voice Mail Management |  <p>Voice Mail</p> <ul style="list-style-type: none"> Add Voice Mail Server List Voice Mail Servers | <p>Add, list, modify, or remove new or existing voice mail servers. If voice mail is enabled in a service package, a customer can select from the following voice mail server types:</p> <ul style="list-style-type: none"> • SIP based - a forward to your voice mail system using SIP. • Trunk based - a Primary Rate Interface (PRI) or Channel Associated Signalling (CAS) trunk is associated with the line. • Line based - a particular phone number is associated to the line. |
| User Management |  <p>Users</p> <ul style="list-style-type: none"> Add User List Users List Aliases List Blended Aliases | <p>Add, list, modify, or delete users, aliases, blended aliases, and i2004 Internet Telephone properties (available from the domain or subdomain menu option).</p> |

Understanding provisioning roles and rights

Administrators can create, or define, all the necessary provisioning roles to support their system. They can allow or restrict provisioning roles to carry out specific actions. The following table provides examples of several administrator provisioning roles and the rights based restrictions associated with various provisioning tasks.

Table 2 Example of roles and rights

| Provisioning Role example | Rights given | Allowed tasks |
|---------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| User administrator | <ul style="list-style-type: none"> User management with read, write, and delete access Domain management with read access only | Can view domain details, and add, delete, or modify users. Does not have access to other parts of the system, for example voice mail, service packages, and so forth. |
| Device administrator | <ul style="list-style-type: none"> Domain management with read access Device management with read, write, and delete access | Cannot add or modify users. Allowed to add, modify or delete devices. |
| System Administrator | <ul style="list-style-type: none"> Full domain access | Can see all domains, regardless of who created the domain, or the list of domains provisioned against the administrator. |

Limiting access to a domain

Complementing the rights-based restrictions, administrators can be provisioned against certain domains in order to limit access to within those domains. This prevents an administrator from domain *xyz* from viewing or modifying details in domain *abc*, even though they are hosted from the same system. Administrators can then delegate work for subdomains to other administrators without giving them access to their own data.

For example, if an administrator assigned to the User Administrator provisioning role was given access to *suba.xyz* (a subdomain of domain *xyz*), this administrator can add, modify, and delete users in the subdomain, but does not have access to the parent domain, domain *xyz*. This allows for the delegation of task on both the rights-level and the domain-level.



Note: Access to any domain implies access to all of its descendant subdomains. For example, access to domain *xyz* implies access to *suba.xyz*. Full domain access will override this domain allocation and allow any administrator assigned with a provisioning role that includes this right, to see all domains regardless of what the administrator is provisioned against.

System administrator provisioning role

The System Administrator (SA) is the highest level administrator and is the only provisioning role that is preloaded when the SIP Provisioning Client is deployed. The SA has full access and rights to all provisioning features and functions.

The SA can designate someone as an Administrative user with the **Add Admin** menu option, and grant specific administrator rights to that administrative user with the **Add Roles** menu option.



Note: The System Administrator role is the only role that cannot be changed.

Creating a new provisioning role

To create a new provisioning role

- 1 Click the **Admins** menu option.
- 2 Click the **Add Roles** sub menu option. The Add a New Role window appears (Figure 6).

Figure 6 Add a New Role window

The screenshot shows the 'Add a New Role' window in the Nortel Networks Provisioning interface. The interface has a blue header with the Nortel logo and 'Unified Network' text. Below the header is a navigation menu with options like HOME and HELP. The main content area is titled 'Add a New Role' and contains the following elements:

- Role Name:** A text input field containing 'Super User'.
- Role Description:** A text input field containing 'All rights'.
- Choose level of access for each Admin privilege:** A table with three columns: Read, Write, and Delete. Each row represents a privilege, and each cell contains a checkbox.
- Add:** A button at the bottom of the form.

| Choose level of access for each Admin privilege | Read | Write | Delete |
|-------------------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|
| Full Domain Access | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| Domain Management | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| IPCM Provisioning | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| Service Package Creation | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| Device Management | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| Telephony Routes | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| Admin | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| Gateway Routes | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| Voice Mail Management | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| User Management | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |

- 3 Enter a name for the provisioning role in the Role Name field (required).
- 4 Enter a description for the provisioning role in the Role Description field.
- 5 To define the level of administrator access, select the checkboxes for Read, Write, or Delete next to each privilege.
- 6 Click **Add**. A message appears that the role has been successfully created.

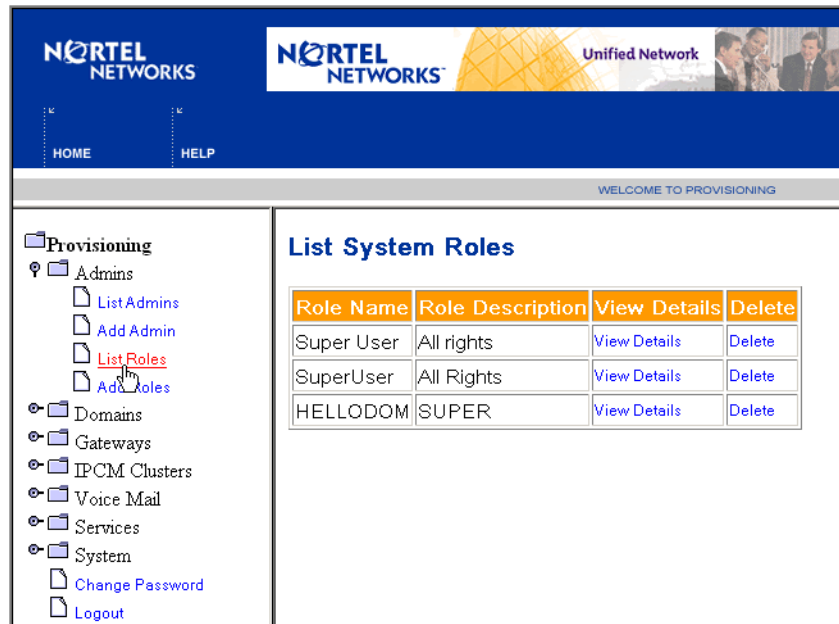
Listing provisioning roles

To view details about a provisioning role to modify or delete the role

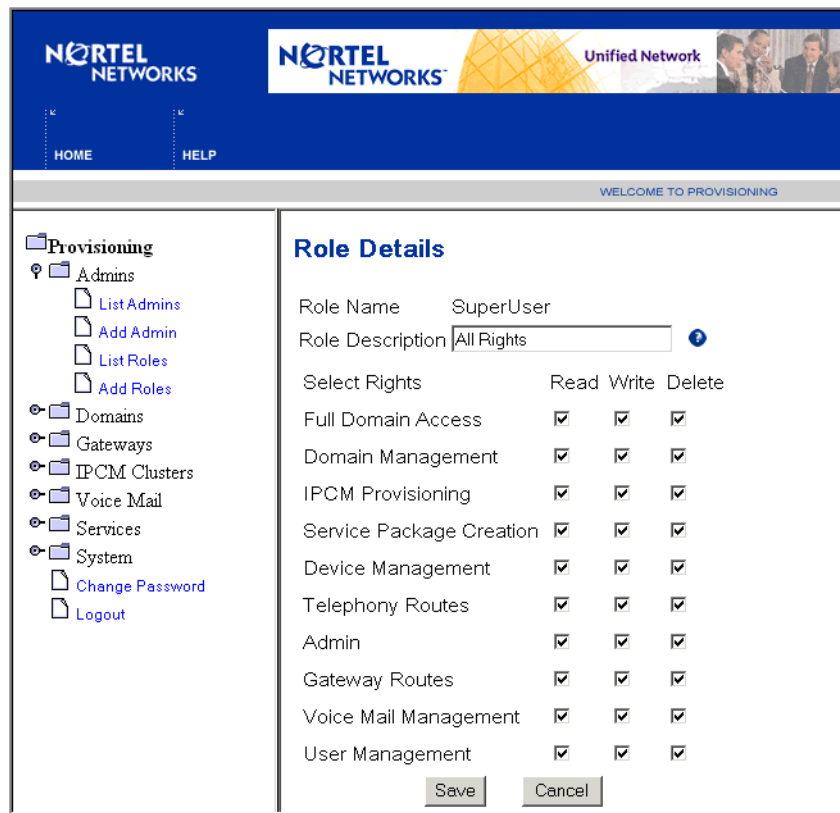
- 1 Click the **Admins** menu option.

- Click the **List Roles** sub menu option. The List System Roles window appears (Figure 7).

Figure 7 List System Roles window



- To view details about a provisioning role or description, click the **View Details** link. The Role Details window appears (Figure 8). To delete a provisioning role, click the **Delete** link next to the role.

Figure 8 Role details window

- 4 To change the provisioning role description, enter a new description in the Role Description box. To modify privileges, select or deselect privileges in the checkbox next to the privilege.
- 5 Click **Save** to save your changes. A message appears that the provisioning role was modified successfully.

Creating a new administrator

To add an administrator

- 1 Click the **Admins** menu option.

- 2 Click the **Add Admin** sub menu option. The Create new admin window appears (Figure 9).

Figure 9 Create new admin

Create new admin

Username: dburns * ?

First Name: david ?

Last Name: burns ?

Password: [masked] ?

Confirm Password: [masked] ?

Email: dburns@xyz.com

Business: 972-123-5544

Phone: 972-111-1199

Cell Phone: 872-223-9901

Pager: [empty]

Fax: [empty]

VPN: [empty]

System Mgr Role: System Administrator ?

Provisioning Role: Super User - All rights ?

Provisionable domains: abc.com, nortelnetworks, otta.abc.com, pstn ?

Time Zone: Eastern Standard Time ?

Locale: English ?

Add

- 3 Enter the username of the administrator (required) and other information in the fields provided.
- 4 Click **Add**. A message appears that the administrator was successfully added.

Listing administrators

To view details about an administrator or to modify or delete an administrator

- 1 Click the **Admins** menu option.

- 2 Click the **List Admins** sub menu option. The Admins window appears (Figure 10). To delete an administrator, click the **Delete** link next to the administrator username.

Figure 10 Admins window



The screenshot displays the Nortel Networks Admins window. The header includes the Nortel Networks logo and the text 'Unified Network'. Below the header, there are navigation links for 'HOME' and 'HELP'. The main content area is titled 'Admins' and contains a table with the following data:

| Username | Full Name | Details | Delete |
|---------------|-------------------------|-------------------------|------------------------|
| demoadmin | Demo Admin | Details | Delete |
| demouseradmin | Demo User Administrator | Details | Delete |
| dburns | dave burns | Details | Delete |
| admin | admin admin | Details | |

- 3 To modify administrator information, enter the changes in the fields provided (Table 11).

Figure 11 Admin Details window

The screenshot shows the 'Admin details' window with the following fields and values:

- Username: dburns
- First Name: david
- Last Name: burns
- Password: [masked]
- Confirm Password: [masked]
- Email: dburns@xyz
- Business: [empty]
- Phone: [empty]
- Cell Phone: [empty]
- Pager: [empty]
- Fax: [empty]
- VPN: [empty]
- System Mgr Role: System Administrator
- Provisioning Role: Super User - All rights
- Provisionable domains: abc.com (selected), nortelnetworks, otta.abc.com, pstn, rich.abc.com
- Time Zone: Eastern Standard Time
- Locale: French
- Save button

- 4 Click **Save** to save your changes. A message appears that the administrator was modified successfully.

Chapter 3

Domain Management

Topics in this section:

- [“Introduction to domains” on page 23](#)
- [“Domain types” on page 24](#)
- [“Adding a local root domain” on page 26](#)
- [“Adding a subdomain” on page 31](#)
- [“Adding a foreign domain” on page 33](#)
- [“Listing/modifying a local or foreign domain” on page 33](#)
- [“Deleting a local domain” on page 35](#)
- [“Banning a user” on page 36](#)
- [“Defining user device status reasons” on page 38](#)

Introduction to domains

A domain is used as a control mechanism of users, services, devices and translation. The SIP Provisioning Client is used to provision a domain. The provisioning of domains allows service providers to define a domain and service packages available for the domain.

Once the domain and service packages are defined, service providers can define subdomains and assign the following:

- services to service packages
- service packages to domain and users
- translation information specific to a domain or a subdomain
- devices for users to an IPCM (IP Client Manager)

If you have full domain access, you can see all provisioned domains. Otherwise, you will only be able to see the domains/subdomains you are provisioned against. The management rights assigned to a domain administrator dictates what you can do with the domains.

Domain types

There are three types of domains:

- root domain
- subdomain
- foreign domain

Root domain

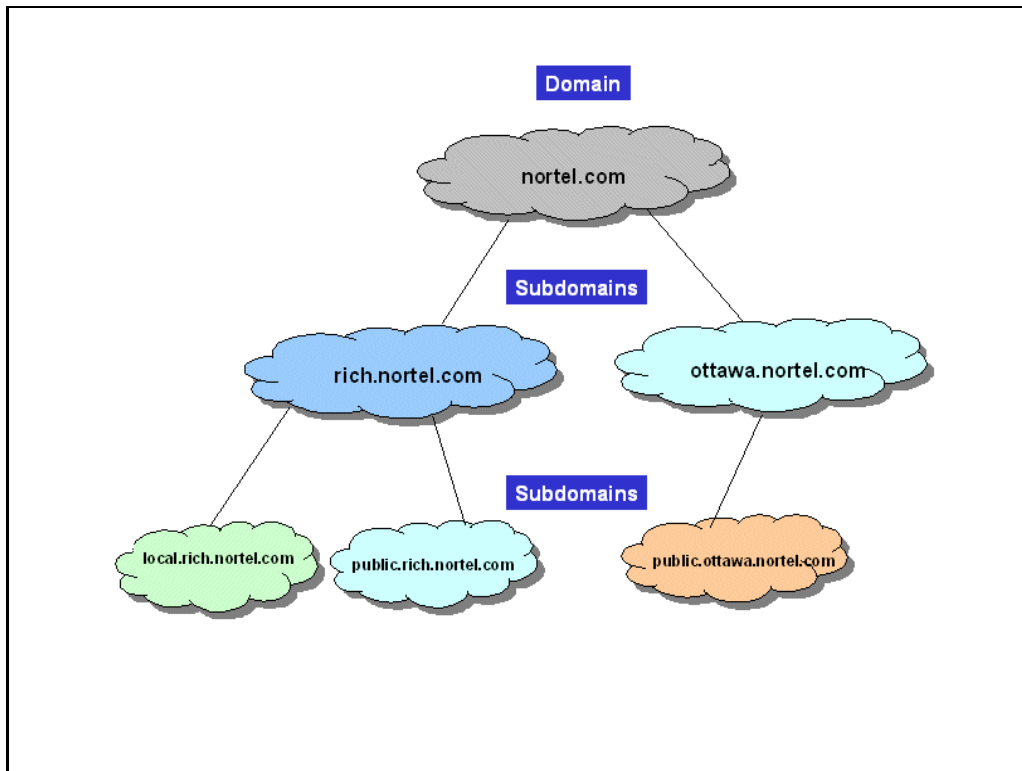
A root domain is the highest level domain. A service provider can create a root domain for each of their customers. An example of a root domain is *nortelnetworks.com*.

Subdomain

A subdomain is a category of domains. Subdomains allow you to divide subscribers under a domain into smaller groups. The groups can be used to control routing and access to services by members of the group.

Under each subdomain, you can add subdomains, service packages, users, devices, and telephony routes. There are no limits to the levels of nesting of subdomains. Each subdomain inherits the parameter values from the root domain.

A typical application of subdomains is based on geographical divisions. For example, using Nortel Networks, you could have *nortelnetworks.com* as the root domain, with subdomains of Richardson and Ottawa. Under each of these subdomains there could be further subdomains as shown in the following illustration (Figure 12).

Figure 12 Domains and subdomains

Foreign domain

A foreign domain is a domain that is not served by any SIP Application Modules in your system. When adding a domain, you must determine whether or not the domain is a foreign domain or not. A foreign domain uses foreign routes, while a local domain uses SIP routing or telephony routes.

A domain that is not local is considered foreign. Usually, a foreign domain is resolved through DNS and would not need to be provisioned. However, since some deployment scenarios may not involve a DNS server, you (as the system administrator) can datafill routes for a foreign domain.

For example, if the SIP Application Module serving *nortelnetworks.com* does not have a DNS server, and there is a need for users to be able to make calls to the *xyz.com* domain, then you could add the *abc.com* domain as a foreign domain and provide a route to *abc*'s proxy.



Note: A foreign domain is not controlled by the system. Therefore it will not be displayed under the domain folder in the menu area, and the you will not be able to provision users, service packages, devices, and so forth to these domains.

Adding a local root domain

To add a local root domain

- 1 Click the **Domains** menu option.
- 2 Click **Add Domain**. The Create new domain window appears (Figure 13).

Figure 13 Create new domain window

Create new domain

Name:

Foreign:

Foreign Dest:

Parameters

Location:

Idle Display:

WD Phone Timer:

WD Timer:

Time FMT:

Date FMT:

PSEIZ Timer:

PDIL Timer:

Alpha: TRUE FALSE

Default Proxy Port:

Behind Firewall: TRUE FALSE

Allow All Codecs: TRUE FALSE

Active Proxy: FALSE TRUE

Default Proxy Host:

Time Zone:

Vocoder Packet Time:

Contrast:

Server Home:

Realm for a domain:

Always Use Media Portal:

Maximum Number of Presence Subscriptions Accepted:

i2004 Locale:

- 3 Enter the name of the new local domain. The domain name must not be more than 60 chars in length and cannot contain the following symbols or characters as described in the following table:

Table 3 Create new domain name restrictions

| | | | | |
|---|----|----|-----|-----|
| ' | \$ | () | | ; |
| ~ | % | _ | [] | / |
| ! | ^ | | { } | ? |
| @ | & | + | “ ” | , |
| # | * | = | : | < > |

- 4 Select **No**. There may be times when you want two domains on different systems to communicate with each other. If **Yes**, enter the URL destination (IP address) of where the domain is located in the **Foreign Dest:** field.



Tip: Domain parameters are not necessary for foreign domains. When adding a foreign domain, the parameters can be ignored.

- 5 Enter parameter information in the fields provided. See the following table for parameter values and descriptions (Table 4).

Table 4 Create new domain parameters

| Parameter | Type | Range | Default | Description |
|--------------|---------------------------------------------|-----------------------------------------------------------|---------|-----------------------------------------------------------------------------------------------------------------------------------|
| i2004 Locale | String; select from drop-down list | Restricted to English in the current release. | N/A | Indicates the language in which the i2004 Internet Telephone will display. Future releases will include more supported languages. |
| Location | String | 0-50 characters | N/A | Text describing location of device. A specific format of this string is scheduled for a future release. |
| Idle Display | String | Up to 30 characters | N/A | Idle telephone display heading. |

Table 4 Create new domain parameters

| Parameter | Type | Range | Default | Description |
|--------------------|---------------------------------------------------------------------------------------------------------------------------------|----------------|---------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| WD Phone Timer | Integer | 13-10000 | 150 | Length of time the i2004 Internet Telephone waits before determining that it has lost connection to the IPCM. Should be greater than WD_Timer by X amount. Where X is difference between MAX expected network latency and MIN expected network. |
| WD Timer | Integer | 30-10000 | 15 | How often (in seconds) the IPCM will send a "reset watchdog timer" message to a connected I2004 device. When the I2004 receives this watch dog message, it knows that it is still connected to the IPCM. |
| Time FMT | Select from drop down list: <ul style="list-style-type: none"> • 12-hour • French • 24-hour | 0,1,2 | 0 | The time format of the proxy used as default for the i2004 Internet telephone 0-> 12 hour clock |
| Date FMT | Numeric Standard MM/DD or Numeric Inverse: DD/MM | 2,3 | 2 | 2-numeric standard (09/16); 3-numeric inverse (16/09). |
| PSEIZ Timer | Integer | N/A | 15 | Time-out for first dialed digit. |
| PDIL Timer | Integer (seconds) | N/A | 4 | Interdigit time-out (seconds). |
| Alpha | Boolean | True/False | False | Enables alphanumeric dialing as the default dialing style. If disabled, I2004 defaults dialing style to numeric. |
| Default Proxy Port | 5060 | Any valid port | 5060 | Default proxy SIP port. No validation is done on this field to ensure validity. |
| Behind Firewall | Boolean | True/False | True | Boolean indicating whether there is a firewall between the IPCM and the i2004 Internet Telephone. |
| Allow All Codecs | Boolean | True/False | False | Allow all codecs (true) or only selected codec (false). See Vocoder parameter (below). |

Table 4 Create new domain parameters

| Parameter | Type | Range | Default | Description |
|--------------------|-------------------------------------|----------------------|---------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Active Proxy | Boolean | True/False | True | Determines if default proxy is used. Used for SIP messaging. If yes, calls will route through the default proxy. No other routing is currently supported for the IPCM. |
| Default Proxy Host | String (30) | Any valid IP address | N/A | IP Address of proxy used as default for the i2004 telephone. No validation is done on this field to ensure validity. |
| Time Zone | Integer; select from drop down list | Up to 30 characters | N/A | Time zone of the user. |
| Vocoder:PacketTime | Integer; select from drop down list | 0,4,8,10,18 | 0 | Default Codec Setting: 0 – G711 Mulaw; 4 – G723\6.3kbps; 8 – G711\Alaw; 10 – L16; 18 – G729A. |
| Contrast | Integer; select from drop down list | 0 - 15 | 8 | i2004 display contrast setting. |
| Server Home | String(60) | Up to 60 characters | N/A | Specifies the SIP Application Module that will host the users in this domain. This is used in an N+M environment. All SIP Application Modules that are not the home SIP Application Module will forward requests for users in this domain to the specified address as appropriate. (Example: 10.0.0.8) If no host address is supplied, or the host address is 0.0.0.0., no homing is performed. |
| Realm for a domain | String (120) | Up to 120 characters | Realm | Identifies the domain for subscribers when they are being authenticated. Subscribers may register in multiple domains. When they are required to enter a password during authentication, they need to know which domain it is so that they can enter the appropriate password. |

Table 4 Create new domain parameters

| Parameter | Type | Range | Default | Description |
|---------------------------------------------------|---------|------------|---------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Always Use Media Portal | String | True/False | False | Directs the Application Server to use the RTP Media Portal when set to true . The purpose is to use the RTP Media Portal function in dealing with special SIP scenarios. For example, a domain that straddles multiple sites contains users who do not have Internet Protocol (IP) connectivity between them due to firewalls at different sites. This domain requires the Always Use Media Portal to be set to True to negotiate between the firewalls to set up SIP sessions. |
| Maximum Number of Presence Subscriptions Accepted | Integer | N/A | 0 | Maximum number of inbound subscriptions to any given user in this domain. The SIP Personal Agent, the SIP Multimedia PC Client, and the SIP Multimedia Web Client will use this limit when allowing a particular username to be added to a subscriber's list of buddies. |

6 Click **Add**.



Tip: In order to see the new root domain added to the list of domains, you must close and reopen the domains folder to refresh the list of domains.

Adding a subdomain

To add a subdomain

- 1 Click the **Domains** menu option
- 2 Under the root domain where you want to add a subdomain, click the **Sub-Domains > Add Sub Domain** menu option. The Create new subdomain window appears with a list of parameters inherited from the parent domain (Figure 14).

Figure 14 Create new subdomain window

Create new Sub Domain

Parent abc.com

Name: .abc.com

Domain Class of Service

Parameters

Always Use Media Portal: false

Location:

Default Proxy Host:

Allow All Codecs: TRUE FALSE

WD Phone Timer:

Contrast:

Idle Display:

Alpha: TRUE FALSE

PSEIZ Timer:

i2004 Locale:

WD Timer:

Maximum Number of Presence Subscriptions Accepted: 0

Vocoder.PacketTime:

Date FMT:

Behind Firewall: TRUE FALSE

Time Zone:

Time FMT:

Realm for a domain: Realm

PDL Timer:

Default Proxy Port:

Server Home:

Active Proxy: FALSE TRUE

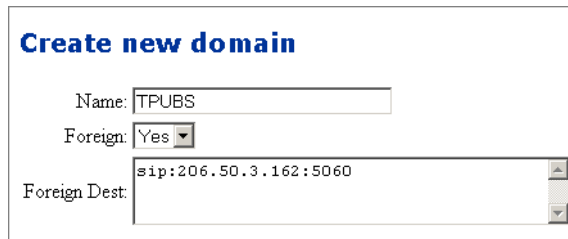
- 3 Enter a name for the subdomain.
- 4 Select a COS (Class of Service) for the subdomain from the drop down list of routing services.
- 5 Click **Save**.

Adding a foreign domain

To add a foreign domain

- 1 Click the **Domain > Add Domain** menu option. The Create new domain window appears (Figure 15). Since parameter fields are not used for foreign domains, only the fields at the top of the window need to be completed.

Figure 15 Create new domain window



Create new domain

Name:

Foreign:

Foreign Dest:

- 2 Enter a name for the foreign domain.
- 3 Select **Yes** from the Foreign drop down box.
- 4 Add a foreign destination into the Foreign Destination box using the following format
`SIP<IP_Addr>:port`
- 5 Click **Add**.

Listing/modifying a local or foreign domain

To list/modify a local or foreign domain

- 1 Click the **Domains > List Foreign Domains** menu option. The Domains window appears (Figure 16).

Figure 16 List Domains window - step 1

The screenshot shows a window titled "Domains". Under the "Criteria Selection" heading, there is a dropdown menu with "Is a foreign domain" selected, followed by an equals sign and another dropdown menu. To the right of these is a text box containing "Tpubs". Below this is an "Add Criteria" button. A horizontal line separates this section from the search options. Below the line, it says "Search by" followed by two radio buttons: "All" (which is selected) and "Any". Below this is a text box containing "Domain name = Tpubs" and a "Remove" button. At the bottom center is a "Search" button.

2 From the Criteria Selection drop down list, select one of the following as a search criteria:

- Domain Name
- Is a Foreign domain
- ID

or enter the name of the domain, foreign domain, or ID in the criteria selection text box.

3 Click **Add Criteria**. The search criteria you selected appears in the lower window and is used to refine your search.

4 Repeat steps 1 -3 to list additional domains.

5 In the lower window area, select whether to search by All or Any of the criteria specified. Note that you do not have to add any criteria if you want to see all the domains in the network.

6 Click **Search** to begin the search, or **Remove** to remove a search criteria from the search list. A list of domains (based on the search criteria specified) appears (Figure 17).

Figure 17 List domains - step 2

| Foreign Domains | | |
|-----------------|-------------------------|------------------------|
| Name | Details | Delete |
| tpubs | Details | Delete |
| tpubs1 | Details | Delete |

- 7 In the Domains window, select a domain and click the **Details** link. The details for the selected domain, or domains, displays.
- 8 Modify domain information in the parameter fields.
- 9 Click **Save**.

Deleting a local domain



Caution: Exercise extreme caution when deleting a domain as this will permanently remove all users, devices, routes, service packages, and so forth, associated with the domain. This is an irreversible operation that could be a service-affecting change. For example, any calls in progress to users or devices in the domain will terminate.

To delete a local domain or subdomain

- 1 Click the **Domains** menu option.
- 2 Click the domain or subdomain you want to delete from the list of domains.



Tip: In order to delete a domain or subdomain you must first delete all of its children subdomains.

- 3 Click **Delete** to delete the domain or subdomain you want to delete.
- 4 Click **OK** in the confirmation dialog box to delete the domain, or **Cancel** to cancel the delete operation.

Deleting a foreign domain



Caution: Exercise extreme caution when deleting a foreign domain as this will permanently remove all users, devices, routes, service packages, and so forth, associated with the domain. This is an irreversible operation that could be a service affecting change. For example, any calls in progress to users or devices in the domain will terminate.

To delete a foreign domain

- 1 Click the **Domains > List Foreign Domains** menu option.
- 2 Click the **Delete** link for the foreign domain you want to delete.
- 3 Click **OK** in the confirmation dialog box to delete the foreign domain, or **Cancel** to cancel the delete operation.

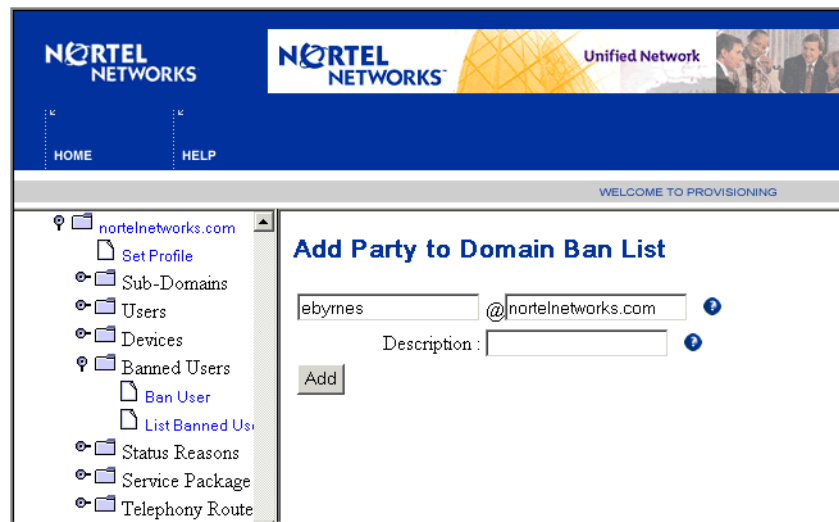
Banning a user

You can ban an individual user or a group of users in a domain or from viewing the presence information (online status) of a user or group of users in a domain. The banning takes effect for all subdomains under the domain.

Setting up ban list

To ban a user or a group of users

- 1 Click the **Domains** menu option.
- 2 Under the domain where you want to ban a user or users, click the **Banned Users > Ban User** menu option. The Add a party to Domain Ban List window appears ([Figure 18](#)).

Figure 18 Add a party to Domain Ban List window

- 3 Enter the party to ban in the text boxes provided. To ban an individual user, use the following format:
`<username>@<domain_name.com>`
 To ban a group of users in a domain, use the following wildcard format:
`<*>@<domain_name.com>`
- 4 Enter a reason for banning the user or group of users to the domain list in the Description text box.
- 5 Click **Save**. A message displays that the party was successfully added to the ban list.

Viewing a list of banned users

To view a list of users banned from viewing the presence information (online status) of a user or a group of users in a domain

- 1 Click the **Domains** menu option.
- 2 Under domain where you want to view a list of banned users, click the **Banned Users > List Banned User** menu option. The Domain Ban List window appears (Figure 19).

Figure 19 Domain Ban List window



| ID | Party | Description | Delete |
|-----|----------------------------|--------------|--------|
| 813 | ebyrnes@nortelnetworks.com | troublemaker | Delete |

- 3 To delete a user or a group of users from the ban list, click the **Delete** link next to the party to ban.
- 4 Click **OK** at the delete confirmation dialog box, or **Cancel** to cancel the delete operation.

Defining user device status reasons

The **Status Reason** menu option allows you to define a list of reasons for an inactive user device status.

When a user is inactive, the user cannot register to the SIP Application Module. This means that the user cannot use any of the services provided by the SIP Application Module, including making calls or subscribing to other users' presence. However, the inactive user can log in to the SIP Personal Agent and view the reason for their INACTIVE status.

Adding a status reason

To define a status reason for a user device

- 1 Click the **Domains** menu option.
- 2 Under the domain where you want to define a status reason, click **Status Reason > Add Reason**. The Add Status Reason window appears (Figure 20).

Figure 20 Add Status Reason window

The screenshot shows the Nortel Networks provisioning interface. The top navigation bar includes 'HOME' and 'HELP' links. The main content area is titled 'Add Status Reason'. On the left, a tree view shows the domain structure under 'nortelnetworks.com', with 'Status Reasons' expanded to show 'Add Reason' (highlighted in red) and 'List Reasons'. The 'Add Status Reason' form contains the following fields:

- Status: ACTIVE (dropdown menu)
- Name: lapsed (text input)
- Description: inactive for at least 6 months (text input)
- URL: www.somecompany.com (text input)
- Save (button)

- 3 From the Status drop down list, select **INACTIVE**.
- 4 In the Name field (required) enter a name to identify the status reason.
- 5 In the Description field, enter a description for the inactive status. For example, if a user has not paid their bill, the reason for the inactive status could be “Inactive for at least 6 months”.
- 6 In the URL field, enter a web page address. This is the web page that the originator of a call to the user will be presented with, along with the status reason description, to explain the called user’s inactive status.
- 7 Click **Save**. A message displays that the status reason was successfully added.

Viewing status reasons

To view a list of status reasons for a user device status

- 1 Click the **Domains** menu option.
- 2 Under the domain where you want to view the status reasons, click the **Status Reason > List Reasons** menu option. The List Reasons window appears (Figure 21).

Figure 21 Status Reasons window

| Status Reasons | | | |
|----------------|------------|------------------------|------------------------|
| Status | Name | Detail | Delete |
| ACTIVE | lapsed | Detail | Delete |
| ACTIVE | non paymnt | Detail | Delete |

- To view or modify status reason information, click the **Details** link for information about the status reason. The Modify Status Reason window appears (Figure 22).

Figure 22 Modify status reason window

| Modify Status Reason | |
|-------------------------------------|------------------------------------------------------------|
| Status: | <input type="text" value="INACTIVE"/> |
| Name : | <input type="text" value="lapsed"/> |
| Description : | <input type="text" value="Unused for at least 6 months"/> |
| URL : | <input type="text" value="http://www.somenewcompany.com"/> |
| <input type="button" value="Save"/> | |

- Enter changes in the fields provided to modify information in the status reasons fields.
- Click **Save**. A message displays that the status reason was successfully modified.
- To delete a status reason, click the **Delete** link.
- At the delete confirmation dialog box, click **OK** to delete the status reason, or **Cancel** to return to the Status Reasons details window.

Chapter 4

IPCMs



Note: You must have IPCM provisioning rights to perform the tasks in this section.

Topics in this section:

- [“What is an IPCM?” on page 41](#)
- [“Adding an IPCM cluster” on page 42](#)
- [“Assigning and modifying IPCMs to a domain” on page 44](#)
- [“Deleting an IPCM cluster” on page 46](#)
- [“Listing physical IPCMs” on page 47](#)

What is an IPCM?

The IP Client Manager (IPCM) is a network application capable of managing many internal i2004 Internet Telephones while maintaining the call state and phone settings. The IPCM also implements features (for example, stock queries and personal directory) on the i2004 Internet Telephones. An IPCM Cluster may span multiple domains and/or contain multiple domains.



Note: Users and devices are linked to IPCM Clusters, and not to individual IPCMs. An IPCM Cluster may contain a single IPCM or a pair of IPCMs. The IPCM cluster contains a primary and standby IPCM for each i2004 Internet Telephone. Clustering of IPCMs ensures that each IPCM retrieves the same user and device data from the database and that each receives notification of database modifications.

In order for the IPCM to manage i2004 Internet Telephones, information pertaining to the IPCM in general, the i2004 telephones, and users that will use i2004s, must be provisioned. For more information, see the *SIP IP Client Manager Basics* documentation.

i2004 Internet Telephone information can be provisioned in the following ways:

- Provision each individual i2004 using the SIP Provisioning Client.
- Set up default device properties for a domain, assign an activation key to the domain, and then use the IPCM to auto-provision each i2004 device whenever a user enters the desired activation key. (See the *SIP IP Client Manager Basics* documentation and the *i2004 Internet Telephone User Guide* for more information.)



Note: Normally, the IPCM is deployed prior to provisioning the IPCM. If this is the case, it is still necessary to assign domains to the IPCM cluster and to assign a provisioned capacity for the IPCM cluster and domain. If the IPCM has been deployed, see [“Assigning and modifying IPCMs to a domain.”](#) If the IPCM has not been deployed, and it is desired to pre-provision the IPCM, see [“Adding an IPCM cluster.”](#)

Adding an IPCM cluster

Adding an IPCM cluster creates a virtual IPCM to which users, i2004 devices, and domains may be assigned. It does not actually create a physical IPCM, nor does it deploy IPCM software onto a server. The association of the physical IPCM to this IPCM cluster is performed when the physical IPCM is deployed and configured.

To add an IPCM cluster

- 1 Click the **IPCM Cluster > Add IPCM Cluster** menu option. The Create new IPCM window displays. The following figure ([Figure 23](#)) shows an example of creating an IPCM named LAB IPCM. The maximum provisioned capacity is 1000, i2004 Internet Telephones. Note that the IPCM name is case sensitive.

Figure 23 Create new IPCM window

The screenshot shows the Nortel Networks provisioning interface. The header includes the Nortel Networks logo and 'Unified Network' branding. Below the header is a navigation bar with 'HOME' and 'HELP' links. A 'WELCOME TO PROVISIONING' message is displayed. The main content area is titled 'Create new IPCM'. It features two input fields: 'IPCM Name' with the value 'LAB IPCM' and 'Prov. Capacity' with the value '1000'. A 'Save' button is positioned below the 'Prov. Capacity' field. On the left, a navigation tree is visible, with 'IPCM Clusters' selected and expanded to show options like 'List IPCM Clusters', 'Add IPCM Cluster', and 'List Physical IPCMs'.

The following table (Table 5) lists the Create new IPCM parameters and values to use when creating a new IPCM.

Table 5 Create new IPCM parameters

| Parameter | Type | Range | Default | Description |
|----------------|-------------|----------------------------|---------|----------------------------------------------------------------------------------------------------------------|
| IPCM Name | String(100) | Up to 100 characters | None | The name of this IPCM. The name of the IPCM must be a unique name. |
| Prov. Capacity | Integer | Maximum of 10,000 per IPCM | None | The number of i2004 Internet Telephones that are allowed to be provisioned for this IPCM (across all domains). |

2 Click **Save**.

Assigning and modifying IPCMs to a domain

An IPCM is not usable until it is assigned to a domain. To assign an IPCM to a domain

- 1 Click the **IPCM Clusters > List IPCM Clusters** menu option. A list of deployed IPCMs appears (Figure 24).



Note: Upon first assignment of an IPCM, the list of deployed IPCMs may be empty. However, in many cases, the IPCM would have already been deployed using the System Management Console and will appear in the list.

Figure 24 List IPCM Clusters window

| IPCM Clusters | | | | |
|---------------|--------------|---------------|-------------------------|------------------------|
| IPCM Cluster | ProvCapacity | AvailCapacity | Details | Delete |
| paul | 1000 | 0 | Details | Delete |
| Test IPCM | 5000 | 5000 | Details | Delete |
| LAB IPCM | 1000 | 1000 | Details | Delete |
| Test | 1000 | 900 | Details | Delete |

- 2 Click the **Details** link for the IPCM you want to assign to a domain. The Modify IPCM window appears (Figure 25).

Figure 25 Modify IPCM window

MODIFY IPCM

| | |
|-------------------------------------|----------------------------------------|
| IPCM Name : | <input type="text" value="Test IPCM"/> |
| Prov. Capacity : | <input type="text" value="5000"/> |
| Avail Capacity: | <input type="text" value="5000"/> |
| <input type="button" value="Save"/> | |

Domains Assigned to this IPCM

| Domain | Activation Key | Assigned Capacity | Current Capacity | Modify | Delete |
|--------|----------------|-------------------|------------------|--------|--------|
| | | | | | |

Assign New Domain

| Available Domains | Activation Key | Assigned Capacity | Current Capacity | Modify |
|-----------------------------------------------------------------------|----------------------|----------------------|------------------|-------------------------------------|
| <input type="text" value="abc.com"/> <input type="button" value="i"/> | <input type="text"/> | <input type="text"/> | 0 | <input type="button" value="Save"/> |

- 3 If necessary, modify the IPCM Name or Prov. Capacity parameters and click **Save**.



Tip: The Avail Capacity field is a read only field calculated by adding the Assigned Capacity of domains to which this IPCM is assigned and subtracting that value from the Prov. Capacity of the IPCM.

- 4 Select a domain from the list of available domains in the Assign New Domain area of the window. Enter information in the following fields for the domain you are assigning to the IPCM:

| Field | Description |
|--------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Assigned Capacity: | The number of i2004 Internet Telephones you want to allow in this assigned domain. |
| Activation Key: | A unique string (number) per IPCM/domain combination that associates default device properties with this particular domain. The administrator must pick a unique string (number) for each domain. If the IPCM is used to auto-provision i2004 devices, the installer will be prompted for this activation key during the installation process. |

- 5 Click **Save**. The IPCM Domain Relation window appears (Figure 26).

Figure 26 IPCM Domain Relation window

IPCM Domain Relation Created

MODIFY IPCM

| | |
|-------------------------------------|----------------------------------------|
| IPCM Name : | <input type="text" value="Test IPCM"/> |
| Prov. Capacity : | <input type="text" value="5000"/> |
| Avail Capacity: | <input type="text" value="500"/> |
| <input type="button" value="Save"/> | |

Domains Assigned to this IPCM

| Domain | Activation Key | Assigned Capacity | Current Capacity | Modify | Delete |
|---------|-----------------------------------|-----------------------------------|------------------|-------------------------------------|---------------------------------------|
| abc.com | <input type="text" value="1234"/> | <input type="text" value="4500"/> | 0 | <input type="button" value="Save"/> | <input type="button" value="Delete"/> |

Assign New Domain

| Available Domains | Activation Key | Assigned Capacity | Current Capacity | Modify |
|------------------------------------------------------------------------------|----------------------|----------------------|------------------|-------------------------------------|
| <input type="text" value="nortelnetworks"/> <input type="button" value="i"/> | <input type="text"/> | <input type="text"/> | 0 | <input type="button" value="Save"/> |

- 6 Enter your changes and click **Save**. A confirmation messages displays confirming your changes.
- 7 To assign another domain to the IPCM, select a domain from the list in the Assign New Domain area. Enter information in the parameter fields and click **Save**.

Deleting an IPCM cluster



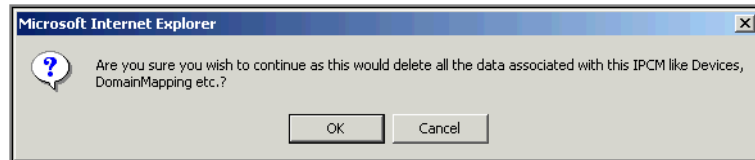
Caution: Exercise extreme caution when deleting an IPCM as this will permanently remove all i2004 Internet Telephones and users' i2004 capabilities associated with the IPCM that you are deleting.

To delete an IPCM cluster

- 1 Click the **IPCM Clusters > List IPCM Clusters** menu option.

- 2 Click the **Delete** link for the IPCM cluster you want to delete.
- 3 Click **OK** in the confirmation dialog box to delete the IPCM cluster, or **Cancel** to cancel the delete operation (Figure 27).

Figure 27 IPCM Delete confirmation dialog box



Listing physical IPCMs

The List Physical IPCMs menu option allows you to view the physical IP Address of the server on which the IPCM resides.

To list the physical IP address of the IPCM

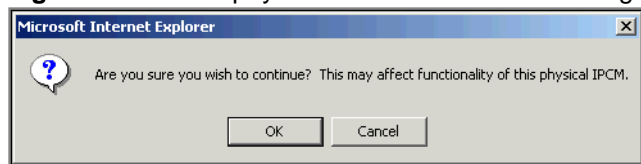
- 1 Click the **IPCM Clusters > List Physical IPCMs** menu option. The Physical IPCMs window appears (Figure 28) listing the following fields:

| Field | Description |
|-----------------------|----------------------------------------------------------------------------|
| IPCM Cluster: | The name that identifies the virtual IPCM cluster. |
| Physical IPCM: | The physical IP Address of the server on which the IPCM resides. |
| Delete Physical IPCM: | Allows you to delete the mapping of the physical IPCM to the IPCM Cluster. |

Figure 28 Listing physical IPCMs

| IPCM Cluster | Physical IPCM | Delete Physical IPCM |
|--------------|----------------|----------------------|
| Test IPCM | Not Found | Not Applicable |
| sean_esm | Not Found | Not Applicable |
| sean_ipcm | 47.103.142.222 | Delete |
| SandeepIPCM | Not Found | Not Applicable |
| Bud IPCM | 47.102.128.177 | Delete |
| 123 | 47.104.12.29 | Delete |
| TaterIPCM2 | Not Found | Not Applicable |
| TaterIPCM | Not Found | Not Applicable |
| andy_esm | 47.102.128.222 | Delete |

- To delete a physical IPCM, click the **Delete** link in the Delete Physical IPCM column. At the delete confirmation dialog box, click **Ok** to delete the physical IPCM or **Cancel** to cancel the delete operation (Figure 29).

Figure 29 Delete physical IPCM confirmation dialog box

Chapter 5

Voice Mail Servers



Note: You must have Voice Mail provisioning rights to perform the tasks in this section.

Topics in this section:

- [“Introduction” on page 49](#)
- [“Voice mail server types” on page 50](#)
- [“Provisioning a SIP-based voice mail server” on page 50](#)
- [“Provisioning a Trunk-based voice mail server” on page 54](#)
- [“Provisioning a Line-based voice mail server” on page 60](#)
- [“Assigning a user to a provisioned voice mail server” on page 67](#)
- [“Listing/modifying/deleting a voice mail server” on page 68](#)
- [“Using CPL scripts with voice mail” on page 69](#)

Introduction

The SIP Provisioning Client allows you to provision third-party voice mail server information to:

- set up and route calls to voice mail
- associate users to a voice mail server
- set up a voice mail server to receive message waiting indication (MWI) notifications

Voice mail server types

The following voice mail server types are supported as described in the following table (Table 6).

Table 6 Voice mail server types

| Voice mail server type | Description |
|------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| SIP-based | An IP based voice mail server that uses the SIP protocol. This server type does not need a PSTN gateway. |
| Line-based | A voice mail server with multiple lines, each identified by a 10 digit phone number that goes into the voice mail server. When people leave messages, the individual lines are used in a round robin manner to connect to the voice mail server to allow someone to leave a message. The SIP Application Module sends an SMDI message to the voice mail server to help establish the connection to the SIP Application Module that uses SMDI to obtain Message Waiting Indication (MWI) information from the voice mail server. |
| Trunk-based | A legacy Public Switched Telephone Network (PSTN) based voice mail server that uses a Primary Rate Interface (PRI) or Channel Associated Signalling (CAS) gateway. The SIP Application Module uses SMDI to send call setup information using a Call Detail Message to get Message Waiting Indication (MWI) information from the voice mail server. |

Provisioning a SIP-based voice mail server

Provisioning a SIP-based voice mail server allows the administrator to configure the SIP Application Module to communicate with a voice mail server using the SIP protocol.

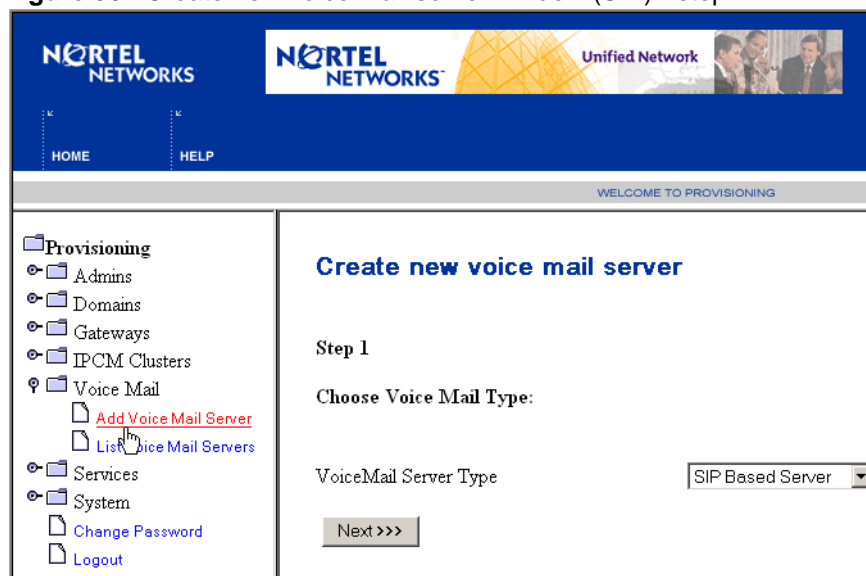
The SIP Application Module is given information on what to query to find the location of the voice mail server. The SIP Application Module will use this information to send SIP calls to the voice mail server when calls are routed to voice mail.

Adding a SIP-based voice mail server

To add a SIP-based voice mail server

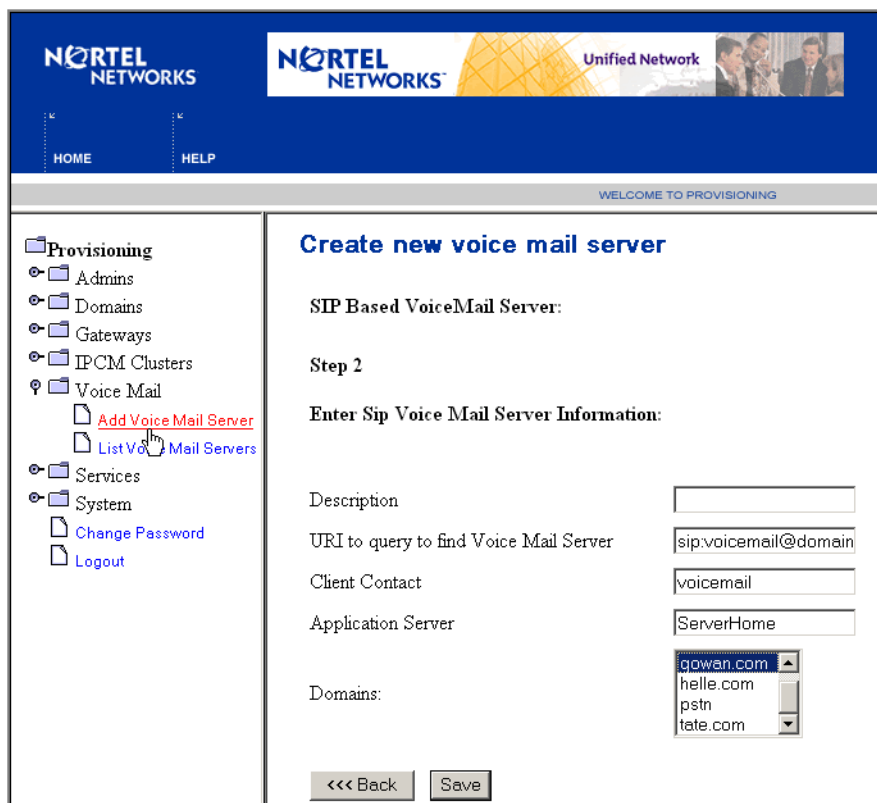
- 1 Click the **Voice Mail > Add Voice Mail Server** menu option. The Create new voice mail server window appears (Figure 30).

Figure 30 Create new voice mail server window (SIP) - step



- 2 Choose **SIP-Based Server** from the drop down and click **Next**. The second page of the Create new voice mail server appears (Figure 31).

Figure 31 Create new voice mail server window (SIP) - step 2



3 Enter information about the SIP-based voice mail server in the parameter fields. The following table (Table 7) lists the SIP-based voice mail parameters.

Table 7 SIP-based voice mail parameters

| Parameter | Type | Range | Default | Description |
|----------------------------------------|-------------|----------------------|---------|------------------------------------------------------------------------------------------------------------------------|
| Description | String(100) | Up to 100 characters | None | The description for the server that identifies the SIP-based voice mail server. |
| URI to query to find Voice Mail Server | String(100) | Up to 100 characters | None | The SIP address of the user that the SIP Application Module will look up to find the address of the voice mail server. |

Table 7 SIP-based voice mail parameters

| Parameter | Type | Range | Default | Description |
|--------------------|-----------------------|----------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Client contact | String(100) | Up to 100 characters | Voice mail | The field used to route voice mail. In most cases, you will use the default (voice mail). If you require a different client contact, contact your Nortel Networks support representative. |
| Application Server | String(60) | Up to 60 characters | ServerHome. With this default, each SIP Application Module will ignore this field and forward SIP requests to the voice mail server instead of forwarding them to a "home" Application Server. | Specifies the SIP Application Module will host the voice mail server being provisioned. This is used in an N+M environment. All SIP Application Modules that are not the home SIP Application Module will forward requests destined to voice mail to the address specified in this field. |
| Domains | String(60) characters | Up to 60 characters | None | The domains that will use this voice mail server. Use the Ctrl-click key to select multiple domains to use the same voice mail server. |

- 4 Click **Save**. The SIP-based voice mail server you added now appears in a list of voice mail servers (Figure 32).

Figure 32 Voice Mail Servers window (SIP)

| Voice Mail Servers | | | | |
|--------------------|----------------|--------------------------|------|--------|
| Description | Client Contact | Host | Type | Delete |
| VMS1 | voicemail | sip.voicemail@domain.com | SIP | Delete |

Adding a phantom voice mail server user

After you add a voice mail server, you then need to create a phantom user named voicemail for the domain that will use the SIP-based voice mail server.

To create a phantom user for the SIP-based voice mail server

- 1 Click the **Domains** menu option.

- 2 Under the domain where you want to add a phantom user, click the **Add User** menu option.
- 3 In the Add User form, enter the text "voice mail" for the User Name, Last Name, First Name, fields. In the password field, enter any password.
- 4 Click **Save**. The User details screen displays a message that the user was added successfully.
- 5 At the top of the window, click the **Routes** link to specify the route to the voice mail server. Enter the SIP address that the voice mail server is expecting, along with the domain and IP address (Figure 33).

Figure 33 User routes window

The screenshot shows a web interface for managing user routes. At the top, there are navigation tabs: 'Main', '2004 properties', 'Routes', and 'Customize service package'. Below the tabs is the title 'User routes'. A text input field contains the value 'sip:voicemail@comverse.com:maddr=47.104.12.24'. To the right of this field is an orange 'ADD' button. Below the input field is a 'Selected:' label and an empty list box. To the right of the list box are orange 'UP' and 'DOWN' buttons. At the bottom of the list box are orange 'REMOVE' and 'REMOVE ALL' buttons. A grey 'Save' button is centered at the bottom of the window.

- 6 Click **Add**.
- 7 Click **Save**.



Note: For CAS/CallPilot, it is recommended that you do not create a phantom user.

Provisioning a Trunk-based voice mail server

Provisioning a Trunk-based voice mail server allows the administrator to configure the SIP Application Module so that it can communicate with a voice mail server using a PRI or CAS gateway.

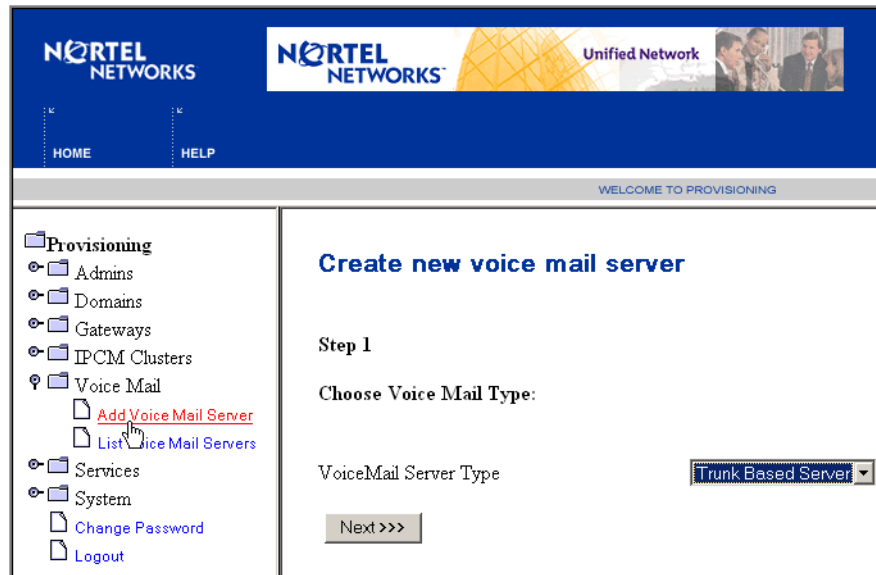
The SIP Application Module is given information on what to query to find the correct gateway to send SIP calls to the voice mail server when calls are routed to voice mail. An SMDI TELNET session is configured to allow the SIP Application Module to send call setup information using a SMDI Call Detail Message to receive SMDI MWI messages.

Adding a Trunk-based voice mail server

To add a Trunk-based voice mail server

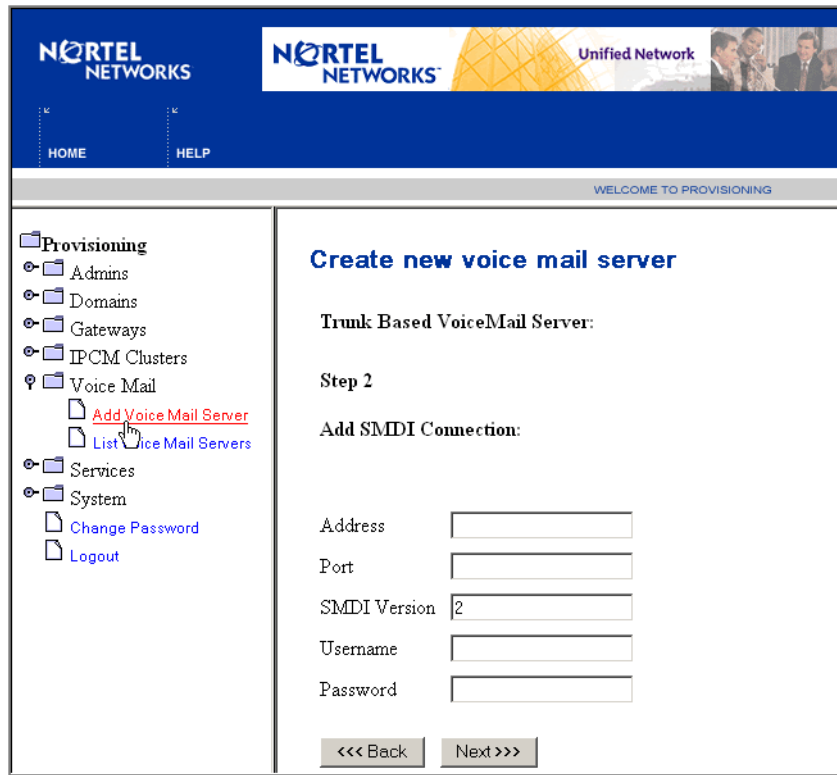
- 1 Click the **Voice Mail > Add Voice Mail Server** menu option. The Create new voice mail server window appears (Figure 34).

Figure 34 Create new voice mail server window (Trunk-based) - step 1



- 2 Choose **Trunk-Based Server** from the drop down and click **Next**. The second page of the Create new voice mail server appears (Figure 35).

Figure 35 Create new voice mail server window (Trunk-based) - step 2



- 3 Enter SMDI parameter information for the Trunk-based voice mail server. The following table (Table 8) lists the SMDI parameters.

Table 8 SMDI parameters

| Parameter | Type | Range | Default | Description |
|--------------|-------------|----------------------|---------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Address | String(100) | Up to 100 characters | N/A | The IP address of the terminal (SMDI) server. |
| Port | Integer | None | None | The port for the SMDI server to TELNET to. |
| SMDI Version | Float | None | None | The SMDI version. There are only 2 versions of the SMDI protocol (Version 1 and Version 2). Version 2 is the latest draft and is supported by most voice mail servers. |
| Username | String(60) | Up to 60 characters | N/A | The TELNET username for logging in to the SMDI TELNET server. Used by the SIP Application Module to login in to the terminal server. |
| Password | String(20) | Up to 20 characters | N/A | The TELNET password for logging in to the SMDI TELNET server. |

- 4 Click **Next** to enter information about the gateway connection. The Create new voice mail server window appears ([Figure 36](#)).

Figure 36 Create new voice mail server window (Trunk-based) - step 3

NORTEL NETWORKS Unified Network

HOME HELP

WELCOME TO PROVISIONING

Provisioning

- Admins
- Domains
- Gateways
- IPCM Clusters
- Voice Mail
 - Add Voice Mail Server**
 - List Voice Mail Servers
- Services
- System
 - Change Password
 - Logout

Create new voice mail server

Add Trunk VoiceMail Server:

Step 2

Enter Trunk Voice Mail Server Information:

Description:

URI to Query to find Gateway:

Type of Trunk:

Client Contact:

Application Server:

Domains:

<<< Back Save

- 5 Enter information about the Trunk-based voice mail server gateway connection in the parameter fields. The following table (Table 9) lists the gateway connection parameters.

Table 9 PRI Gateway connection parameters

| Parameter | Type | Range | Default | Description |
|------------------------------|-------------|----------------------|------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Description | String(100) | Up to 100 characters | None | The description for the server that identifies the Trunk-based voice mail server. |
| URI to query to find gateway | String(100) | Up to 100 characters | None | <p>For PRI Gateway: The address (in either 10 digit phone number and domain name format or in SIP:voice mail@nortelnetworks.com format) of the machine that will host this voice mail server. The SIP Application Module performs translations on the phone number and domain to understand which gateway to use. You can change the translations to use a different gateway, and do not have to change this field.</p> <p>For CAS Gateway (CallPilot): The SIP address for the CAS Gateway that routes to the CallPilot. The address is made up of a DN that the gateway routes to the CallPilot, and an maddr specification for the CAS Gateway. (For example, sip:66250@nortelnetworks.com;maddr=46.10.11.219).</p> |
| Type of Trunk | Drop List | NA | PRI_TRUNK | A specification of whether the voice mail Gateway is PRI or CAS. |
| Client contact | String(100) | Up to 100 characters | Voice mail | The field used to route voice mail. In most cases, you will use the default (voice mail). If you require a different client contact, contact your Nortel Networks support representative. |

Table 9 PRI Gateway connection parameters

| Parameter | Type | Range | Default | Description |
|--------------------|------------|---------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Application Server | String(60) | Up to 60 characters | ServerHome. With this default, each SIP Application Module will ignore this field and forward SIP requests to the voice mail server instead of forwarding them to a "home" Application Server. | Specifies the SIP Application Module will host the voice mail server being provisioned. This is used in an N+M environment. All SIP Application Modules that are not the home SIP Application Module will forward requests destined to voice mail to the address specified in this field. |
| Domain name | String(60) | Up to 60 characters | None | The domains that will use this Trunk-based voice mail server. Use the Ctrl-click key to select multiple domains to use the same voice mail server. |

- Click **Save**. The Trunk-based voice mail server you added now appears in the list of voice mail servers (Figure 37).

Figure 37 Voice Mail Servers window (Trunk-based)

| Voice Mail Servers | | | | |
|--------------------|----------------|---------------------------|-------|--------|
| Description | Client Contact | Host | Type | Delete |
| VMS1 | voicemail | sip.voicemail@domain.com | SIP | Delete |
| Trunk VMS | voicemail | sip:1234567890@domain.com | Trunk | Delete |

Provisioning a Line-based voice mail server

Provisioning a Line-based voice mail server allows the administrator to configure the application so that it can communicate with a voice mail server using a Lines Gateway. The SIP Application Module is given information on which Lines Gateway to use and which line numbers to use for sending calls to the gateway when routing a call to voice mail.

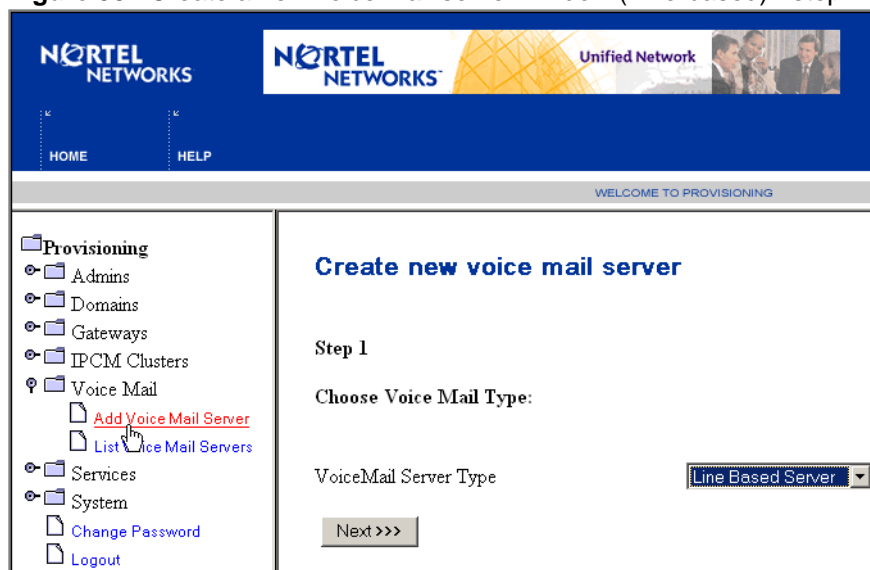
In addition, an SMDI TELNET session is configured to allow the SIP Application Module to directly communicate call setup information to the voice mail server. The SMDI link is also used by the SIP Application Module to receive SMDI MWI messages.

Adding a Line-based voice mail server

To add a Line-based voice mail server

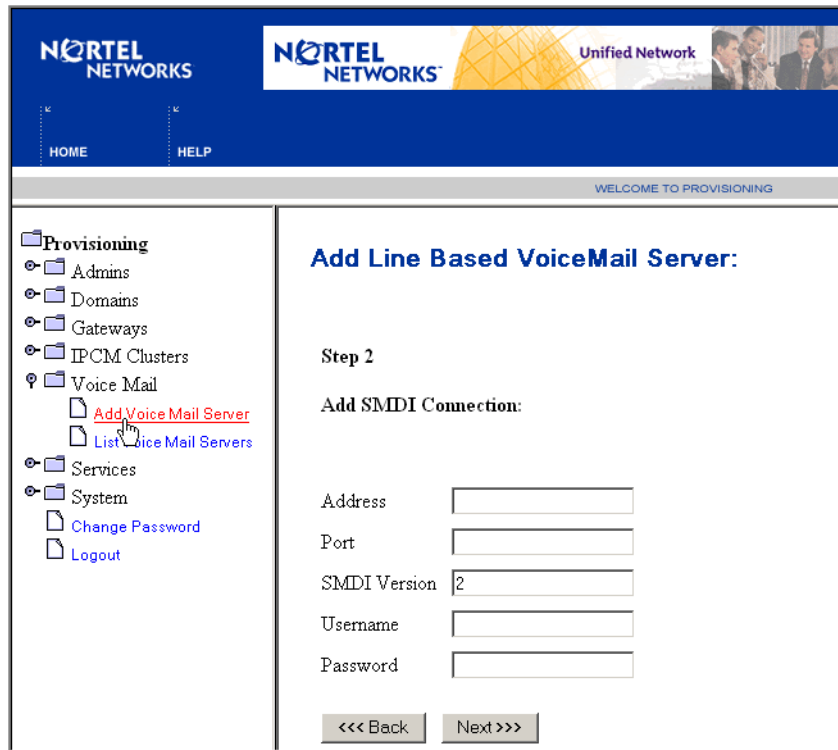
- 1 Click the **Voice Mail > Add Voicemail Server** menu option. The Create new voice mail server window appears (Figure 38).

Figure 38 Create a new voice mail server window (Line-based) - step 1



- 2 Choose **Line-Based Server** from the drop down and click **Next**. The second window of the Create new voice mail server appears (Figure 39).

Figure 39 Create new voice mail server window (Line-based) - step 2



- 3 Enter SMDI parameter information about the Line-based voice mail server. The following table (Table 10) lists the Line-based voice mail server SMDI parameters.

Table 10 Line-based voice mail server SMDI parameters

| Parameter | Type | Range | Default | Description |
|--------------|-------------|----------------------|---------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Address | String(100) | Up to 100 characters | N/A | The IP address of the terminal (SMDI) server. |
| Port | Integer | None | None | The port for the SMDI server to TELNET to. |
| SMDI Version | Float | None | None | The SMDI version. There are only 2 versions of the SMDI protocol: Version 1 and Version 2. (Version 2 is the latest draft and is supported by most voice mail servers.) |
| Username | String(60) | Up to 60 characters | N/A | The TELNET username for logging in to the SMDI TELNET server. |
| Password | String(20) | Up to 20 characters | N/A | The TELNET password for logging in to the SMDI TELNET server. |

- 4 Click **Next** to enter general information about the Line-based server ([Figure 39](#)).

Figure 40 Create new voice mail server window (Line-based) - step 3



- 5 Enter general information about the Line-based voice mail server in the parameter fields. The following table lists the Line-based parameter information.

Table 11 Line-based voice mail server general parameter information

| Parameter | Type | Range | Default | Description |
|-----------------------|-------------|----------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Description | String(100) | Up to 100 characters | None | The description for the server that identifies the Line-based voice mail server. |
| Client contact | String(100) | Up to 100 characters | Voice mail | The field used to route voice mail. In most cases, you will use the default (voice mail). If you require a different client contact, contact your Nortel Networks support representative. |
| Application Server | String(60) | Up to 60 characters | ServerHome. With this default, each SIP Application Module will ignore this field and forward SIP requests to the voice mail server instead of forwarding them to a "home" Application Server. | Specifies the SIP Application Module will host the voice mail server being provisioned. This is used in an N+M environment. All SIP Application Modules that are not the home SIP Application Module will forward requests destined to voice mail to the address specified in this field. |
| Maximum Call Duration | Digit | 1-99 | None | The period of time that the SIP Application Module allows a caller to stay connected to the Line-based voice mail server when someone is trying to leave a voice mail message. |
| Domain name | String(60) | Up to 60 characters | None | The domains that will use this voice mail server. Use the Ctrl-click key to select multiple domains to use the same voice mail server. |

- 6 Click **Next** to provision the lines for the gateway (Figure 41).

Figure 41 Create new voice mail server window (Line-based) - step 4

Add Line Based VoiceMail Server:

Step 4

Set Line Data:

Lines Previously Provisioned

None

Gateway

Message Desk Line Position Number

<> Lines Added

| Line | Message Desk | Position Number |
|------|--------------|-----------------|
| 33 | 333 | 1234 |
| 222 | 12 | 222 |

- 7** Enter the Line-based parameter information to provision the lines for the gateway. The following table (Table 12) lists the Line-based voice mail server parameters.

Table 12 Line-based voice mail server parameters

| Parameter | Type | Range | Default | Description |
|-----------------|----------|------------------|--------------------------------------------------|---------------------------------------------------------------------------------------------|
| Gateway | String | 1-100 characters | domain.com; norteltrkgrp=1;ma addr=0.0.0.0 | The trunk group, domain name and IP address of the gateway. |
| Line | Digit | 0-38 digits | None | The 10-digit telephone number that identifies the line from the gateway's point of view. |
| Message Desk | 3 digits | 000-999 | None | A 3-digit number used by the SMDI protocol to determine where to store an incoming message. |
| Position Number | 4 digits | 0000-9999 | None | A 4-digit number used to determine where to put an incoming message. |

- 8 Click **Add Line**. To add multiple lines, continue to enter the gateway parameter information and click **Add Line**.
- 9 When you are finished adding lines for this gateway, click **Set New Line** to have more than one gateway with lines going into the same voice mail server.
- 10 Click **Save**. The Line-based voice mail server you added now appears in the list of voice mail servers (Figure 42).

Figure 42 Voice Mail Servers window (Line-based)

| Voice Mail Servers | | | | |
|--------------------|----------------|---------------------------|-------|--------|
| Description | Client Contact | Host | Type | Delete |
| LB VMS | voicemail | UnAvailable | Line | Delete |
| VMS1 | voicemail | sip.voicemail@domain.com | SIP | Delete |
| Trunk VMS | voicemail | sip:1234567890@domain.com | Trunk | Delete |

Assigning a user to a provisioned voice mail server

In order for a user to have voice mail, you must associate a voice mail server and mail box for that user.

To associate a user with a voice mail server that has been previously provisioned

- 1 Click the **Domains** menu option
- 2 Click the domain for the user.
- 3 Click the **Users > List Users** menu option.
- 4 Click **Search** at the bottom of the screen to display a list of users in the domain.
- 5 Click the **Details** link for the user you want to associate with a voice mail server.
- 6 Scroll down the screen to the voice mail server field. Select a voice mail server to use when sending calls for this user to voice mail from the drop down list of provisioned voice mail servers for this domain.
- 7 Enter the information for the user's voice mail box in the Voice mail field.
- 8 Click **Save**.

Listing/modifying/deleting a voice mail server

To view details of a provisioned SIP, Trunk, or Line-based voice mail server

- 1 Click the **Voice Mail > List Voice Mail Servers** menu option. The Voice Mail Server window appears (Figure 43).

Figure 43 List of voice mail servers

| Voice Mail Servers | | | | |
|--------------------|----------------|---------------------------|-------|------------------------|
| Description | Client Contact | Host | Type | Delete |
| LB VMS | voicemail | UnAvailable | Line | Delete |
| VMS1 | voicemail | sip.voicemail@domain.com | SIP | Delete |
| Trunk VMS | voicemail | sip.1234567890@domain.com | Trunk | Delete |

- 2 To view the details of a voice mail server, click the link of the voice mail server in the Description column. The Create new voice mail server form appears where you can modify information.
- 3 Click **Save** to save your changes or **Back** to return to the list of voice mail servers.

- 4 To delete a voice mail server, click the **Delete** link for the voice mail server you wish to delete. At the confirmation dialog box, click **OK** to delete the voice mail server, or **Cancel** to cancel the operation.

Using CPL scripts with voice mail

A user will only have voice mail service if there is a Call Processing Language (CPL) script that specifies a route to *sip:voicemail@domain.com* (usually the last route).

To specify a route, you need to provision the following voice mail parameters in the SIP Provisioning Client:

| Voice mail parameter | Description |
|------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Client Contact | Must match the username in the CPL script and should always be set to voicemail (unless you are working with Nortel Networks technical support). If this parameter is set to something else, properties in the IPCM and Web Client will require changes. |
| URI to Query to Find Gateway | Point to a static route, a user with a static route, or have a number that can be translated by the Telephony routing to go to the correct gateway. |

To create a CPL script that has *sip:voicemail@domain.com* as the last route

- 1 Log in to the SIP Personal Agent.
- 2 Click the **Call Manager** icon.
- 3 In the Default Behavior window, select the **Try to Find Me** option.
- 4 Click **Save**



Tip: Unless a user logs into the SIP Personal Agent and clicks **Save** in the Call Manager window, they will not have a CPL Script and thus calls to them will never route to voice mail as a final route.

Chapter 6

Defining and Assigning Services



Note: You must have Services provisioning rights to perform the tasks in this section.

Topics in this section:

- [“Defining service parameters” on page 71](#)
- [“Assigning services to a domain” on page 75](#)

Defining service parameters

The SIP Provisioning Client allows you to define and assign services to domains. You can assign all or a subset of the pre-defined services to a domain.

When you define the service parameters for service package, you are essentially defining its contents. Services are set up much like convenience packages for the domain. The service provider then has the flexibility to set up services and create a standard set of services for their customers according to their own specifications.

To define service parameters

- 1 Click the **Services > Define Service Parameters** menu option.

The list of available services window appears ([Figure 44](#)).

Figure 44 List of available services window

The screenshot shows the 'List of available services' window in the Nortel Networks Provisioning interface. The window is titled 'List of available services' and contains a table of service parameters and their values. The parameters are grouped into sections: 'Enable Voicemail', 'Enable Audio Conference', 'Enable Presence', 'Enable Video', 'Simultaneous Network Call Restriction', 'Enable Advanced Screening', and 'Enable QoS'. Each parameter has a 'Values' field with a dropdown menu and an '[Edit]' link.

| Service(s) | Parameter | Values | [Edit] |
|---------------------------------------|-------------------------------------------|--------|--------|
| Enable Voicemail | Maximum Number of Ports | 4 | [Edit] |
| | Enable Audio Conference | | |
| Enable Presence | Maximum Number of Subscriptions Generated | 10 | [Edit] |
| | Maximum size of client buddy list | 10 | [Edit] |
| | Enable Video | | |
| Simultaneous Network Call Restriction | Maximum Number of Ringlist entries | 3 | [Edit] |
| Enable Advanced Screening | | | |
| Enable QoS | QoS DiffServ Code for Signalling | 8 | [Edit] |
| | QoS DiffServ Code for Audio | 10 | [Edit] |
| | QoS DiffServ Code for Video | 10 | [Edit] |
| | QoS 802.1p Service Priority | 1 | [Edit] |
| Blended User | | | |

- From the list of available services window, select a value for a service parameter from the drop down list. The following table (Table 13) lists the service parameters and options.

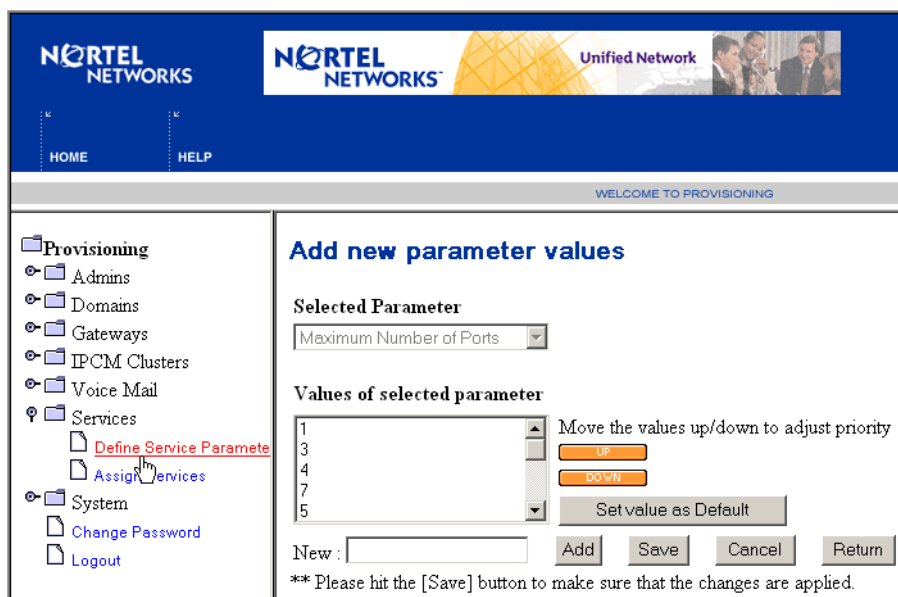
Table 13 Services parameters and options

| Service Option | Parameter | Description | Client enforced | Server enforced |
|---------------------------------|-------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|-----------------|-------------------------------------------------------|
| Enable/disable Voicemail | None | Retrieve voice mail or for other subscribers, to leave a voice mail message. Voice mail is retrieved from a network server. The user is aware of deposited voice mail since their client's MWI (message waiting indicator) is enabled. | Yes. | Optional. Enabled through the SIP Application Module. |
| Enable/disable Audio Conference | Maximum number of ports | Initiate an audio conference call utilizing a SIP audio server. The number of users in the conference call is limited by the maximum number of ports assigned. Values beyond above thirty two are not allowed. | Yes. | Optional. Enabled through the SIP Application Module. |
| Enable/disable Presence | Maximum size of client list of buddies | Specifies the number of people in a list of buddies. This is the number of people that a user is subscribing to in order to view their presence. Although a user can be logged into numerous devices, the list of buddies is a shared resource. For example, if the max list of buddies size is set to 10, there can be 10 buddies on each of the client devices (the same 10 buddies, not 10 per device). The client has the option of sending any of the buddies. | Yes | |
| | Maximum number of subscriptions generated | Allows a user to be logged into multiple devices (i2004 Internet Telephone, SIP Multimedia PC Client, SIP Multimedia Web Client). Each such device performs messaging to determine the presence (online status) of each of the buddies in the list. There is a need for limiting the number of devices and buddies to limit resource contention issues. The maximum number of presence subscriptions is defined by the following formula: maximum number of subscriptions generated = (maximum # of clients) * (maximum number of allowed buddies). | | Yes. Mandatory. |

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| Service Option | Parameter | Description | Client enforced | Server enforced |
|---------------------------------------|------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|-----------------|-----------------|
| Enable/disable Video | None | Use of point to point video service between web clients and PC clients. | Yes. | |
| Simultaneous Network Call Restriction | Maximum number of ringlist entries | Specifies the maximum number of users that are defined in a route in the Call Manager portion of the SIP Personal Agent. The users in the ringlist are simultaneously rung. | Yes. | |
| Enable/disable Advanced Screening | - | Enables the ability to perform address screening (based on the user's current address book entries) and time screening (time of day, day of week) in the Call Manager area of the SIP Personal Agent. | Yes. | |
| Enable/disable QoS | QoS DiffServ Code for Signalling | Specifies quality of service for client signaling, audio and video (up to 64). | Yes. | |
| | QoS DiffServ Code for Audio | | | |
| | QoS DiffServ Code for Video | | | |
| | QoS 802.1p Service Priority | | | |
| Enable/disable Blended User | None | Enables the use of a traditional phone for voice calls and the SIP Multimedia PC for multimedia communication and collaboration. | Yes | |

- 3** To create new values for the service parameter, click **[Edit]**.
The Add new parameter values window appears ([Figure 45](#)).

Figure 45 Add new parameter values window

- 4 For the selected parameter, select a value from the list, using the **UP** and **DOWN** buttons to set the priority of the value.
- 5 To add a new value, click **Add**. The new value appears in the list of parameter values.
- 6 Click **Set as Default** to set the value in the list as the default value for the service parameter.
- 7 Click **Save** to save your service parameter changes.

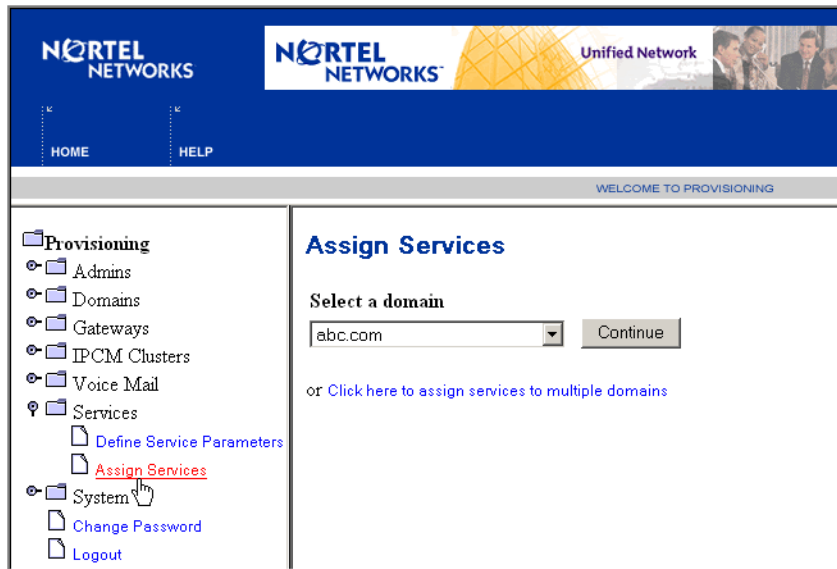
Assigning services to a domain

Once you add a value for a service parameter, you can select a domain to assign the services to.

To assign services to a domain

- 1 Click the **Services > Assign Services** menu option. The Assign Services window appears (Figure 46).

Figure 46 Assign services window



- 2 Select a domain from the drop down list and click **Continue**. To assign services to more than one domain, click the link to assign services to multiple domains in the Assign Services window. The Assign services to domains window appears (Figure 47).

Figure 47 Assign services to a domain window

Assign services to domains

Select Domain(s)
abc.com

Select Service(s)

Choose All Services

Enable Voicemail

Enable Audio Conference
Maximum Number of Ports: 4

Enable Presence
Maximum Number of Subscriptions Generated: 10
Maximum size of client buddy list: 10
Report when inactive:
Inactivity Timer (in minutes):
Report when on the phone:

Enable Video

Simultaneous Network Call Restriction
Maximum Number of Ringlist entries: 3

Enable Advanced Screening

Enable QoS
QoS DiffServ Code for Signalling: 8
QoS DiffServ Code for Audio: 10
QoS DiffServ Code for Video: 10
QoS 802.1p Service Priority: 1

Blended User
Preferred Audio Device

Save Cancel

- 3 Select the services and service parameter values you want to assign to the selected domain or multiple domains (Use the **Ctrl-click** key to choose multiple domains.) To enable all services, check the **Enable all Services** checkbox.
- 4 Click **Save**. A confirmation message displays that your service was assigned successfully.

Once services have been defined and assigned to domains, you can list all the services that are available for a domain (or multiple domains) and create and assign packages accordingly. For a subdomain, services are assigned from the root domain.

Chapter 7

Assigning Services and Creating Service Packages

Topics in this section:

- [“Understanding service packages” on page 79](#)
- [“Listing available services” on page 79](#)
- [“Assigning services to a service package” on page 83](#)
- [“Creating a service package” on page 86](#)
- [“Listing/modifying/deleting service packages” on page 87](#)
- [“Assigning service packages to domains and subdomains” on page 89](#)
- [“Upgrading and downgrading service packages” on page 90](#)

Understanding service packages

Once services are provisioned for a domain, you then have the flexibility to create service packages for a domain or subdomain based on a set of predefined services.

Each service package is a collection of predefined services. Including a service in a service package enables the service, or feature, for the domains associated with the package. You can assign packages that are currently created in the domain down to a subdomain.

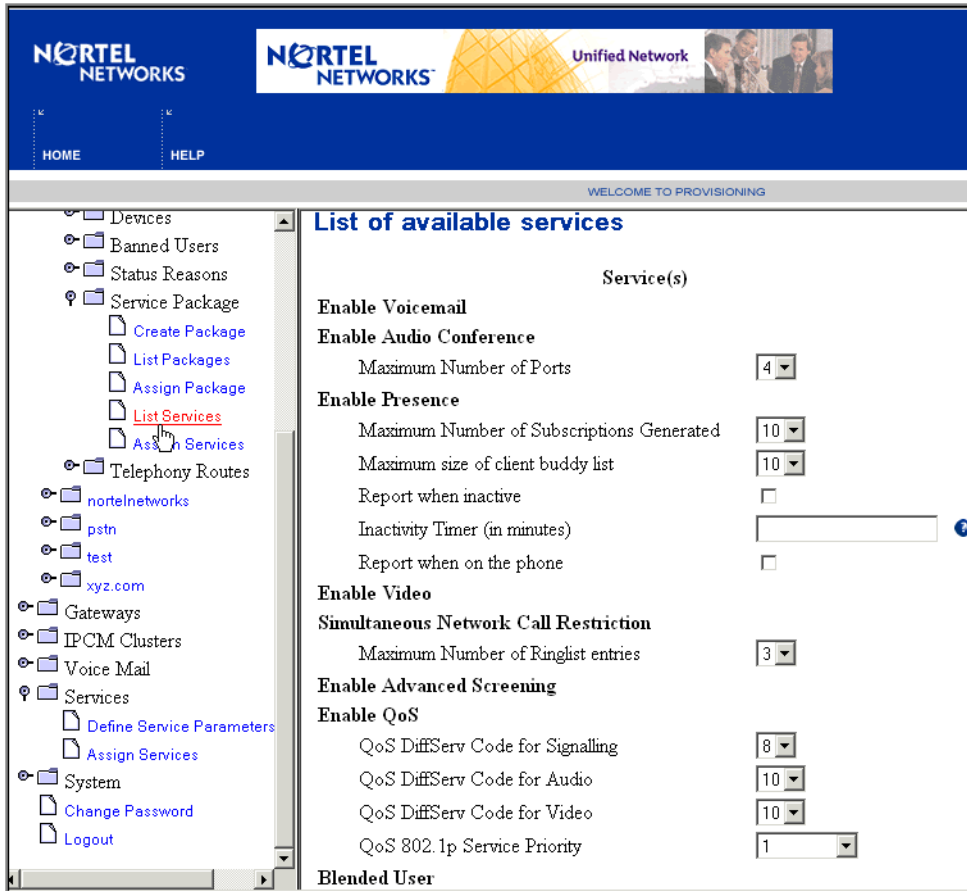
Listing available services

To view a list of all services and parameters available in a service package

- 1 Click the **Domains** menu option.

- Under the domain or subdomain for which you wish to list services, click the **Service Package > List Services** sub menu option. The List of available services window appears (Figure 48).

Figure 48 List of available services



- Review the list of services and service parameters. The following table (Table 14) lists service packages, service parameters, and the values available for each service parameter.

Table 14 Services parameters and options

| Service Option | Parameter | Description | Client enforced | Server enforced |
|---------------------------------|-------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|-----------------|-------------------------------------------------------|
| Enable/disable Voicemail | None | Retrieve voice mail or for others, to leave a voice mail message. Voice mail is retrieved from a network server. The user is aware of deposited voice mail since their client's MWI is enabled. | Yes. | Optional. Enabled through the SIP Application Module. |
| Enable/disable Audio Conference | Maximum number of ports | Initiate an audio conference call utilizing a SIP audio server. The number of users in the conference call is limited by the maximum number of ports assigned. Values beyond 32 is not allowed. | Yes. | Optional. Enabled through the SIP Application Module. |
| Enable/disable Presence | Maximum size of client list of buddies | Specifies the number of people in a list of buddies. This is the number of people that a user is subscribing to in order to view their presence. Although a user can be logged into numerous devices, the list of buddies is a shared resource. For example, if the max list of buddies size is set to 10, there can be 10 buddies on each of the client devices (the same 10 buddies, not 10 per device). The client has the option of sending any of the buddies. | Yes. Mandatory | No. |
| | Maximum number of subscriptions generated | Allows a user to be logged into multiple devices (i2004 Internet Telephone, SIP Multimedia PC Client, SIP Multimedia Web Client). Each such device performs messaging to determine the presence (online status) of each of the buddies in the tree. There is a need for limiting the number of devices and buddies to limit resource contention issues. The maximum number of presence subscriptions is defined by the following formula: maximum number of subscriptions generated = (maximum # of clients) * (maximum number of allowed buddies) | No. | Yes. Mandatory. |

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| Service Option | Parameter | Description | Client enforced | Server enforced |
|---------------------------------------|------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|-----------------|----------------------------------------------------------------------------------------------------------------------|
| | Report when inactive | A checkbox to report a user's inactivity (a user is inactive when they do not move their mouse or type on their keyboard within a configurable amount of time.) | Yes. | No. |
| | Inactivity Timer (in minutes) | If enabled, an inactivity timer (in minutes) that PC-based clients will observe before reporting "Inactive" to the Presence Module. Values range from 10 to 600, with 15 being recommended. If this service package parameter is enabled, then the user must explicitly enable the feature on the client itself. | Yes. | Yes/No (Enabled as a privilege, lower bound of what a user can select is determined in the SIP Provisioning Client.) |
| | Report when on the phone | If enabled, then every call made by subscribers can have "On the Phone" presence reported to the Presence Module. If this service package parameter is enabled, then the user must explicitly enable the feature on the client itself. | Yes. | Yes/No. (Enabled as a privilege in the SIP Provisioning Client.) |
| Enable/disable Video | None | Use of point to point video service between web clients and PC clients. | Yes. | |
| Simultaneous Network Call Restriction | Maximum number of ringlist entries | Specifies the maximum number of users that are defined in a route in the Call Manager portion of the SIP Personal Agent. The users in the ringlist are simultaneously rung. | Yes. | |
| Enable/disable Advanced Screening | - | Enables the ability to perform address screening (based on the user's current address book entries) and time screening (time of day, day of week) in the Call Manager portion of the SIP Personal Agent. | Yes. | |

| Service Option | Parameter | Description | Client enforced | Server enforced |
|--------------------|----------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|-----------------|-----------------|
| Enable/disable QoS | QoS DiffServ Code for Signalling | Specifies quality of service for client signaling, audio and video (up to 64). | Yes. | |
| | QoS DiffServ Code for Audio | | | |
| | QoS DiffServ Code for Video | | | |
| | QoS 802.1p Service Priority | | | |
| Blended user | Preferred Audio Device | If the Blended User service option is checked, the Preferred Audio Device is required so that the regular clients (SIP Multimedia PC Client, SIP Multimedia Web Client, i2004 Internet Telephone controlled via IPCM) can call a Blended PC Client by their SIP address (not just by the PSTN equivalence number). | | Yes. |

- 4 Select a Service Package sub menu option to continue setting up a service package.

Assigning services to a service package

The **Assign Services** menu option allows you to select services and service parameter values for a domain or subdomain (or multiple domains or subdomains).



Tip: Changes to service package services and parameter values will take effect immediately or upon the next login, depending on the client.

To assign services to a domain or subdomain

- 1 Click the **Domains** menu option.

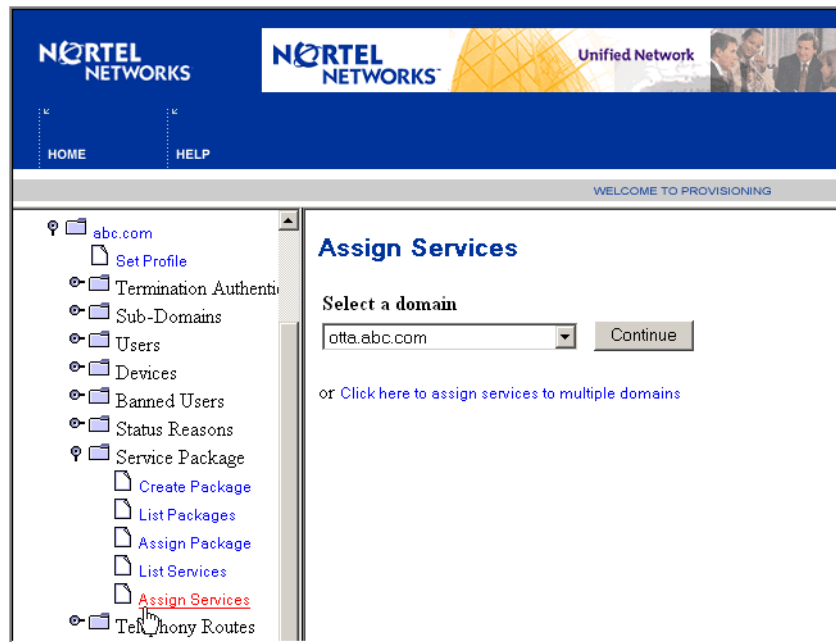
- 2 Click the domain or subdomain for which you wish to assign services.



Tip: When adding a package to a subdomain, you must first have defined a service package for the parent domain.

- 3 Click the **Service Package > Assign Services** sub menu option. The Assign Services window appears (Figure 49).

Figure 49 Assign services window



- 4 From the drop down list, select a domain to assign services and service parameters to and click **Continue**. The Assign services to domains window appears (Figure 50).
- 5 To assign services and service parameters to multiple domains, use the **Ctrl-click** key to select multiple domains from the drop down list and click the **Click here to assign services to multiple domains** link.

Figure 50 Assign services to domains window

Assign services to domains

Select Domain(s)
otta.abc.com

Select Service(s)

Choose All Services

Enable Voicemail

Enable Audio Conference
Maximum Number of Ports: 4

Enable Presence
Maximum Number of Subscriptions Generated: 10
Maximum size of client buddy list: 10
Report when inactive:
Inactivity Timer (in minutes): ⓘ
Report when on the phone:

Enable Video

Simultaneous Network Call Restriction
Maximum Number of Ringlist entries: 3

Enable Advanced Screening

Enable QoS
QoS DiffServ Code for Signalling: 8
QoS DiffServ Code for Audio: 10
QoS DiffServ Code for Video: 10
QoS 802.1p Service Priority: 1

Blended User

- 6 To assign services for the selected domain, subdomain, or multiple domains or subdomains, select the checkbox next to the service option (see Table 14 on page 81 for service and parameter descriptions).
- 7 To modify a service parameter, select a value from the drop down list.
- 8 To select all services, check the **Choose All Services** checkbox.
- 9 To select particular services for a domain, select the checkbox next to the service.
- 10 Click **Save** to assign the services.

Creating a service package

Domains, subdomains, and users are assigned service packages to control their access to network services.

To create a service package

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain for which you wish to assign services, click the **Service Package > Create Package** menu option. The Create new package window appears (Figure 51).

Figure 51 Create new package window

Create new package

Name of the Package *

Set package as default for domain

Select Service(s)

Choose All Services

Enable Voicemail

Enable Audio Conference
Maximum Number of Ports

Enable Presence
Maximum Number of Subscriptions Generated
Maximum size of client buddy list
Report when inactive
Inactivity Timer (in minutes)
Report when on the phone

Enable Video

Simultaneous Network Call Restriction
Maximum Number of Ringlist entries

Enable Advanced Screening

Enable QoS
QoS DiffServ Code for Signalling
QoS DiffServ Code for Audio
QoS DiffServ Code for Video
QoS 802.1p Service Priority

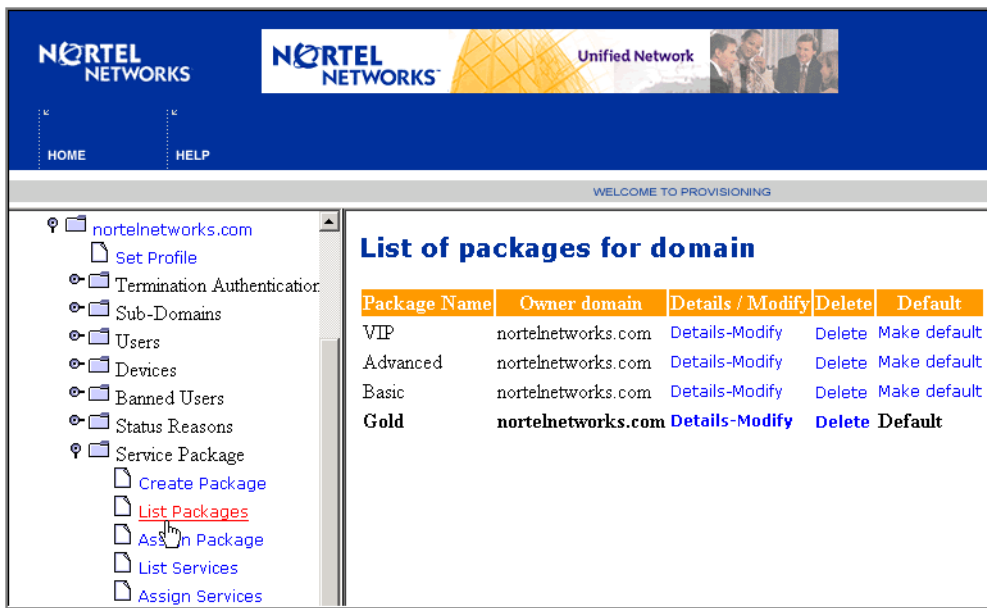
Blended User
Preferred Audio Device

- 3 Enter a name for the service package in the text box.
- 4 Select whether or not to make this package the default for the domain or subdomain. The default is NO.
- 5 Create the service package by selecting:
 - a the checkbox next to the service option to select services for the service package (see [Table 14 on page 81](#) for service and parameter descriptions).
 - b a service parameter value from the drop down list.
 - c the Choose **All Services** checkbox to select all services.
 - d the checkbox next to the service to select particular services.
- 6 Click **Save** to create the service package. A message displays stating that the package was successfully created.

Listing/modifying/deleting service packages

To view a list of all service packages provisioned in a domain or subdomain

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain for which you wish to view the service package details, click the **Service Package > List Packages** menu option. The Details of the package window appears ([Figure 52](#)).

Figure 52 List of packages domain window

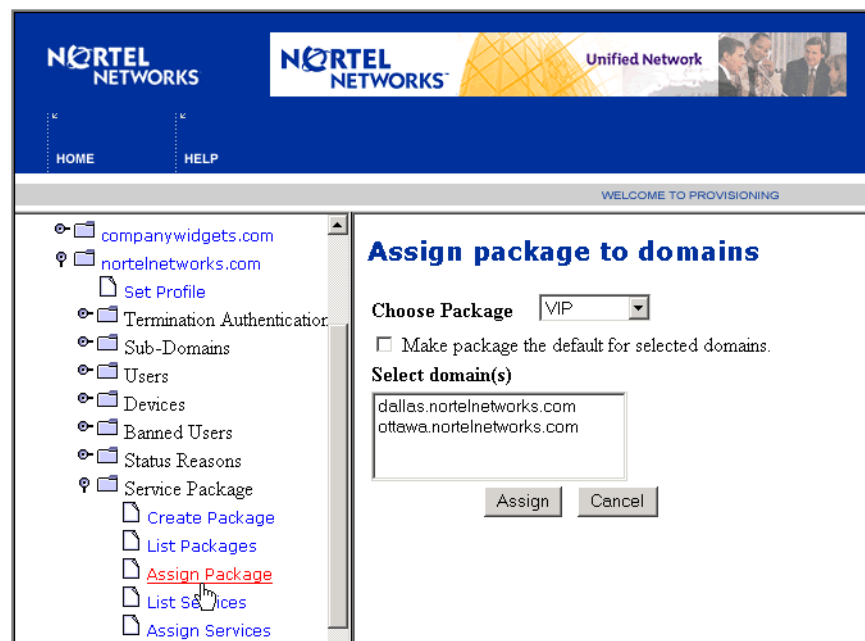
- 3 From the list of packages
 - a Click the **Details-Modify** link to view service package details and modify service parameters and values (see Table 14 on page 81 for service and parameter descriptions). If a service package is modified, then all users who currently have this service package are affected. If service attributes are modified for a domain, then existing packages utilizing this service remain as is. Any new packages created for this domain utilize the new parameters.
 - b Click the **Delete** link to delete the service package from the domain. A service package can only be deleted if no users are associated with the service package.
 - c Click the **Make default** link to make the selected package the default package for the domain.
- 4 Click **Save** from the Details screen to save your changes, or click a menu option on the left side of your screen to continue with another operation.

Assigning service packages to domains and subdomains

To assign a service package to a domain or subdomain

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain for which you wish to assign a service package, click the **Service Package > Assign Packages** sub menu option. The Assign package to domains window appears (Figure 53).

Figure 53 Assign package to domains window



- 3 From the drop down list, select a service package.
- 4 To make the selected package the default, check the **Make package the default for selected domains** checkbox.
- 5 Select the domain from the list of domains in the display area. Use the **Ctrl-click** key to select multiple domains.
- 6 Click **Assign**.

Upgrading and downgrading service packages

Upgrading a service package is allowed (as in the case when you want to add services) as long as the service is available at the domain. Upgrading a service package to update parameter values is also allowed, but only to the limit assigned to the domain.

Downgrading a service package (for services with parameters) is allowed only to the upper limit of the parameter values for the user and subdomains.

Chapter 8

User Management

Topics in this section:

- [“Managing user functions” on page 91](#)
- [“Adding a user to a domain or subdomain” on page 92](#)
- [“Adding a blended user” on page 95](#)
- [“Adding i2004 Internet Telephone capability for a user” on page 99](#)
- [“Adding a conference server user and route” on page 101](#)
- [“Customizing service packages” on page 104](#)
- [“Listing/modifying/deleting users” on page 105](#)
- [“Listing/modifying/deleting aliases” on page 107](#)
- [“Listing/modifying/deleting blended aliases” on page 109](#)

Managing user functions

Once a domain or subdomain is provisioned, an administrator is responsible for managing the following user functions:

- adding, viewing, modifying, or deleting users (including blended users) from the domain or subdomain
- adding, viewing, modifying, or deleting i2004 Internet Telephone device properties
- assigning users to i2004 Internet Telephones
- customizing service packages (for example, to add a blended user if available as a service package option)
- viewing, modifying, or deleting aliases and/or blended aliases

Adding a user to a domain or subdomain

To add a user to a domain or subdomain

- 1 Click the **Domains** option.
- 2 Under the domain or subdomain, click the **User > Add User** menu option. The Add new user window appears (Figure 54).

Figure 54 Add new user

Add new user

User Name: * ?

First Name: ?

Last Name: ?

Password: *

Confirm Password: * ?

Service package:

Aliases:

Blended Aliases:

Status Reason:

email:

Business Phone: ?

Home Phone: ?

Cell Phone: ?

Pager: ?

Fax: ?

Voicemail Server:

Private Charge ID:

Public Charge ID:

Voicemail:

Class of Service:

Time Zone:

Locale:

- 3 Enter information about the new user in the parameter fields as described in the following table (Table 15).

Table 15 Add user parameter descriptions

| Parameter | Type | Range | Default | Description |
|------------------|----------------------------|----------------------|---------|----------------------------------------------------------------------------------------------------------------------------|
| User Name | String (60) | Up to 60 characters | N/A | Username of this user. Use lowercase. Does not have to be the same name as the First Name. |
| First Name | String (30) | Up to 30 characters | N/A | First name of the user. Case sensitive. |
| Last Name | String (30) | Up to 30 characters | N/A | Last name of this user. Case sensitive. |
| Password | String (200) | Up to 200 characters | N/A | Password of this user. |
| Confirm Password | String (200) | Up to 200 characters | N/A | Password of this user. |
| Service package | Select from drop down list | - | N/A | The list of service packages assigned to the user's domain or subdomain. |
| Aliases | String (40) | - | N/A | The list of aliases for this user. Aliases associate a PSTN phone number to a username. |
| Blended Aliases | String | - | N/A | Alias for a blended user, a person who uses the traditional phone but is equipped with multimedia services at the desktop. |
| Status Reason | Select from drop down list | N/A | N/A | The status value of this user. |
| email | String (60) | Up to 60 characters | N/A | Email address of this user. |
| Business Phone | String (60) | Up to 60 characters | N/A | Business phone number. |
| Home Phone | String (30) | Up to 30 characters | N/A | Home phone number. |
| Cell Phone | String (30) | Up to 30 characters | N/A | Cell phone number. |
| Pager | String (30) | Up to 30 characters | N/A | Pager number. |
| Fax | String (30) | Up to 30 characters | N/A | Fax number of this user. |

Table 15 Add user parameter descriptions

| Parameter | Type | Range | Default | Description |
|-------------------|----------------------------|---------------------|---------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Voicemail Server | Select from drop down list | - | None selected | The voice mail server available to this user. |
| Private Charge ID | String (30) | Up to 30 characters | N/A | The Private Charge ID of this user. |
| Public Charge ID | String (30) | Up to 30 characters | N/A | The 10-digit national dial plan number associated with the subscriber. This is required for the interworking with Public Switched Telephone Network (PSTN) switches in the Time Division Multiplex (TDM) network. |
| Voicemail | String (30) | Up to 30 characters | N/A | The voice mail address of this user. |
| Class of Service | Select from drop down list | Up to 30 characters | None selected | The name of the class of service assigned to the domain and user. This is displayed to the administrator when modifying or assigning the COS to a domain/subdomain/user. |
| Time Zone | Select from drop down list | - | N/A | The time zone that this user belongs to. |
| Locale | Select from drop down list | - | N/A | The location of this user. |

4 Click **Save**.

Adding a blended user

Setting up a user as a blended user allows the end user to use their PCs for the multimedia portion of their communication, while using their existing telephony system for voice.



Note: When a user receives their service package (at SIP Multimedia PC Client startup and on any changes to the subscriber data made by the administrator), and that service package contains Blended User, then that PC Client becomes a SIP Multimedia Blended PC Client.

To add a user as a blended user to a domain or subdomain

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain where you wish to set up a blended user, click the **User > Add User** menu option. The Add new user window appears.

Figure 55 Add new user - blended

Add new user

User Name: *

First Name:

Last Name:

Password: *

Confirm Password: *

Service package:

Aliases:

Blended Aliases:

Status Reason:

email:

Business Phone:

Home Phone:

Cell Phone:

Pager:

Fax:

Voicemail Server:

Private Charge ID:

Public Charge ID:

Voicemail:

Class of Service:

Time Zone:

Locale:

- 3 Complete the fields on the Add new user form.
- 4 Add a blended alias for the user. This alias is used to identify calls that come from the SimRing feature that is assigned to the user on an existing switching

system. It must match the called party number that will be sent to the SIP Application Module on these SimRing calls.



Note: Blended aliases are per subscriber data and are entered as list of telephony numbers into the Blended Aliases text box. The blended aliases must not match any of the normal aliases entered into the Aliases text box on the same screen.

- 5 Add an alias (not blended alias) that matches the calling party number that will be sent to the SIP Application Module when the user calls another blended user. This is the calling party number that is sent to the SIP Application Module on the SimRing call.
- 6 If calls from users (non-blended) to blended users are routed to the existing switching system over a public route, add an alias (not blended alias) that matches the public charge ID that you intend to provision for the user. If calls from users (non-blended) to blended users are routed to the existing switching system over a private route, add an alias (not blended alias) that matches the private charge ID you intend to provision for the user.

Adding these aliases allows the SIP Application Module to identify the calling party on SimRing calls that are the result of a call origination, transfer or redirect from the SIP Application Module back to a blended user's Preferred Audio Device on the existing switching system.



Note: This step should also be done for all non-blended users that may call a blended user in order to identify SimRing calls that were originated by non-blended users.

- 7 Click **Save**. A User added successfully message appears.

Figure 56 Blended User added successfully confirmation



- 8 Click the **Customize service package link** at the top of the form to create a service package containing specific services for the blended user. The Service package customization window appears (Figure 57).

- 9 Enter the Preferred Audio Device for the user. The number entered here must be the number that is required to route a call to the user's phone and trigger the user's SimRing feature. This number should be unique for each user. This number is required so that regular clients (SIP Multimedia PC Client, SIP Multimedia Web Client, and the i2004 Internet Telephone controlled through the IPCM) can call a Blended PC Client by their SIP address (and not only by the PSTN equivalence number).

Figure 57 Service package customization for a blended user

Service package customization

You are customizing the **blended user** package for this user

Enable Voicemail

Enable Audio Conference

| | |
|-------------------------|---|
| Maximum Number of Ports | 4 |
|-------------------------|---|

Enable Presence

| | |
|-------------------------------------------|---------------------------------------------------------|
| Maximum Number of Subscriptions Generated | 10 |
| Maximum size of client buddy list | 10 |
| Report when inactive | <input checked="" type="checkbox"/> |
| Inactivity Timer (in minutes) | <input style="width: 100px;" type="text" value="30"/> ⓘ |
| Report when on the phone | <input checked="" type="checkbox"/> |

Enable Video

Simultaneous Network Call Restriction

| | |
|------------------------------------|---|
| Maximum Number of Ringlist entries | 3 |
|------------------------------------|---|

Enable Advanced Screening

Enable QoS

| | |
|----------------------------------|----|
| QoS DiffServ Code for Signalling | 8 |
| QoS DiffServ Code for Audio | 10 |
| QoS DiffServ Code for Video | 10 |
| QoS 802.1p Service Priority | 1 |

Blended User

| | |
|------------------------|--------------------------------------------------------------------------|
| Preferred Audio Device | <input style="width: 150px;" type="text" value="41000@nortelnetworks."/> |
|------------------------|--------------------------------------------------------------------------|

- 10 Click **Save**. A message displays that the user was added successfully.

Adding i2004 Internet Telephone capability for a user

The IP Client Manager requires the i2004 properties of a user to enable that user to log into, receive calls on, or make calls from an i2004 Internet Telephone.

To add i2004 Internet Telephone capability

- 1 After successfully adding a new user, click the **i2004 properties** link at the top of the User details window. The i2004 User details window appears (Figure 58).

Figure 58 User details window (i2004)

Main | i2004 properties | Routes | Customize service package

User details

Display Name:

Registration Length:

Registration Unit:

IPCМ:

- 2 Enter i2004 Internet Telephone properties in the parameter fields as described in the following table.

Table 16 i2004 properties

| Parameter | Type | Range | Default | Description |
|---------------------|--------------|-----------------------------------------------|---------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Display Name | String (30) | Up to 30 characters | N/A | The display string that is displayed on the i2004 Internet Telephone. |
| Registration Length | Integer | Max 86400 seconds or 1440 minutes or 24 hours | 3600 | This value, along with the Registration Unit parameter value, is used to determine when the i2004 Internet Telephone should register with the IPCM (see Note). It is recommended to use Registration Length in the range of 24 - 48. |
| Registration Unit | String (10) | Up to 10 characters | Seconds | The unit used by the i2004 Internet Telephone to know when to register with the IPCM. It is recommended to use Registration Unit as Hours. |
| IPCM | String (100) | Up to 100 characters | N/A | Drop down list of IPCM names to which this user belongs. Varies depending on which IPCMs are associated with the root domain. |



Note: The Registration Length and Registration Unit fields combined together make up the registration expiration time. The smaller the value, the more SIP traffic generated. If you were to select a registration expiration of every 2 minutes, then the IPCM would only register the user for 2 minute time periods and would have to re-register the user every 2 minutes.

For security reasons, it is recommended to not keep the registration expiration time too high. If you selected a registration expiration of 10 days, then the registration authentication process is only going to occur every 10 days. If the user changes a password, the change would not be noted for up to 10 days (if only registration on authentication is being used). Also, in the unlikely event that the registration is somehow lost, a reasonable length registration expiration would allow the IPCM to re-register the user.

3 Click **Save**.

Adding a conference server user and route

To enable conferencing capability, a conference user and route must be provisioned using the SIP Provisioning Client or the SIP Personal Agent.

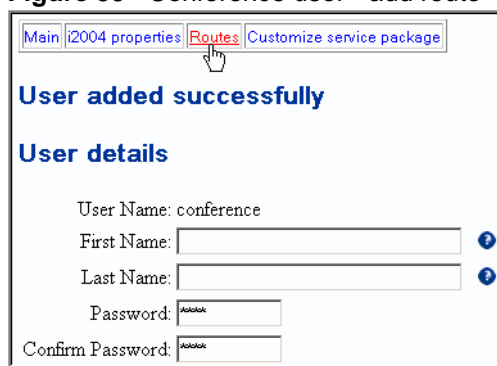
| Use this client... | To set up... |
|-------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| SIP Provisioning Client | a conference user, and, if there is only one conference server, a route to the conference server. (Routes added through the SIP Provisioning Client are always attempted simultaneously.) |
| SIP Personal Agent | sequential routes and rules for multiple conference servers that include each conference server listed in a separate ringlist. |

Adding a conference server user and route using the SIP Provisioning Client

To add a single conference server user and route using the SIP Provisioning Client

- 1 Click the **Domains** menu option.
- 2 Click the domain where you want to add a conference server user.
- 3 Click the **Add User** menu option.
- 4 In the Add User form, enter the text “conference” for the User Name. In the password field, enter any password.
- 5 Click **Save**. The User details screen displays a message that the user was added successfully.
- 6 At the top of the screen, click the **Routes** link to specify the route to the conference server.

Figure 59 Conference user - add route



- 7 In the **User Routes** text box, enter the SIP address that the conference server is expecting using the following format:

```
sip:conference@<conference_server_IP_address>;nt_info=proxy_foreign;  
nortelservice=conference
```
- 8 Click **Add**. The route now appears in the window of selected routes.

Figure 60 Conference user - route added

The screenshot shows the 'User routes' configuration page in the SIP Provisioning Client. At the top, there are navigation tabs: 'Main', '2004 properties', 'Routes', and 'Customize service package'. The 'Routes' tab is selected. Below the tabs, the title 'User routes' is displayed. A text input field contains the SIP address: 'sip:conference@47.02.117.5060;nt_info=proxy_foreign;nortelservice=conference'. To the right of this field is an 'ADD' button. Below the input field, a 'Selected:' list contains the same SIP address. To the right of this list are 'UP' and 'DOWN' buttons. At the bottom of the list area are 'REMOVE' and 'REMOVE ALL' buttons. A 'Save' button is located at the bottom center of the page.

- 9 Click **Save** to save the route.

Adding conference server routes using the SIP Personal Agent

To add multiple conference server routes using the SIP Personal Agent

- 1 Log in to the SIP Personal Agent. (The login name and password is the user name and password that was provisioned for the conference user in the SIP Provisioning Client.)
- 2 Click the **Call Manager** icon on the left of the SIP Personal Agent screen. The Welcome to Call Manager home screen displays.
- 3 Click the **ListRoutes** tab from the Call Manager Home screen. The Ring List Details screen displays.
- 4 Enter a name for the conference server route in the **Ring list name** text box (for example, conference route).
- 5 From the drop down list, select the maximum number of times to ring the phone.
- 6 In the first text box in the column, enter the SIP address that the conference server is expecting using the following format:

```

sip:conference@<conference_server_IP_address>;nt_info=proxy_foreign;
nortelservice=conference

```
- 7 Select SIP address from the drop down list.

- 8 To set up routes for multiple conference servers, repeat steps 6 and 7 for each sequential ring list in columns 2 and 3.
- 9 Click **Add Route**. The **Add Route** button updates the online user database with the conference server routing information.

Setting up rules for the conference server route

Once a route has been set up, a screening rule must be created to define the activation of the conference servers. The conference server route is not activated until a rule has been applied.

To set up a screening rule for the first time in the SIP Personal Agent

- 1 Click **ListRules** from the Call Manager screening and routing screen. The Welcome to Call Manager screen displays.
- 2 Click the link, **create a screening rule**. The Call Manager Rule Details screen displays.
- 3 Enter a name for the conference server rule in the **Rule name** text box (for example, conference rule).
- 4 Click the **Active** check box to select that the rule should appear in the Active Rules window.
- 5 Select the conference route from the pull-down menu next to the Active route field.
- 6 Click **Save Rule**.

Customizing service packages

Once a user is added, you may want to customize the service package on a per user basis.

To customize a user's service package

- 1 After adding or listing details for a user, click the **Customize service package** link.

Figure 61 Customize service package link

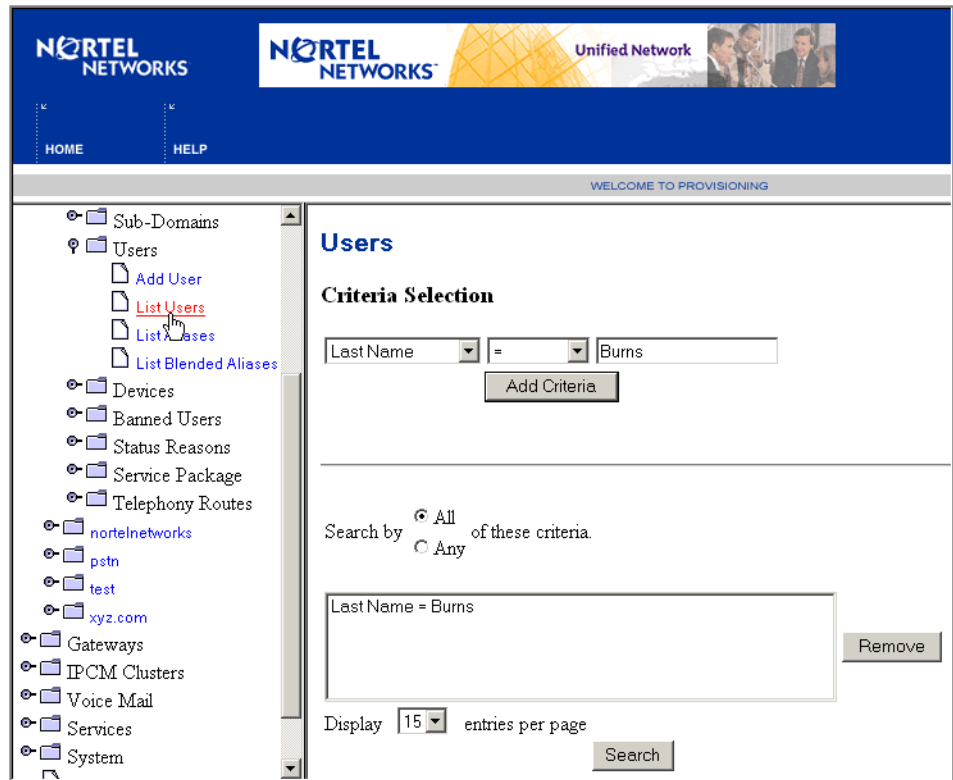
- 2 Customize the options in the user's service package by completing the available selections.
- 3 Click **Save** to save the changes to the service package. A message appears that the user package was customized successfully.

Listing/modifying/deleting users

To list, modify, or delete users (including blended users)

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain, click the **User > List User** menu option. The Users window appears (Figure 62). Select whether to search on **Phone Number, Last Name, User Name, or First Name** from the drop down list
- 3 Click **Add Criteria**. The search criteria you selected appears in the lower window and is used to refine your search.
- 4 Repeat steps 2 and 3 to list additional users.
- 5 In the lower window area, select whether to search by **All** or **Any** of the criteria specified. Note that you do not have to add any criteria if you want to see all the users in the selected domain. This will also find all users in any of the selected domain's subdomains.

Figure 62 List Users window



- 6 Define the size of your search list from the drop down list.
- 7 Click **Search** to begin the search, or **Remove** to remove search criteria from the search list. A list of users appears.

Figure 63 List users search results

| Users | | | | | |
|----------|--------------------|-----------|-------|-------------------------|------------------------|
| Username | Domain | Name | Phone | Details | Delete |
| dburns | nortelnetworks.com | null null | null | Details | Delete |

- 8 Click the **Details** link to modify user information. The User details window appears (Figure 64). To delete the user from the domain, click the **Delete** link. At the delete confirmation dialog box, click **OK** to delete the user, or **Cancel**.

Figure 64 User details window

Main | 2004 properties | Routes | Customize service package

User details

User Name: dburns

First Name: ⓘ

Last Name: ⓘ

Password:

Confirm Password:

Service package:

Aliases:

Blended Aliases:

Status Reason:

email:

Business Phone:

Home Phone:

Cell Phone:

Pager:

Fax:

Voicemail Server:

Private Charge ID:

Public Charge ID:

Voicemail:

Class of Service:

Time Zone:

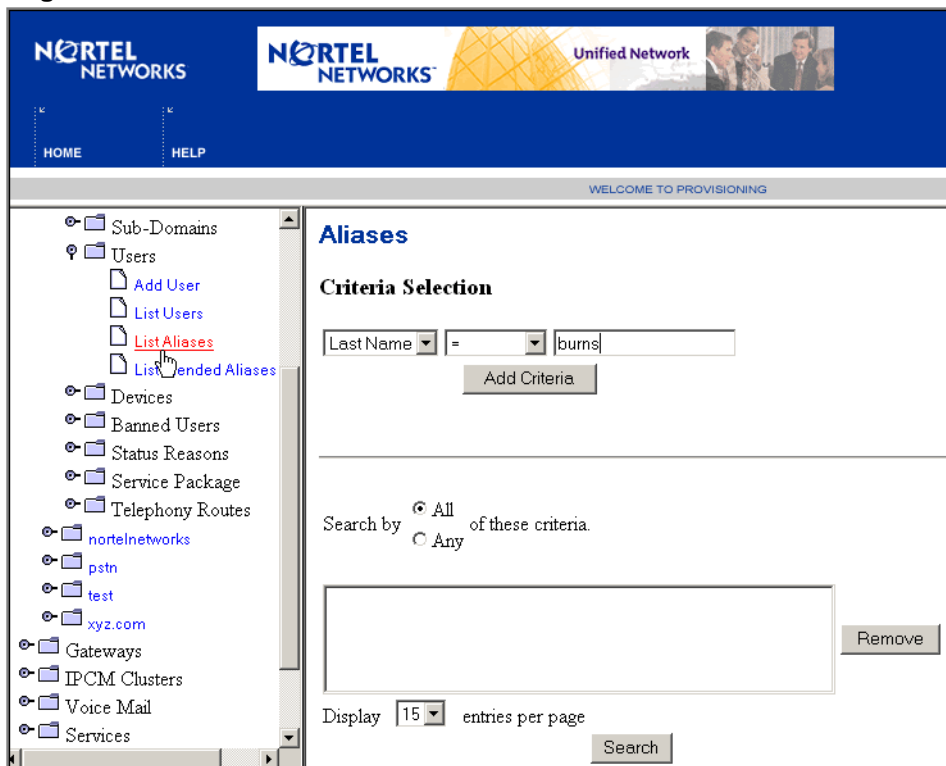
Locale:

9 Click **Save**. A User Updated Successfully message appears.

Listing/modifying/deleting aliases

To list all aliases in a domain

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain, click the **Users > List Aliases** menu option. The Aliases window appears (Figure 65). Select whether to search on **Last Name**, **Alias**, or **Domain** from the drop down list
- 3 Click **Add Criteria**. The search criteria you selected appears in the lower window and is used to refine your search.
- 4 Repeat steps 1-3 to list additional users.
- 5 In the lower window area, select whether to search by **All** or **Any** of the criteria specified. Note that you do not have to add any criteria if you want to see all the aliases in the selected domain. This will also find all aliases in any of the selected domain's subdomains.

Figure 65 List aliases window

- 6 Define the size of your search list from the drop down list.

- Click **Search** to begin the search, or **Remove** to remove the search criteria from the search list. The Aliases search results window appears (Figure 66).

Figure 66 List aliases search results

| Aliases | | | | | |
|---------|--------------------|----------|-------------|-------------------------|------------------------|
| Alias | Domain | Username | Name | Details | Delete |
| elena | nortelnetworks.com | ebyrnes | elena burns | Details | Delete |

- Click the **Details** link to modify alias information. The User details window appears. Click **Save** to save your changes.
- To delete an alias, click the **Delete** link on the List aliases search results screen to delete the user from the domain. At the delete confirmation dialog box, click **OK** to delete the user, or **Cancel**.
- Click **Save**.



Tip: If you delete an alias, the username associated with the alias is deleted.

Listing/modifying/deleting blended aliases

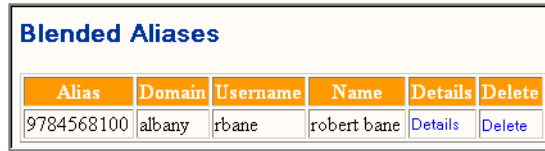
To list all blended aliases in a domain

- Click the **Domains** menu option.
- Select a domain (or a subdomain of a domain).
- Click the **Users > List Blended Aliases** menu option. The Blended Aliases window appears. Select whether to search on **Last Name**, **Alias**, or **Domain** from the drop down list
- Click **Add Criteria**. The search criteria you selected appears in the lower window and is used to refine your search.
- Repeat steps 1-4 to list additional last names, aliases, or domains.
- In the lower window area, select whether to search by **All** or **Any** of the criteria specified. Note that you do not have to add any criteria if you want to

see all the blended aliases in the selected domain. This will also find all blended aliases in any of the selected domain's subdomains.

- 7 Click **Save**. The Blended Alias search results window displays (Figure 67)

Figure 67 Blended Aliases search results window



| Alias | Domain | Username | Name | Details | Delete |
|------------|--------|----------|-------------|-------------------------|------------------------|
| 9784568100 | albany | rbane | robert bane | Details | Delete |

- 8 Click the **Details** link to modify alias information. The User details window appears (Figure 64). To delete the user from the domain, click the **Delete** link. At the delete confirmation dialog box, click **OK** to delete the user, or **Cancel**.

Chapter 9

Managing i2004 Internet Telephones

Topics in this section:

- [“Introduction” on page 111](#)
- [“Adding an i2004 Internet Telephone” on page 111](#)
- [“Listing an i2004 Internet Telephone” on page 116](#)
- [“Deleting an i2004 Internet Telephone” on page 119](#)

Introduction

i2004 Internet Telephones are provisioned by creating an i2004 device entity and assigning this device to a domain or subdomain.

Devices are assigned to a domain based on the domain in which it is created. An administrator can use the **Devices** menu option to add, view, modify or delete i2004 Internet Telephones.

Adding an i2004 Internet Telephone

- 1 Click the **Domains** menu option.
- 2 Click the domain or subdomain for which you wish to provision an i2004 Internet Telephone.

- Click the **Devices > Add Device** sub menu option. The Adding a Device window appears ([Figure 68](#))



Tip: Several fields in the Adding a Device window are populated with the default property values that have been previously set up for the domain or subdomain.

Figure 68 Adding a device

Adding a Device in abc domain

Label: ⓘ

MAC Address: ⓘ

Available IPCM List: ⓘ

Default Proxy Host:

Def Proxy Port:

Active Proxy: FALSE TRUE

Allow All Codecs: TRUE FALSE

Behind Firewall: TRUE FALSE

Alpha: TRUE FALSE

Contrast:

VocoderPacketTime:

PDIL Timer:

PSEIZ Timer:

DATE FMT:

TIME FMT:

Location:

IdleDisplay:

Time Zone:

Locale:

- Enter device parameters as described in the following table ([Table 17](#)).

- 5 Notice that the most of the values that appear on the form are default values that are set from what was entered when the domain was provisioned.

Table 17 i2004 Internet Telephone parameters

| Parameter | Type | Range | Default | Description |
|---------------------|----------------------------|----------------------|---------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Label | String (30) | N/A | N/A | User defined text label that uniquely describes this device. This label is used by the System Management Console when performing maintenance operations on queried devices (for example, Tom's Desk). Defaults to owner display name. |
| MAC Address | String (100) | Up to 100 characters | N/A | 12 Digit Hex string. Permanently burned into the i2004 device at manufacture time. Read from tag on back of device. This string is converted to lower case. Must be unique per IPCM. |
| Available IPCM List | Select from drop down list | N/A | N/A | Drop down list of named IPCMs. |
| Def Proxy Host | String (30) | N/A | N/A | IP Address of Proxy used as default for the i2004 Internet Telephone. |
| Def Proxy Port | Integer | Any valid port | 5060 | Default Proxy SIP Port. No validation is done on this field to ensure validity. |
| Active Proxy | Boolean | True/False | True | If True, calls will route through the default proxy. For future use. No other routing is currently supported for the IPCM. |
| Allow All Codecs | Boolean | True False | True | Allow all codecs (True) or only selected codec (False). See Vocoder parameter. |
| Behind Firewall | Boolean | True/False | True | Indicates whether there is a firewall/NAT between the i2004 Internet Telephone and the system. |
| Alpha | Boolean | True/False | True | Enables alphanumeric dialing as the default dialing style. If disabled, the i2004 Internet Telephone dialing style defaults to numeric. |
| Contrast | Integer | 0-15 | 8 | Telephone display contrast setting. |
| Vocoder:Packet Time | Integer (msec) | 10/20/30/40/60 | 20 | Packet time in milliseconds {10\ms, 20\ms, 30\ms, 40\ms, 60\ms}. |
| PDIL Timer | Integer (seconds) | N/A | 4 | Interdigit time-out. |
| PSEIZ Timer | Integer | N/A | 15 | Time-out for first dialed digit. |

Table 17 i2004 Internet Telephone parameters

| Parameter | Type | Range | Default | Description |
|--------------|------------------------------------------------------------------------------------------------------------------------------|----------------------------------------|---------|--------------------------------------------------------------------------------------------------------|
| DATE FMT | Numeric Standard MM/DD or Numeric Inverse DD/MM | 2,3 | 2 | Only formats 2 and 3 are supported. 2-numeric standard (09/16); 3-numeric inverse (16/09). |
| TIME FMT | Select from drop down list: <ul style="list-style-type: none"> • 12-hour • French • 24-hour | 0,1,2 | 0 | The time format of the proxy used as the default for the I2004 Internet Telephone 0 > 12 hour clock. |
| Location | String (50) | Up to 50 characters | | Text describing location of device. A specific format of this string is scheduled for a future release |
| Idle Display | String (30) | Up to 30 characters | N/A | Idle telephone display heading. |
| Time Zone | String (20) | Any valid entry in table TIMEZONE INFO | N/A | The time zone that this user belongs to. |
| Locale | String (20) | | N/A | The location of this user. |

Table 18 Click **Save**. The IPCM Device Created window appears (Figure 69).

Figure 69 Add device - IPCM device created

IPCMDevice Created

Modifying a Device in abc domain.

Assign Users

Label: Desktop

MAC Address: 0060381695

IPCM List: paul:AC(1000)::CC(0)

Default Proxy Host: EnterDefaultProxyHost

Def Proxy Port: 5060

Active Proxy: FALSE TRUE

Allow All Codecs: TRUE FALSE

Behind Firewall: TRUE FALSE

Alpha: TRUE FALSE

Vocoder:PacketTime : G723/6.3Kbps:PT(30)

Contrast: Contrast(8)

PDIL Timer: 6

PSELZ Timer: 15

DATE FMT: NumericStandard(MM/DD)

TIME FMT: 12-hour

Idle Display: Hello world

Location: PhysicalLocation

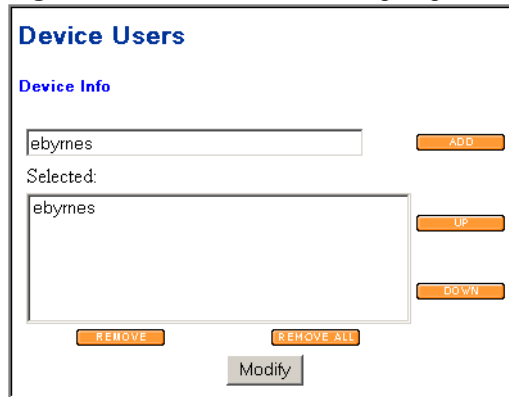
Time Zone: Saskatchewan

Locale: English

Save

- The i2004 is now provisioned with the parameters that were entered. The i2004 is ready for login. If you want to remotely place a user on an i2004 Internet Telephone, click **Assign Users** at the top of the screen. (This is the equivalent to a user logging in.) The Device Users window appears (Figure 70).

Figure 70 Device Users - assigning users



- 7 Enter the user name in the text field and click **Add**. (If you do not add a domain name for the user, the domain or subdomain of the device is assumed.)



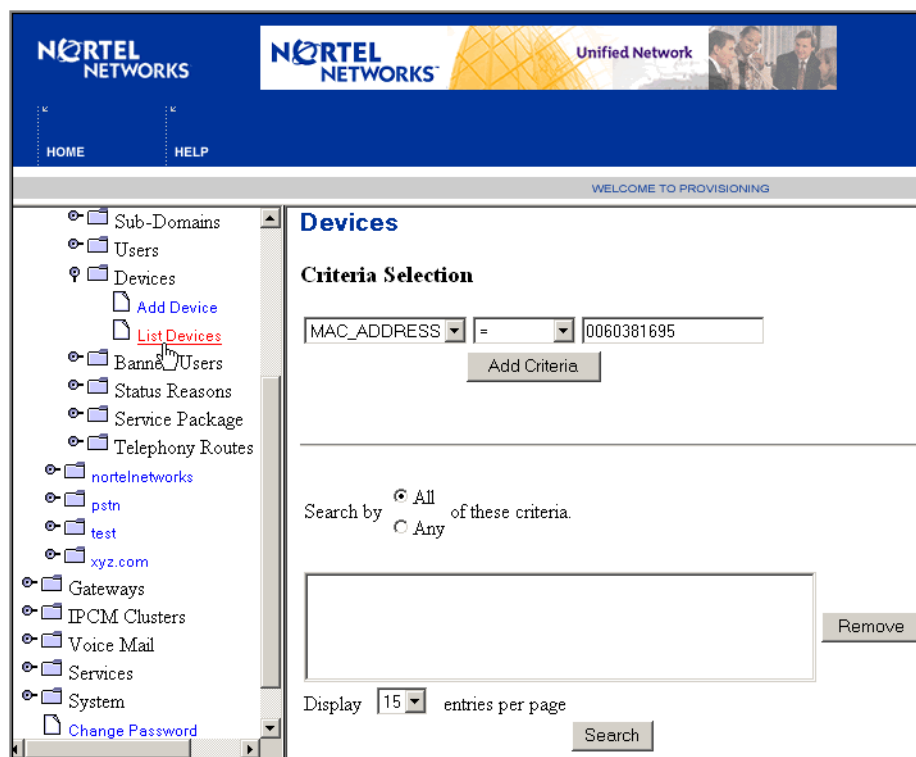
Tip: You can add up to 6 different users per i2004 Internet Telephone.

- 8 To move users up or down in the selected window, select the user and click **Up** or **Down**. To remove a name, click **Remove** or **Remove All**.
- 9 Click **Modify** to submit your changes.

Listing an i2004 Internet Telephone

To view, or list a particular device (or all devices) in a domain or subdomain

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain where you want to view a device, click the **Devices > List Device** menu option. The Devices window appears (Figure 71).

Figure 71 List Devices window

- 3 Select whether to search on the **MACaddress** or **ID** of the i2004 Internet Telephone from the drop down list.
- 4 Click **Add Criteria**. The search criteria you selected appears in the lower window.
- 5 Repeat to list additional devices.
- 6 In the lower window area, check whether to search by **All** (devices), or **Any**, using the device criteria you specified.
- 7 Click **Search** to begin the search, or **Remove** to remove the device from the search list. The Devices in Domain window appears (Figure 72).

Figure 72 List devices search results

Devices in Domain:

| Device Label | MACAddress | Domain | IPCM | Details | Delete |
|--------------|------------|---------|------|-------------------------|------------------------|
| Desktop | 0060381695 | abc.com | paul | Details | Delete |

- 8 Click the **Details** link to modify device information. The Modifying a Device in a domain window appears (Figure 73).

Figure 73 Modifying a device in a domain window

Modifying a Device in abc.com domain.

Assign Users

Label:

MAC Address: 0060381695

IPCM List:

Default Proxy Host:

Def Proxy Port:

Active Proxy: FALSE TRUE

Allow All Codecs: TRUE FALSE

Behind Firewall: TRUE FALSE

Alpha: TRUE FALSE

Vocoder:PacketTime :

Contrast:

PDIL Timer:

PSEIZ Timer:

DATE FMT:

TIME FMT:

Idle Display:

Location:

Time Zone:

Locale:

- 9 Click **Save**.

Deleting an i2004 Internet Telephone

To delete an i2004 Internet Telephone

- 1 Click the **Domains** menu option.
- 2 Click the domain or subdomain for which you wish to delete an i2004 Internet Telephone.
- 3 Click the **Devices > List Devices** sub menu option. The Devices window appears. Select whether to search on the MACAddress or ID of the i2004 Internet Telephone from the drop down list.
- 4 Click **Add Criteria**. The search criteria you selected appears in the lower window.
- 5 Repeat to list additional devices.
- 6 In the lower window area, select whether to search by **All** (devices), or **Any**, using the criteria specified.
- 7 Click **Search**. The Devices in Domain window appears with the results of your search (Figure 74).

Figure 74 Devices in Domain window (search results)

| Devices in Domain: | | | | | |
|--------------------|------------|---------|------|-------------------------|------------------------|
| Device Label | MACAddress | Domain | IPCM | Details | Delete |
| Desktop | 0060381695 | abc.com | paul | Details | Delete |

- 8 Click the **Delete** link to delete the i2004 Internet Telephone. At the delete confirmation dialog box, click **OK** to delete the device, or **Cancel**.

Chapter 10

Telephony Routes



Note: Telephony routing enables customers to configure their dial plans and to set Virtual Private Networks (VPNs). You must have telephony routing and domain management provisioning rights to perform the tasks in this section.

Topics in this section:

- [“Adding Class of Service \(COS\) information” on page 122](#)
- [“Adding a telephony route” on page 124](#)
- [“Listing a telephony route” on page 128](#)
- [“Adding a route list” on page 128](#)
- [“Listing a route list” on page 130](#)
- [“Using the translation verification tool” on page 131](#)
- [“Setting up an interworking profile with the Communications Server 2000 \(CS2000\)” on page 133](#)

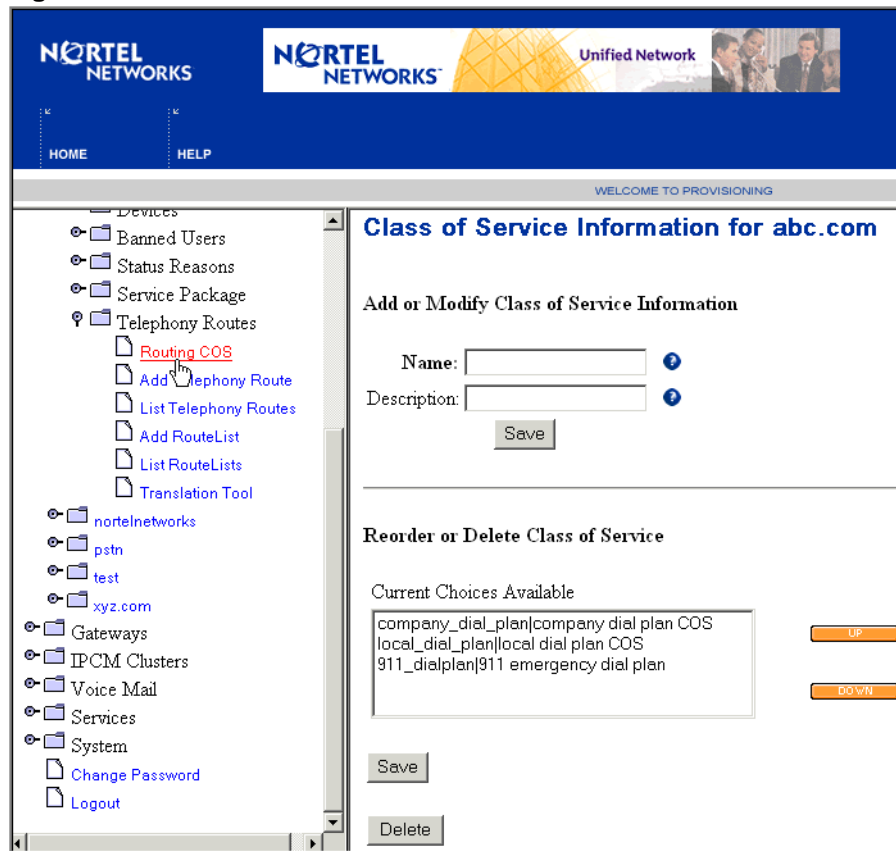
For more information and examples about telephony routing and translation plans, see [Appendix B, “Understanding Telephony Routing and Translations,” on page 149.](#)

Adding Class of Service (COS) information

COS is assigned to users, domains, and routing lists. It is an integer value, and the lower the number, the greater, or higher, the COS value. For example, if a user has a COS value of 1, and a domain has a COS of 2, 3, and 4, the user with a COS of 1 can use all the domains to make a call. The lower the number, the higher access to making calls.

To create a COS to associate with a domain, user, or route list

- 1 Click the **Domains** menu option.
- 2 Under the domain for which you wish to add COS information, click the **Telephony Routes > Routing COS** menu option. The Class of Service information window for the domain appears ([Figure 75](#)).

Figure 75 Class of Service information for domain window

- 3 Enter a name for the COS in the Name text box.
- 4 Enter a description for the COS in the Description text box.
- 5 Click **Save**. The COS now appears in the window below. Repeat this step for each new COS you wish to create.
- 6 Click **UP** or **DOWN** to reorder the COS in the Current Choices Available window.



Tip: The higher the position of the COS in the reorder list indicates that the COS has a lower number, which means a higher COS value, or service.

- 7 To delete a COS from the Current Choices window, click **Delete**.

Adding a telephony route

To add a new telephony route

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain for which you wish to add a new telephony route, click the **Telephony Routes > Add Telephony Routing** menu option. The Create New Telephony Route window appears (Figure 76).

Figure 76 Create New Telephony Route window

The screenshot shows the 'Create New Telephony Route' window. The left sidebar contains a tree view with the following items: Devices, Banned Users, Status Reasons, Service Package, Telephony Routes (expanded), Routing COS, Add Telephony Route (highlighted), List Telephony Routes, Add RouteList, List RouteLists, Translation Tool, nortelnetworks, pstn, test, xyz.com, Gateways, IPCM Clusters, Voice Mail, Services, System, Change Password, and Logout. The main area contains the following form fields:

- Name:
- Description:
- From Digits:
- To Digits:
- Min Number of Digits:
- Max Number of Digits:
- Route Type:
- Remove:
- Prefix:
- Recursive:
- Route List:

Buttons: Save, Clear

- 3 Enter telephony route parameter information and values in the fields provided. Note that the Route List window will be empty the first time you

create a telephony route. The following table (Table 19) lists the telephony route list parameter values and descriptions.

Table 19 Telephony route parameter information

| Parameter | Type | Range | Default | Description |
|----------------------|--------------|-------|---------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Name | String (20) | N/A | N/A | The name of this telephony route. |
| Description | String (100) | N/A | N/A | A description for this telephony route. |
| From Digits | String (32) | N/A | N/A | If you take out the most significant digits (MSD) from the number to be translated, its value should be equal to or greater than this field's value. The number of MSDs to be compared is the number of digits you have entered in this field. |
| To Digits | String (32) | N/A | N/A | If you take out the MSDs from the number to be translated, its value should be equal to or less than this field's value. The number of MSDs to be compared is the number of digits you have filled in this field. |
| Min Number of Digits | Integer | N/A | N/A | The number to be translated should have digits equal to or more than the Min Number of Digits parameter value. |
| Max Number of Digits | Integer | N/A | N/A | The number to be translated should have digits equal to or less than the Max Number of Digits parameter value. |
| Route Type | Integer | | | Select one of the following route types <ul style="list-style-type: none"> Private (for example, a private dial plan not going out to a gateway) SIP (does not have to go through a gateway, but can go to a different application module or domain) Gateway |
| Remove | Integer | N/A | 0 | This is the number of MSDs, which will be removed while performing digit manipulation/translation. |
| Prefix | String (24) | N/A | N/A | This is the number that would be prefixed to the number coming in for translation. |
| Recursive | Boolean | True | False | If set to Y, the digit manipulation process is carried out recursively after each iteration. |

- Click **Save** to create the telephony route. The Telephony Route Listing window appears with the new route listed added to the domain and parameter options based on the telephony route type (Figure 77).

Figure 77 Telephony Route Listing window

| Name | From Digits | To Digits | Min Digits | Max Digits | Route Type | Remove | Prefix | Recursive | Delete |
|------------------------------|-------------|-----------|------------|------------|------------|--------|--------|-----------|------------------------|
| 5_digit_plan | 5 | 5 | 5 | 5 | Private | 0 | 97268 | No | Delete |

[Change Parameters](#)

- Click the **Change Parameters** link.

- Provision the parameters for the new telephony route based on the whether the route is a Private, SIP, or Gateway route. The following telephony route parameters windows appear as described in the following table (Table 20).

Table 20 Telephony route parameters window (Private, SIP, Gateway)

| Telephony route parameters | Description |
|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <p>Parameters</p> <p>Subscriber Not Found: <input type="text" value="STOP"/> <input type="button" value="Save"/> <input type="button" value="Clear"/></p> | <p>For a Private Route:</p> <p>From the drop down list, specify what to do if the subscriber is not found:</p> <p>Stop: Stop at the private route. Do not go to SIP or Gateway. The call is dropped, since subscriber does not exist.</p> <p>Continue: Continue to search for SIP or gateway route without translating the number.</p> <p>Pre Translate: Continue to SIP or gateway route with translating the numbers.</p> <p>Click Save.</p> |
| <p>Parameters</p> <p>SIP Route: <input type="text"/></p> <p>Subscriber Not Found: <input type="text" value="STOP"/> <input type="button" value="Save"/> <input type="button" value="Clear"/></p> | <p>For a SIP Route:</p> <p>From the drop down list, specify a route name and what to do if the subscriber is not found:</p> <p>Stop: Stop at the SIP route. Do not go to gateway route. Since the subscriber does not exist, the call is dropped.</p> <p>Continue: Continue to gateway route without translating the number.</p> <p>Pre Translate: Continue to gateway route and translate the numbers.</p> <p>SIP Route: Specify the domain you want the call to go to or a different appserver address.</p> <p>Click Save.</p> |
| <p>Parameters</p> <p>OverRide Charge ID: <input type="text" value="N"/></p> <p>Gateway Route: <input type="text" value="sl100_route"/></p> <p>Gateway RouteType Option: <input type="text" value="PRIVATE"/></p> <p>OverRide Charge Value: <input type="text"/></p> <p><input type="button" value="Save"/> <input type="button" value="Clear"/></p> | <p>For a Gateway Route</p> <p>OverRide Charge ID: If set to Yes then the charge ID described in the OverRide Charge Value field is used instead of either public or private.</p> <p>Gateway Route: Select the name of the gateway route from the list to which you want to route the call to.</p> <p>Gateway RouteType Option: Specifies what charge ID to be used: Public or Private. If OverRide Charge ID is set to Yes, then this is not used.</p> <p>OverRide Charge Value: The value of the charge ID to be overridden with.</p> <p>Click Save.</p> |

Listing a telephony route

To view details about a telephony route

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain for which you wish to view telephony route details, click the **Telephony Routes > List Telephony Routing** menu option. The Telephony Route Listing for the domain window appears (Figure 78).

Figure 78 Telephony route listing window

| Telephony Route Listing for - abc.com | | | | | | | | | |
|---------------------------------------|-------------|-----------|------------|------------|------------|--------|--------|-----------|--------|
| Name | From Digits | To Digits | Min Digits | Max Digits | Route Type | Remove | Prefix | Recursive | Delete |
| 911 private route | 911 | 911 | 3 | 3 | Private | 0 | | No | Delete |
| Subscriber Not Found: PRE-TRANSLATE | | | | | | | | | |
| Change Parameters | | | | | | | | | |
| 5_digit_plan | 5 | 5 | 5 | 5 | Private | 0 | | No | Delete |
| Change Parameters | | | | | | | | | |
| route_to_xyz | 6 | 6 | 8 | 8 | SIP | 1 | 214 | No | Delete |
| SIP Route: xyz.com | | | | | | | | | |
| Subscriber Not Found: STOP | | | | | | | | | |
| Change Parameters | | | | | | | | | |
| pstn_route | 9 | 9 | 11 | 11 | Gateway | 1 | | No | Delete |

- 3 To view or modify a telephony route, click the telephony route name. The Create New Telephony Route window appears.
- 4 Modify the fields. Click **Save** to return to the Telephony Route Listing window.
- 5 To change the telephony route parameter, click the **Change Parameters** link for the telephony route. The telephony parameters window appears. Based on the whether the route is a Private, SIP, or Gateway route, modify the telephony parameters (Table 20).

Adding a route list

To add a new telephony route list

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain for which you wish to add a route list, click the **Telephony Routes > Add RouteList** menu option. The Create a new route list window appears (Figure 79).

Figure 79 Create a New Route List window

The screenshot displays the Nortel Networks provisioning interface. The top navigation bar includes the Nortel Networks logo and a 'Unified Network' banner. Below the navigation bar, there are links for 'HOME' and 'HELP'. The main content area is titled 'WELCOME TO PROVISIONING' and features a 'Create a New Route List' window. On the left, a navigation tree shows the following structure:

- Sub-Domains
- Users
- Devices
- Banned Users
- Status Reasons
- Service Package
- Telephony Routes
 - Routing COS
 - Add Telephony Route
 - List Telephony Routes
 - Add RouteList** (highlighted)
 - List RouteLists
 - Translation Tool
- nortelnetworks
- pstn
- test

The 'Create a New Route List' form contains the following fields and options:

- Name:** [Text input field]
- Description:** [Text input field]
- Incoming Foreign Domain Call Routing:** [Dropdown menu set to 'ALLOW']
- Incoming Local Domain Call Routing:** [Dropdown menu set to 'ALLOW']
- Class of Service:** [Dropdown menu set to 'local_dial_plan']
- Telephony Route(s):** [Text area containing:
 - 911 private route||911||911
 - 5_digit_plan||5||5
 - route_to_xyz||6||6
 - pstn_route||9||9
 - backup_plan||411||411]

At the bottom of the form, there are 'Save' and 'Clear' buttons.

- 3 Enter a name for the new route list in the text field provided.
- 4 Enter a text description of the route list in the text field provided.
- 5 From the drop down list next to the Incoming Foreign Domain Call Routing (from another domain or subdomain) and Incoming Local Domain Call

Routing (coming from the same domain) fields, choose one of the following call routing options as described in the following table (Table 21).

Table 21 Call routing options

| Call routing option | Description |
|---------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Allow | Allow all incoming sessions from other domains and subdomains. |
| Block | Block all incoming calls from other domains and subdomains. |
| Redirect | Redirect sessions to the originator's domain. Use this option to redirect an incoming session from another domain that is routing to a restricted route list. |

- 6 From the drop down list, select a **Class of Service** for the new route list.
- 7 In the Telephony Route(s) window, highlight and select one or more telephony routes to add to the route list (each telephony route must go in a route list). To select multiple telephony routes, use the **Ctrl-click** key.

Listing a route list

To view details about a route list

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain you wish for which you wish to view route list details, click the **Telephony Routes > List RouteLists** menu option. The List of RouteLists for the domain window appears ([Figure 80](#)).

Figure 80 List route lists window

The screenshot shows the 'Route Lists' window in the Nortel Networks provisioning interface. The left sidebar contains a navigation tree with categories like Sub-Domains, Users, Devices, and Telephony Routes. The main content area is titled 'Route Lists' and shows a 'List of RouteLists for abc.com'. It contains a table with columns: Name, Description, Other Domain Action, Local Domain Action, Class of Service, and Delete. Below the table are two sections of 'Associated Telephony Routes', each with a table showing Route Name, From Digits, To Digits, Remove, and Prefix.

| Name | Description | Other Domain Action | Local Domain Action | Class of Service | Delete |
|------------------------------------|------------------------|---------------------|---------------------|-------------------|------------------------|
| company route list | highest COS route list | BLOCK | ALLOW | company_dial_plan | Delete |

Associated Telephony Routes

| Route Name | From Digits | To Digits | Remove | Prefix |
|------------------------------|-------------|-----------|--------|--------|
| route_to_xyz | 6 | 6 | 1 | 214 |

| Name | Description | Other Domain Action | Local Domain Action | Class of Service | Delete |
|----------------------------------|----------------------------|---------------------|---------------------|------------------|------------------------|
| local_route_list | local dial plan route list | BLOCK | ALLOW | local_dial_plan | Delete |

Associated Telephony Routes

| Route Name | From Digits | To Digits | Remove | Prefix |
|-----------------------------------|-------------|-----------|--------|--------|
| 911 private route | 911 | 911 | 0 | |

- 3 Click the link for the route list name to modify the route list. The Create New Route List window appears. Modify route list information and click **Save**.
- 4 To delete an existing route list, click the **Delete** link in the Delete column next to the route list you wish to delete.
- 5 To delete a telephony route associated with a route list, click the **Delete** link in the Remove column.

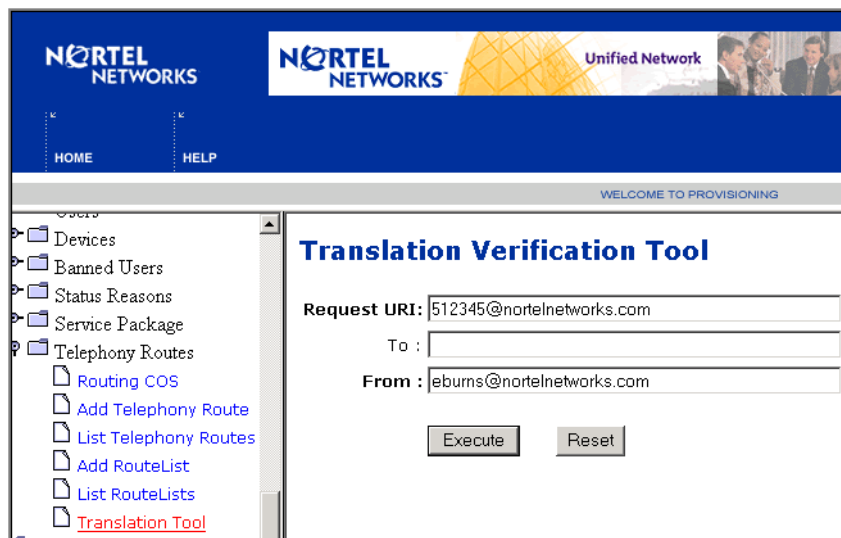
Using the translation verification tool

The translation tool allows you to verify that a call is provisioned correctly.

To use the translation verification tool

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain, click the **Telephony Routes > Translation Tool** menu option. The Translation Verification Tool window appears (Figure 81).

Figure 81 Translation Verification Tool window



- 3 In the Request URI field, enter a number to call using the following format
 $\langle \text{number} \rangle @ \langle \text{domain_or_subdomain_name} \rangle . \text{com}$
- 4 In the To field, enter the same name as in the Request URI field, or leave blank.
- 5 In the From field, enter the calling person's number using one of the following formats
 $\langle \text{name} \rangle @ \langle \text{domain_or_subdomain_name} \rangle . \text{com}$
 $\langle \text{number} \rangle @ \langle \text{domain_or_subdomain_name} \rangle . \text{com}$
- 6 Click **Execute**. The following window appears and lists the translation (read only) verification results (Figure 82). If the window displays some, or all, null fields, then the route has been misprovisioned and must be provisioned again.

Figure 82 Translation tool results window

| | |
|-----------------------|--------------------|
| Route | gw_1 |
| RouteType | PRIVATE |
| RouteList | another test |
| Class of Service | vip |
| RouteList Domain | nortelnetworks.com |
| Gateway Route Type | Gateway |
| OverRideCharge ID | N |
| OverRide Charge Value | 0 |
| Destinations | |

Setting up an interworking profile with the Communications Server 2000 (CS2000)

The Set Profile menu option is for interworking with the CS2000.

To set the profile

- 1 Click the **Domains** menu option.
- 2 Under the domain or subdomain that you want to set up to interwork with the CS2000, click the **Set Profile** menu option. The Set Profile for the domain window appears ([Figure 83](#)).

Figure 83 Set Profile for Domain window

Profile for domain - abc.com

Domain Name: abc.com

Profile:

- 3 For the domain selected, enter a name for the profile in the text box provided.
- 4 Click **Save**. A message appears that the domain profile was modified successfully.

Chapter 11

Gateways



Note: You must have gateway management rights to perform the tasks in this section.

Topics in this section:

- [“Introduction to gateways” on page 135](#)
- [“Adding a gateway” on page 136](#)
- [“Listing a gateway” on page 137](#)
- [“Adding a gateway route” on page 138](#)
- [“Listing a gateway route” on page 140](#)
- [“Adding a gateway trunk group” on page 141](#)
- [“Listing a gateway trunk group” on page 142](#)

Introduction to gateways

A gateway is a server that connects the SIP network with the PSTN. The gateway handles the negotiation for calls between PRI signaling and SIP signalling. The **Gateways** menu option is used by an administrator to provision a PRI gateway or the gateway’s translations. The data submitted is validated by the provisioning system before being stored in the database.

The data entered in the SIP Provisioning Client should match the data entered of the gateway itself done in the System Management Console. For the PRI Gateway, the following items need to match:

- IP address of the gateway
- trunk group on the gateway
- domain assigned to the trunk group

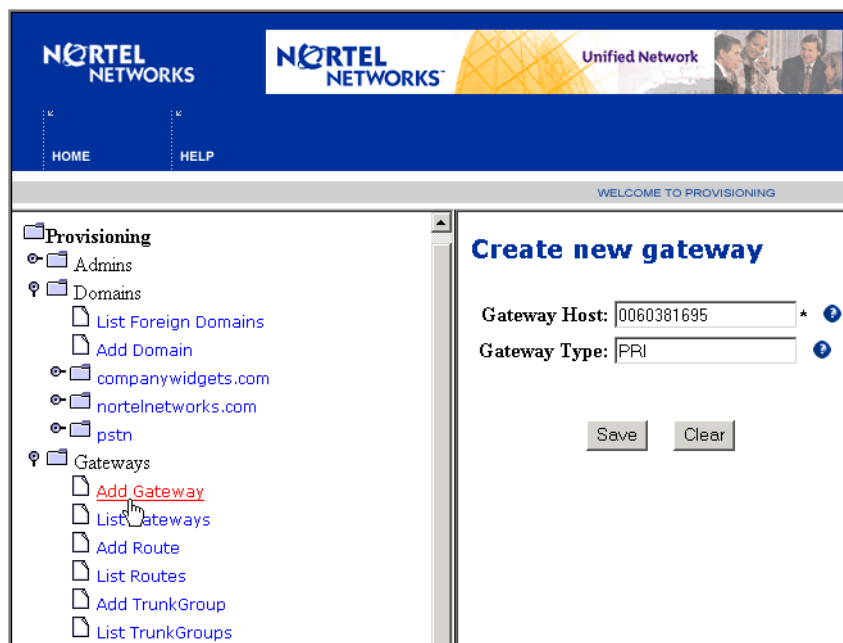
Adding a gateway

The **Add Gateway** menu option allows an administrator to provision a new gateway.

To add a new gateway

- 1 Click the **Gateways > Add Gateway** menu option. The Create new gateway window appears (Figure 84).

Figure 84 Create new gateway window




- 2 Enter the IP address or MADDR of the gateway in the text box (required). For example:

nortelnetworks.com;maddr=<IP_address>

- 3 Enter a description the gateway in the Gateway Type text box.
- 4 Click **Save**. (To clear the text fields, click **Clear**.) The Gateways window appears with a list of all provisioned gateways, including the gateway just added (Figure 85).

Figure 85 Gateways window



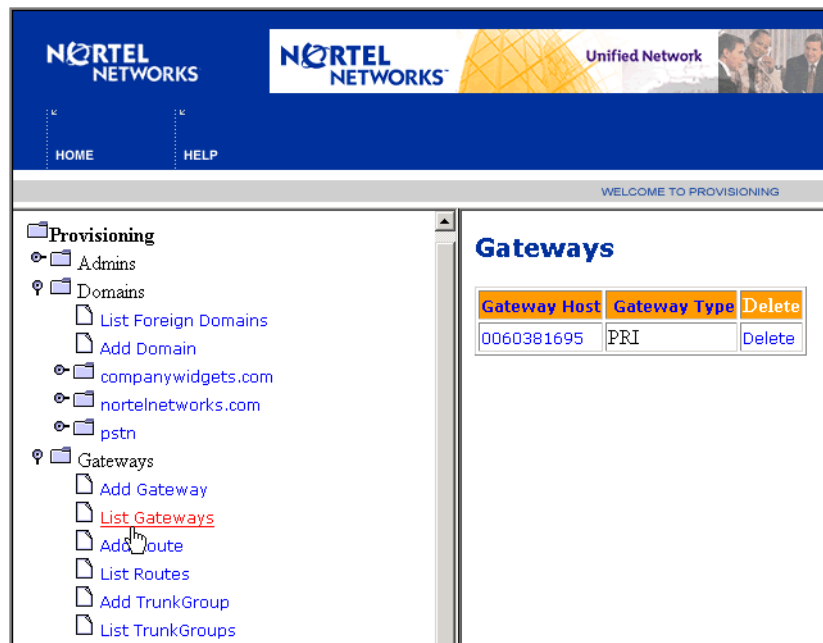
| Gateway Host | Gateway Type | Delete |
|--------------|--------------|--------|
| 0060381695 | PRI | Delete |

Listing a gateway

The List Gateway option allows an administrator to view information of all the available provisioned gateways.

To list a gateway

- 1 Click the **Gateways > List Gateways** menu option. The Gateways window appears (Figure 86).

Figure 86 Gateways window (List Gateways)

- 2 Click the gateway host name to view the details corresponding to that particular gateway.
- 3 To modify gateway information, enter the IP address or MADDR of the gateway in the textbox (required). For example,

nortelnetworks.com;maddr=<IP_address>
- 4 To modify gateway description, enter changes in the Gateway Type text box.
- 5 Click **Save**. (To clear the text fields, click **Clear**.) The Gateways window appears with a list of all provisioned gateways, including the gateway modified.
- 6 To delete a gateway, click the **Delete** link next to the gateway. This removes the provisioned gateway from the database and the system.

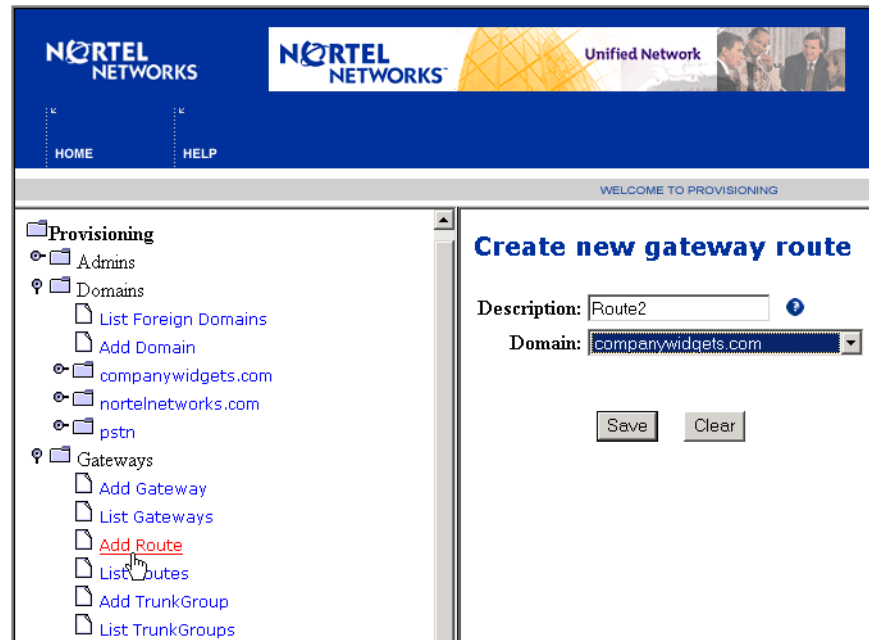
Adding a gateway route

This option allows an administrator to add gateway route information to associate the route with a domain.

To add a new gateway route

- 1 Click the **Gateways > Add Route** menu option. The Create new gateway route appears (Figure 87).

Figure 87 Create new gateway route window



- 2 Enter a description for the new route in the Description text box.
- 3 From the drop down list, select a domain to associate with the route.
- 4 Click **Save**. (To clear the text fields, click **Clear**.) The Gateway Routes window appears (Figure 88).

Figure 88 Gateway routes window

| Gateway Routes | | |
|----------------|----------------------|--------|
| Description | Trunk Groups Reorder | Delete |
| Nortel | Re-Order | Delete |
| CompanyRoute | Re-Order | Delete |
| Route2 | Re-Order | Delete |
| DallasRoute | Re-Order | Delete |

Listing a gateway route

The List Gateways menu option allows the administrator to view routing information for a particular gateway.

To view details for a gateway route

- 1 Click the **Gateways > List Routes** menu option. The Gateway Routes window appears (Figure 89).

Figure 89 Gateway routes window (List Routes)



- 2 Click on a route to view gateway route information corresponding to that particular route.
- 3 To modify the gateway route information, enter a description for the route in the Description text box.
- 4 From the drop down list, select a domain to associate with the route.
- 5 Click **Save**. (To clear the domain field, click **Clear**.) The Gateway Routes window appears.
- 6 A gateway can have multiple trunk groups. To reorder trunk groups associated to a route, click the **Re-Order** link next to the gateway route. The Reorder Trunk Groups window appears (Figure 90).

Figure 90 Reorder trunk groups window

- 7 In the Current Choices Available window, click **UP** or **DOWN** to reorder the trunk group.
- 8 Click **Reorder** to submit the trunk group reorder to the database.

Adding a gateway trunk group

The **AddTrunkGroup** menu option allows an administrator to add gateway trunk group information for a particular gateway. Multiple trunk groups can be assigned to a gateway route. If the first trunk group is busy or not available, then the next trunk group in the route is used. This option allows an administrator to view information for all available gateway trunk groups for a particular gateway.

To add a new trunk group

- 1 Click the **Gateways > AddTrunkGroup** menu option. The Create new trunkgroup window appears ([Figure 91](#)).

Figure 91 Create new trunkgroup window

The screenshot shows the Nortel Networks Provisioning interface. On the left is a navigation tree under 'Provisioning' with sub-items: Admins, Domains, Gateways, Add Gateway, List Gateways, Add Route, List Routes, Add TrunkGroup, and List TrunkGroups. The main content area is titled 'Create new trunkgroup' and contains the following fields and buttons:

- Gateway: 47.20.25.25 (dropdown menu)
- Route: Nortel (dropdown menu)
- Trunk Group: Local (text input field)
- Save button
- Clear button

- 2 Select a gateway from the Gateway drop down list.
- 3 Select a route from the Route drop down list.
- 4 Enter a description for the new trunk group in the text box provided.
- 5 Click **Save**. (To clear information from any of the fields, click **Clear**.) The TrunkGroups window appears (Figure 92).

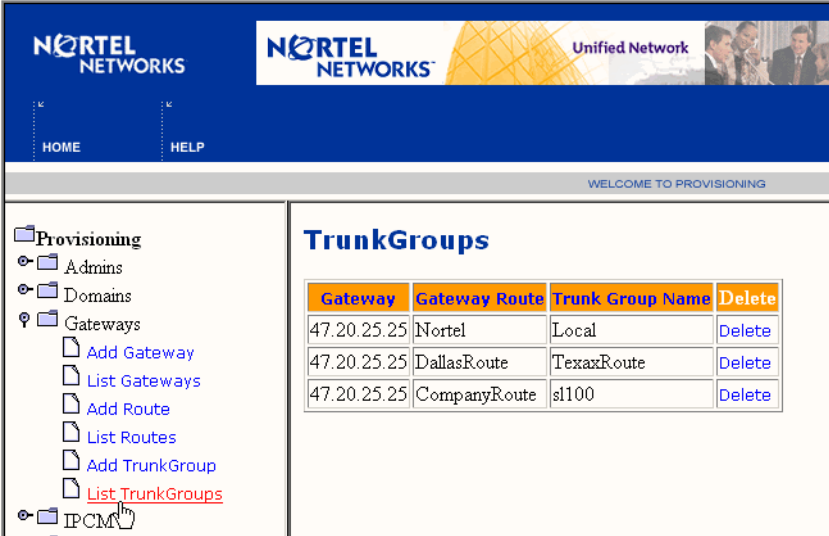
Figure 92 TrunkGroups window

| TrunkGroups | | | |
|-------------|---------------|------------------|--------|
| Gateway | Gateway Route | Trunk Group Name | Delete |
| 47.20.25.25 | Nortel | Local | Delete |

Listing a gateway trunk group

To view trunk group information

- 1 Click **Gateways > List TrunkGroups**. The TrunkGroups window appears (Figure 93).

Figure 93 TrunkGroups window (List TrunkGroup)

The screenshot shows the Nortel Networks Provisioning Client interface. The top header features the Nortel Networks logo and the text "Unified Network". Below the header, there are navigation links for "HOME" and "HELP". A "WELCOME TO PROVISIONING" message is displayed. The left sidebar contains a tree view with the following items:

- Provisioning
 - Admins
 - Domains
 - Gateways
 - Add Gateway
 - List Gateways
 - Add Route
 - List Routes
 - Add TrunkGroup
 - List TrunkGroups
 - IPCM

The main content area is titled "TrunkGroups" and displays a table with the following data:

| Gateway | Gateway Route | Trunk Group Name | Delete |
|-------------|---------------|------------------|------------------------|
| 47.20.25.25 | Nortel | Local | Delete |
| 47.20.25.25 | DallasRoute | TexasRoute | Delete |
| 47.20.25.25 | CompanyRoute | s1100 | Delete |

- 2 To delete a gateway trunk group, click the **Delete** link next to the gateway.
- 3 At the confirmation dialog box, click **OK** to delete the gateway trunk group information from the database for that particular gateway, or **Cancel** to return to the Trunk Groups window.

Appendix A Provisioning Sequence

This section describes the sequence of provisioning tasks.

Sequence of tasks

The following table provides a step by step provisioning sequence.

Table 22 SIP Provisioning Client provisioning sequence

| Step | Description | Task(s) | Menu option |
|------|---------------------------------------------------------------|-----------------------------------|--------------------------------------------------|
| 1. | Define roles and rights and use to create a new Administrator | Add role and assign rights | Admins Add Role |
| | | Add Admin | Admins Add Admin |
| 2. | Define new domain(s) | Add domain(s) | Domains Add Domain |
| | | Add sub domain(s) if required | Domains <i>domainxx.com</i> Add Sub-domain |
| 3. | Define service parameters and assign to domain(s) | Define service parameters | Services Define Service Parameters |
| | | Assign services to primary domain | Services Assign Services |

146 Provisioning Sequence

| Step | Description | Task(s) | Menu option |
|------|--------------------------------------------------------------------------------------------|-------------------------------------------------------------------------|---------------------------------------------------------------------------------|
| 4. | Define domain service package | Create service package | Domains <i>domainxx.com</i> Service Package Create Package |
| | | Assign service package(s) to domain(s) and sub-domain(s) | Domains <i>domainxx.com</i> Service Package Assign Package |
| | | Assign services to subdomain(s) | Domains <i>domainxx.com</i> Service Package Assign Services |
| 5. | Define voice mail servers and assign to domain(s) | Add voice mail server (SIP-/ Trunk-/Line-based) and assign to domain(s) | Voice Mail Add Voice Mail Server |
| 6. | If not deployed using the System Management Console, add IPCM cluster and assign to domain | Add IPCM cluster | IPCM Add IPCM Cluster |
| | | Assign IPCM to domain | IPCM List IPCM Cluster>Details>Assign to Domain>Assign Domain |
| 7. | Add domain status reason(s) | Add reason | Domains <i>domainxx.com</i> Status Reasons Add Reason |
| 8. | Add users to domain(s) | Add user(s) | Domains <i>domainxx.com</i> Users Add User |
| | | Add user(s) to subdomains | Domains <i>domainxx.com</i> SubDomain SubDomainYY Users Add User |
| | | Add i2004 device properties | |

| Step | Description | Task(s) | Menu option |
|------|------------------------------------------------------|------------------------------------------------|----------------------------------------------------------------------------|
| 9. | If not autoprovisioning, assign devices to domain(s) | Add device | Domains <i>domainxx.com</i> Devices Add Device |
| | | Assign users to a device | Domains <i>domainxx.com</i> Devices List Device Assign User |
| 10. | Define gateway, gateway routes, and trunk groups | Add gateway | Gateways Add Gateway |
| | | Add gateway route | Gateways Add Route |
| | | Add trunkgroup | Gateways Add TrunkGroup |
| 11. | Define domain telephony routes and parameters | Add routing Class of Service (COS) | Domains <i>domainxx.com</i> Telephony Routes Routing COS |
| | | Add telephony routes: Private, SIP, or Gateway | Domains <i>domainxx.com</i> Telephony Routes Add Telephony Route |
| | | Change routing parameters | Domains <i>domainxx.com</i> Telephony Routes List Telephony Route |
| | | Add route list | Domains <i>domainxx.com</i> Telephony Routes RouteList |
| 12. | Define banned users for a domain | Ban users | Domains <i>domainxx.com</i> Banned Users Ban User |

Appendix B

Understanding Telephony Routing and Translations

This section provides a basic understanding of how to provision telephony style translations interworking with traditional circuit-switched network routing.

Topics in this section:

- [“Telephony routing concepts” on page 149](#)
- [“Telephony routing terminology” on page 152](#)
- [“Sample dial plan for telephony routes” on page 155](#)
- [“Telephony route provisioning scenario and examples” on page 158](#)

Telephony routing concepts

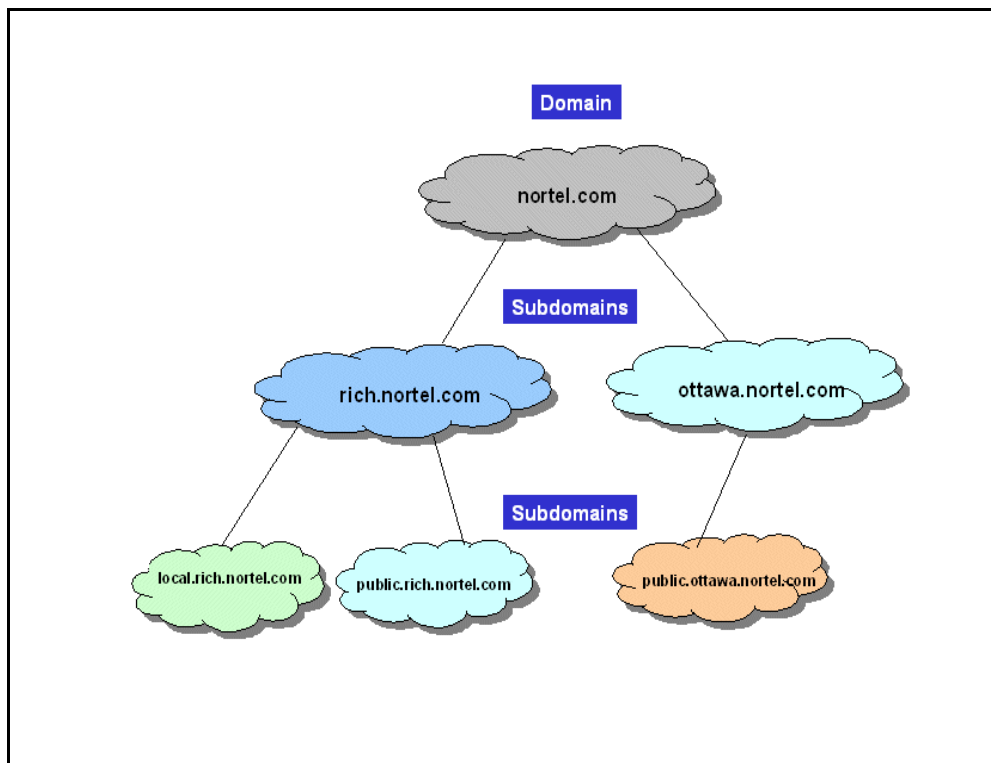
The basic function of any communication equipment is routing, and SIP-based systems add additional requirements to routing when compared to traditional circuit-switched network-based routing. Routing in the VoIP networks adds the complexity of integrating with the traditional circuit-switched networks and domain-based VoIP/SIP networks.

Subdomains

The concept of subdomains allows the domain administrator to group users into a smaller routing groups. For example, Nortel Networks can have a single system supporting a Richardson domain and an Ottawa domain. Users within Nortel Networks are assigned to subdomains, but can be reached by just using the top domain. The concept of SIP subdomains is similar to the email domains used within an enterprise.

The following figure shows an example of the subdomain concept (Figure 94).

Figure 94 Example domain tree



Tip: Subscribers do not have to be aware of subdomains. Subdomains are used only to group subscribers for partitioning telephony translations.

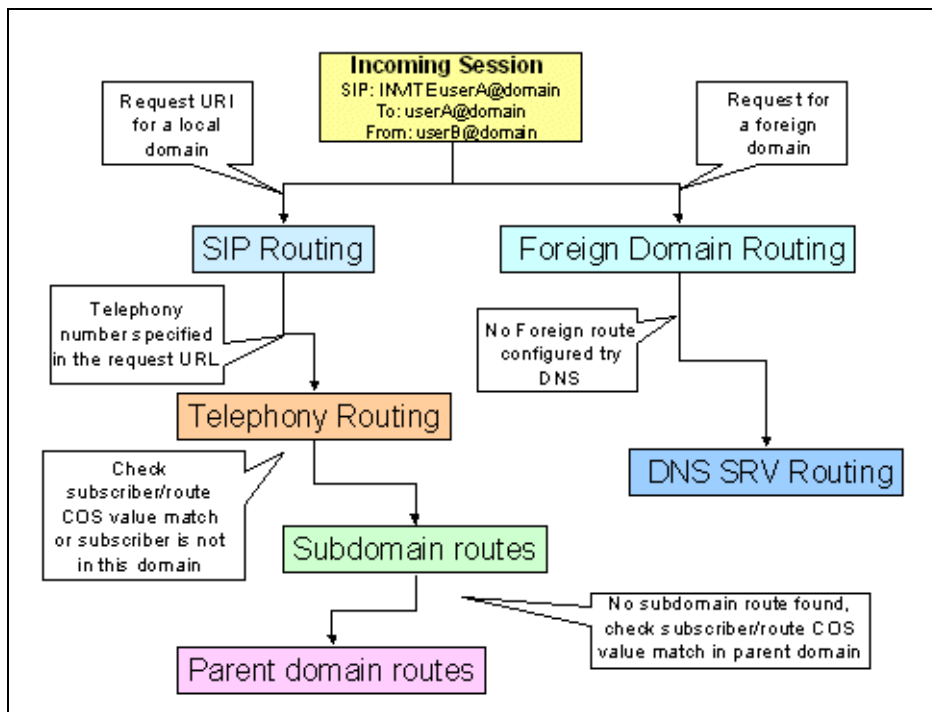
Class of Service

Class of Service (COS) screens and restricts users from accessing certain telephony routes within a domain and subdomain. The following restrictions are imposed on the use of COS:

- A single COS value can be assigned to a list of routes, domain, subdomain or a subscriber.
- A domain, subdomain or subscriber has access to the routes if the COS value equal or higher is assigned to a list of routes.
- The domain based COS value is used to determine the telephony routes for incoming sessions from gateways or other SIP networks. In this scenario the domain will not host the originator of the call so the domain's routing behavior is applied.
- If the **From:** header and **Request URI:** do not have the same domain, then the domain specified in the **Request URI:** is used to determine the telephony routes.
- If the **From:**, **To:**, and **Request URI:** domains are the same, then the subscriber's COS and the domain assigned to the subscriber is used to determine the telephony routes. If the user does not have any COS value then the domain's COS value in which the user actually exists is applied.
- The same COS values can be assigned to a domain, subdomain and a subscriber. This allows or restricts a subscriber in the subdomain to access routes in the parent and other subdomains.

SIP Application Module routing stages

The following figure (Figure 95) shows the relationship between the telephony routing stages provided by the SIP Application Module. If the COS value of the subscriber and subdomain route do not match, then the parent and subsequent subdomains are checked for routes with the same, or higher, COS value.

Figure 95 SIP Application Module routing stages

Telephony routing terminology

This section describes the following telephony routing terms:

- route lists
- routes

Route lists

Route lists consist of specific telephony routes for a local domain, gateways, or another SIP domain. The following additional options can be applied to routes in the route list:

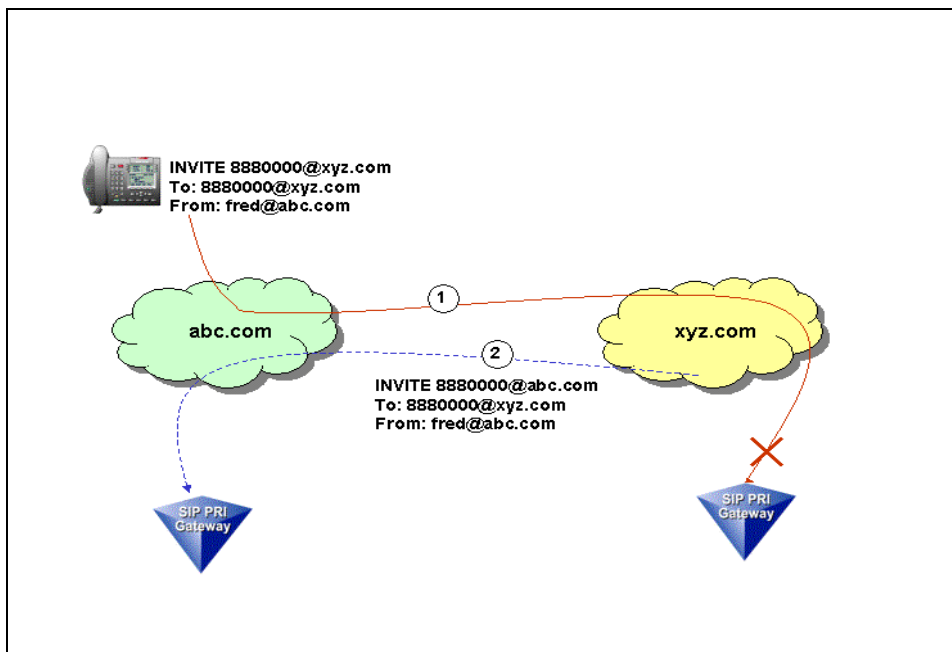
- allow all incoming sessions from other domains

- block all incoming sessions from other domains
- allow all incoming sessions from other subdomains
- block all incoming sessions from other subdomains

Example of telephony route redirection between domains

The following figure (Figure 96) shows an example of a telephony redirect imposed by a route list in the *xyz.com* domain when a session originates from the *abc.com* domain.

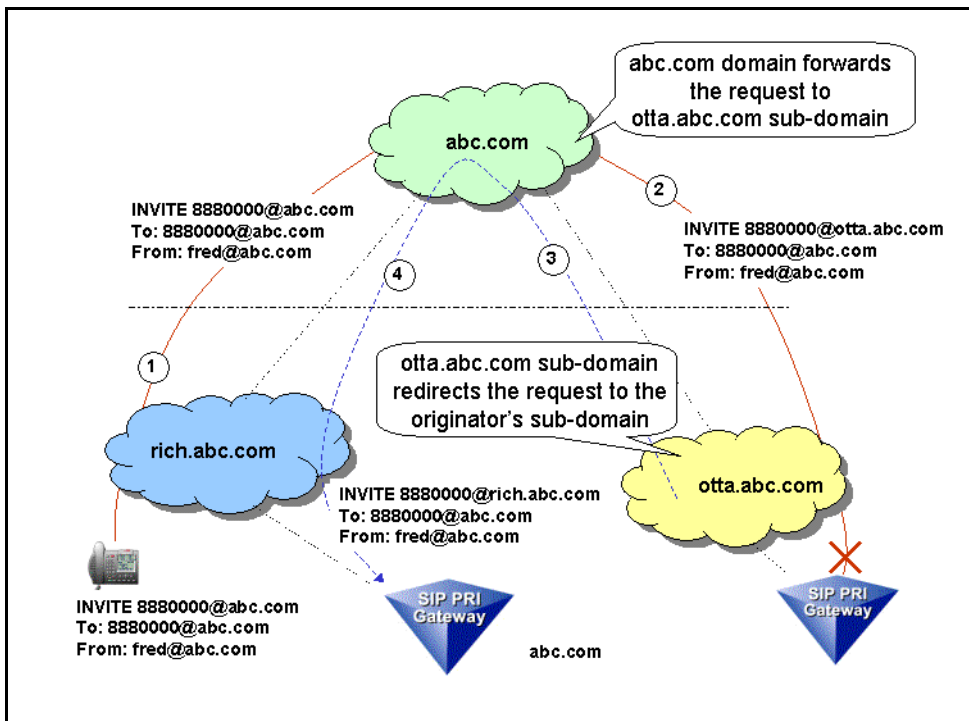
Figure 96 Route list redirect (domain to domain)



Example of telephony route redirection between subdomains

The following figure (Figure 97) shows an example of a telephony redirect between two subdomains. In this scenario the originating subscriber's subdomain is checked when a route list in the subdomain *otta.abc.com* sets the subdomain redirect option.

Figure 97 Route list redirect (between subdomains)



Routes

Telephony routes provide the actual dial plan to a particular destination. The telephony routing includes routes for a private digit dial plan, routes to gateways, and telephony style routing between SIP domains and subdomains. The three types of telephony routes available are (in order of telephony route selection):

- Private
- SIP
- Gateway

Private telephony routes

Private telephony routes are used for abbreviated telephony style digit dial plans. For example, private routes can be set up for a 5-digit dial plan between selected subscribers in a domain.

SIP and gateway routes

SIP and gateway routes provide access to a gateway, and SIP routes pointing to other SIP domains and subdomains for inter-SIP domain routing using telephony style dial plans.

Sample dial plan for telephony routes

This section provides a sample dial plan for provisioning the following telephone routes:

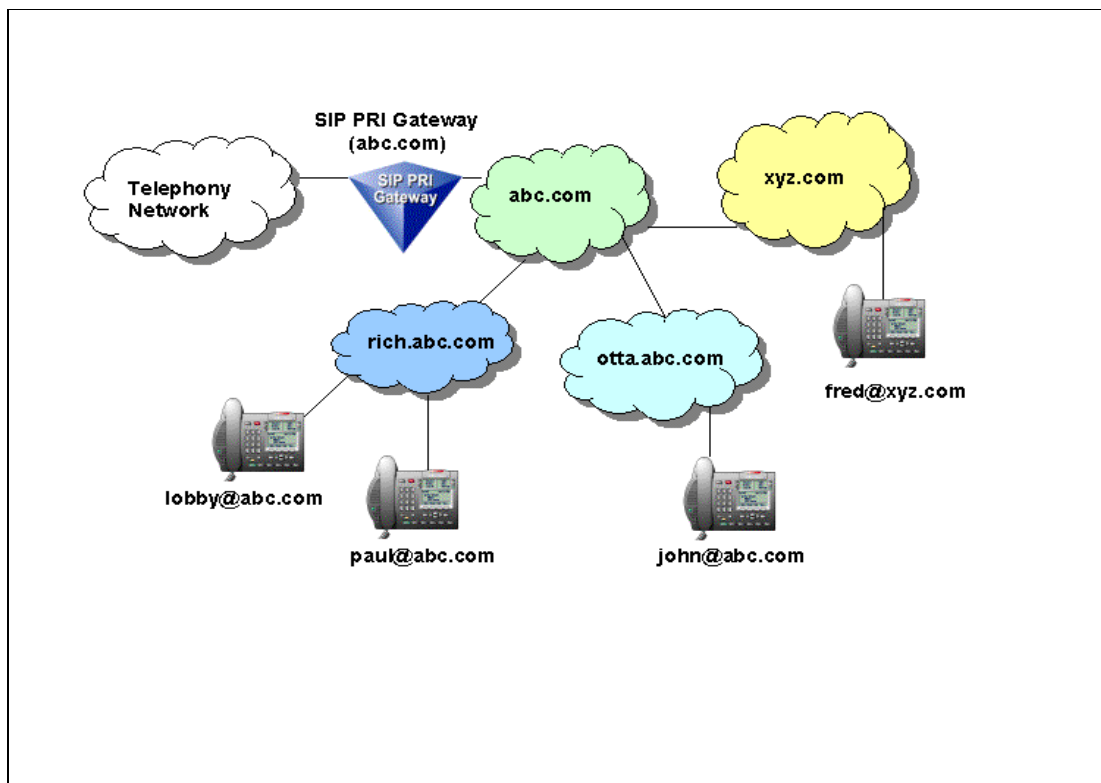
- Private telephony route for using 5-digit translations between subscribers.
- SIP telephony route for using PSTN style number plan to call between different domains.
- Gateway route to make local (PSTN) calls using 9+ prefix.

Provisioning domains, subscribers, and COS

Before implementing the dial plan a list of domains, subscribers, and COS must be provisioned for the following domains in the example:

- *abc.com* - (includes the *rich.abc.com* and *otta.abc.com* subdomains)
- *xyz.com*

The following figure ([Figure 98](#)) shows the domains and subscribers in the sample dial plan.

Figure 98 Domains/subscribers used for the sample dial plan

The domain *abc.com* defines the following three types of COS:

- lobby_dial_plan (lowest COS)
- local_dial_plan
- company_dial_plan (highest COS)

A subscriber that has *company_dial_plan* COS will also have access to routes defined for *lobby_dial_plan* and *local_dial_plan*. The domain *xyz.com* defines a COS, *all_dial_plans*, that is available to all subscribers in this domain.



Tip: It is assumed that each subscriber is assigned a 10-digit alias in all the domains.

Example dial plan assignments

The following table (Table 23) provides an example list of subscribers, subscriber aliases, and COS assignments in the domains.

Table 23 Example dial plan data

| Subscriber | Alias | Domain | Subdomain | COS |
|------------|------------|----------------|---------------------|-------------------|
| lobby | 9726850000 | <i>abc.com</i> | <i>rich.abc.com</i> | lobby_dial_plan |
| paul | 9726851000 | <i>abc.com</i> | <i>rich.abc.com</i> | local_dial_plan |
| john | 9726852000 | <i>abc.com</i> | <i>otta.abc.com</i> | company_dial_plan |
| fred | 2146842000 | <i>xyz.com</i> | | all_dial_plans |

Dial plan description

The dial plan for the two parent domains (*abc.com* and *xyz.com*) uses the various digit ranges as described in the following table (Table 24).

Table 24 Dial plan for *abc.com* and *xyz.com*

| Route Name | Description | From | To | Number digits | Domain | Subdomain | Type of Route |
|------------------|------------------------------------------------------------------------------------|------|----|---------------|----------------|---------------------|---------------|
| 5_digit_plan | 5+4 digit to reach subscribers in <i>rich.abc.com</i> | 5 | 5 | 5 | <i>abc.com</i> | <i>rich.abc.com</i> | Private |
| pstn_route | 9+10 digits to reach a PSTN subscriber. Strip the 9 before reaching the subscriber | 9 | 9 | 11 | <i>abc.com</i> | | Gateway |
| route_to_xyz.com | 6+7 digits to reach <i>xyz.com</i> from <i>abc.com</i> | 6 | 6 | 8 | <i>abc.com</i> | | SIP |
| route_to_abc.com | 6+7 digits to reach <i>abc.com</i> from <i>xyz.com</i> | 6 | 6 | 8 | <i>xyz.com</i> | | SIP |

The following provides further description of the dial plans:

- The 5+4 digit dial plan allows the subscribers in subdomain *rich.abc.com* to reach other subscribers in the same subdomain using only 5 digit dial plan (for example, 5+1000). Since the 5 digit dial plan is only required in the subdomain, it is assigned as a private route.
- The 9+10 digit dial plan is used to access the SIP PRI Gateway to terminate calls to a PSTN network, so this is assigned a gateway route.
- The 6+7 digit dial plan is used for communicating between two parent domains, therefore this digit range is assigned to SIP routes.

Associating COS and the route

The COS and routes described earlier in this section are tied together to provide the user restrictions in the following table (Table 25). This table shows that the *lobby_dial_plan* COS will only have access to the 5+4 digit dialing, therefore, restricting phones in a lobby to only 5 digit dialing only. The *company_dial_plan*, having the highest COS, will have access to all three routes..

Table 25 Association between COS and Routes

| COS | Routes |
|--------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <i>lobby_dial_plan</i> | 5+4 digit to reach subscribers in <i>rich.abc.com</i> |
| <i>local_dial_plan</i> | 5+4 digit to reach subscribers in <i>rich.abc.com</i> 9+10 digits to reach a PSTN subscriber. Strip the 9 before reaching the subscriber |
| <i>company_dial_plan</i> | 5+4 digit to reach subscribers in <i>rich.abc.com</i> 9+10 digits to reach a PSTN subscriber. Strip the 9 before reaching the subscriber 6+7 digits to reach <i>xyz.com</i> from <i>abc.com</i> |
| <i>all_dial_plans</i> | 6+7 digits to reach <i>abc.com</i> from <i>xyz.com</i> |

Telephony route provisioning scenario and examples

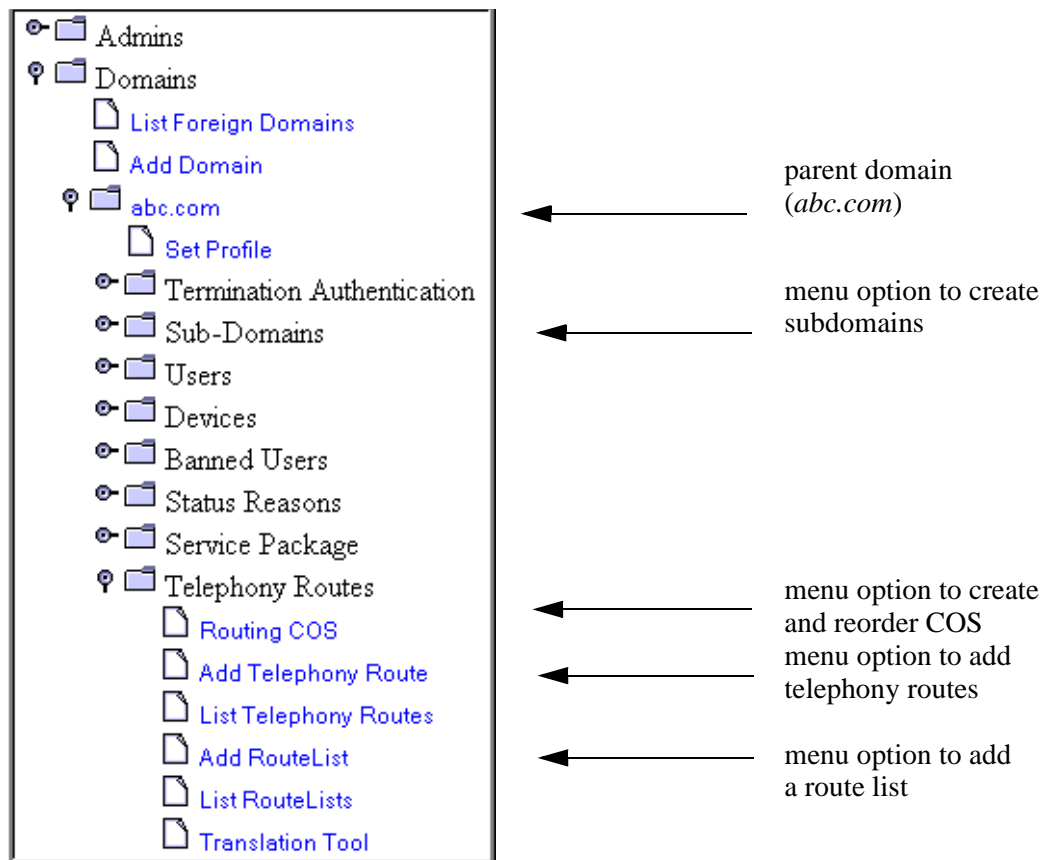
The steps in this section provide a scenario and sequence to follow when using the sample dial plan to provision telephony routing.

- “Step 1: Create the parent and subdomains” on page 159
- “Step 2: Create the COS” on page 161
- “Step 3: Create the subscribers and assign the COS” on page 162
- “Step 4: Create gateways, gateway routes and trunk groups” on page 165
- “Step 5: Create the telephony routes” on page 166
- “Step 6: Create the telephony route lists and assign COS” on page 174
- “Step 7: Test the telephony routing configuration” on page 178

Step 1: Create the parent and subdomains

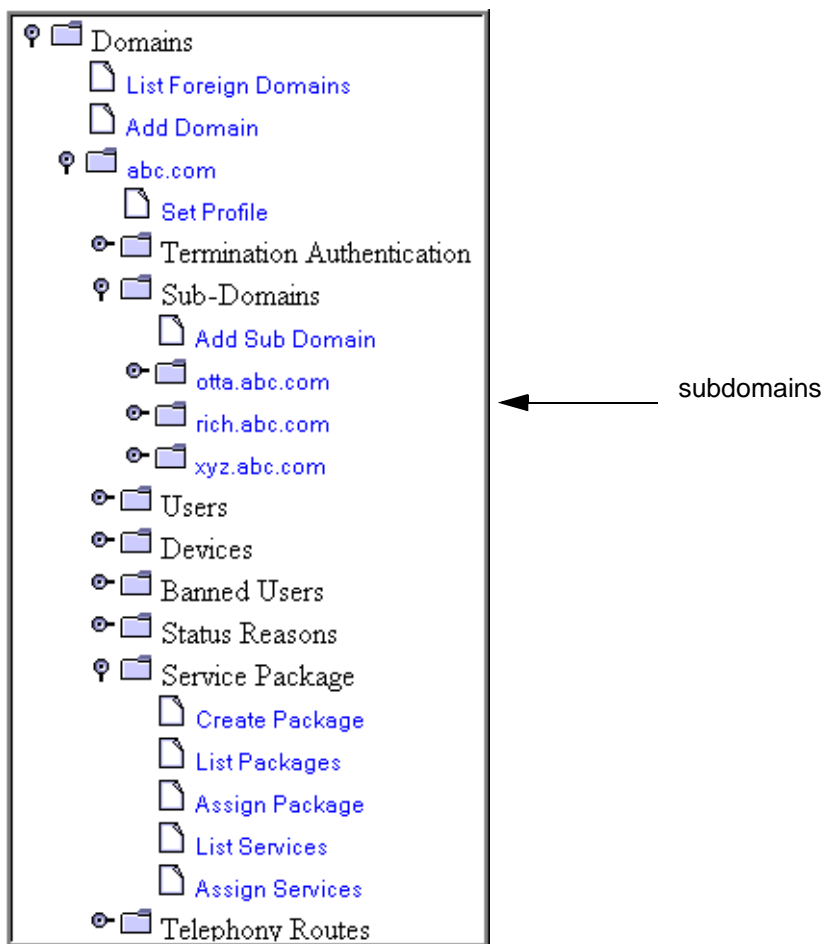
To create the parent and subdomains

- 1 Login in to the SIP Provisioning Client.
- 2 Click the **Domains > Add domain** menu option.
- 3 Create a domain called *abc.com*. See the following figure (Figure 99) and locate the parent domain in the SIP Provisioning Client.

Figure 99 Parent domain provisioning menu**4** Create the following subdomains:

- *rich.abc.com*
- *otta.abc.com*
- *xyz.abc.com*

The subdomains assume the default parameters from the parent domain. After adding the subdomains, the SIP Provisioning Client will look similar to the following figure (Figure 100).

Figure 100 Subdomain provisioning

Step 2: Create the COS

To create the COS

- 1 Click the **Domains > abc.com > Telephony Routes > Routing COS** menu option.

- 2 Add three COS to the parent domain *abc.com*, and re-order the COS as shown in the following figure (Figure 101). The highest COS is *company_dial_plan*.



Tip: The COS list is only created in the parent domain.

Figure 101 Creating COS

Add or Modify Class of Service Information

Name: ⓘ

Description: ⓘ

Reorder or Delete Class of Service

Current Choices Available

| | |
|-----------------------------------------|-------------------------------------|
| company_dial_plan company dial plan COS | <input type="button" value="UP"/> |
| local_dial_plan local dial plan COS | |
| lobby_dial_plan 5 digit dial plan | <input type="button" value="DOWN"/> |

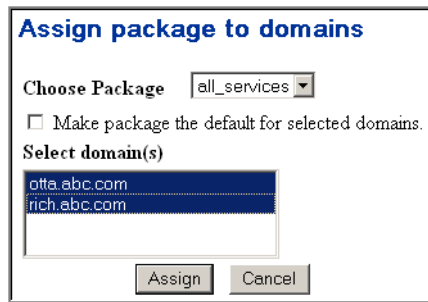
- 3 Click **Save**.

Step 3: Create the subscribers and assign the COS

To create the subscribers and assign the COS to the subscribers

- 1 Create a service package and assign the service package to the parent and subdomains. An example service package assignment is shown in the following figure (Figure 102).

Figure 102 Assigning service packages to subdomains



- 2 Create the subscribers for *rich.abc.com*, *otta.abc.com* and *xyz.com*. The following figure (Figure 103) shows a user called lobby added to the subdomain *rich.abc.com*.

Figure 103 Adding a subscriber with COS

Add new user

User Name: lobby * ?

First Name: Build 1 ?

Last Name: Lobby ?

Password: * ?

Confirm Password: * ?

Service package: all_services

Aliases: 972.6850000

Blended Aliases:

Status Reason: ACTIVE

email:

Business Phone: ?

Home Phone: ?

Cell Phone: ?

Pager: ?

Fax: ?

Voicemail Server: -None Selected-

Private Charge ID: 6850000

Public Charge ID: 9726850000

Voicemail: 9726850000

Class of Service: lobby_dial_plan

Time Zone: Pacific Standard Time

Locale: English

Save

SIP user

10 digit alias

10 digit calling number display used for PSTN calls

COS assigned to this user

- 3 Click **Save**.
- 4 Add the remaining subscribers to *rich.abc.com*, *otta.abc.com* and *xyz.com* domains.
- 5 Click **Save**.

Step 4: Create gateways, gateway routes and trunk groups

Before a gateway route can be assigned to a dial plan, the gateway and trunk groups must be created. A SIP PRI gateway is used with following parameters provisioned on the gateway:

- the gateway's IP address is 192.12.12.12.
- the trunk group assigned to domain *abc.com* is *pstn_trunk_group* for facing with the PSTN network.

This example uses a gateway route to terminate 9+10 digits to PSTN subscribers.

To create a gateway, gateway route and trunk group

- 1 Click the **Gateways > Add Gateway** menu option to create a logical link to the PRI Gateway. Use the data above to create the gateway as shown in the following figure (Figure 104).

Figure 104 Adding a gateway to *abc.com*



Create new gateway

Gateway Host: ⓘ

Gateway Type: ⓘ

- 2 Click the **Gateways > Add Route** menu option to provision a gateway route for *abc.com* (for example, Route to *abc.com*). This addition is shown in the following figure (Figure 105). Note that the route created here is used later in the dial plan to route calls to this gateway.

Figure 105 Adding a gateway route

- 3 Click the **Gateways > Add TrunkGroup** menu option to assign the gateway trunk group to this route as shown in the following figure (Figure 106).

Figure 106 Adding a trunk group to a gateway route

- 4 Click **Save**.

Step 5: Create the telephony routes

Using the dial plan defined earlier (Table 24), you can now create the telephony routes. First, create the private routes so that subscribers in subdomain *rich.abc.com* can reach each other using 5+xxxx digits.

Using the subscriber information in the following table (Table 26), a lobby phone (user = lobby) can reach user the user named paul by dialing 51000.

Table 26 Subscribers in *rich.abc.com*

| Subscriber | Alias | Subdomain | COS |
|------------|------------|---------------------|-----------------|
| lobby | 9726850000 | <i>rich.abc.com</i> | lobby_dial_plan |
| paul | 9726851000 | <i>rich.abc.com</i> | local_dial_plan |

Adding a private route to subdomain rich.abc.com

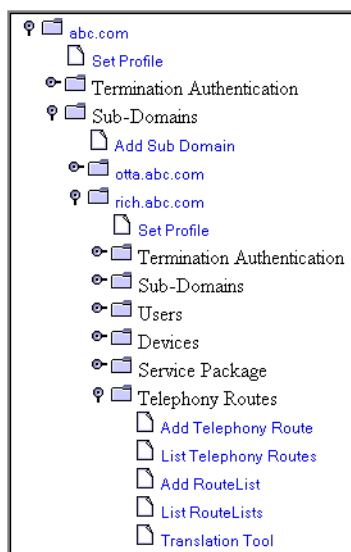


Note: Privates routes in this example are for the *rich.abc.com* subdomain only.

To add a private route

- 1 Click the **Domains > abc.com > rich.abc.com > Add Telephony Route** sub menu option as shown in the following figure (Figure 107).

Figure 107 Menu to add routes in subdomain



- 2 Provision a 5 digit dial plan as shown in the following figure (Figure 108). The prefix 97268 is used to create the 10 digit alias assigned to the subscriber.

Figure 108 Adding a private route to *rich.abc.com*

3 Click **Save**. The following window appears (Figure 109).

Figure 109 Private route created

| Name | From Digits | To Digits | Min Digits | Max Digits | Route Type | Remove | Prefix | Recursive | Delete |
|--------------|-------------|-----------|------------|------------|------------|--------|--------|-----------|--------|
| 5_digit_plan | 5 | 5 | 5 | 5 | Private | 0 | 97268 | No | Delete |

[Change Parameters](#)

4 Click the **Change Parameters** link.

5 Set the Subscriber Not Found option to **STOP**. The **STOP** option terminates the call if a subscriber with the prefix 97268+5 digits cannot be found in the *abc.com* domain.

Figure 110 Setting the Subscriber Not Found option

- 6 Click **Save**. The following window appears (Figure 111) indicating a 5-digit private route is created for subdomain *rich.abc.com*.

Figure 111 Listing a private route

| Telephony Route Listing for - rich.abc.com | | | | | | | | | |
|--------------------------------------------|-------------|-----------|------------|------------|------------|--------|--------|-----------|--------|
| Name | From Digits | To Digits | Min Digits | Max Digits | Route Type | Remove | Prefix | Recursive | Delete |
| 5_digit_plan | 5 | 5 | 5 | 5 | Private | 0 | 97268 | No | Delete |
| Subscriber Not Found: STOP | | | | | | | | | |
| Change Parameters | | | | | | | | | |

Adding SIP routes to abc.com and xyz.com

In this example, two SIP routes are added. The routes use the dial plan defined in the previous sections to communicate between *abc.com* and *xyz.com* domains using PSTN style dialing. The two SIP routes created to reach each domain are described in the following table (Table 27).

Table 27 Data for SIP routes

| Route Name | Description | From | To | Number digits | Domain | Type of Route |
|------------------|------------------------------------------|------|----|---------------|---------|---------------|
| route_to_xyz.com | 6+7 digits to reach xyz.com from abc.com | 6 | 6 | 8 | abc.com | SIP |
| route_to_abc.com | 6+7 digits to reach abc.com from xyz.com | 6 | 6 | 8 | xyz.com | SIP |



Tip: The SIP routes are added to the parent domains and not to the subdomains. This option allows the subdomains to access the common routes defined in the parent domain. For this example, *rich.abc.com* and *otta.abc.com* can use the same route defined in *abc.com*.

To create a SIP route

- 1 Click the **Domains > abc.com > Telephony Routing > Add Telephony Route** menu option. In this route, the 6 prefix is removed and 214 is prefixed

to determine the user alias in *xyz.com*. The following figure shows a SIP route added to the *abc.com* domain (Figure 112).

Figure 112 Adding a SIP route to abc.com

Create New Telephony Route

Name:

Description:

From Digits:

To Digits:

Min Number of Digits:

Max Number of Digits:

Route Type:

Remove:

Prefix:

Recursive:

Route List:

2 Click the **Change Parameters** link to set the route list options (Figure 113).

Figure 113 SIP route change parameters

Telephony Route Listing for - abc.com

| Name | From Digits | To Digits | Min Digits | Max Digits | Route Type | Remove | Prefix | Recursive | Delete |
|-----------------------------------|-------------|-----------|------------|------------|------------|--------|--------|-----------|------------------------|
| 5_digit_plan | 5 | 5 | 5 | 5 | Private | 0 | | No | Delete |
| Change Parameters | | | | | | | | | |
| route_to_xyz | 6 | 6 | 8 | 8 | SIP | 1 | 214 | No | Delete |
| Change Parameters | | | | | | | | | |

3 Provision the SIP route to point to the *xyz.com* domain as shown in the following figure (Figure 114). The **STOP** option terminates the call if a subscriber with the prefix 214+7 digits cannot be found in *xyz.com*.

Figure 114 SIP route options

Parameters

SIP Route:

Subscriber Not Found:

- 4 Click **Save**.
- 5 Click the **Domains > abc.com > Telephony Route > List TelephonyRoute** menu option to view a list of the telephony routes as shown in the following figure (Figure 115).

Figure 115 Listing SIP routes in *abc.com*

Telephony Route Listing for - abc.com

| Name | From Digits | To Digits | Min Digits | Max Digits | Route Type | Remove | Prefix | Recursive | Delete |
|-----------------------------------------------------------------------------------------------------|-------------|-----------|------------|------------|------------|--------|--------|-----------|--------|
| 5_digit_plan | 5 | 5 | 5 | 5 | Private | 0 | | No | Delete |
| Change Parameters | | | | | | | | | |
| route_to_xyz | 6 | 6 | 8 | 8 | SIP | 1 | 214 | No | Delete |
| SIP Route: xyz.com Subscriber Not Found: STOP Change Parameters | | | | | | | | | |

- 6 Add a SIP route to *xyz.com* to point to *abc.com* using the data shown in the following figure (Figure 116).

Figure 116 SIP route to *abc.com*

Telephony Route Listing for - xyz.com

| Name | From Digits | To Digits | Min Digits | Max Digits | Route Type | Remove | Prefix | Recursive | Delete |
|-----------------------------------------------------------------------------------------------------|-------------|-----------|------------|------------|------------|--------|--------|-----------|--------|
| route to abc | 6 | 6 | 8 | 8 | SIP | 1 | 972 | No | Delete |
| SIP Route: abc.com Subscriber Not Found: STOP Change Parameters | | | | | | | | | |

Adding a gateway route to *abc.com*

Add the dial plan to gateway route as shown in [Table 24](#) to domain *abc.com* (9+ 10 digits).

To add a gateway route to *abc.com*

- 1 Click the **abc.com > Telephony Routing > Add Telephony Route** menu option to assign the gateway route to a dial plan, In this route the 9 prefix is removed before sending the request to the PRI Gateway as shown in the following figure ([Figure 117](#)).

Figure 117 Adding a gateway route

Create New Telephony Route

Name: ?

Description: ?

From Digits: ?

To Digits: ?

Min Number of Digits: ?

Max Number of Digits: ?

Route Type: ?

Remove: ?

Prefix: ?

Recursive: ?

Route List:

- 2 Click **Save**.
- 3 Click the **Change Parameters** link on the route listing window ([Figure 118](#)).

Figure 118 Gateway change parameters

Telephony Route Listing for - xyz.abc.com

| Name | From Digits | To Digits | Min Digits | Max Digits | Route Type | Remove | Prefix | Recursive | Delete |
|-----------------------------------------------------------------------------------------------------|-------------|-----------|------------|------------|------------|--------|--------|-----------|--------|
| route_to_abc | 6 | 6 | 8 | 8 | SIP | 1 | 972 | No | Delete |
| SIP Route: abc.com Subscriber Not Found: STOP Change Parameters | | | | | | | | | |
| route_to_xyz | 6 | 6 | 8 | 8 | SIP | 1 | 972 | No | Delete |
| SIP Route: abc.com Subscriber Not Found: STOP Change Parameters | | | | | | | | | |
| pstrn_route | 9 | 9 | 11 | 11 | Gateway | 1 | | No | Delete |
| Change Parameters | | | | | | | | | |

- 4 Provision the gateway route and options as shown in the following figure (Figure 119). By selecting the **PUBLIC** option, the public charge ID assigned to a subscriber in *abc.com* is sent as the calling number to the PSTN.

Figure 119 Gateway route options

Parameters

OverRide Charge ID:

Gateway Route:

Gateway RouteType Option:

OverRide Charge Value:

- 5 Click the **List Telephony Routes** menu option to view the list of telephony routes. Notice that there are now two routes in this listing (Figure 120).

Figure 120 Listing *abc.com* routes

| Name | From Digits | To Digits | Min Digits | Max Digits | Route Type | Remove | Prefix | Recursive | Delete |
|----------------------------------------------------------------------------------------------------------------------------------------------------------------|-------------|-----------|------------|------------|------------|--------|--------|-----------|--------|
| 5_digit_plan | 5 | 5 | 5 | 5 | Private | 0 | | No | Delete |
| Change Parameters | | | | | | | | | |
| route_to_xyz | 6 | 6 | 8 | 8 | SIP | 1 | 214 | No | Delete |
| SIP Route: xyz.com Subscriber Not Found: STOP Change Parameters | | | | | | | | | |
| pstn_route | 9 | 9 | 11 | 11 | Gateway | 1 | | No | Delete |
| Gateway Route: Route for abc.com Gateway RouteType Option: PUBLIC OverRide Charge ID: N OverRide Charge Value: 0 Change Parameters | | | | | | | | | |

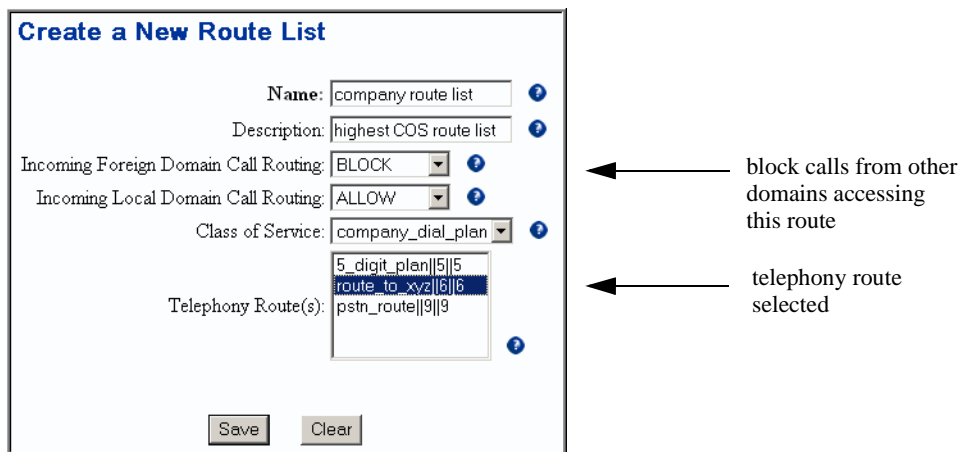
Step 6: Create the telephony route lists and assign COS

At this point, all the subscribers with COS and telephony routes have been created. However, there is no association between the subscribers and telephony routes. The route lists provide the association between subscribers COS and telephony routes that can be used by the subscribers.

To create the telephony route lists and assign COS to each list

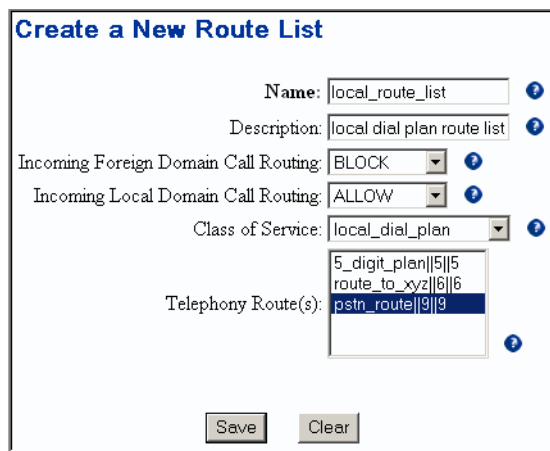
- 1 Create a route list named `abc_route_list` using the COS `company_dial_plan` and assign the two routes in *abc.com* to this route list.
- 2 Click the **Domains > abc.com > Telephony Routes > Add RouteList** menu option.
- 3 Enter the data as shown in the following figure (Figure 121). The **BLOCK** option blocks calls from other domains accessing the routes in this Route List. For example, a subscriber in *xyz.com* will be blocked if `9972685121@abc.com` is sent. Since the COS of this route list is `company_dial_plan`, only the subscriber named john in the *abc.com* domain will have access to this route list.

Figure 121 Adding a route list to *abc.com*



- 4 Create a route list for the local calling plan and assign it the local_dial_plan COS. This route will be available to subscribers named john and paul in the *abc.com* domain. Since local_dial_plan is a lower COS than company_dial_plan, both route lists are available to the subscriber named john (Figure 122).

Figure 122 Adding a route list with PSTN route to *abc.com*



- 5 Click the **Domains > abc.com > Telephony Routes > List RouteLists** menu option to view the route list as shown in the following figure (Figure 123).

Figure 123 Listing of *abc.com* route lists

| List of RouteLists for abc.com | | | | | |
|------------------------------------|----------------------------|---------------------|---------------------|-------------------|------------------------|
| Name | Description | Other Domain Action | Local Domain Action | Class of Service | Delete |
| company route list | highest COS route list | BLOCK | ALLOW | company_dial_plan | Delete |
| Associated Telephony Routes | | | | | |
| Route Name | From Digits | To Digits | Remove | Prefix | |
| route_to_xyz | 6 | 6 | 1 | 214 | |
| local_route_list | local dial plan route list | BLOCK | ALLOW | local_dial_plan | Delete |
| Associated Telephony Routes | | | | | |
| Route Name | From Digits | To Digits | Remove | Prefix | |
| pstn_route | 9 | 9 | 1 | | |

- 6 Create a route list in subdomain *xyz.com* and assign it the COS of *all_dial_plans*. The Route List of *xyz.com* is shown in the following figure (Figure 124).

Figure 124 Listing of *xyz.com* route lists

| Route Lists | | | | | |
|------------------------------------|------------------------|---------------------|---------------------|------------------|------------------------|
| List of RouteLists for xyz.abc.com | | | | | |
| Name | Description | Other Domain Action | Local Domain Action | Class of Service | Delete |
| for_all_users | all dial plans in here | ALLOW | ALLOW | all_dial_plans | Delete |
| Associated Telephony Routes | | | | | |
| Route Name | From Digits | To Digits | Remove | Prefix | |
| route_to_abc | 6 | 6 | 1 | 972 | |

- 7 Click the **Domains > abc.com > rich.abc.com > Telephony Routes > Add RouteList** menu option to add a route list that contains the private route

created earlier for the 5 digit dialing between the subscribers in the *rich.abc.com* subdomain. The following figure (Figure 125) shows the new route list.

This route will be available to subscribers named paul and lobby in the subdomain *rich.abc.com* only. Since the COS for lobby_dial_plan is lower than the COS for local_dial_plan, subscriber paul has access to this route list.

Figure 125 Adding a route list to *rich.abc.com*

Create a New Route List

Name:

Description:

Incoming Foreign Domain Call Routing:

Incoming Local Domain Call Routing:

Class of Service:

Telephony Route(s):

8 Click **Save**. The following figure appears (Figure 126).

Figure 126 Listing of *rich.abc.com* route lists

| Route Lists | | | | | |
|-------------------------------------|-------------------------------|---------------------|---------------------|------------------|--------|
| List of RouteLists for rich.abc.com | | | | | |
| Name | Description | Other Domain Action | Local Domain Action | Class of Service | Delete |
| 5_digit_plan | 5_digit_plan for rich.abc.com | BLOCK | BLOCK | lobby_dial_plan | Delete |
| Associated Telephony Routes | | | | | |
| Route Name | From Digits | To Digits | Remove | Prefix | |
| 5_digit_plan | 5 | 5 | 0 | 97268 | |

Step 7: Test the telephony routing configuration

At this point all subscribers, COS, route, and route lists have been provisioned. Before making any calls, telephony routing can be verified using the Translations Tool menu option available in the domains and subdomains.

To check the telephony routing

- 1 Click the **abc.com > rich.abc.com > TelephonyRoutes > Translations Tool** menu option. Verify that the subscriber named lobby can reach the subscriber named paul by dialing 51000.

The data for translations trace and results are shown in the following figure (Figure 127).

Figure 127 Subdomain 5 digit translations trace

The screenshot shows the 'Translation Verification Tool' interface. It has three input fields: 'Request URI' with '51000@abc.com', 'To' with '51000@abc.com', and 'From' with 'lobby@abc.com'. Below these are 'Execute' and 'Reset' buttons. To the right, two arrows point to the '51000@abc.com' and 'lobby@abc.com' fields with labels 'Paul's 5 digit extension' and 'Calling from subscriber lobby' respectively. Below the form is a table with the following data:

| | |
|------------------|-----------------------------|
| Route | 5_digit_plan |
| RouteType | Private |
| RouteList | 5_digit_plan |
| Class of Service | lobby_dial_plan |
| RouteList Domain | rich.abc.com |
| Destinations | sip:9726851000@rich.abc.com |

- 2 Verify that the subscriber paul can reach the PSTN network using the 9+10 digit dial plan. The data for translations trace and results are shown in the following figure (Figure 128).

Figure 128 Gateway route translations trace

Translation Verification Tool

Request URI:

To:

From:

| | |
|-----------------------|----------------------------------------------------------------------------------|
| Route | pstn_route |
| Route Type | Gateway |
| RouteList | local_route_list |
| Class of Service | local_dial_plan |
| RouteList Domain | abc.com |
| Gateway Route Type | PUBLIC |
| OverRideCharge ID | N |
| OverRide Charge Value | 0 |
| Destinations | sip:9724440000@abc.com;norteldevice=pri;norteltrkgrp=pstn_trunk_group;user=phone |

calling number to PSTN PRI gateway trunk group IP address of the PRI gateway

- 3 Verify that subscriber named john in subdomain *otta.abc.com* can reach the SIP domain *xyz.com* subscriber named fred with the 214684200 alias using the 6+8 digit dial plan. The data and results are shown in the following figure (Figure 128).

Figure 129 SIP route trace from *abc.com* to *xyz.com*

Translation Verification Tool

Request URI:

To :

From :

| | |
|------------------|------------------------|
| Route | route_to_xyz |
| RouteType | SIP |
| RouteList | company route list |
| Class of Service | company_dial_plan |
| RouteList Domain | abc.com |
| Destinations | sip:2146842000@xyz.com |

Fred's 10 digit alias

- 4 Verify that subscriber named paul in subdomain *rich.abc.com* cannot reach the SIP domain *xyz.com* subscriber named fred with the 214684200 alias using the 6+8 digit dial plan. The data and results are shown in the following figure (Figure 130). Paul only has a COS of *local_dial_plan*, therefore the *route_to_xyz* route list is not available.

Figure 130 Restricted route trace

Translation Verification Tool

Request URI:

To :

From :

No Destinations Found For The Information Entered

Appendix C

911 Emergency Notification



Note: You must have telephony routing and domain management provisioning rights to perform the tasks in this section.

This section describes how to set up 911 emergency notification using the SIP Provisioning Client and the SIP Personal Agent.

Topics in this section:

- [“911 emergency notification terms and concepts” on page 181](#)
- [“Setting up Instant Message Notifications” on page 182](#)
- [“Using the SIP Provisioning Client to add emergency subscribers and telephony routes” on page 183](#)
- [“Using the SIP Personal Agent to route 911 emergency notifications” on page 192](#)

911 emergency notification terms and concepts

The SIP Application Module supports sending Instant Message (IM) notifications to a specified On Site Notification (OSN) subscriber whenever a user makes a call to an emergency number such as 911. This service is provided by using the same mechanism that allows users to push web pages and/or email links back to the originator of a call.

A Public Safety Answering Point (PSAP) is an agency or group of agencies designated and authorized to receive and respond to 911 emergency calls requiring one or more public services: Police, Fire, Emergency Medical Service, or all three.

The OSN is required so that the OSN staff can prepare for the arrival of the emergency response team, and direct them to the location of the 911 caller when they arrive on site.

An emergency subscriber is the logical representation of a PSAP/OSN pair. When a user dials an emergency number (for example, 911), the call is mapped (using telephony routing rules) to the correct emergency subscriber for that subdomain. The routing rules for that emergency subscriber (created in the SIP Personal Agent) then instruct the SIP Application Module to both route the call to the correct PSAP and to send an Instant Message notification to the correct OSN subscriber. There is, therefore, a one-to-one relationship between a subdomain and the PSAP/OSN pair.

Setting up Instant Message Notifications

In order to properly set up IM notifications to an OSN subscriber, the domain administrator should divide the domain into geographical areas. A subdomain should then be created for each of these areas. If there is only one geographical area, no subdomains need to be created.

When a subscriber places an emergency call, the OSN subscriber assigned to the same subdomain (area) as the originator is notified. It is up to the administrator to decide how large or small to make the subdomain (area).

To set up Instant Message (IM) notifications, the following information must be provisioned using both the SIP Provisioning Client and the SIP Personal Agent.

| Use this client... | To set up... |
|-------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| SIP Provisioning Client | <ul style="list-style-type: none"> • an emergency subscriber per subdomain • an OSN subscriber per subdomain • the 911 route in each subdomain to point to each emergency subscriber for that subdomain • the PSAP route to route to the correct PSAP for each subdomain |
| SIP Personal Agent | routing rules to the emergency subscriber for <ul style="list-style-type: none"> • an emergency telephone number to route the call to the proper gateway serving the PSAP. • the OSN subscriber to receive the IM notification. See the <i>SIP Personal Agent Getting Started Guide</i> for more information on using the SIP Personal Agent. |



Tip: Since telephony routes are only unique within a subdomain, you cannot have more than one OSN location per subdomain.

Using the SIP Provisioning Client to add emergency subscribers and telephony routes

Use the SIP Provisioning Client for the following:

- [“Creating a COS for the 911 notification” on page 184](#)
- [“Adding a gateway, gateway route, and trunk group” on page 185](#)
- [“Adding a private telephony route” on page 187](#)
- [“Adding a gateway telephony route” on page 189](#)
- [“Adding an emergency subscriber” on page 190](#)
- [“Adding an OSN subscriber” on page 192](#)
- [“Setting up a route list for the emergency subscriber” on page 192](#)

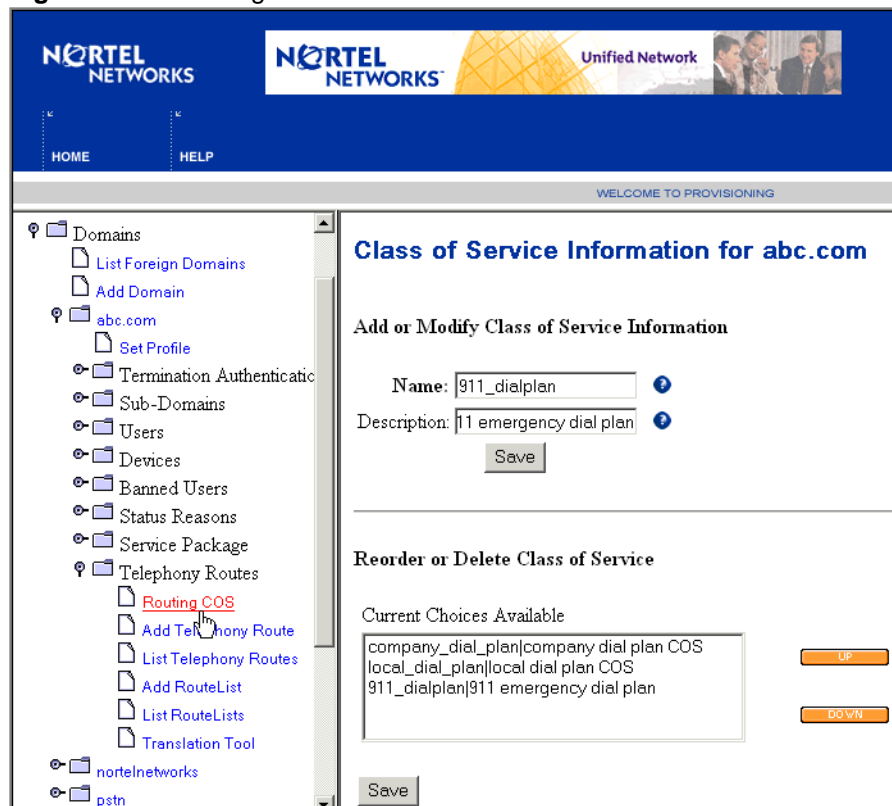
- [“Setting up a rule for the emergency notification route” on page 193](#)

Creating a COS for the 911 notification

As a first step, create a new COS for the 911 notification and assign it to the parent domain.

To create the COS

- 1 Click the **Domains** menu option. Click the subdomain for which you want to set up the COS (for example, *rich.abc.com*)
- 2 Click the **Telephony Routes > Routing COS** menu option.
- 3 Enter a name and a description for the COS.
- 4 Click **Save**. The COS appears in the Current Choices Available window ([Figure 131](#)).

Figure 131 Creating a COS for the 911 notification

- 5 Click **UP** or **DOWN** to reorder the COS in the Current Choices Available window. The COS for 911 emergency notification should be the highest in the reorder list. This indicates that the COS has a lower number, which means a higher COS value, or service, and ensures that everyone will receive 911 emergency notification support.

Adding a gateway, gateway route, and trunk group

To create a PSAP gateway, gateway route, and trunk group per domain or subdomain

- 1 Click the **Gateways > Add Gateway** menu option to create a logical link to the PRI gateway.
- 2 Enter the IP address or MADDR (short address) of the gateway (Figure 132).

Figure 132 Adding a new gateway

Create new gateway

Gateway Host: ⓘ

Gateway Type: ⓘ

3 Click **Save**. The new gateway appears in the list of gateways ([Figure 133](#)).

Figure 133 List of gateways

| Gateway Host | Gateway Type | Delete |
|---------------------------------------------------|--------------|------------------------|
| rich.abc.com;maddr=47.102.128.182 | pri | Delete |
| abc.com:maddr=192.12.10.12 | PRI | Delete |
| 47.102.128.75 | pri | Delete |

4 Click the **Gateways > Add Route** menu option to provision a gateway route for the parent domain, for example, Route for *abc.com* ([Figure 134](#)). The route created here will route 911 calls to this gateway.

Figure 134 Creating a new gateway route

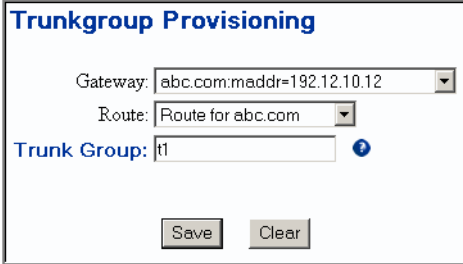
Create new gateway route

Route Name: ⓘ

Domain:

5 Click **Save**.

6 Click the **Gateways > Add TrunkGroup** menu option to assign the gateway trunk group to this route ([Figure 135](#)).

Figure 135 Creating a new trunkgroup

The screenshot shows a web form titled "Trunkgroup Provisioning". It contains three main input fields: "Gateway:" with a dropdown menu showing "abc.com:maddr=192.12.10.12", "Route:" with a dropdown menu showing "Route for abc.com", and "Trunk Group:" with a text input field containing "t1" and a blue help icon to its right. At the bottom of the form are two buttons: "Save" and "Clear".

Adding a private telephony route

An emergency telephony route must be set up that is based on the unique emergency subscriber name provisioned per domain or subdomain.

To create a private telephony route for when a user dials 911

- 1 Click the **Domains** menu option.
- 2 Click the domain subdomain for which you want to set up the private telephony route (for example, *rich.abc.com*)
- 3 Click the **Telephony Routes > Add Telephony Route** menu option. Complete the fields for the new telephony route in the The Create New Telephony Route window (Figure 136). The prefix field (this is the number that goes in front of the 911) must be changed based on the username of the pseudo emergency subscriber. The prefix field is only field that is different between every telephony route in every subdomain.

Figure 136 Create new telephony route window

Create New Telephony Route

Name: 911 private route

Description: 911 private route

From Digits: 911

To Digits: 911

Min Number of Digits: 3

Max Number of Digits: 3

Route Type: Private

Remove: 0

Prefix: 55

Recursive: No

Route List: 911_routelist, 5_digit_plan

Save Clear

- 4 Click **Save**. The Telephony Route Listing for the domain appears (for example, *abc.com* domain).
- 5 Click the **Change Parameters** link and select **PRE-TRANSLATE** from the drop down list. Click **Save** (Figure 137).

Figure 137 Change parameters

Parameters

Subscriber Not Found: PRE-TRANSLATE

Save Clear

- 6 Click the **Telephony Routes > Add RouteList** menu option. Complete the fields for the new telephony route in the The Create New Telephony Route window (Figure 138).

Figure 138 Creating a new route list

Create a New Route List

Name: 911 routelist

Description: 911 route list

Incoming Foreign Domain Call Routing: ALLOW

Incoming Local Domain Call Routing: ALLOW

Class of Service: 911_dialplan

Telephony Route(s):

- 911 private route|911|911
- 911 PSAP Route|911|911

Save Clear

Adding a gateway telephony route

To add a gateway telephony route that routes to the proper gateway serving the PSAP for the OSN location when a user dials 911

- 1 Click the **abc.com > Telephony Routing > Add Telephony Route** menu option to assign the gateway telephony route for the PSAP route.
- 2 Complete the fields for the telephony gateway route to the PSAP in the Create New Telephony Route window (Figure 139). The number in the From Digits and To Digits fields (for example, 911) corresponds to the number that will be provisioned in the SIP Personal Agent route list.

Figure 139 Create gateway telephony route to PSAP

Create New Telephony Route

Name: 911 PSAP Route

Description: 911 PSAP route

From Digits: 6911

To Digits: 6911

Min Number of Digits: 4

Max Number of Digits: 4

Route Type: Gateway

Remove: 1

Prefix:

Recursive: No

Route List:

- 3 Click **Save**.
- 4 Click the **Change Parameters** link on the route listing window.
- 5 Provision the gateway route and options as shown in the following figure (Figure 140). By selecting the **PUBLIC** option, the public charge ID assigned to a subscriber in abc.com is sent as the calling number to the PSTN.

Figure 140 Gateway route options

Parameters

OverRide Charge ID: N

Gateway Route: Route for abc.com

Gateway RouteType Option: PUBLIC

OverRide Charge Value:

Save Clear

- 6 Click the **List Telephony Routes** menu option to view the list of telephony routes. Notice that there are now two routes in this listing.

Adding an emergency subscriber

To add an emergency subscriber to a domain or subdomain

- 1 Click the **Domains** menu option.

- 2 Select a domain (or a subdomain of a domain).
- 3 Click the **User > Add User** sub menu option. The Add new user window appears (Figure 141).

Figure 141 Add user - emergency subscriber

Add new user

User Name: 55911 * ?

First Name: ?

Last Name: ?

Password: * ?

Confirm Password: * ?

Service package: all_services

Aliases:

Blended Aliases:

Status Reason: ACTIVE

email:

Business Phone: ?

Home Phone: ?

Cell Phone: ?

Pager: ?

Fax: ?

Voicemail Server: -None Selected-

Private Charge ID:

Public Charge ID:

Voicemail:

Class of Service: 911_dialplan

Time Zone: Pacific Standard Time

Locale: English

Save

- 4 Enter a name for the emergency subscriber (for example, 55911).
- 5 Enter and confirm a password for the subscriber.
- 6 Enter additional information for the emergency subscriber (optional).
- 7 Select the COS for the subscriber that you created earlier.

- 8 Click **Save**. A message appears that the user was added successfully.

Adding an OSN subscriber

To add an OSN subscriber to a domain or subdomain

- 1 Click the **Domains** option.
- 2 Select a domain (or a subdomain of a domain).
- 3 Click the **User > Add User** sub menu option. The Add new user window appears.
- 4 Enter a name for the OSN subscriber (for example, guarddeskA).
- 5 Enter and confirm a password for the OSN subscriber (required).
- 6 Enter additional information for the OSN subscriber (optional).
- 7 Click **Save**. A message appears that the user was added successfully.

Using the SIP Personal Agent to route 911 emergency notifications

The instant message sent to the Guard Desk is provisioned through the Personal Agent just as you would if you wanted to push a web page back to the originator while the terminator is being alerted.

Use the SIP Personal Agent to add a new rule to the emergency subscriber's Call Manager route list.

Setting up a route list for the emergency subscriber

To set up a route and ring list to send an IM to a specific OSN location and to call an emergency number at the same time

- 1 Log in to the SIP Personal Agent.
- 2 Click the **Call Manager** icon on the left of the SIP Personal Agent screen. The Welcome to Call Manager home screen displays.

- 3 Click **ListRoutes** tab from the Call Manager Home screen. The Ring List Details screen displays.
- 4 Enter a name for the emergency route in the **Ring list name** text box (for example, Emergency notification).
- 5 From the drop down list, select the maximum number of times to ring the phone.
- 6 In the first text box in the column, enter a unique emergency number. This number must correspond to the number provisioned in the gateway telephony route that routes to the proper gateway serving the PSAP for the OSN location (for example, 6911).
- 7 Select **SIP address** from the drop down list.
- 8 In the second text box in the first column, enter the SIP address of the OSN subscriber. This is name of the target user who receives the IM notification, for example:

 sip:guarddeskA@abc.com.
- 9 Select Web address from the drop down list.
- 10 Click **Add Route**. The **Add Route** button updates the online user database with your the emergency subscriber and routing information.

Setting up a rule for the emergency notification route

Once a route is set up, a screening rule must be created that uses the defined emergency notification ring list to define the activation of 911 calls.



Tip: The 911 notification is not activated until a rule has been applied.

To set up a screening rule for the first time

- 1 Click **ListRules** from the Call Manager screening and routing screen. The Welcome to Call Manager screen displays.

- 2 Click the [create a screening rule](#) link. The Call Manager Rule Details screen displays.
- 3 Enter a name for emergency rule in the **Rule name** text box (for example, 911 rule).
- 4 Click the **Active** check box to select that the rule should appear in the Active Rules window.
- 5 Select the emergency route from the pull-down menu next to the Active route field.
- 6 Click **Save Rule**.



Tip: If you selected the Active check box in the Call Manager Rule Details screen, then the rules you just programmed now appear in the Active rules window in the Call Manager Home screen. Every time 911 is called, the Call Manager checks the rules to match and route the 911 calls appropriately.

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