SIPxNano

IP-PBX Server

Getting Started Guide

Important

This is a Class A device and is intended for use in a light industrial environment. It is not intended nor approved for use in an industrial or residential environment.

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Trademark Statement

Important Information

To use virtual private network (VPN) and/or AES/DES/3DES encryption capabilities with the SIPxNano, you may need to purchase additional licenses, hardware, software, network connection, and/or service. Contact sales@patton.com or +1 (301) 975-1000 for assistance.

Warranty Information

The software described in this document is furnished under a license and may be used or copied only in accordance with the terms of such license. For information about the license, go to **www.patton.com**.

Patton Electronics warrants all SIPxNano components to be free from defects, and will—at our option—repair or replace the product should it fail within one year from the first date of the shipment.

This warranty is limited to defects in workmanship or materials, and does not cover customer damage, abuse or unauthorized modification. If the product fails to perform as warranted, your sole recourse shall be repair or replacement as described above. Under no condition shall Patton Electronics be liable for any damages incurred by the use of this product. These damages include, but are not limited to, the following: lost profits, lost savings and incidental or consequential damages arising from the use of or inability to use this product. Patton Electronics specifically disclaims all other warranties, expressed or implied, and the installation or use of this product shall be deemed an acceptance of these terms by the user.

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About this guide

This guide describes the SIPxNano hardware, installation and basic configuration.

Audience

This guide is intended for the following users:

- Operators
- System administrators
- Network engineers

Structure

This guide contains the following chapters and appendices:

- Chapter 1, "General Introduction" on page 6 provides information about hardware and factory defaults.
- Chapter 2, "Setting Up the SIPxNano" on page 9 contains information on setting up the SIPxNano.
- Chapter 3, "Users" on page 14 explains how to set up users and user groups.
- Chapter 4, "Devices" on page 25 provides information on setting up devices.
- Chapter 5, "Features" on page 40 contains an overview of setting up and configuring features.
- Chapter 6, "System" on page 56 contains information about system settings.
- Chapter 7, "Diagnostics" on page 73 explains diagnostic functions.
- Chapter 8, "Voicemail" on page 78 provides information about using voicemail.
- Chapter 9, "Contacting Patton for assistance" on page 83 contains information on how to contact Patton for assistance with the SIPxNano.
- Appendix A, "Session Initiation Protocol (SIP)" on page 86 contains an overview of SIP.
- Appendix B, "Firewalls and NAT" on page 95 provides information on firewalls and NAT.
- Appendix C, "Configuration APIs" on page 100 describes Configuration APIs.

For best results, read the contents of this guide *before* you install the server.

Precautions

Notes, cautions, and warnings, which have the following meanings, are used throughout this guide to help you become aware of potential problems. ***Warnings*** are intended to prevent safety hazards that could result in personal injury. ***Cautions*** are intended to prevent situations that could result in property damage or impaired functioning.

Note A note presents additional information or interesting sidelights.

IMPORTANT

The alert symbol and IMPORTANT heading calls attention to important information.

CAUTION

The alert symbol and CAUTION heading indicate a potential hazard. Strictly follow the instructions to avoid property damage.

CAUTION

The shock hazard symbol and CAUTION heading indicate a potential electric shock hazard. Strictly follow the instructions to avoid property damage caused by electric shock.

WARNING

The alert symbol and WARNING heading indicate a potential safety hazard. Strictly follow the warning instructions to avoid personal injury.

WARNING

The shock hazard symbol and WARNING heading indicate a potential electric shock hazard. Strictly follow the warning instructions to avoid injury caused by electric shock.

Safety when working with electricity

WARNING

This device contains no user serviceable parts. The equipment shall be returned to Patton Electronics for repairs, or repaired by qualified service personnel.

WARNING

The external power adapter shall be a listed Limited Power Source. Ensure that the power cable used meets all applicable standards for the country in which it is to be installed, and that it is connected to a wall outlet which has earth ground. The mains outlet that is utilized to power the devise shall be within 10 feet (3 meters) of the device, shall be easily accessible, and protected by a circuit breaker.

WARNING

Hazardous network voltages are present in WAN ports regardless of whether power to the SIPxNano is ON or OFF. To avoid electric shock, use caution when near WAN ports. When detaching cables, detach the end away from the SIPxNano first.

WARNING

Do not work on the system or connect or disconnect cables during periods of lightning activity.

In accordance with the requirements of council directive 2002/96/EC on Waste of Electrical and Electronic Equipment (WEEE), ensure that at end-of-life you separate this product from other waste and scrap and deliver to the WEEE collection system in your country for recycling.

General observations

- Clean the case with a soft slightly moist anti-static cloth
- Place the unit upright in the stand (included) to ensure free air circulation
- Avoid exposing the unit to direct sunlight and other heat sources
- Protect the unit from moisture, vapors, and corrosive liquids

Typographical conventions used in this document

This section describes the typographical conventions and terms used in this guide.

General conventions

The procedures described in this manual use the following text conventions:

Table 1. General conventions

<table>
<tr><th>Convention</th><th>Meaning</th></tr>
<tr><td>Garamond blue type</td><td>Indicates a cross-reference hyperlink that points to a figure, graphic, table, or section heading. Clicking on the hyperlink jumps you to the reference. When you have finished reviewing the reference, click on the Go to Previous View button in the Adobe® Acrobat® Reader toolbar to return to your starting point.</td></tr>
<tr><td>Futura bold type</td><td>Commands and keywords are in boldface font.</td></tr>
<tr><td>Futura bold-italic type</td><td>Parts of commands, which are related to elements already named by the user, are in boldface italic font.</td></tr>
<tr><td>Italicized Futura type</td><td>Variables for which you supply values are in italic font</td></tr>
<tr><td>Futura type</td><td>Indicates the names of fields or windows.</td></tr>
<tr><td>Garamond bold type</td><td>Indicates the names of command buttons that execute an action.</td></tr>
<tr><td>< ></td><td>Angle brackets indicate function and keyboard keys, such as <SHIFT>, <CTRL>, <C>, and so on.</td></tr>
<tr><td>[ ]</td><td>Elements in square brackets are optional.</td></tr>
<tr><td>{a | b | c}</td><td>Alternative but required keywords are grouped in braces ({ }) and are separated by vertical bars (|)</td></tr>
<tr><td><code>screen</code></td><td>Terminal sessions and information the system displays are in <code>screen font</code>.</td></tr>
<tr><td>node</td><td>The leading IP address or nodename of a SIPxNano is substituted with node in boldface italic font.</td></tr>
<tr><td>SN</td><td>The leading SN on a command line represents the nodename of the SIPxNano</td></tr>
<tr><td>#</td><td>An hash sign at the beginning of a line indicates a comment line.</td></tr>
</table>

Chapter 1 General Introduction

Chapter contents

Welcome

Welcome to the SIPxNano! This guide provides conceptual information on the Patton SIPxNano IP voice system, which runs on the CentOS operating system.

This chapter includes:

- "Factory defaults" on page 7
- "Hardware included" on page 8
- "What you will need" on page 8

Factory defaults

When you first access the SIPxNano, the following defaults are activated. You have the option to change these defaults when the system first boots.

- **Hostname:** sipx.patton.com
- **IP:** 192.168.200.200
- **Netmask:** 255.255.255.0

Note Make sure to set the netmask to the same subnet as the PC you will be accessing the SIPxNano webpage interface with.

- **Gateway:** 192.168.200.1
- **Nameserver:** 0.0.0.0
- **Sipx Login:** root
- **Admin Password:** superuser
- **Admin Email:** superuser@patton.local
- **Time Zone:** U.S. Eastern
- **SIP Domain Name:** patton.local

Note If you want to change the factory defaults, you will need to run the reset script. For more information, refer to "Configuring and Running the Reset Script" on page 11.

GUI Defaults

- **User ID:** superadmin
- **PIN:** patton

Hardware included

The following items are included with your SIPxNano:

- NanoServ system computer
- Power adapter
- Stand for the NanoServ
- Documentation CD
- Quick Start guide

What you will need

Note The following items are **NOT** included, but you will still need them to use the SIPxNano.

You will need the following items to use the SIPxNano:

- Monitor
- Keyboard
- Mouse
- Separate PC
- Ethernet cable

About the NanoServ

The NanoServ™ is a unique tiny embedded system for a wide range of applications, from industrial to office to home.The system offers multi-server features to function as firewall, mail-server, print server, and many other single task applications. It is suitable for a space-conscious environment, with dimensions of 17x124x38mm (Ultra-Thin System) (or 58mm - Thin System), equivalent to 4.32”x3.15”x0.96” (Ultra-Thin System) (or 1.47” - Thin System).

For more information about the NanoServ hardware, see the *NanoServ User Guide,* which is available at http://www.patton.com.

Chapter 2 **Setting Up the SIPxNano**

Chapter contents

Introduction

This chapter will help you set up the SIPxNano for the first time. The factory has already installed the CentOS operating system, but you will need to initially configure the system the first time you use it.

Before you begin...

Before you turn on the SIPxNano, follow these steps (see figure 1):

Note The NanoServ chassis is within thermal and design limits but it does run warm. For proper operation and cooling the system must have 2 cm or 1 inch of space for pen air around both side vents. In other words, it cannot be laid down flat on a table, and it cannot be standing flat against a partition, etc.. If mounted in some place, it must be mounted with stand off space about 1 inch. If mounted in a closed enclosure, it will need 1 or 2 fans on the outer enclosure for proper air flow inside.

1. Place the SIPxNano upright in the metal stand (included).
2. Attach a VGA monitor (not included) into the VGA port on the rear panel.
3. Attach the keyboard (not included) into the PS/2 port on the rear panel.
4. Attach the mouse (not included) into the other PS/2 port on the rear panel.
5. Attach the Ethernet cable (not included) into the RJ-45 port and connect the other end to the PC (not included).
6. Attach the power adapter (included), and plug it into the closest electrical outlet.
7. Turn on the power supply switch on the rear panel.

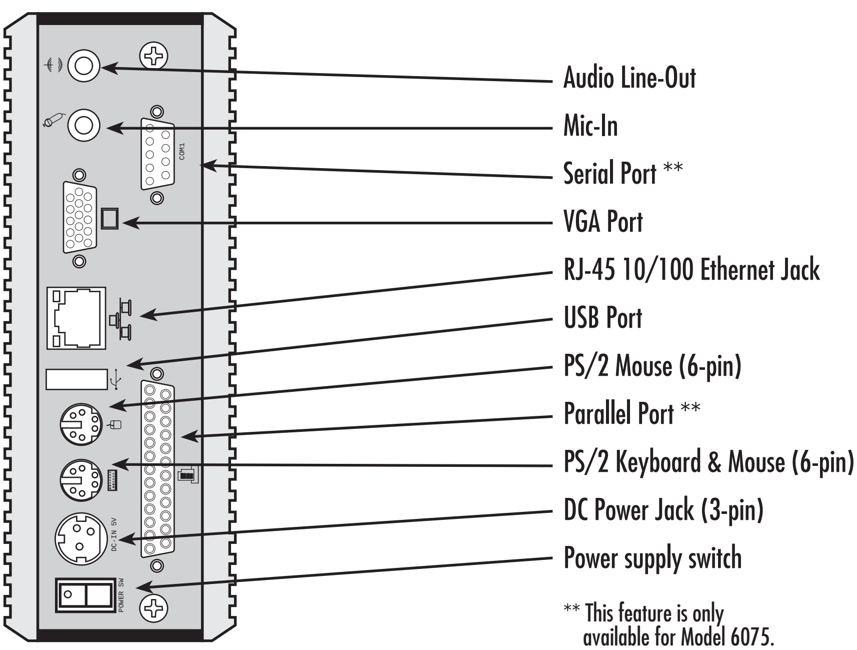


Figure 1. Rear panel of the SIPxNano (Model 6075 shown)

Configuring and Running the Reset Script

When you turn on the SIPxNano for the first time, you will need to run and configure a reset script before you can access the webpage interface.

When you boot the system, a CLI prompt will appear asking for a login and password:

- **login:** root
- **password:** superuser
- [**root@SIPxNano** ~]# will appear. To run the reset script, type:

  /recovery/linux/sipx_reset_cd <Press enter>.

Note Do **not** press "I" for Interactive Setup while running the reset script. Allow the reset script to run without interruption.

After running the reset script, you will have the option to change the default settings such as the hostname, IP address, netmask, gateway, nameserver, password, and timezone. (If you would like to keep the default settings, press <enter> at each prompt). Type '**yes**', then press <enter>. The system will reboot.

Note If you make a mistake while running the reset script or changing the default settings, type '**CTRL+C**' to interrupt the reset script and start over.

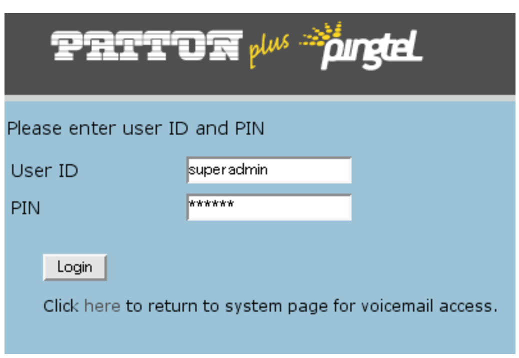
After the system reboots for the second time, enter the login and password you set (or the default settings, if you did not make any changes).

Accessing the system

Enter the **IP address** of the SIPxNano (in this example, 10.10.200.1) into a web browser on a separate PC. Be sure that the PC is on the same subnet as the SIPxNano.

To log into the SIPxNano:

1. After typing the IP address into a web browser, a welcome screen will appear. Click on the '**Configuration**' link.
2. Log into the webpage interface. The default login is:
   - **User ID:** superadmin
   - **PIN:** patton



3. The SIPxNano homepage will appear.

Figure 2. SIPxNano homepage

4. From the main page, on the main navigation menu at the top, you can access the configuration pages for setting up **Users**, **Devices**, **Features**, **System**, or **Diagnostics**.

Configuration Overview

The following chapters in the manual provide information on how to set up and configure users, devices, features, system settings, diagnostics, and voicemail.

Users

From the **Users** menu, you can:

- Add and configure settings for new users
- Add and configure user groups
- Configure the user extension pool

For more information on **Users**, see Chapter **3**, "Users" on page 14.

Devices

From the **Devices** menu, you can:

- Add and configure settings for phones
- Add and configure phone groups
- Add and configure gateways
- Add and configure files for devices

For more information on **Devices**, see Chapter **4**, "Devices" on page 25.

Features

From the **Features** menu, you can:

- Add and configure auto attendants
- Add and configure hunt groups
- Add and configure call park extensions

For more information on **Features**, see Chapter **5**, “Features” on page 40.

System Settings

From the **System** menu, you can:

- Add and configure dial plans and dialing rules
- Configure general system settings
- Import CSV files
- Backup SIPxNano system configurations

For more information on **System Settings**, see Chapter **6**, “System” on page 56.

Diagnostics

From the **Diagnostics** menu, you can:

- Refresh registrations and view the primary registrar
- View and edit the status of jobs
- Start, stop, restart, and refresh services
- Configure snapshot settings for log files, the Apache configuration, and credentials

For more information on **Diagnostics**, see Chapter 7, “Diagnostics” on page 73.

Voicemail

To access the voicemail page, click ‘**Voice Mail**’ on the main screen.
Click the ‘**Inbox**’ to show the login prompt for **Voicemail**.

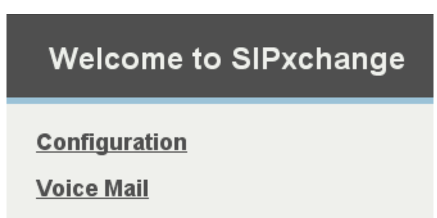


Figure 3. Voice Mail

From the **Voicemail** screen, you can:

- Configure voicemail settings for specific users who have logged in

For more information on **Voicemail**, see Chapter **8**, “Voicemail” on page 78.

Chapter 3 **Users**

Chapter contents

Overview

This chapter provides information on SIPxNano user functions. Menu options include:

- "Users" on page 15
- "User Groups" on page 22
- "Extension Pool" on page 24

To help you accurately deliver configuration settings and management capabilities where they are needed in your organization, SIPxNano distinguishes between the settings and management that:

- Apply to the operation of a specific device and its ability to function in your network
- A particular end user prefers, or is permitted to have, when using the device

Examples of device-level configuration settings include:

- The location of network servers and other entities, including a time server, SNMP stations, and SIP servers
- Activation of debugging tools, such as console output or Telnet access
- The location of hosts and ports for the device to use when receiving, or sending, data through a firewall

When you send profiles to a device, SIPxNano delivers settings only to the SIP phone associated with that device.

Users

This section covers:

- "Adding Users" on page 16
- "Editing users" on page 21
- "Filtering users" on page 21
- "Deleting users" on page 21
- "More Actions" on page 21

For a user, you can:

- Grant or deny permissions, such as permission to dial long distance or international numbers
- Define certain configuration settings, such as activation of services for handling incoming calls, or entries in a personal speed dial directory
- Assign devices

Note Permissions are not delivered to SIP phones. The Comm Server and Media Server use the Permission database to authorize calls and enable features such as voicemail for the user.

Note Settings delivery and device management, outlined in this chapter, apply to Patton certified/managed phones, which currently include the Cisco 7900 series of phones and the Patton SIP Softphone. All other phones are managed from the respective phone after adding user(s) and user groups to SIPx-Nano.

Adding Users

To add a new user:

1. From the main menu, click on **Users** > **Users,** then the **Add User** hyperlink. The **New User** screen appears.

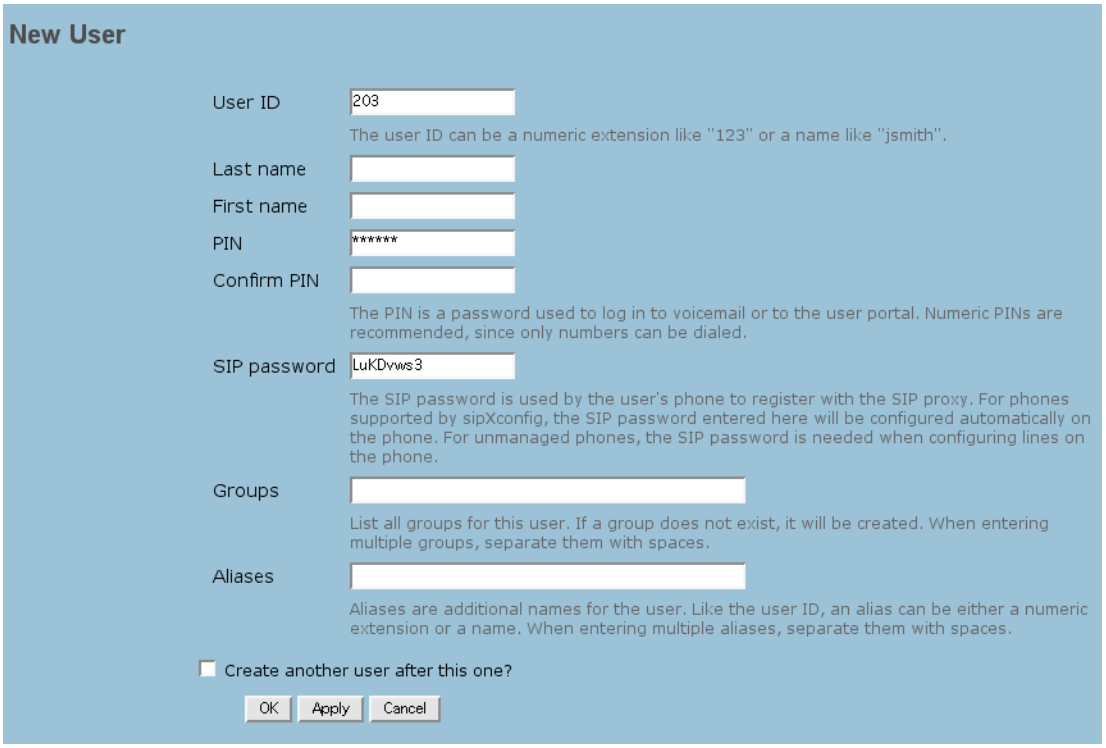


Figure 4. Adding a new user

2. Enter a unique User ID. The User ID can be a numeric extension or a name. (See “User IDs” on page 17).
3. Enter the user’s **Last Name.** In the next field, enter the user’s **First Name.**
4. Create a numeric **PIN** for the user to access voicemail. In the next field, type the PIN again.
5. The **SIP password** is used by the user's phone to register with the SIP proxy. For phones supported by sipXconfig, the SIP password entered here will be configured automatically on the phone. For unmanaged phones, the SIP password is needed when configuring lines on the phone.
6. List any existing **User Groups** that the user will belong to (optional).
7. In the **Aliases** field, list any additional names for the user (optional). (See “Aliases” on page 19).
8. Click **OK.**

User IDs

For each SIPxNano user, you must supply a unique, alphanumeric User ID. This User ID, along with a numeric Personal Identification Number (PIN), is required for end user access to the SIPxNano interface.

> **Note** SIPxNano requires a unique User ID for each user. Patton recommends that you assign a unique PIN to each user, and also encourage end users to change their PINs frequently.

In addition, you can supply a numeric extension number and one or more aliases for each user. These additional identifiers provide flexibility for both call addressing and user interface access.

You assign a numeric-only identifier to your SIPxNano users so that:

- Callers can dial numbers, rather than full SIP URLs, to address calls
- End users can access SIPxNano features, including voicemail, from the phone top

Either a user's required User ID or the optional extension can be the numeric-only identifier. If you assign numeric User IDs to your users, an additional extension is not needed.

> **Note** If you use extensions, all User IDs and extensions must be unique (the same number cannot be used as both a User ID and an extension).

You can employ an alphanumeric naming convention for your SIPxNano User IDs. For example, a User ID can be made up of a first initial and last name (jsmith), or from first and last names (jane_smith); as a result, end users can have the same identifier that they use for email. Alternatively, User IDs can reflect extension numbers, Direct Inward Dialing (DID) telephone numbers, or some other numbering scheme that you administer.

When you add a user, SIPxNano automatically sets up a user line with a SIP URL based on the new User ID so that calls can be directed to that user. This line automatically registers with the Comm Server's SipRegistrar component for each SIP phone assigned to the user. SIPxNano stores all registered SIP URLs in the Registration database. For more information on Registrations, see the section "Registrations" on page 74 in Chapter 7, "Diagnostics" .

Extensions

In general, if you assign User IDs that include alphabetic characters you will also set up a numeric extension for each user. If you assign numeric-only User IDs to your users, an additional extension may not be needed.

When you set up an extension or alias for a User ID, SIPxNano automatically adds them to a database of User IDs and all of the extensions and aliases associated with each one. The components of the Comm Server use this Alias database, as well as the lines stored in the Registration database, to help the Comm Server route incoming calls: regardless of whether a call is addressed to a User ID, an extension number, or an alias, SIPx-Nano routes the call correctly to the user's assigned device(s) (see table 2).

Table 2. User IDS and SIP URLs

User information:	Dialed SIP URL:
User ID = jsmith	sip:jsmith@example.com
Extension = 123	sip:123@example.com
Alias = jane_smith	sip:jane_smith@example.com

SIPxNano also stores all extensions in its Extension database. If an end user logs in to a voicemail inbox from the phone top interface with an extension number, the Media Server uses this database to confirm the log in data.

If you use extensions to identify users, you can use extension pools to manage the set of usable extensions and keep track of numbers that have already been assigned to users.

- Extension pools are not hierarchical, and each extension number can only be assigned to a single pool. You might set up just one pool for your installation, or several pools to reflect different geographical locations, departments, or other organizational structures.
- When you add a new user, you can supply an extension either by typing in an extension number, or by choosing the next available extension from a pool. Since SIPxNano extensions must be unique to each user, using an extension pool can help speed up the assignment and data entry process.

For information on setting up extension pools, see "Extension Pool" on page 24.

> **Note** To prevent certain values from being assigned to users as either extensions or User IDs, you can place numbers in the "reserved" extension pool. For example, you can add the extensions that identify the Voicemail and Auto Attendant applications to the reserved extension pool. If you do so, those numbers cannot be inadvertently used to identify users.

Aliases

Unlike User IDs and extensions, SIPxNano aliases do not need to be unique. End users cannot use an alias to log in to the SIPxNano interface or the phone top interface.

You can set up zero, one, or more aliases for a user. Because they are not unique, you can use an alias to associate a single identifier with several different users. For example, you can assign an alias of “sales” to each of the users in the sales department.

> **Note** For the entity “sales” to have a dedicated voicemail inbox, you must set up a separate SIPxNano user with a User ID of sales and a unique extension. An end user can then use the extension to access voicemail for the sales user.

Another example of an alias is one that resembles an email address, which callers from VoIP phones may find easier to remember than a phone number. That is, for a user with a SIPxNano User ID of 2435 and an email address of `rsherman@example.com`, you might supply an alias of `rsherman`. Callers from PSTN phones could dial 2435; callers from SIP phones could dial the SIP URL `sip:rsherman@SIPxNano.example.com` (or `sip:rsherman@example.com`, using the correct domain for your SIPxNano server).

When you set up an alias for a User ID, SIPxNano automatically adds it to the Alias database. The components of the Comm Server use the Alias database to route incoming calls.

Permissions

With the SIPxNano interface for administrators, you authorize or prohibit access to certain system features by enabling permissions for users or groups of users.

Table 3. Permissions

Feature:	Description:
900 Dialing	User can dial 900 numbers
Auto Attendant	Include this user in the dial by name directory presented by the Auto Attendant
International Dialing	User can dial international number
Local Dialing	User can dial local numbers
Long Distance Dialing	User can dial long distance numbers
Voicemail	User has a voicemail inbox and can receive voicemail messages
PSTN forwarding	External, PSTN numbers can be defined as forwarding addresses for this user's calls
System prompts	When logged in to a voicemail inbox from a phone top, user can record system-wide prompts and greetings

For example, to limit access to 900 numbers, you disable the 900 Dialing permission for most users. Whenever an end user dials a 900 number, that user's permissions in the Permission database are checked against the Outbound Authorization database. Only users with the required permission (900 Dialing) enabled are able to complete calls to those numbers successfully.

The Outbound Authorization database associates permissions with dialed numbers as follows:

Table 4. Outbound Authorization Permissions

To dial:	Permission needed for authorization:
911 or sos	None
1900xxxxxxx. or 900xxxxxxx.	900 Dialing
1[2-9]xxxxxxxxx. or [2-9]xxxxxxxxx.	Long Distance Dialing
numbers that start with the international dialing prefix	International Dialing
1800xxxxxxx. or 800xxxxxxx. 1888xxxxxxx. or 888xxxxxxx. 1877xxxxxxx. or 877xxxxxxx. 411 0 00	Local Dialing

SIPxNano does not deliver permissions to phones. Instead, SIPxNano incorporates permissions into several internal databases, including the Permission and Outbound Authorization databases.

Editing users

After creating a new user, you can adjust certain settings such as call forwarding and permissions.
From the Users page, click on the **User ID** of the user you'd like to configure settings for.

1. Under Identification on the left side of the screen, there are links to **Identification**, **Call Forwarding**, and **Permissions.** (You are currently on the Identification screen).
2. To edit the call forwarding settings for a user, click on **Call Forwarding** > **Add Number**.
3. Enter your desired instructions for call forwarding, and click **OK.**
4. To edit specific access rights for a user, click on **Permissions.**
   From this screen, you can add or remove permissions for a specific user by checking or unchecking the boxes.

Note To change the SIP password, you will need to click on the "**Show Advanced Settings**" link to view and modify the SIP password settings.

Filtering users

You can filter users in the users list by using the **Filter by...** drop-down menu.
The following options are available in the **Filter by...** menu:

- - **all** -: Shows all existing users
- - **search** -: Shows a field where you can enter search terms to filter specific users
- *Group*: Shows only users in a certain group

Deleting users

To delete a user:

1. From the main menu, click on **Users** > **Users**. The user list appears.
2. Click the check box next to the user you want to delete.
3. Click on the **Delete** button.
4. A confirmation prompt will appear. Click **OK.**

More Actions

The **More actions...** drop-down menu is located next to the Delete button. With the **More Actions...** menu, you can add users to an existing group.

To use the **More Actions...** function:

1. Click the check box next to the user you want to add to an existing group.
2. From the **More Actions...** drop-down menu, select the group you want to add the user to.
3. The page will refresh, and a confirmation message will appear at the top of the screen.

User Groups

This section covers:

SIPxNano user groups help you deliver configuration settings to users efficiently; you can create groups of users through the SIPxNano interface for administrators. When you specify settings for a group, they automatically apply to all of the members assigned to the group, speeding configuration and making selected features available consistently.

For example, all of the users who are customer service representatives can be grouped together.

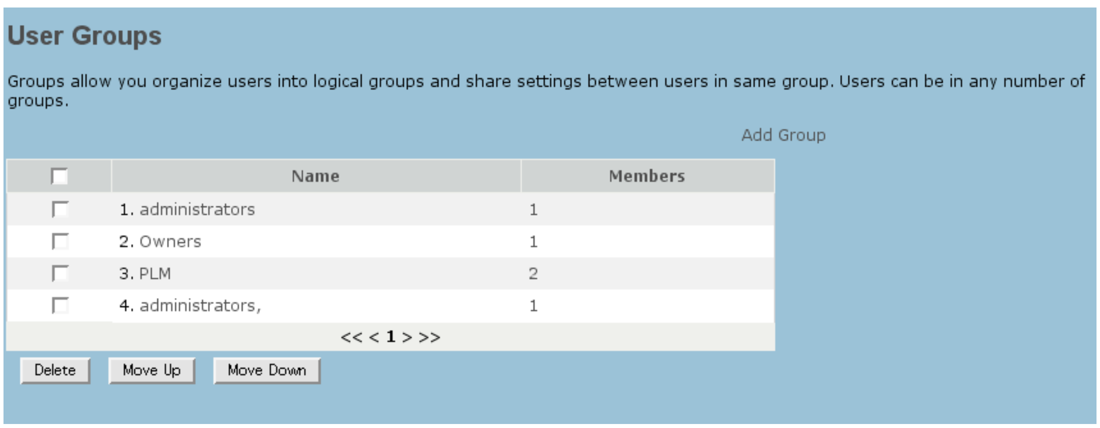


Figure 5. User Groups main screen

Since every user in the customer service department uses the same applications, you can assign applications once, to the Customer Service group, rather than assigning the same applications to each user who works in that department, individually.

Adding User Groups

To add a new user group:

1. From the main menu, click **Users** > **User Groups** > **Add Group**.
2. Enter a user group name (e.g., CustomerService) in the "**Name**" field.
   The **Name** of the user group cannot contain any spaces or whitespace characters.
3. Enter a description for the group (optional).
4. Click **OK**.

The User Groups window displays the group or groups you added in the name column; these group names are clickable.

Adding Users to a User Group

After adding a user group, you can add users to the group from the **Users** menu.
For information on how to add existing users to an existing user group, see "More Actions" on page 21.

Editing User Groups

When you add a new group, you are only creating a group - you are not configuring it.

To configure or edit an existing group:

1. From the list on the **User Groups** screen, click on the name of the group you want to edit.
2. The **Group Settings** page appears. To enable a setting, check the box next to it. To disable a setting, uncheck the box next to it.

   > **Note** When you created the new group, the group's settings were automatically set to the default permissions for user groups.

3. To save your changes, click **OK.**

Deleting User Groups

To delete a user group:

1. From the list on the **User Groups** screen, check the box next to the group you want to delete.
2. Click the **Delete** button.
3. A confirmation prompt will appear. Click **OK**.

Moving User Groups

On the **User Groups** page, you can change the order of the user groups in the list.

To move a user group in the user group list:

1. Check the box next to the user group you want to move.
2. Click the **Move Up** button to move the group closer to the top (click as many times as necessary).
3. Click the **Move Down** button to move the group closer to the bottom (click as many times as necessary).

Extension Pool

The **Extension Pool** link allows you to edit the range of extensions that can be used by users.

To edit the Extension Pool:

1. From the main menu, click on **Users** > **Extension Pool.**
2. Type the lowest number of the range you desire in the **First pool extension** field.
3. Type the highest number of the range you desire in the **Last pool extension** field.
4. Click **Apply.**

The User Extension Pool page will refresh and a confirmation message will appear at the top.

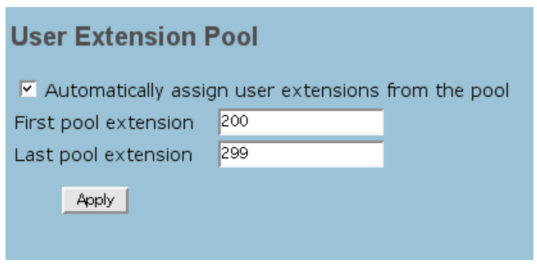


Figure 6. Configuring the extension pool

Chapter 4 **Devices**

Chapter contents

Overview

This chapter provides information on SIPxNano devices. Menu options include:

- "Phones" on page 26
- "Phone Groups" on page 31
- "Gateways" on page 35
- "Files" on page 38

Phones

This section covers:

- "Adding phones" on page 27
- "Editing phones" on page 28
- "Filtering phones" on page 30
- "Deleting phones" on page 30
- "Restarting phones" on page 30

You can add or edit phones by accessing the SIPxNano Configuration server interface through a Web browser and clicking **Phones** under Devices from the main navigation. The following screen displays:

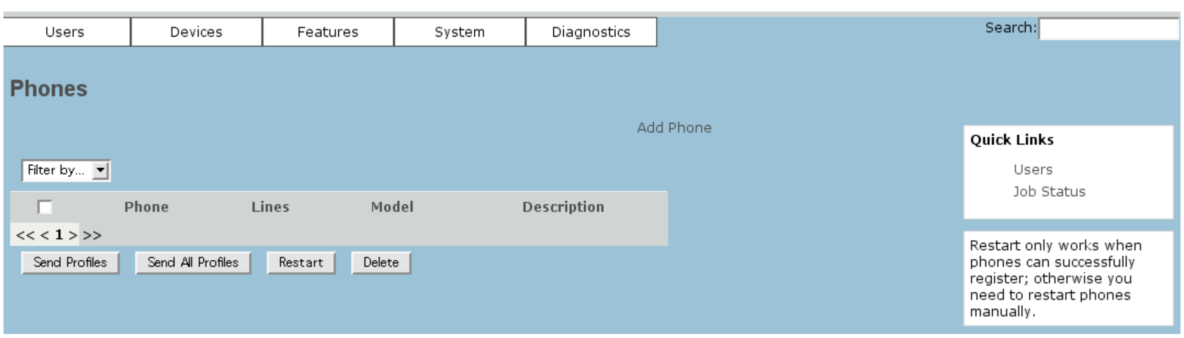


Figure 7. Phones main screen

Note From this screen you have the ability to Send Profiles to a particular user or users after selecting the check box to the left of the Phone column and then pressing the **Send Profiles** button. Alternatively, you can press the **Send All Profiles** button without selecting any checkboxes; profiles are sent to all phones that are listed.

Adding phones

To set up a phone, you supply a serial number and description, then select a phone model and associate one or multiple groups with that phone.

To set up a phone:

1. From the main screen, click **Devices** > **Phones** > **Add Phone**.

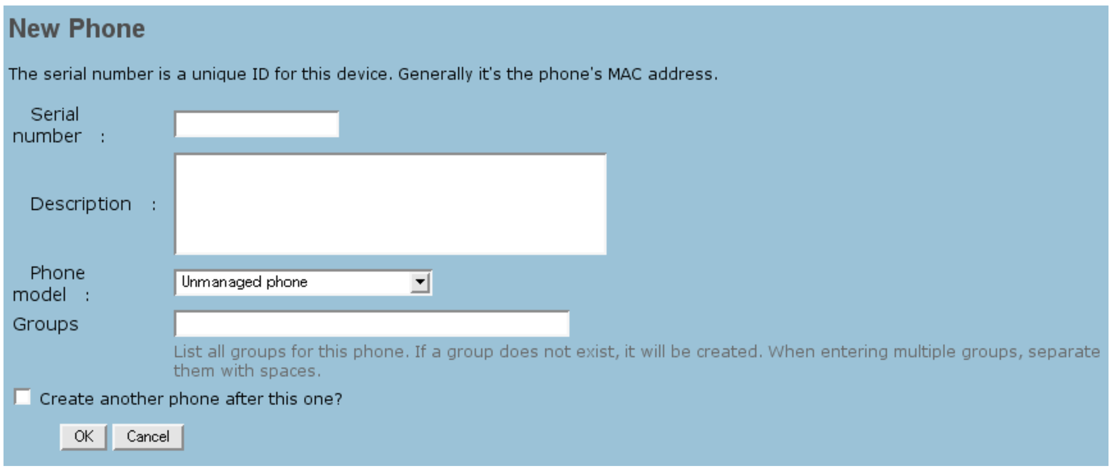


Figure 8. Adding a new phone

2. Enter the phone's serial number in the **Serial Number** field.
3. Enter a description about the phone (i.e., user or lines associated with the phone, phone's purpose, location, etc.).
4. Select a **Phone Model**.

> **Note** If you cannot find the phone in the **Phone Model** drop-down list, the phone is not directly managed through the Configuration Server. Select **Unmanaged Phone** from the drop-down list.

5. Enter a phone group or groups to associate the phone with in the groups field by choosing a group from Existing Groups, displayed at the right of the New Phone screen, or create a new group by simply typing its name into the Groups field on the New Phone screen; the group is automatically created and available for other additions.
6. Click **OK**.

Editing phones

The **Edit Phone** page has links to additional parameters that can be configured for a particular phone.

Note The optional links that you can edit phone settings for are: Identification, Lines, Date/Time, User Preferences, DTMF, Sound Effects, Voice/Codecs, Quality of Service, SNTP, RTP, Web Server, Call Handling, Hold Reminder, Directory Resources, Keys, Basic Logging, Security, Request, Features, MicroBrowser, SIP Servers, SIP, SIP Settings in DHCP, Dial Plan, Messaging, and NAT.

To edit the parameters of a particular phone:

1. From the main menu, click on **Devices** > **Phones**.
2. From the list of phones, click on the **serial number** of the phone you want to edit.

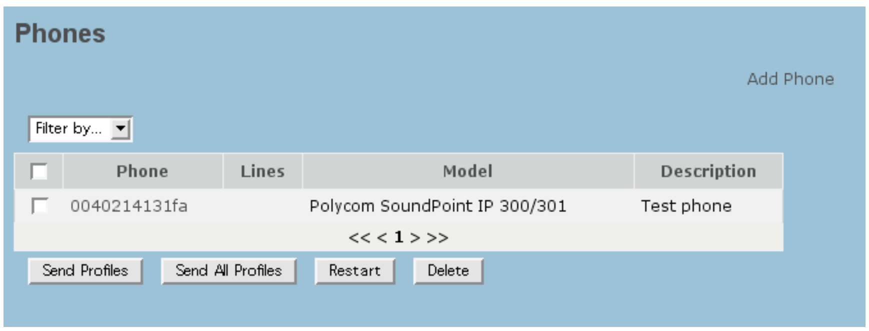


Figure 9. List of phones

3. The **Edit Phone** page will display. You can edit parameters including **Serial Number, Description**, **Phone Model**, and enter or create additional groups. You can also click links at the left-hand side of this screen and edit additional parameters available for a particular phone (e.g., Lines, Call Handling, Dial Plan, etc.). Defaults are always set and noted on additional parameter screens. Change parameters accordingly.
4. Click **OK.**

Adding lines to an existing phone

You can also add a line from the Identification screen by clicking **Lines** in the Quick Links section, then click the '**Add Line**' hyperlink at the top. The Add Line screen displays:

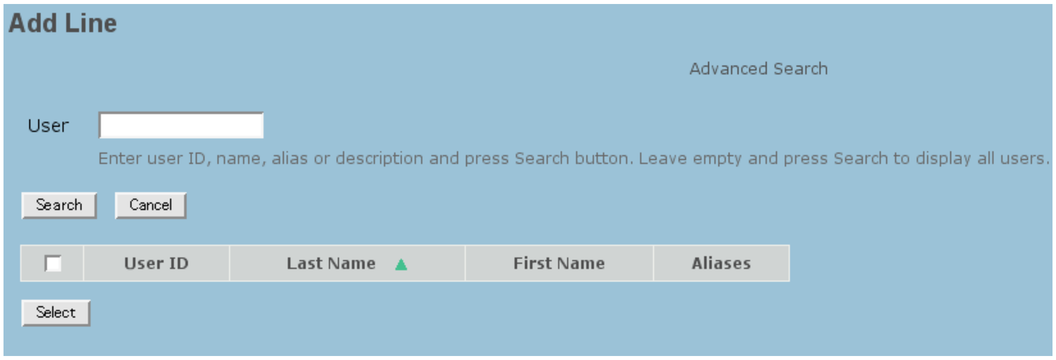


Figure 10. Adding a line to a phone

Perform a search by entering a partial or complete user ID, first and or last name, alias, or description in the User field and click the **Search** button. You can switch to an advanced search by clicking the Advanced Search link, which then becomes the Simple Search link, allowing you to switch back to a simple search. Simple search is for general broad-based searching or if you are unsure about the user information. Advanced Searching allows you to quickly find what you are looking for if simple searching returns too much unnecessary information.

Associated matches appear in the table below the **Search** and **Cancel** buttons. If the search has returned the correct user, select that user by selecting the check box to the left of the User ID column and clicking the **Select** button.

The Lines page displays with the appearance of a new line for the user selected on the Add Line screen:

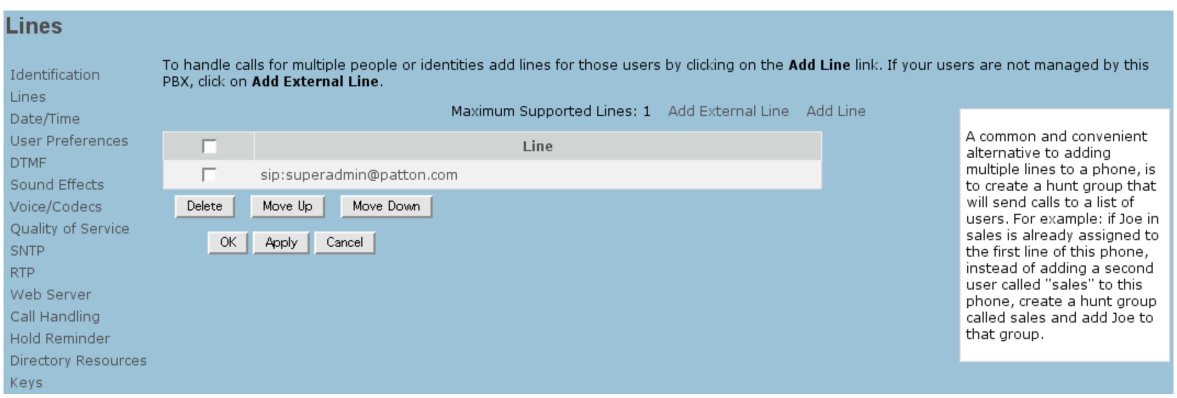


Figure 11. Phone lines

Adding a user to the phone adds a line appearance on that phone, managing calls for that user. You can add users to many phones, including different phone models. Settings regarding how that user is configured on this particular phone can be set after adding the user.

You can choose to add another line from the lines screen by clicking the Add Line link above the table. You can add an external line for users not managed by the SIPxNano you are currently working with.

Note From the Lines screen you can also edit the parameters for the associated phone by clicking a link in the left-hand navigation on the Lines screen.

Filtering phones

You can filter phones in the phones list by using the **Filter by...** drop-down menu.
The following options are available in the **Filter by...** menu:

- - **all** -: Shows all existing users
- - **search** -: Shows a field where you can enter search terms to filter specific phones

In addition to displaying all phones in the system, you can select a particular group from the **Filter by...** drop down menu above the table and display only the phones that are in that group. You can also select -**search**- from the **Filter by...** drop down menu and search for a phone by serial number, model, or description. Additionally, you can choose to add phones to or remove phones from a particular group at the Phones screen by selecting -**all**- or a particular group from the **Filter by...** drop-down menu, selecting the checkbox next to the respective phone's serial number, and selecting the group to add to or remove the phone from in the **More actions...** drop down menu below the table.

Deleting phones

To delete a phone:

1. From the main menu, click on **Devices** > **Phones**. The phones list appears.
2. Click the check box next to the phone you want to delete.
3. Click on the **Delete** button.
4. A confirmation prompt will appear. Click **OK.**

Restarting phones

To restart a phone:

1. From the list of phones on the **Phones** page, click the box next to the phone you want to restart.
2. Click the **Restart** button.
3. A confirmation message will appear at the top of the **Phones** screen.
4. You can check the status of the restart by clicking on **Job Status** in the **Quick Links** box on the right, or by click **Diagnostics** (on the main menu) > **Job Status**.

Phone Groups

This section covers:

- "Adding Phone Groups" on page 32
- "Editing Phone Groups" on page 33
- "Deleting Phone Groups" on page 34
- "Moving Phone Groups" on page 34

Groups allow you to organize phones into logical groups. Phones can be in any number of groups. You can add or edit Phone Groups by accessing the SIPxNano Configuration server interface through a Web browser and clicking **Phone Groups** under Devices from the main navigation. The following screen displays:

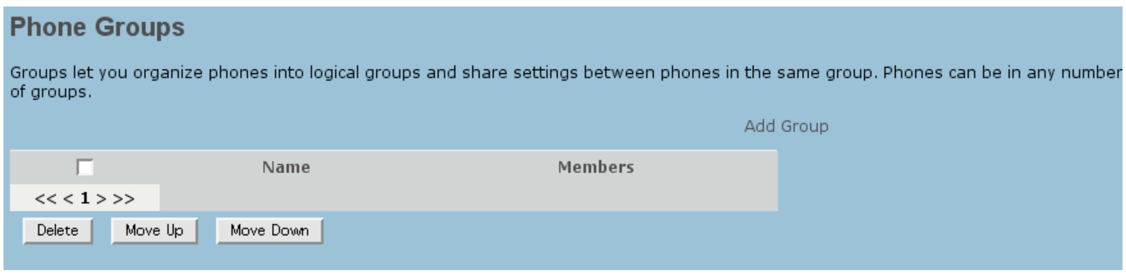


Figure 12. Phone Groups main screen

From this screen you can edit existing Phone Groups within the table. You can also click the number in an associated Phone Group Members column to view and edit the phones that are in a particular group. Select the checkbox to the left of the Name column on the Phone Groups screen and click **Delete** to delete that particular Phone Group.

Adding Phone Groups

To add a phone group:

1. From the main screen, click on **Devices** > **Phone Groups** > **Add Group**.

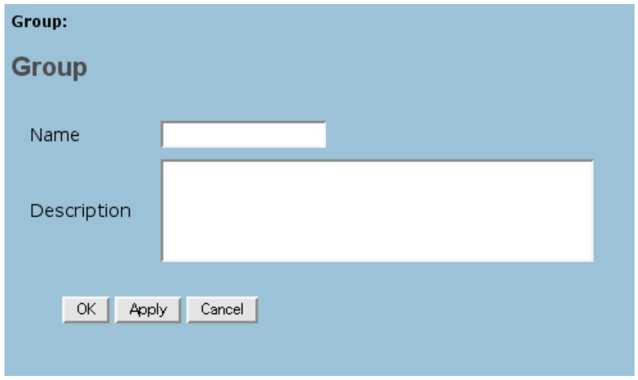


Figure 13. Adding a phone group

2. Enter a phone group name (e.g., Support) in the **Name** field.
   The **Name** of the phone group cannot contain any spaces or whitespace characters.
3. Create a description of this group in the **Description** field (e.g., Support Group phones).
4. Click **OK** to save phone group information and return to the Phone Groups screen; click **Apply** to save phone group information and remain at the current screen; click **Cancel** to return to the Phone Groups screen without saving any of the information you created.

Editing Phone Groups

To edit a phone group:

1. From the main menu, click on **Devices** > **Phone Groups**.
2. Click on the name of the phone group you want to edit.

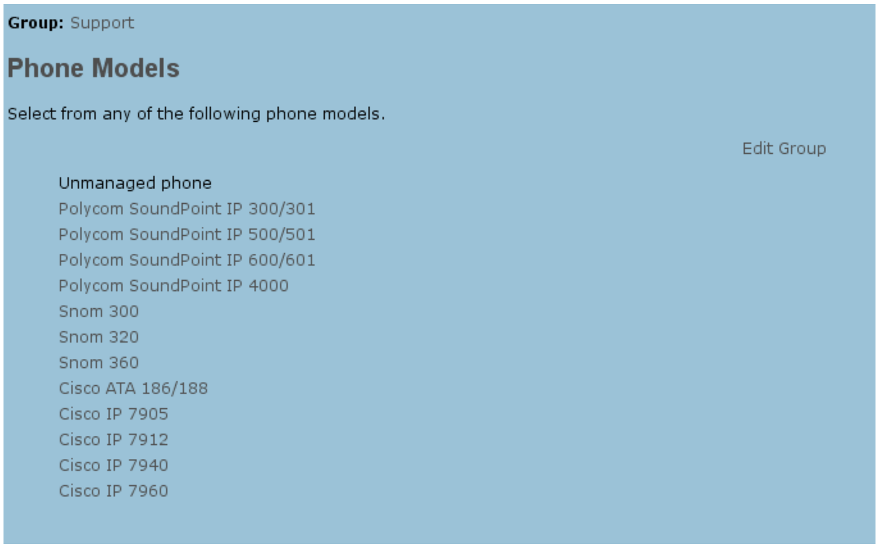


Figure 14. Phone model groups

Note You can also edit the group parameters associated with a phone or set of phones associated with a particular phone type from the **Phone Models** screen by clicking on a specific phone type name in the displayed list.

3. Click **Edit Group** to edit the group name and/or description.

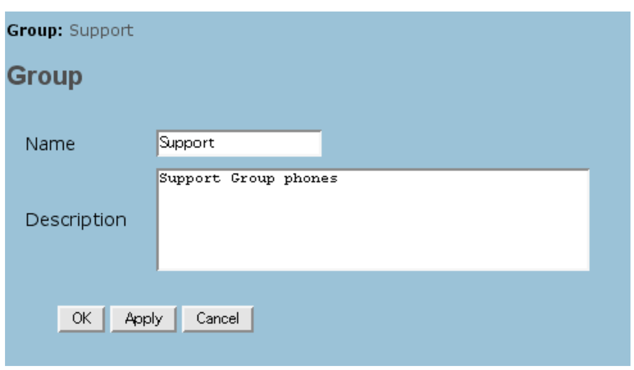


Figure 15. Editing a phone group

Deleting Phone Groups

To delete a phone group:

1. From the list on the **Phone Groups** screen, check the box next to the group you want to delete.
2. Click the **Delete** button.
3. A confirmation prompt will appear. Click **OK**.

Moving Phone Groups

On the **Phone Groups** page, you can change the order of the phone groups in the list.

To move a phone group in the phone group list:

1. Check the box next to the phone group you want to move.
2. Click the **Move Up** button to move the group closer to the top (click as many times as necessary).
3. Click the **Move Down** button to move the group closer to the bottom (click as many times as necessary).

Gateways

This section covers:

- “Adding gateways” on page 35
- “Editing gateways” on page 36
- “Deleting gateways” on page 37

You can define gateways that the SIPxNano will use to route outbound calls. You can specify gateways to use for routine calls sepertaely from gateways to use for emergency calls.

To access the Gateways page, click **Devices** > **Gateways**.

Adding gateways

To add a gateway:

1. From the **Add a new gateway...** drop-down menu, select **Unmanaged gateway** or **SIP trunk**.

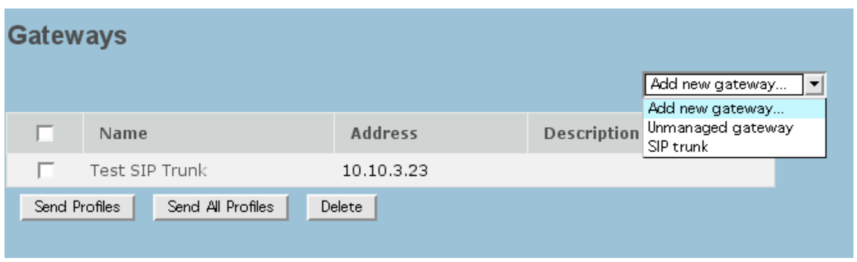


Figure 16. Adding a gateway

2. Enter details for the following options:

– **Name:** The name of the gateway.

– **Address:** IP address (example: 10.1.1.1) or a fully qualified hostname (example: gateway.example.com)

– **Serial Number (Unamanaged gateway):** Serial number contains 12 hexadecimal digits (0-9 and a-f), for example: 0040214131fa. Usually the serial number is set to the device's MAC address. For an unmanaged gateway, the serial number is only displayed, not used.

– **Description:** A description of the gateway.

3. Click **OK** to save and return to the Gateways screen; click **Apply** to save and remain at the current screen; click **Cancel** to return to the Gateways screen without saving any of the information you created.

Editing gateways

To edit an existing gateway:

1. On the Gateways screen, click on the name of the gateway in the Name column.
2. The Gateway Details screen will appear. Using the links on the left, you can edit the gateway details.
3. **Configuration:** Edit the name, address, and description of the gateway.
4. **Dial Plan:** Enter a prefix that will be added to all numbers for calls connected through this gateway.
5. **Caller ID:** Specify caller ID settings for calls connected through the gateway.
   - **Default Caller ID:** Caller ID used for all the calls connected through this gateway, unless more specific caller ID is specified for a caller.
   - **Block Caller ID:** If checked, all calls connected through this gateway will have Caller ID blocked, unless more specific caller ID is specified for the user.
   - **Ignore user Caller ID:** If checked only gateway default caller ID and Block Caller ID options are consider for this gateway.
   - **Transform extension:** If checked gateway will produce Caller ID by transforming user extension. If not checked caller ID specified on a group level will be used.
   - **Caller ID prefix:** Optional prefix added to caller externsion to create Caller ID.
   - **Keep digits:** Number of extension digits that are kept before adding Caller ID prefix. If caller extension has more digits than configured here, leading digits are dropped when creating Caller ID.

Figure 17. Caller ID Details for SIP Trunk

6. Click **OK** to save and return to the Gateways screen; click **Apply** to save and remain at the current screen; click **Cancel** to return to the Gateways screen without saving any of the information you created.

Deleting gateways

To delete an existing gateway:

1. On the Gateways screen, select the checkbox next to the name of the gateway(s) you want to delete.
2. Click **Delete**.

Files

This section covers:

The files section pertains to various phone files for Polycom SoundPoint phones and Cisco 7940 and 7960 series phones. Device files you can upload include firmware, the directory.xml file for Polycom phones (commonly dialed numbers/speed dialing) and the dialplan.xml file for Cisco 7940 and 7960 series phones (circumvents the need to press the dial button on the phone after entering the dialed number), and ringtones that can be uploaded into SIPxNano and then distributed to all phones associated with the device name (e.g., Polycom).

Unmanaged TFTP files can also be uploaded. Use the Unmanaged TFTP (Trivial File Transfer Protocol) option to upload files for phones not listed in the Add files... drop-down menu. You must have a clear understanding of the particular files you can associate with your phones. Reference your phones’ manual for information.

Adding files

1. From the main screen, click on **Devices** > **Files.**

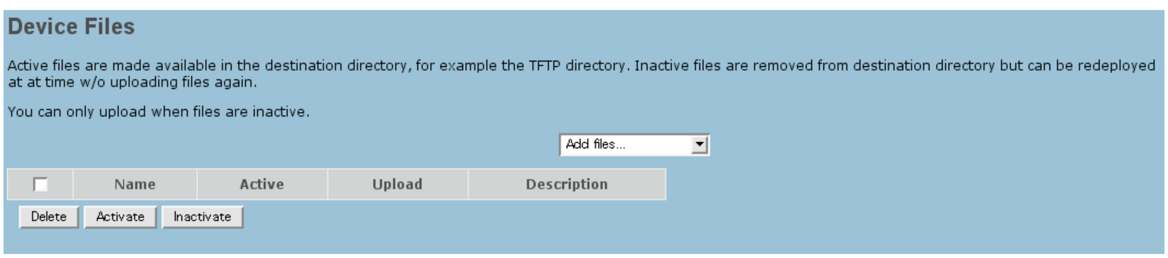


Figure 18. Adding a device file

2. Select a file type from the **Add files...** drop-down menu.
3. Enter the file **Name** and **Description**.
4. Click **OK**.

Editing files

To edit a file:

1. On the **Files** page, click on the name of the file you want to edit.
2. Make your changes and click **OK.** The main **Files** page will display.
3. To activate the file, click the box next to the file name and click the **Activate** button. Click **OK**.
4. To deactivate a file, click the box next to the file name and click the **Inactivate** button. Click **OK**.

Deleting files

To delete a file:

1. From the list on the **Files** screen, check the box next to the file you want to delete.
2. Click the **Delete** button.
3. A confirmation prompt will appear. Click **OK**.

Device Profiles

When you send profiles to IP phones:

- The phones store the newly updated data in configuration files
- New configuration settings do not take effect until the phone restarts
- The Job Status log records the activity (see “Gateways” on page 35)

If the target phone is in use, it will not restart until it is idle. When an idle phone receives new profiles and a restart request, it displays a message and restarts automatically after approximately one minute.

Note You should plan to send profiles only when call volume is low.

To deliver new configuration settings to a Cisco 7900 series IP phone or a Patton SIP Softphone you send one, two, or three profiles to that phone (profiles of device or user settings). To access the Send Profiles function:

1. Click on the device link (either **Phones, Gateways,** or **Files**) under the **Devices** link on the main menu.
2. Select the specific path or item from the list and click the **Send Profiles** button.

   A confirmation message will appear at the top of the page. (In this example, the gateway device was used).

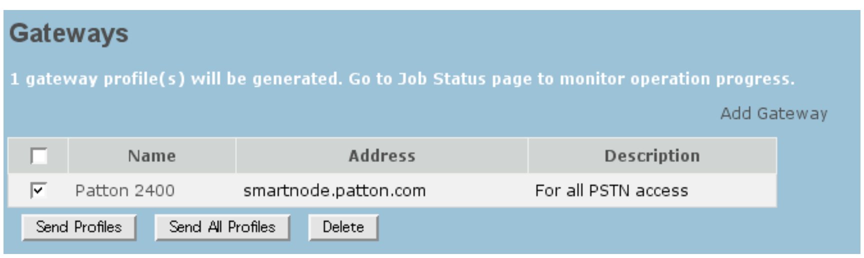


Figure 19. Sending profiles

Chapter 5 **Features**

Chapter contents

Overview

The Feature Management section of the SIPxNano Configuration Server interface allows you to configure and view SIPxNano settings that directly impact users. Menu options include:

- "Auto Attendants" on page 41
- "Intercom" on page 48
- "Hunt Groups" on page 49
- "Call Park" on page 51
- "Music on Hold" on page 55

Auto Attendants

This section covers:

- "Adding auto attendants" on page 42
- "Editing auto attendants" on page 44
- "Deleting auto attendants" on page 44
- "Adding auto attendants to Dial Plans" on page 44
- "Setting the Special Auto Attendant" on page 47

The Auto Attendant feature allows you to create multiple automatic answering operators within a single SIPx-Nano system. When a live person is unavailable, callers need to be directed to a specific section of an organization, or there are so many calls that callers must wait in a queue. The routing rules for which menu to use for a specific number are defined using the Dial Plan functionality of SIPxNano. An auto attendant's dialing rules are set within the AutoAttendant dial plan on the AutoAttendant Dialing screen, accessed by clicking **Dial Plans** under System from the main navigation and then clicking **AutoAttendant** from the Name column in the table on the Dial Plans screen.

Adding auto attendants

To set up a new auto attendant:

1. From the main screen, click on **Features** > **Auto Attendants** > **Add Attendant.**

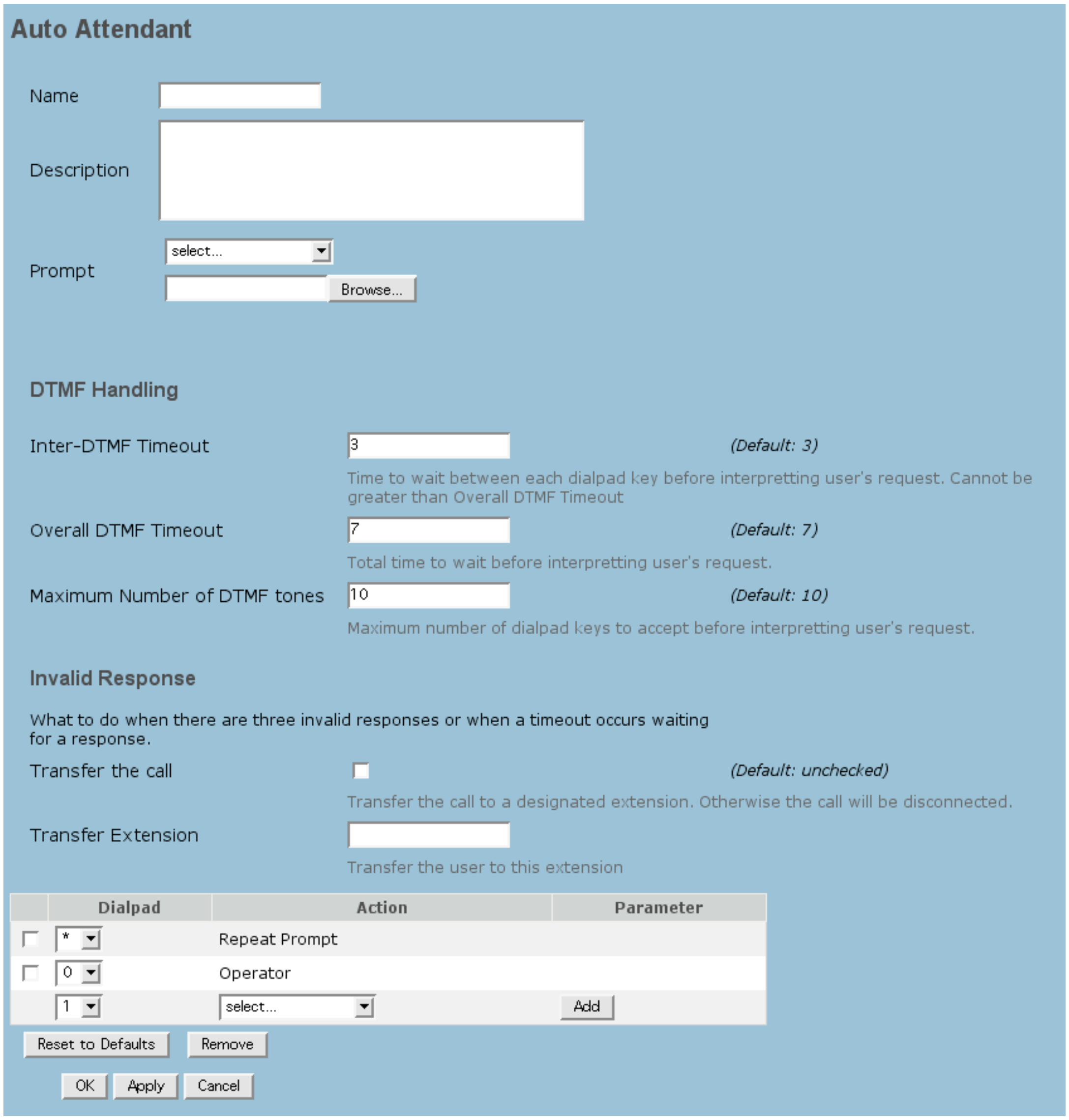


Figure 20. Adding an auto attendant

2. Enter a name for the Auto Attendant in the **Name** field.
3. Enter a description in the **Description** field.
4. Select a standard prompt (i.e., autoattendant.wav, afterhours.wav) using the **Prompt** section's '**select...**' drop-down menu, or record and upload an Auto Attendant prompt using the **Prompt** section's **Browse...** button.
   Prompts can be recorded from any phone connected to the SIPxNano voicemail system and are immedi-

ately selectable from the **Prompt** drop-down list after they are recorded. You can click **Listen** to listen to a standard prompt after you make your selection or browse to upload a new prompt.

Note **Prompts** must be recorded in **.wav file format** (mono mode with 8kHz sampling rate and 16 bit PCM wave format). For more information on recording .wav files see "Recording .WAV Files (Windows OS)" on page 53.

Note The **Listen** link only appears after you have made a selection from the **Prompt** section '**select...**' drop-down menu, or after you have browsed to and uploaded a custom prompt that you recorded.

5. Set the **Inter-DTMF Timeout** value. The default is set to **3**.
(The **Inter-DTMF Timeout** is the time to wait between each dialpad key before interpreting a user's request. This value cannot be greater than the **Overall DTMF Timeout**).
6. Set the **Overall DTMF Timeout** value. The default is set to 7.
(The **Overall DTMF Timeout** is the total time to wait before interpreting a user's request).
7. Set the **Maximum Number of DTMF tones** value. The default is set to **10**.
(The **Maximum Number of DTMF tones** is the maximum number of dialpad keys to accept before interpreting a user's request).
8. Enter a transfer extension in the **Transfer Extension** field of the Invalid Response section, and choose to transfer calls, or not, using the **Transfer the Call** checkbox.
(The settings in the **Invalid Response section** determine what happens to a call after three invalid system responses or when a timeout occurs while waiting for a response).
9. Create dialpad actions that are associated with the auto attendant you are creating by editing actions from the **Dialpad** column drop-down lists.

   You can remove a dialpad action by selecting its associated checkbox and clicking **Remove**. You can add new dialpad actions by selecting an action from the drop-down list in the action column, defining a numeric or character driven action, and clicking **Add** to add it to the dialpad action table.
10. Click **OK** to save auto attendant information and return to the Auto Attendants screen; click **Apply** to save auto attendant information and remain at the current screen; click **Cancel** to return to the Auto Attendants screen without saving any of the information you created.

Note Defaults for all fields in both the **DTMF Handling** section and the **Invalid Response** section are noted to the right of the respective fields. They can also be changed by clicking the **Defaults** link next to the **Add Attendant** link on the Auto Attendants screen. Customized defaults are set automatically when you create a new Auto Attendant.

Editing auto attendants

To edit an existing auto attendant:

1. From the main screen, click on **Features** > **Auto Attendant**.
2. On the Auto Attendants screen, click on the **name** of the auto attendant you want to edit.
3. From this screen, you can **edit** any of the settings you previously set for this particular auto attendant.
4. To save your changes, click the **OK** button.

Deleting auto attendants

To delete an auto attendant:

1. From the main screen, click on **Features** > **Auto Attendants**.
2. Check the box next to the auto attendant that you want to delete.
3. Click the **Delete** button. A confirmation prompt will appear. Click **OK**.

Adding auto attendants to Dial Plans

Your newly created auto attendant must be added to Dial Plans.
To add an auto attendant to Dial Plans:

1. From the main screen, click **System** > **Dial Plans**.
2. Click the **Add Dial Rules** link near the top of the screen.

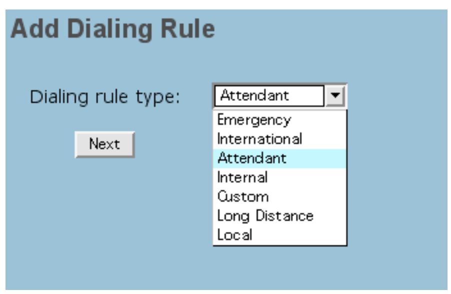


Figure 21. Adding a dialing rule

3. Select **Attendant** from the Dialing Rule Type drop-down list and click **Next**.

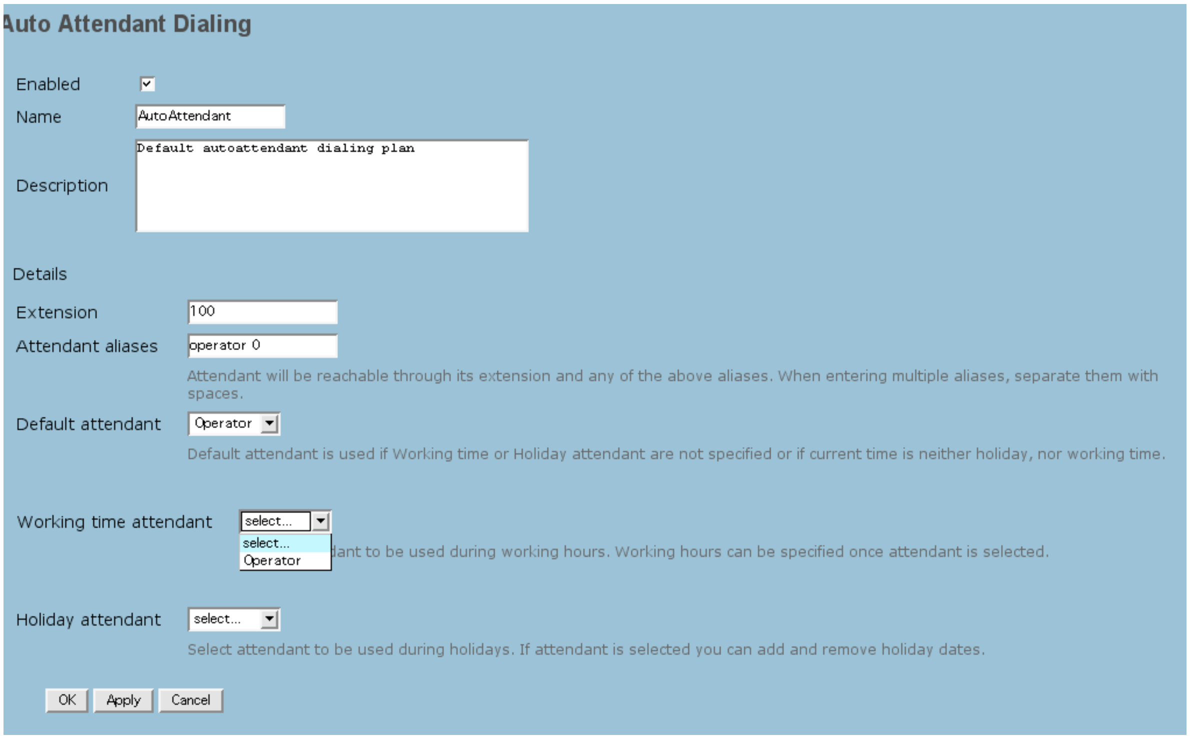


Figure 22. Editing the auto attendant dialing rule

4. Select the **Enabled** check box.
5. Enter a **Name** for the auto attendant dial plan.
6. Enter a **description** for the dial plan.
7. Enter the auto attendant's extension in the **Extension** field.
8. Enter any attendant aliases in the **Attendant Aliases** field.
9. Select a default attendant to be used when the working time attendant or holiday attendant are not applicable (e.g., after normal business hours) from the **Default Attendant** drop-down menu.
10. Select an attendant to be played during normal business hours form the **Working Time Attendant** drop-down menu. When you make a selection from this drop-down menu, a settings form displays (shown next) where you can set the days of the week/weekends and timeframes during which the Working Time Attendant should be played.
11. Select an attendant to be played during holidays from **Holiday Attendant** drop-down menu. When you make a selection from this drop-down menu, an **Add Holiday** link displays. When you click **Add Holiday**, a form field displays with the current date selected. You can set the current day by moving on to step 12 or you can click the calendar icon to the right of the form field and select a different day as well as moving on to a different month. Click **Add Holiday** again to add more days during which time the holiday attendant should be played. You can delete days by clicking **Delete Holiday** next to a respective form field.

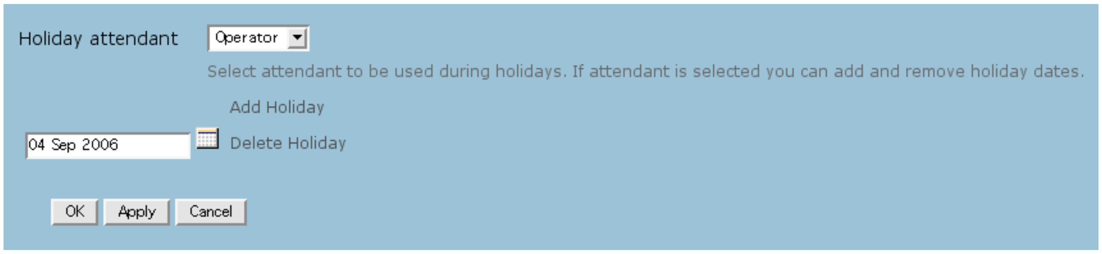


Figure 23. Setting up the holiday attendant

12. Click **OK** to save auto attendant dial plan information and return to the Dial Plans screen; click **Apply** to save auto attendant dial plan information and remain at the current screen; click **Cancel** to return to the Dial Plans screen without saving any of the auto attendant dial plan information you created.

> **Note** Dial plans are sensitive to order. You may want to move the **AutoAttendant** dial plan closer to the bottom of the dial plans list.

13. On the **Dial Plans** screen, select your newly created dial plan for the auto attendant and click **Activate** in the Dial Plan Activation section. A confirmation screen will appear. Click **OK**.

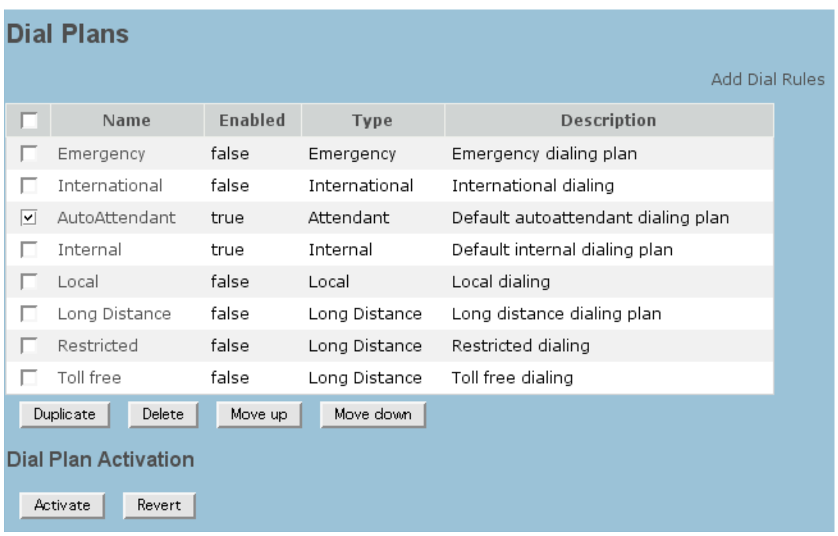


Figure 24. Activating dial plans

Setting the Special Auto Attendant

You can use the **Special Auto Attendant** feature to temporarily overwrite auto attendant configuration. By default, the auto attendant is selected based on the auto attendant dialing rules. If the **Special Auto Attendant** is activated, it is used to handle all calls.

To set up the Special Auto Attendant:

1. From the main screen, click on **Features** > **Auto Attendants.**
2. In the **Special Auto Attendants** section on the Auto Attendants screen, click on '**Use special auto attendant.**'
3. Choose an auto attendant from the '**select...**' drop-down menu.
4. Click **Apply.**

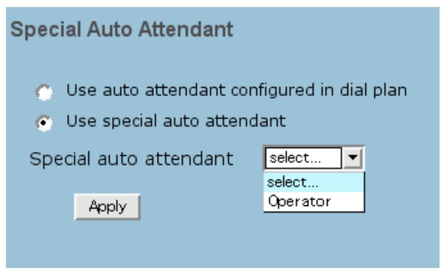


Figure 25. Setting up the special auto attendant

To turn the Special Auto Attendant off:

1. From the main screen, click on **Features** > **Auto Attendants.**
2. In the **Special Auto Attendants** section on the Auto Attendants screen, click on '**Use auto attendant configured in dial plan.**'
3. Click **Apply.**

Intercom

The Intercom feature is supported only if a Polycom phone is receiving the call, while Intercom calls can be initiated by any phone by dialing the proper intercom prefix.

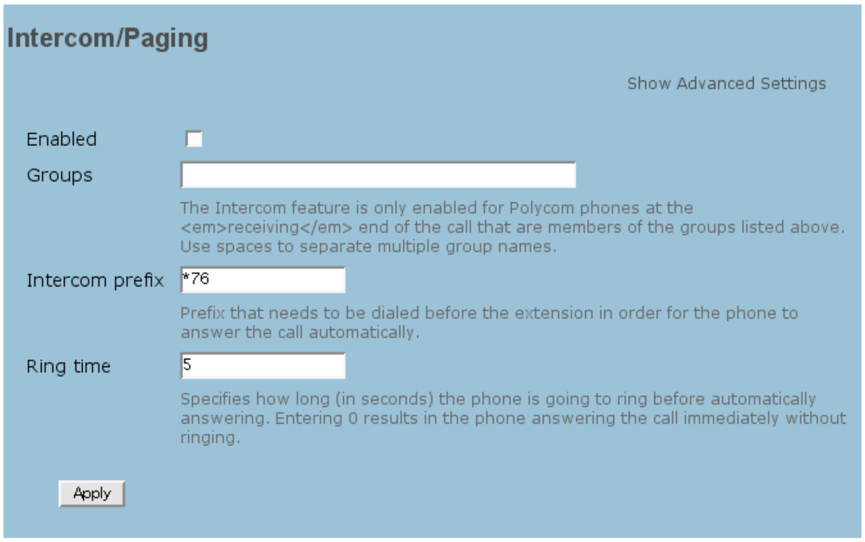


Figure 26. Configuring the intercom feature

Configuring intercom

1. Create a new phone group that only contains the Polycom phones that should be configured to answer phones automatically. (For more information on creating phone groups, see “Phone Groups” on page 31).
2. From the main screen, click **Features** > **Intercom**.
3. Enter the group name in the **Groups** field. Use spaces to separate multiple group names.

> **Note** The Intercom feature is only enabled for Polycom phones at the *receiving* end of the call that are members of the groups listed.

4. Check the **Enabled** box.
5. In the **Intercom prefix** field, enter the desired prefix that needs to be dialed before the extension in order for the phone to answer the call automatically.
6. In the **Ring time** field, enter how many seconds the phone is going to ring before automatically answering. To configure the phone to answer the call immediately, enter “0’ as the Ring time.
7. Click **Apply**.

Changes to to the **Intercom** page require re-activating the dial plan and re-sending configuration profiles to the affected phones.

Using intercom

To use the intercom feature, dial the **Intercom prefix**, then the extension of the phone.

Hunt Groups

This section covers:

- "Adding hunt groups" on page 49
- "Editing hunt groups" on page 50
- "Deleting hunt groups" on page 50

Hunt Groups allow you to set up an extension that encompasses a group of users. When a hunt group extension is dialed, such as 400, a group of users associated with this master extension fall into a call sequence. You can dial all the users within the hunt group simultaneously or set them up in a successive format so that when there is no response by one user, another user is dialed. You can also define the initial user to dial in the hunt group.

Adding hunt groups

To create a hunt group:

1. From the main screen, click **Features** > **Hunt Groups** > **Add Hunt Group**.

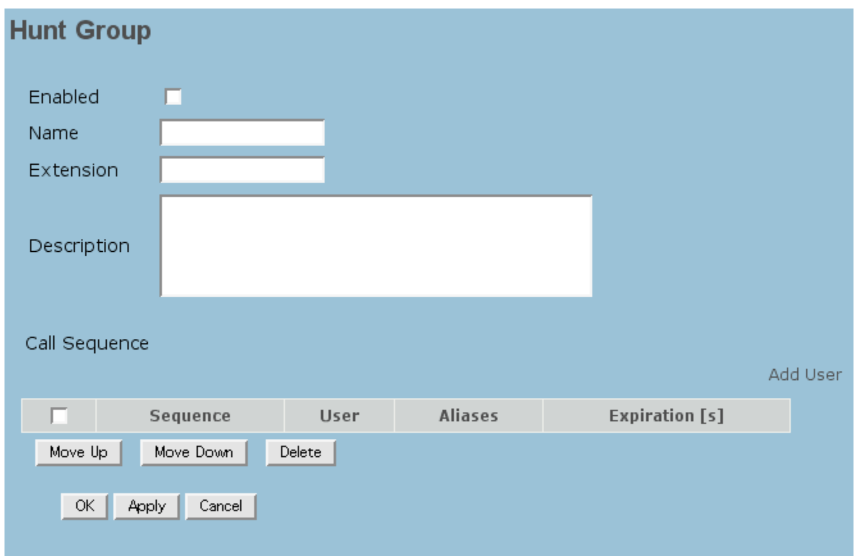


Figure 27. Adding a hunt group

2. Click the **Enabled** box to enable the hunt group.
3. Enter a name for the hunt group in the **Name** field
4. Enter an extension for the hunt group in the **Extension** field.
5. Create the call sequence for this hunt group by clicking **Add User**. Use the **search** bar to find the user you want to add to the hunt group. Click the box next to the user(s) you want to add, and click **Select**.
6. On the **Hunt Groups** screen, define the hierarchy of users by selecting an associated user's check box and clicking **Move Up** or **Move Down**. You can also delete a user from a hunt group by selecting the user's associated checkbox and clicking **Delete**.

7. Click **OK** to save hunt group information and return to the Hunt Groups screen; click **Apply** to save hunt group information and remain at the current screen; click **Cancel** to return to the Hunt Groups screen without saving any of the user information you created.

Editing hunt groups

To edit an existing hunt group:

1. From the main screen, click on **Features** > **Hunt Groups**.
2. On the Hunt Groups screen, click on the **name** of the hunt group you want to edit.
3. From this screen, you can add more users to the group, delete users from the group, or edit any of the settings you previously set.
4. To save your changes, click the **OK** button.

Deleting hunt groups

To delete a hunt group:

1. From the main screen, click on **Features** > **Hunt Groups**.
2. Check the box next to the hunt group that you want to delete.
3. Click the **Delete** button. A confirmation prompt will appear. Click **OK**.

Call Park

This section covers:

The Call Park feature, when enabled by selecting the Enabled check box on the Call Park Extension screen, provides the ability to transfer calls to call park extensions In the transfer state, the caller hears background music while waiting for someone to pick up at the extension. Calls can be retrieved by pressing *4 followed by the extension number. Select a music file, or upload new files to provide background music while calls are parked.

Adding Call Park Extensions

1. From the main screen, click **Features** > **Call Park** > **Add Call Park Extension**.

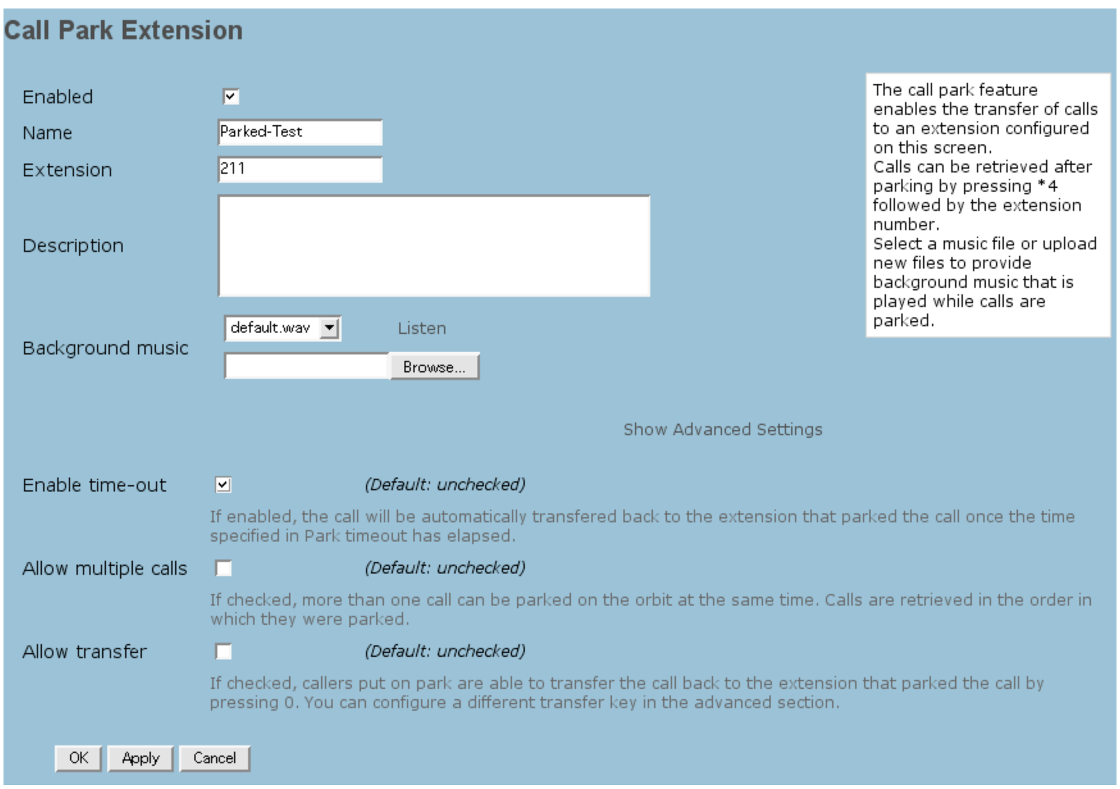


Figure 28. Adding a call park extension

2. Select the **Enabled** checkbox.
3. Enter a name for the call park extension in the **Name** field.

4. Enter an extension in the **Extension** field.
5. Enter a description in the **Description** field.
6. Select background music from the **Background Music** drop-down list. If there are no files to select or you want to upload new background music, click **Browse** and upload new music. You can create .wav files to upload as background music. For information on recording .wav files see "Recording .WAV Files (Windows OS)" on page 53.
7. Listen to background music that you have selected by clicking **Listen**.
(Ensure that your headphones or PC speakers are turned on to listen).
8. If desired, click the **Enable time-out** checkbox. If enabled, the call will automatically be transferred back to the extension that parked the call once the time specified in the **Park time-out** field has elapsed. Click the **Show Advanced Settings** link to set **Park time-out.**
9. If desired, click the **Allow multiple calls** checkbox. If enabled, more than one call can be parked on the orbit at the same time. Calls are retrieved in the order in which they were parked.
10. If desired, click the **Allow transfer** checkbox. If checked, callers put on park are able to transfer the call back to the extension that parked the call by pressing 0. To configure a new **Transfer key**, click the **Show Advanced Settings** link.
11. Click **OK** to save your changes and return to the Call Park screen; Or, click **Cancel** to return to the Call Park screen without saving changes. Click **Apply** to save changes and remain at the Call Park Extension screen.

Editing Call Park Extensions

To edit a call park extension:

1. From the main screen, click **Features** > **Call Park**.
2. On the Call Park screen, click on the **name** of the call park you want to edit.
3. Make any necessary edits such as selecting or uploading new background music.
Select background music from the **Background Music** drop-down list. If there are no files to select or you want to upload new background music, click **Browse** and upload new music. You can create .wav files to upload as background music. For information on recording .wav files see "Recording .WAV Files (Windows OS)" on page 53.
Listen to background music that you have selected by clicking **Listen**.
(Ensure that your headphones or PC speakers are turned on).
4. Click **OK** to save your changes and return to the Call Park screen; Or, click **Cancel** to return to the Call Park screen without saving changes. Click **Apply** to save changes and remain at the Call Park Extension screen.

Using Call Park Extensions

To park a call:

1. Transfer the caller to the enabled call park extension (see "Adding Call Park Extensions" on page 51).

To access a call on a call park extension:

1. Dial ***4** + <**Call Park Extension**>. For example, if the call park extension is 401, to pick up the call that is parked, you will dial ***4401**.

Deleting Call Park Extensions

To delete a call park extension:

1. From the main screen, click on **Features** > **Call Park**.
2. Check the box next to the call park extension that you want to delete.
3. Click the **Delete** button. A confirmation prompt will appear. Click **OK**.

Recording .WAV Files (Windows OS)

WAV files are recorded, uploaded, and set in SIPxNano for the following reasons:

- Auto Attendant Prompts
- Call Park Background Music

All .wav files must be recorded in mono mode with an 8kHz sampling rate and 16 bit PCM wave format.

To record .wav files using a Windows PC:

1. Select **Start** > **All Programs** > **Accessories** > **Entertainment** > **Sound Recorder**.

Figure 29. Windows Sound Recorder

2. From the file menu, select **Properties**.

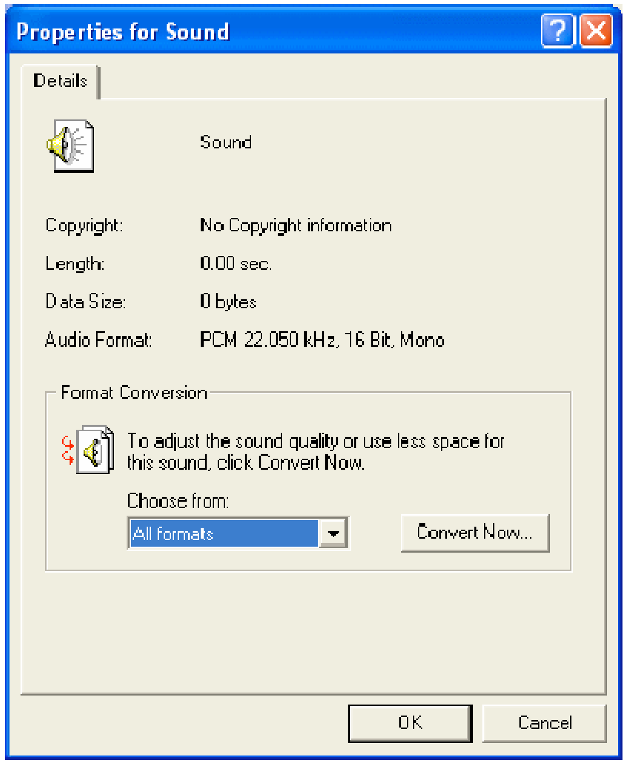


Figure 30. Properties for sound

3. Click **Convert Now**.

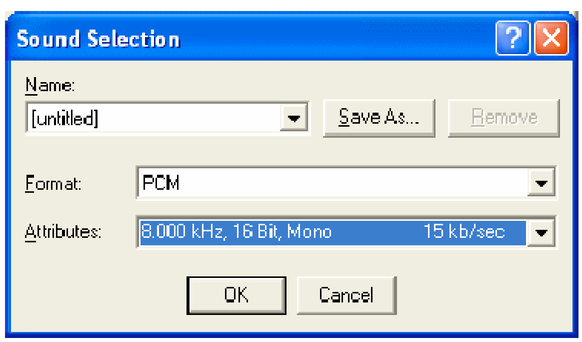


Figure 31. Sound Selection properties

4. Select **PCM** from the **Format** drop-down list.
5. Select 8.000 kHz, 16 Bit, Mono from the **Attributes** drop-down list.
6. Click **Save As...** and save the .wav file to a location you can navigate to from the SIPxNano Configuration server. Browse from the **Auto Attendant** and **Call Park** sections of the Configuration server to locate .wav files that you have recorded and upload them to SIPxNano.

Note If you use another recording device to record .wav files, ensure that the files are recorded in mono mode with an 8kHz sampling rate and 16 bit PCM wave format, and that you save the files to a location that is accessible from the SIPxNano Configuration server (e.g., a network location that you can browse to).

Music on Hold

The Music on Hold feature is supported for all phones that implement the Music on Hold service IETF draft.

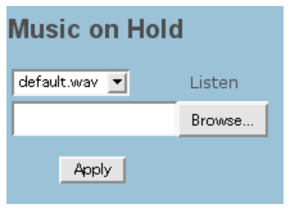


Figure 32. Configuring Music on Hold

To configure the Music on Hold feature:

1. From the main screen, click **Features** > **Music on Hold**.
2. Choose a .wav file from the drop-down menu, or click **Browse** to upload a new .wav file. (For more information on .wav files, see “Recording .WAV Files (Windows OS)” on page 53).
3. Click **Apply**.

Chapter 6 **System**

Chapter contents

Overview

The System Management section of the SIPxNano Configuration Server interface allows you to configure and view SIPxNano settings that impact SIPxNano behind the scenes (not directly apparent to users) and are important to administrators for monitoring system performance status as well as providing the ability to take precautionary measures in backing up the system. Menu options include:

- "Dial Plans" on page 57
- "Permissions" on page 60
- "General" on page 61
- "Import" on page 67
- "LDAP" on page 68
- "Backup" on page 70
- "Domain" on page 72

Dial Plans

This section covers:

- "Adding dial rules" on page 58
- "Editing and deleting dial rules" on page 58
- "Activating dial rules" on page 59
- "Setting up emergency routing" on page 59

You define system dial plans to establish how SIPxNano will route incoming and outgoing calls. SIPxNano uses these dial plans to route calls to:

- Internal and external call routing
- Voicemail
- Auto Attendant
- Local number dialing
- International number dialing

Adding dial rules

To add a dial rule:

1. Click **System** > **Dial Plans** > **Add Dial Rules**.

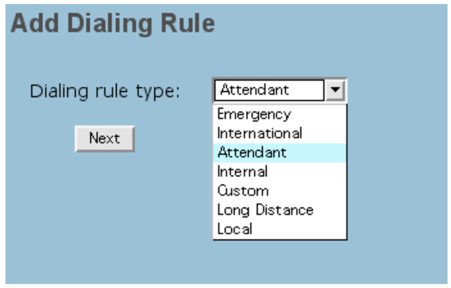


Figure 33. Add dialing rule

2. From the **Add Dialing Rule** drop-down menu, select the type of dialing rule you want to add.
3. Click the **Enabled** box.
   Make sure that you select the Enabled check box on the associated dialing rule's screen before saving an added or modified dial plan.
4. Type a name for the dialing rule in the **Name** field.
5. Type a description of the dialing rule in the **Description** field.
6. Each dialing rule has different parameters that need to be set up that are specific to that type of dialing rule. Enter your desired values for your specific dial rule.
7. If you desire, you may add a new or existing gateway by clicking on the respective gateway link.
8. Click **OK**.
9. On the main **Dial Plans** screen, click the box next to the name of the dialing rule you want to activate.
10. Click the **Activate** button under Dial Plan Activation.
11. A confirmation screen will appear. Click **OK**.

Editing and deleting dial rules

You can edit an existing dial plan by clicking on its name in the name column on the Dial Plans screen. You can also duplicate and modify a plan (e.g., create two different local dial plans with separate parameters by selecting the existing plan's associated checkbox and clicking **Duplicate**), delete dial plans by selecting the checkbox to the left of the dial plan name column and clicking **Delete**, and organize the order that your dial plan list appears by selecting a respective dial plan's checkbox and clicking the **Move Up** or **Move Down** buttons.

Note The order of dial rules is very sensitive. Be sure to put the primary Dial Rule at the top of the list, and continue organizing the list of dial rules in that manner. (For example, the AutoAttendant dial rule should be near the bottom of the list, and the Local and Long Distance dial rules should be near the top of the list).

Note The **AutoAttendant** dial rule should **never** be at the top of the **Dial Rules** list because it will catch all the other dial plans and prevent calls out.

Activating dial rules

When you have completed your dial plan configurations, select the respective dial plan's checkbox on the **Dial Plans** page and click **Activate** in the Dial Plan Activation section to activate the dial plan(s) or **Revert** to discard all changes made to a dial plan.

Setting up emergency routing

You should only use this feature if your phone does not support direct forwarding. If this feature is enabled, all emergency calls are forwarded to the default gateway. Optionally, you can configure a set of phone numbers to be forwarded to an alternative gateway

1. From the main screen, click **System** > **Dial Plans**, and then click **Emergency** from the Name column on the Dial Plans screen.

Figure 34. Emergency dialing

2. Enter the emergency number in the **Emergency Number** field (in the Details section).
3. Select a gateway from the **Gateway** list.
4. Click **OK** to save emergency routing information and return to the Emergency Dialing screen; click **Apply** to save emergency routing information and remain at the current screen; click **Cancel** to return to the Emergency Dialing screen without saving any of the routing information you created.

Permissions

To access the Permissions page, click **System** > **Permissions.**

Permissions defined on the **Permissions** page under System can be used when defining dialing plan rules.Built-in permissions cannot be changed or removed. You can enable and disable permissions for User and User Groups. (For more information on setting permissions for specific users, see "Permissions" on page 20 in Chapter 3, "Users").

Table 5. Default Permissions

Feature:	Description:
900 Dialing	User can dial 900 numbers
Attendant Directory	List user in Auto Attendant
Forward Calls External	User can forward calls to external numbers
International Dialing	User can dial international numbers
Local Dialing	User can dial local numbers
Long Distance Dialing	User can dial long distance numbers
Mobile Dialing	User can dial mobile numbers
Record System Prompts	User can record system prompts
Toll Free	User can dial toll free numbers
Voicemail	User has a voicemail inbox and can receive voicemail messages

Adding permissions

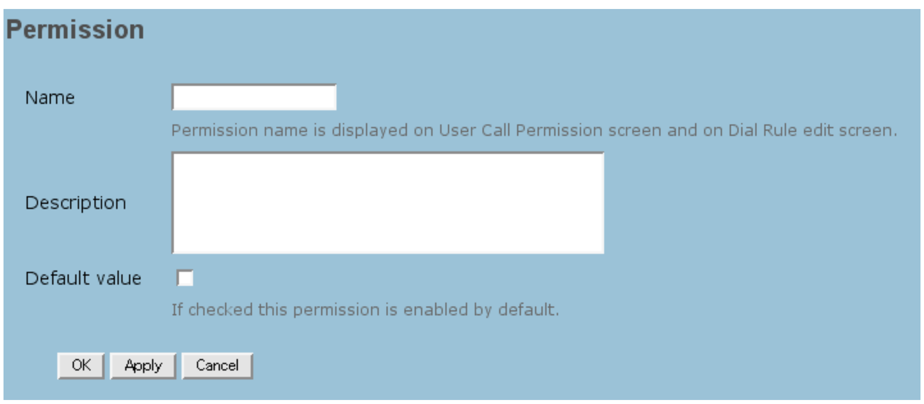


Figure 35. Adding a permission

To add a permission:

1. Click **System** > **Permissions**. Then, click on the **Add Permission** hyperlink.
2. Enter a name for the permission in the **Name** field. The Permission name is displayed on the User Cal Permission screen and on the Dial Rule edit screen.
3. Enter a description for the permission in the **Description** field.
4. Check the **Default value** box to enable the new permission by default.
5. Click **Apply**, then **OK**.

General

This section covers:

To access the **General** screen, click **System** > **General**. General configuration of SIPxNano includes SIP paramters, voicemail, call pickup, presence server, logging changes and settings, and call detail records (CDRs).

Note When you click **Apply** after making changes to the sections under **General**, the **Restart Services** screen will display. You can click **Continue** and restart SIPxNano services or if the work you are currently performing requires you to continue without restarting services, you can select the Do Not Restart Services Now checkbox. In either case, services must be restarted at some point for the Configuration changes to take effect, but settings will be saved until that time.

SIP Parameters

To display editable SIP Parameters, click on the **Show Advanced Settings** link. You can edit the following:

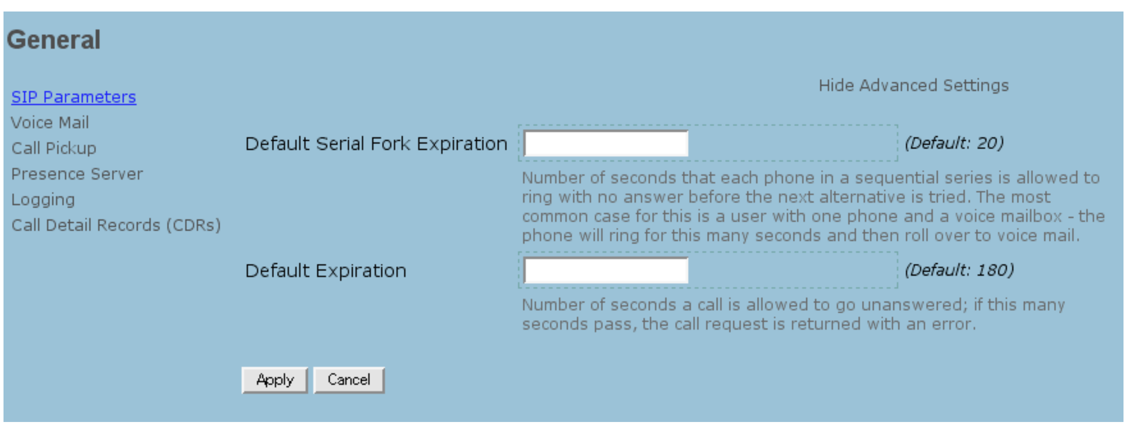


Figure 36. SIP Parameters

- **Default Serial Fork Expiration:** Number of seconds that each phone in a sequential series is allowed to ring with no answer before the next alternative is tried. The most common case for this is a user with one phone and a voice mailbox - the phone will ring for this many seconds and then roll over to voice mail. The default value is 20.
- **Default Expiration:** Number of seconds a call is allowed to go unanswered; if this many seconds pass, the call request is returned with an error. The default value is 180.

Voice Mail

The Voice Mail section under General has configurable fields including: VoiceMail expiration (which defines the number of days voicemail messages should remain on the system before they are deleted) and Maximum internal extension length (which defines the longest internal extension allowed in the system dialing plan). This is pertinent to the auto attendant prompt for determining whether a user has entered the entire extension.

For more information about Voice Mail, see Chapter 8, “Voicemail” on page 78.

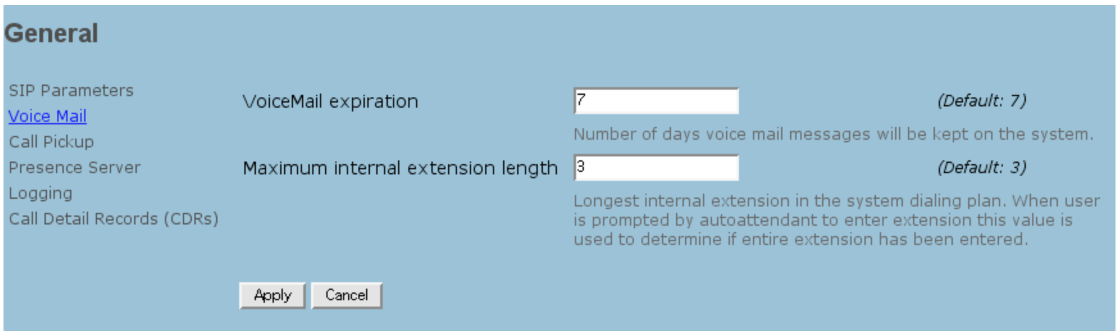


Figure 37. Voicemail settings in the General menu

The settings you can configure are:

- **VoiceMail expiration:** Number of days voice mail messages will be kept in the system. (Default = 7)
- **Maximum internal extension length:** Longest internal extension in the system dialing plan. When a user is prompted by auto attendant to enter an extension, this value is used to determine if the entire extension has been entered. (Default = **3**).
- **SMTP server:** IP address or DNS name of SMTP server used for Voice Mail notification. (No default).

Call Pickup

Enter dialing codes for global call pickup, directed call pickup, and call park retrieve. Click Show Advanced Settings enter the setting for Call Pickup Timeout, in seconds, that the original call will ring after the pickup code is dialed.

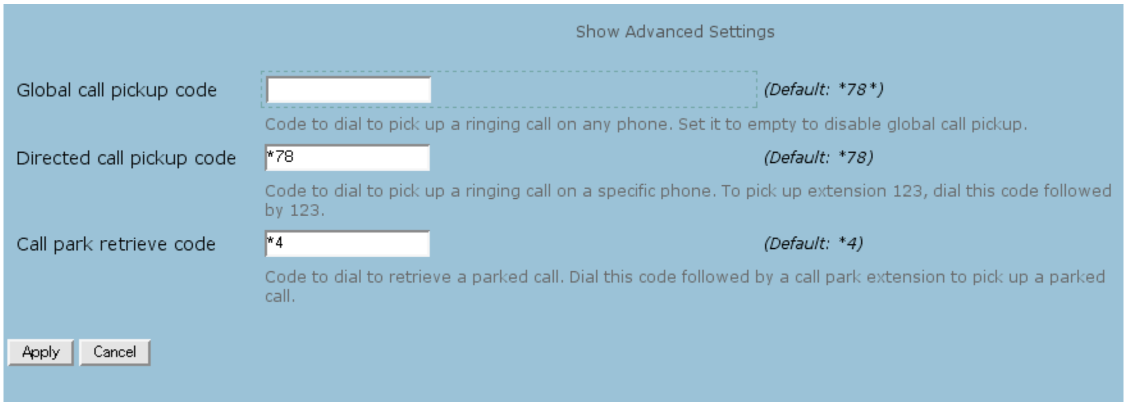


Figure 38. Call pickup

The settings you can configure are:

- **Global pickup code:** Code to dial to pick up a ringing call on any phone. Set it to empty to disable global call pickup. (Default = *78*)
- **Directed call pickup code:** Code to dial to pick up a ringing call on a specific phone. To pick up extension 123, dial this code followed by 123. (Default = *78).
- **Call park retrieve code:** Code to dial to retrieve a parked call. Dial this code followed by a call park extension to pick up a parked call. (Default = *4).
- **Call pickup timeout** (Advanced Setting)**:** Number of seconds that the original call will ring after the pickup code is dialed. (Default = 2).

Presence Server

This server determines the code that people use to log in and out of the system at phones. The default login is *88 and the default logout is *86, but you can change these based on your requirements.

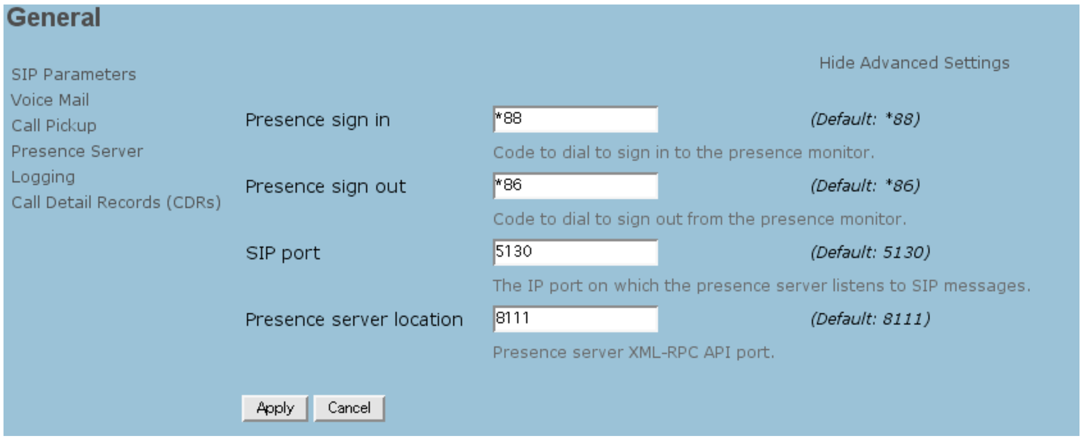


Figure 39. Presence server

The settings you can configure are:

- **Presence sign in:** Code to dial to sign in to the presence monitor. (Default = ***88**)
- **Presence sign out:** Code to dial to sign out from the presence monitor. (Default = ***86**).
- **SIP port** (Advanced Setting)**:** The IP port on which the presence server listens to SIP messages. (Default = **5130**).
- **Presence server location** (Advanced Setting)**:** Presence server XML-RPC API port. (Default = **8111**).

Note To view Advanced Settings, click on the '**Show Advanced Settings**' link.

Logging

Logging parameters set here determine the level of log detail for associated components, including Authorization Proxy, Proxy, Registrar, Park Server, Status Server, and the Presence Server.

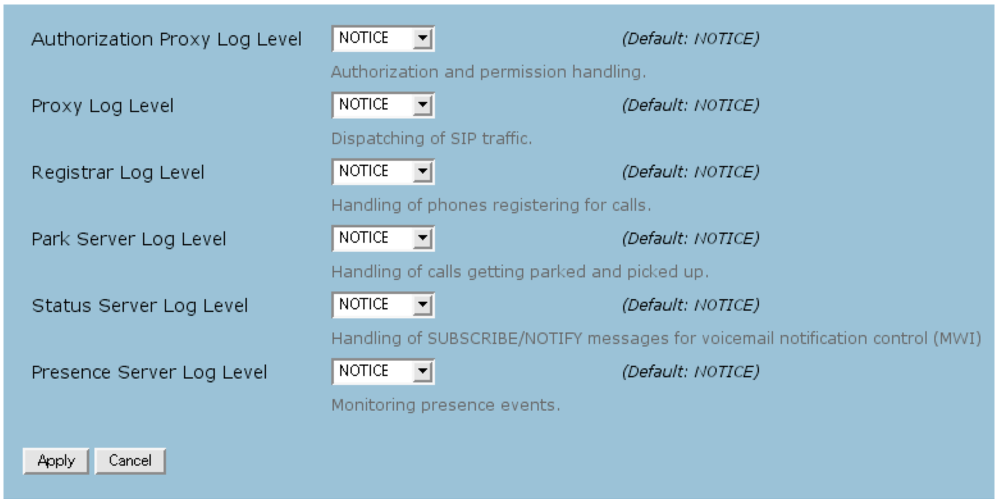


Figure 40. Logging

The settings you can configure are:

- **Authorization Proxy Log Level:** Authorization and permission handling. (Default = **NOTICE**)
- **Proxy Log Level:** Dispatching of SIP traffic. (Default = **NOTICE**).
- **Registrar Log Level:** Handling of phones registering for calls. (Default = **NOTICE**).
- **Park Server Log Level:** Handling of calls getting parked and picked up. (Default = **NOTICE**).
- **Status Server Log Level:** Handling of SUBSCRIBE/NOTIFY messages for voicemail notification control (MWI). (Default = **NOTICE**).
- **Presence Server Log Level:** Monitoring presence events. (Default = **NOTICE**).

Note If you use '**DEBUG**', be sure to change the '**DEBUG**' value back to '**NOTICE**' when you are done using it. Leaving a logging parameter set at '**DEBUG**' for an extended period of time **will cause severely adverse effects on system performance.**

Call Detail Records (CDRs)

Call Detail Records capture information about calls handled by your SIPxNano ECS system. They can be used for billing and cost accounting.

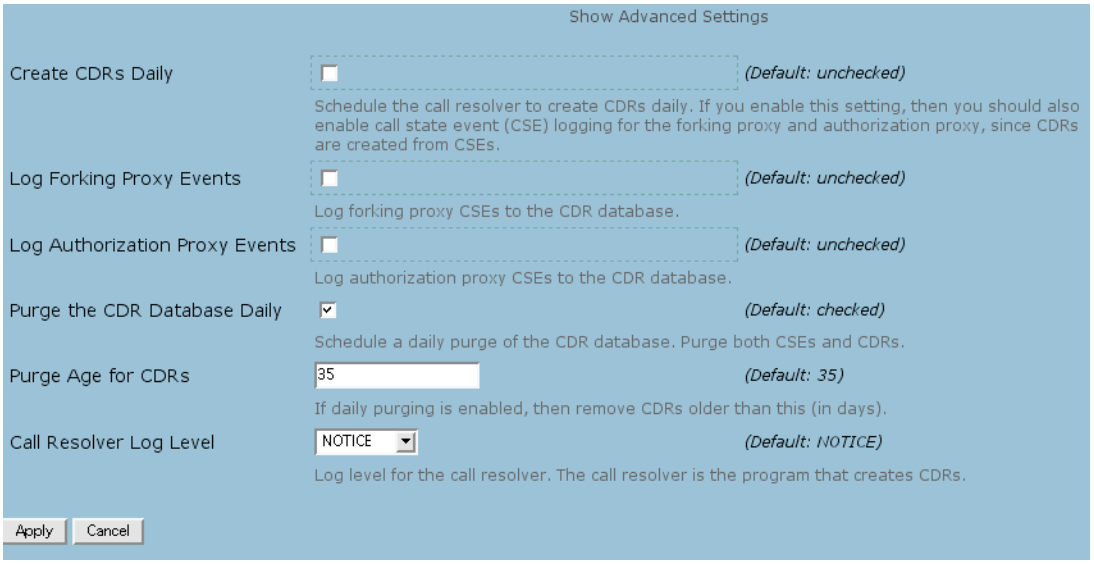


Figure 41. Call Detail Records (CDR)

The settings you can configure are:

- **Create CDRs Daily:** Schedule the call resolver to create CDRs daily. If you enable this setting, then you should also enable call state event (CSE) logging for the forking proxy and authorization proxy, since CDRs are created from CSEs. (Default = **unchecked**)
- **Log Forking Proxy Events:** Log forking proxy CSEs to the CDR database. (Default = **unchecked**).
- **Log Authorization Proxy Events:** Log authorization proxy CSEs to the CDR database. (Default = **unchecked**).
- **Purge the CDR Database Daily:** Schedule a daily purge of the CDR database. Purge both CSEs and CDRs. (Default = **checked**).
- **Purge Age for CDRs:** If daily purging is enabled, then remove CDRs older than this (in days). (Default = **35**).
- **Purge Age for CSEs** (Advanced Setting)**:** If daily purging is enabled, then remove CSEs older than this (in days). (Default = 7).
- **Call Resolver Log Level:** Log level for the call resolver. The call resolver is the program that creates CDRs. (Default = **NOTICE**).
- **Call Direction** (Advanced Setting)**:** Compute call direction. Call direction labels calls as incoming, outgoing, or intranetwork based on whether the call came from a PSTN gateway, went to a PSTN gateway, or neither, respectively. (Default = **unchecked**).

Import

You can generate a comma separated values file (**CSV**) by exporting data from your favorite spreadsheet application (e.g., Microsoft Excel).

Figure 42. Importing CSV files

Importing CSV files

1. From the main screen, click **System** > **Import**.
2. Click **Browse** to select the CSV file you want to import.
   The CSV file should have a title line and the following fields:
   - User Name
   - Voicemail PIN
   - SIP Password
   - First Name
   - Last Name
   - User Alias
   - User Group
   - User's Phone Serial Number
   - User's Phone Manufacturer
   - User's Phone Model
   - User's Phone Group
   - User's Phone Description
3. Click **Import**.

Each line from an imported file results in the creation of the phone and the user assigned to that phone. If the user group or phone group fields are not already in the system, the user and phone groups are created when the CSV file is imported and the associated users and phones are placed in those groups.

LDAP

Lightweight Directory Access Protocol (LDAP) can be used to import information on users and their attributes. Currently, LDAP on the SIPxNano works only with OpenLDAP. Before you can import LDAP, you must first configure the LDAP server.

Configuring the LDAP server

To configure the LDAP server:

1. Click on **System** > **LDAP**. Then, click on the **LDAP Server** link under **Quick Links.**

Figure 43. LDAP Server link

2. On the LDAP Server screen, enter the following information:

- **Host:** Enter the IP address of the name of the host on which the LDAP server is running.
- **Port:** Enter the port number on which the LDAP server is listening for requests. The default port number is 389.
- **User:** Enter the distinguished name of the user to bind to the LDAP directory. Leave this field empty is accessing anonymously.
- **Password** and **Confirm Password:** Enter and re-enter the password for simple authentication.

3. Click **Continue**.
4. On the LDAP attribute mapping configuration screen, the following fields can be mapped:

Table 6. LDAP Mapping Fields

Field	LDAP Attribute	Description
user id	uid	*This field is required.* A unique user identification. A user's extension can be used, or other identifiers can be used.
firstname	gn	User's first name.
lastname	sn	User's last name.
aliases	telephoneNumber	Multiple attributes can be mapped to this field. Any values that are not unique will be dropped.
Voicemail PIN	-	Secret personal identification number used by user to access voice mail.
SIP Password	-	The password used by the phone to register with the SIPx-Nano. You have the option to: a) map this field to the LDAP attribute, b) set the initial value for all fields, or c) let SIPx-Nano randomly generate a value.
Group	-	Multi-value attribute containing user group name.

Import LDAP

To setup an LDAP import:

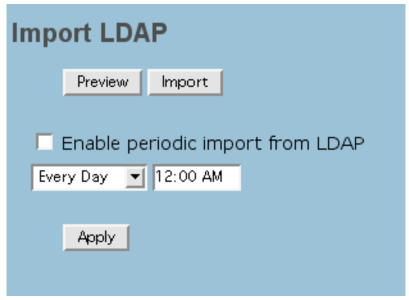


Figure 44. Import LDAP

1. From the main screen, click **System** > **LDAP**.
2. Click **Preview** to see the example of the user imported from the LDAP server. Click **OK**.
   If you receive an error, then the LDAP server is configured incorrectly and you will need to fix the server settings. See "Configuring the LDAP server" on page 68.
3. On the Import LDAP screen, click **Import**. Click **Job Status** under **Quick Links** to monitor the import.
4. To enable periodic imports from LDAP, click the **Enable periodic import from LDAP** checkbox. Choose the timing of the periodic imports from the drop-down menu, then click **Apply**.

Backup

This section covers:

- "Performing a backup" on page 70
- "Scheduling backups" on page 71
- "Restoring the system" on page 71

System administrators should regularly create backup copies of the SIPxNano data files so that site-specific SIPxNano data can be readily restored in the event of a disk failure or other loss of data. Administrators can back up configuration and profile information as well as voicemail.

Performing a backup

To perform a backup:

1. From the main screen, click **System** > **Backup**.

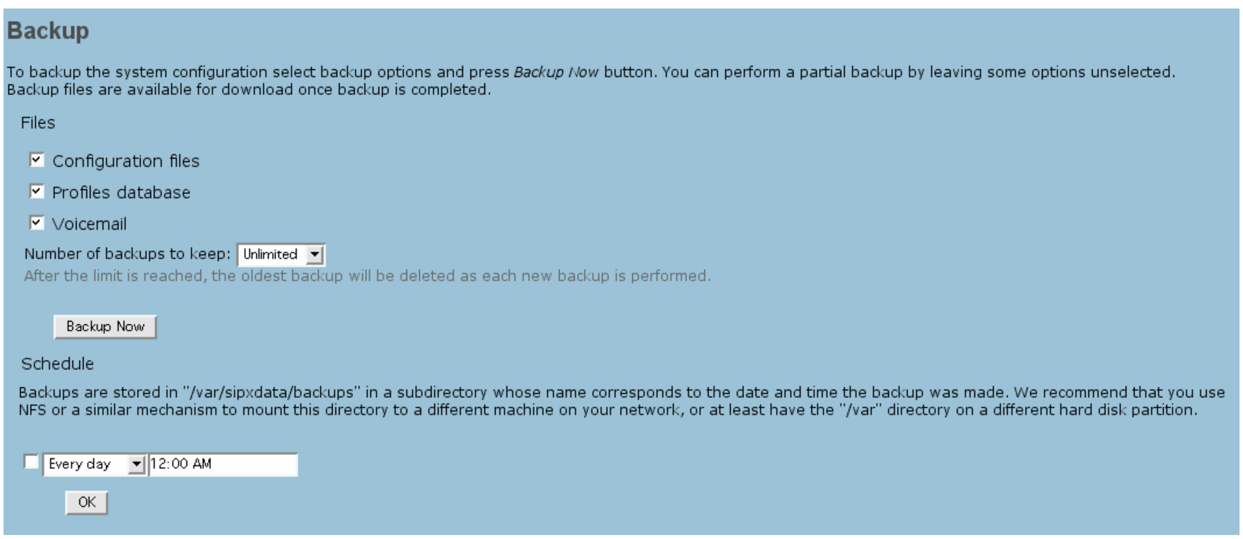


Figure 45. Backing up system files

2. Under **Files,** check the boxes next to the files you want to backup.
3. From the drop-down menu, select the **Number of backups to keep**.

4. Click the **Backup Now** button.
A message is displayed below the Backup Now button indicating whether the backup completed successfully or not. If the backup is successful, you can download the archived data by clicking the displayed directory path links, which can include *backup-configs/fs.tar.gz*, *backup-configs/pds.tar.gz*, and *backup-mailstore/mailstore.tar.gz*.

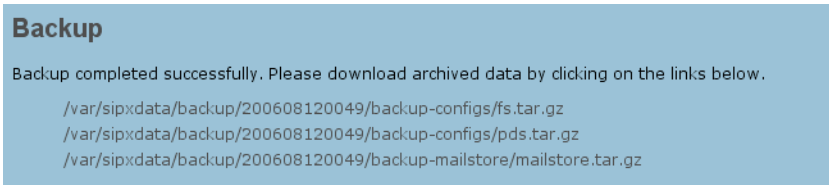


Figure 46. Successful backup

Note SIPxNano backup files should be stored on media or a machine other than the SIPxNano server.

Scheduling backups

In addition to performing a manual backup, you can schedule backups.

To schedule a backup:

1. From the main screen, click **System** > **Backup**.
2. Select the **Schedule** checkbox (next to the '**Every day**' menu).
3. Select a specific day or every day from the '**Every day**' drop-down menu, and enter the time you want to backup.
4. Click **OK**.

Restoring the system

Before restoring the system, transfer the backup files from the media or storage machine back to the relative folder, either backup-configs or backup-mailstore.

To restore the system:

1. In a command-line prompt, type **restore-configs.sh**.
This will restore files from the **fs.tar.gz** and **pds.tar.gz** files.
2. To restore voicemail data, in a command-line prompt, type **restore-mailstore.sh.**
This will restore files from the **mailstore.tar.gz** file.

Note The restore files must be started from the proper directory or you will corrupt the system. For example, voicemails **must** be restored from the voicemail directory, etc..

Domain

To access the Domain page, click **System** > **Domain**.

Note If the Domain Name setting is changed in a production system, all the phone profiles have to be regenerated. Use with caution.

Managing domain

The domain name field contains the domain name for handling SIP. If DNS SRV is not configured to send all SIP traffic to the SIP Proxy on the machine, enter the FQDN (Fully Qualified Domain Name).

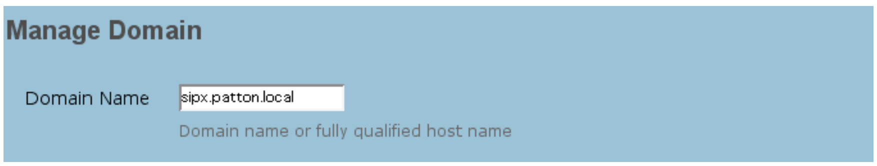


Figure 47. Managing the domain name

Note This is the domain responsible for handling SIP. Enter the fully qualified host name if DNS SRV is ***not*** configured to send all SIP traffic to the SIP proxy on this machine. This controls how SIP addresses are built and used through the system. For example, sip:user@old-value becomes sip:user@new-value. Changing this value can have various side effects depending on how your system is setup.

Domain aliases

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

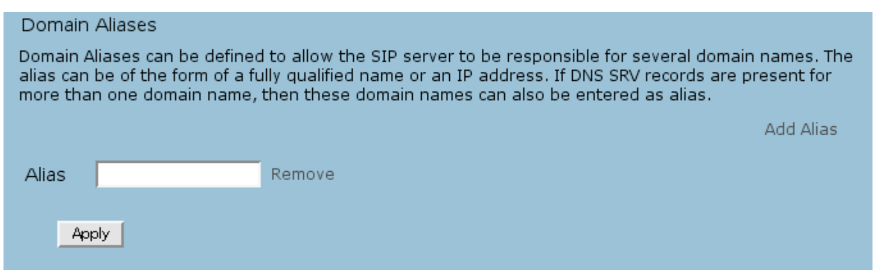


Figure 48. Adding a domain alias

To add a domain alias, click the **Add Alias** link, then enter a name in the **Alias** field and click **Apply**.

Click the **Remove** link to remove a domain alias.

Chapter 7 **Diagnostics**

Chapter contents

Overview

This chapter provides information on SIPxNano diagnostics. Menu options include:

Registrations

From the **Registrations** screen, you can monitor all phones registered with SIPxNano as well as showing or hiding primary registrar by clicking on that link in its current state.

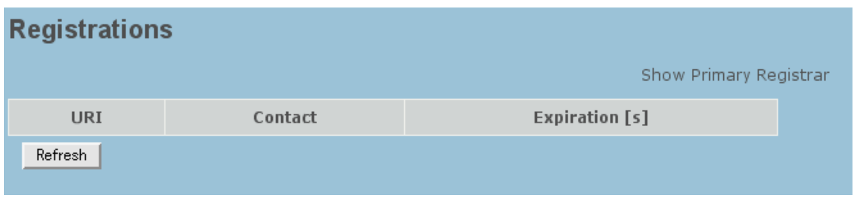


Figure 49. Registrations

Viewing registrations

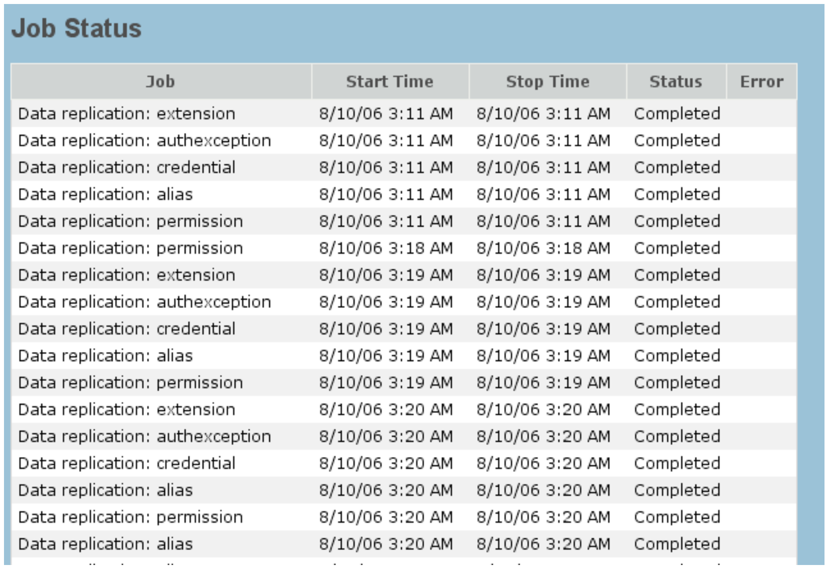
To view active registrations:

1. From the main screen, click **Diagnostics** > **Registrations**.
2. Click **Refresh**.

If you have high availability configured then you can display an additional Primary Registrar column. It contains the name of the registration server that handled the initial registration for a contact.

Job Status

When you send profiles to managed phones, the Job Status log records and displays information, including the date and time of your request and its status.



Job Status

Job	Start Time	Stop Time	Status	Error
Data replication: extension	8/10/06 3:11 AM	8/10/06 3:11 AM	Completed	
Data replication: authexception	8/10/06 3:11 AM	8/10/06 3:11 AM	Completed	
Data replication: credential	8/10/06 3:11 AM	8/10/06 3:11 AM	Completed	
Data replication: alias	8/10/06 3:11 AM	8/10/06 3:11 AM	Completed	
Data replication: permission	8/10/06 3:11 AM	8/10/06 3:11 AM	Completed	
Data replication: permission	8/10/06 3:18 AM	8/10/06 3:18 AM	Completed	
Data replication: extension	8/10/06 3:19 AM	8/10/06 3:19 AM	Completed	
Data replication: authexception	8/10/06 3:19 AM	8/10/06 3:19 AM	Completed	
Data replication: credential	8/10/06 3:19 AM	8/10/06 3:19 AM	Completed	
Data replication: alias	8/10/06 3:19 AM	8/10/06 3:19 AM	Completed	
Data replication: permission	8/10/06 3:19 AM	8/10/06 3:19 AM	Completed	
Data replication: extension	8/10/06 3:20 AM	8/10/06 3:20 AM	Completed	
Data replication: authexception	8/10/06 3:20 AM	8/10/06 3:20 AM	Completed	
Data replication: credential	8/10/06 3:20 AM	8/10/06 3:20 AM	Completed	
Data replication: alias	8/10/06 3:20 AM	8/10/06 3:20 AM	Completed	
Data replication: permission	8/10/06 3:20 AM	8/10/06 3:20 AM	Completed	
Data replication: alias	8/10/06 3:20 AM	8/10/06 3:20 AM	Completed	

Figure 50. Job Status log

Viewing the job status log

To view the job status log:

1. From the main menu, click **Diagnostics** > **Job Status**.
   If the Configuration server is in the process of running a job, you can view its Job Name, Start Time, Stop Time, Status, and any errors that occur.
2. To refresh the job status log, click **Refresh**.

Clearing the job status log

To clear the job status log:

1. From the main menu, click **Diagnostics** > **Job Status**.
2. To clear completed jobs, click the **Clear Completed** button.
3. To clear all jobs, click the **Clear All** button.

Services

Services allows you to start, stop, restart, and refresh all component services of the SIPxNano server, including CommServer, ConfigServer and ACD Server, and MediaServer. You can also start, stop, and restart parts of SIPxNano components, such as SIPAuthProxy, SIPProxy, SIPRegistrar, and SIPStatus, all parts of the CommServer.

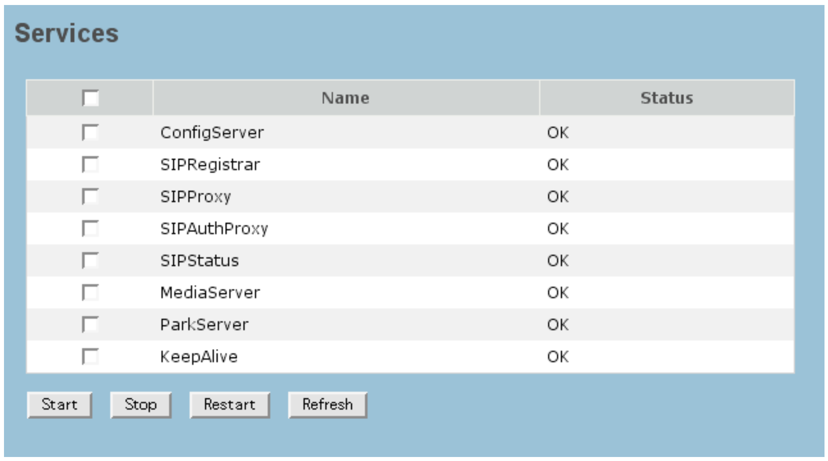


Figure 51. SIPxNano Services

Snapshot

Snapshot allows you to capture **Log files**, **Apache configuration files**, and if you are diagnosing a problem that requires real credential information, you can select the **Credentials** checkbox to capture password information from the credentials file.

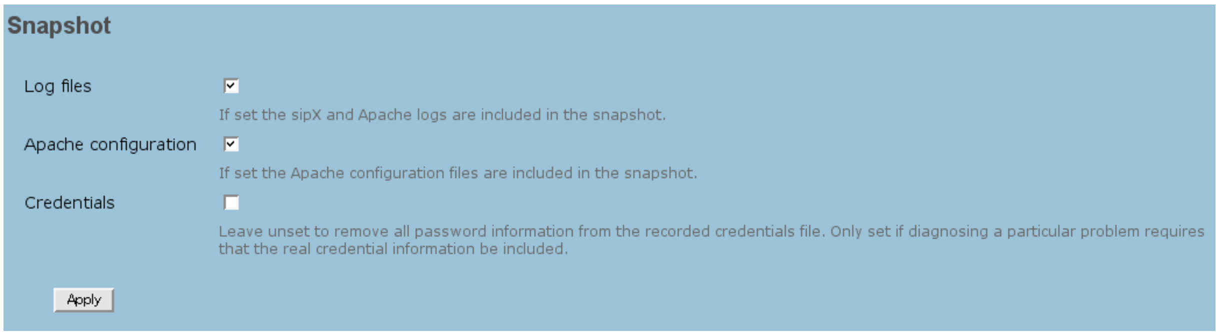


Figure 52. Snapshot

After selecting or deselecting the appropriate checkboxes, click **Apply**.
The **sipx-configuration.tar.gz** archive file is created and made available as a link on the Snapshot screen.

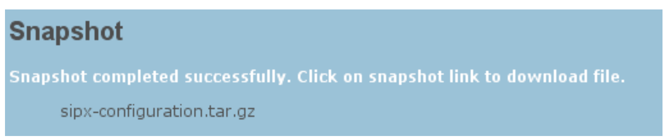


Figure 53. Successful snapshot

You can download this archive by clicking the **sipx-configuration.tar.gz link** and clicking **Save** on the File Download pop-up window that displays.

Chapter 8 **Voicemail**

Chapter contents

Overview

Each existing user has voicemail access, but the voicemail page is separate from the main screen.

Note You can only access the voicemail webpage with an active **User ID** and **PIN.** For more information about adding new users and setting up user voicemail privileges, see Chapter **3,** "Users" on page 14.

Accessing the Voicemail webpage

To access the Voicemail webpage, from the main screen, click:

- **Voice Mail** > **Inbox**

A prompt will appear asking for a user name and password.

- Enter your **Extension** and **PIN**, and click **OK.**

Note To **logout** of the voicemail page, the user must exit the web browser.

Note Online help is available for the **Voicemail** webpage. To access voicemail help, click on the blue **Help** link at the top of the Voicemail webpage.

Managing Voicemail Messages

When you first log in to the **Voicemail** interface, click on the **Inbox** to access the login prompt.

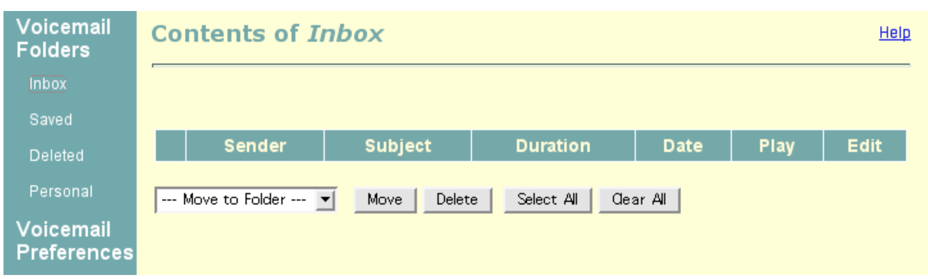


Figure 54. Voicemail inbox

From the **Inbox**, you can:

- Listen to messages
- Move messages to other folders
- Delete messages

To access Voicemail folders other than the Inbox, click on the respective link under **Voicemail Folders** in the left-hand menu. The **Saved, Deleted,** and **Personal** folders are defaults.

Listening to messages

To listen to voice messages:

1. Click on the folder you'd like to access (Inbox, Saved, Deleted, or Personal).
2. Under the **Play** column, click on the speaker icon by the message you want to hear.

Moving messages

To move a message:

1. Check the box next to the message you want to move.
2. In the drop-down menu, select the folder you'd like to move the message to.
3. Click the **Move** button.

Deleting messages

To delete a message:

1. Check the box next to the message you want to delete.
2. Click the **Delete** button.

Editing messages

Every new voice message has the default name,"Voice Message." To organize your messages, you may want to rename each subject. To rename the subject of a message:

1. Click on the pencil icon in the **Edit** column of the message you want to rename.
2. Type the new subject.
3. Click **Save.**

Editing Voicemail Preferences

Under the **Voicemail Preferences** section on the Voicemail webpage, the following options are available: **Manage Folders**, **Manage Greetings**, **Manage Distribution**, and **Mange Notifications.**

Manage Folders

From the **Manage Folders** page, you can add, delete, and edit personal folders.

To add a folder

1. Click on the **Create Folder** button.
2. Type a name for the folder in **Folder Name** field.
3. Click **Save.**

To delete or edit a folder

1. Click on the **Manage Folders** link.
2. To **edit** an existing personal folder, click on the pencil icon in the **Edit** column of the folder you want to edit. Click **Save.**
3. To **delete** a personal folder, click on the trash can icon in the **Delete** column of the folder you want to delete. (This will delete all of the messages in that folder, along with the folder itself).

Manage Greetings

From the **Manage Greetings** folder, you can play a greeting or select an active greeting. The default greeting is, "The owner of extension (your extension number or user ID) is not available." You can change the default greeting by recording your name and a personal greeting. You can record up to three (3) personal greetings.

To play a greeting

1. Click on the **Manage Greetings** link under Voicemail Preferences.
2. Click on the speaker icon in the **Play** column of the greeting you want to hear.

To make a greeting active

1. Click on **Manage Greetings** under Voicemail Preferences.
2. From the **Change Active Greeting** drop-down menu, choose the greeting you want to make active.
3. Click **Save.** The page will refresh and a check mark will appear next to the greeting you selected.

To record a greeting

1. Log in to voice mail on your phone.
2. Select voice mail options to record your name or greeting.

Manage Distributions

Distribution lists are used for delivering the messages to multiple parties.

To add a distribution list

1. Click on the **Manage Distributions** link under Voicemail Preferences.
2. Click **Add Distribution** to create your distribution lists.
3. Enter the mailbox address(es) in the **Destinations** field. Use a comma to separate addresses in the distribution list.
4. Click **Save.**

Manage Notifications

Notifications can be sent to your email when you receive a new voice mail.

1. Click on the **Manage Notifications** link under Voicemail Preferences.
2. Click **Add Notification** to specify your email address.
3. Enter your email address in the **Email Address** field.
4. If you want to attach a voice message, select the **Attach voice message** checkbox.
5. Click **Save.**

Note The **Email Notification Feature** must be enabled by your SIPxNano administrator in order for individual users to set up email notifications.

Chapter 9 Contacting Patton for assistance

Chapter contents

Introduction

This chapter contains the following information:

- "Contact information"—describes how to contact Patton technical support for assistance.
- "Warranty Service and Returned Merchandise Authorizations (RMAs)"—contains information about the warranty and obtaining a return merchandise authorization (RMA).

Contact information

Patton Electronics offers a wide array of free technical services. If you have questions about any of our other products we recommend you begin your search for answers by using our technical knowledge base. Here, we have gathered together many of the more commonly asked questions and compiled them into a searchable database to help you quickly solve your problems.

Patton support headquarters in the USA

- Online support: available at **www.patton.com**
- E-mail support: e-mail sent to **support@patton.com** will be answered within 1 business day
- Telephone support: standard telephone support is available five days a week—from **8:00 am** to **5:00 pm EST** (**1300** to **2200 UTC/GMT**)—by calling **+1 (301) 975-1007**
- Fax: **+1 (253) 663-5693**

Alternate Patton support for Europe, Middle East, and Africa (EMEA)

- Online support: available at **www.patton-inalp.com**
- E-mail support: e-mail sent to **support@patton-inalp.com** will be answered within 1 business day
- Telephone support: standard telephone support is available five days a week—from **8:00 am** to **5:00 pm CET** (**0900** to **1800 UTC/GMT**)—by calling **+41 (0)31 985 25 55**
- Fax: **+41 (0)31 985 25 26**

Warranty Service and Returned Merchandise Authorizations (RMAs)

Patton Electronics is an ISO-9001 certified manufacturer and our products are carefully tested before shipment. All of our products are backed by a comprehensive warranty program.

> **Note** If you purchased your equipment from a Patton Electronics reseller, ask your reseller how you should proceed with warranty service. It is often more convenient for you to work with your local reseller to obtain a replacement. Patton services our products no matter how you acquired them.

Warranty coverage

Our products are under warranty to be free from defects, and we will, at our option, repair or replace the product should it fail within one year from the first date of shipment. Our warranty is limited to defects in workmanship or materials, and does not cover customer damage, lightning or power surge damage, abuse, or unauthorized modification.

Out-of-warranty service

Patton services what we sell, no matter how you acquired it, including malfunctioning products that are no longer under warranty. Our products have a flat fee for repairs. Units damaged by lightning or other catastrophes may require replacement.

Returns for credit

Customer satisfaction is important to us, therefore any product may be returned with authorization within 30 days from the shipment date for a full credit of the purchase price. If you have ordered the wrong equipment or you are dissatisfied in any way, please contact us to request an RMA number to accept your return. Patton is not responsible for equipment returned without a Return Authorization.

Return for credit policy

- Less than 30 days: No Charge. Your credit will be issued upon receipt and inspection of the equipment.
- 30 to 60 days: We will add a 20% restocking charge (crediting your account with 80% of the purchase price).
- Over 60 days: Products will be accepted for repairs only.

RMA numbers

RMA numbers are required for all product returns. You can obtain an RMA by doing one of the following:

- Completing a request on the RMA Request page in the *Support* section at **www.patton.com**
- By calling **+1 (301) 975-1007** and speaking to a Technical Support Engineer
- By sending an e-mail to **returns@patton.com**

All returned units must have the RMA number clearly visible on the outside of the shipping container. Please use the original packing material that the device came in or pack the unit securely to avoid damage during shipping.

Shipping instructions

The RMA number should be clearly visible on the address label. Our shipping address is as follows:

Patton Electronics Company
RMA#: xxxx
7622 Rickenbacker Dr.
Gaithersburg, MD 20879-4773 USA

Patton will ship the equipment back to you in the same manner you ship it to us. Patton will pay the return shipping costs.

Appendix A **Session Initiation Protocol (SIP)**

Chapter contents

Overview

The Session Initiation Protocol (SIP) establishes a standard methodology for setting up, maintaining, and ending interactive communication sessions. To perform these tasks, SIP, like HTTP (Hypertext Transfer Protocol), uses a request-response model in which messages are exchanged by system components.

The particular content of a session is described in, but not included with, SIP messages. Phones that use SIP to manage the transmission of voice data during phone calls can also be used for the transmission of other types of data, such as video, fax, and multimedia.

> **Note** SIP messages may actually travel over different networks than the audio or video packets they describe.

This section introduces the contents of SIP messages and describes how the messages are organized into transactions, sessions, and calls. The Internet Engineering Task Force (IETF) Request for Comments (RFC) 3261 provides the specification for the SIP protocol.

SIP Messages

SIP is ASCII-based. All SIP messages are formatted as text using HTTP syntax. SIP messages contain call control methods for requests or response codes for replies, and are exchanged to:

- Initiate a session between an originator and a target. This involves:
  - Locating the target
  - Determining whether or not the target is available
  - Determining the media capabilities of the target
- Establish a connection between the originator and the target
- End the session by terminating or transferring the connection

Descriptions of the SIP methods and response codes, the required fields in SIP message headers, SIP URLs, and examples of SIP messages and message flows follow.

Methods

SIP call control methods identify the type of request that is being made. The following table lists SIP methods and their functions.

Table 7. SIP Methods and Functions

Method	Function
INVITE	Invites a target to participate in a session; establishes a connection. Also used to change call state or capabilities, such as the codec used.
ACK	Confirms receipt of a final response to an INVITE request.
BYE	Indicates that either the originator or the target wishes to end the call; terminates a connection or declines an invitation.
REFER	Indicates that the recipient should contact a third party using provided contact information; initiates a transfer.
CANCEL	Cancels a pending request; does not affect a completed request.
REGISTER	Registers a user's address with a SIP location server; resolves a public address to a specific address. Not related to specific session.
OPTIONS	Solicits information about features supported by SIP servers such as supported methods and media capabilities.
NOTIFY	Provides information about a state change; not related to a specific session. Used for message waiting communications with a voicemail server, to indicate the outcome of transfers, and for configuration.
SUBSCRIBE	Indicates the desire for NOTIFY (state change) requests. Used for message waiting communications with a voicemail server and for configuration.

SIP Phones support these SIP methods:

Table 8. Supported Methods

Methods Initiated	Methods Received
INVITE	INVITE
ACK	ACK
BYE	BYE
REFER	REFER
CANCEL	CANCEL
REGISTER	
OPTIONS	OPTIONS
NOTIFY	NOTIFY
SUBSCRIBE	

Response Codes

SIP response codes indicate the status of a session. Response codes are generated and sent in outgoing messages, and accepted when received in incoming messages.

Table 9. Response codes

Code	Function
1xx	Informational: trying, ringing, forwarding, queuing, in progress
2xx	Successful: Ok
3xx	Redirection: indicate additional information for call forwarding
4xx	Request Failure: indicate request errors such as missing information
5xx	Server Failure: time outs, unavailable services, and other server errors
6xx	Global Failures: busy, declined, not found, not acceptable

Responses with a 1xx response code are also called provisional responses, while the remaining response codes (2xx, 3xx, 4xx, 5xx, and 6xx) indicate final responses.

Refer to Request for Comments 3261 for standard SIP status code definitions. Additional codes may also be listed in Internet Draft draft-ietf-sip-rfc2543bis-05.txt.

Message Headers

Each SIP message is accompanied by a header with these required fields:

Table 10. Message Headers

Field:	Contains:
From	The address of the session originator, expressed as a SIP URL
To	The address of the session target, expressed as a SIP URL
Call-ID	A unique identifier assigned to all of the SIP messages related to a call
Cseq	The SIP call control method and an identifying sequence number

Sample SIP Message

Here's an example of the SIP message that is sent when one phone dials another:

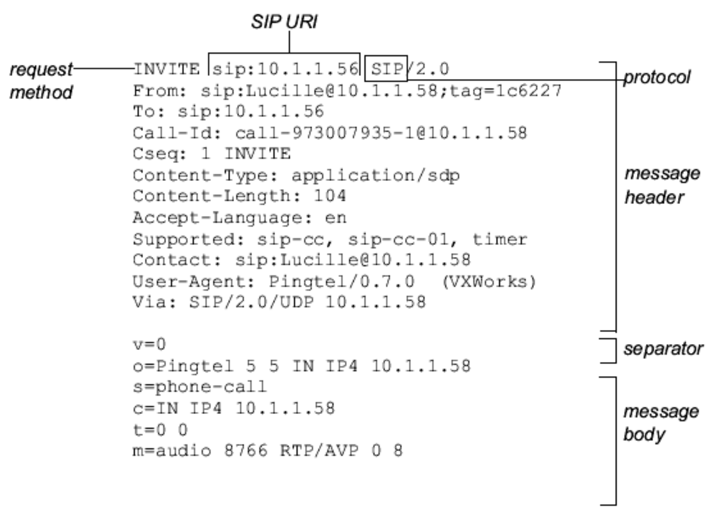


Figure 55. Sample SIP Message

Message Flow Examples

To illustrate the sequence and direction in which SIP call control methods and response codes are sent, message flow examples follow for:

- Call setup
- Call teardown
- Successful transfer (blind)
- Successful transfer (consultative)

Call setup

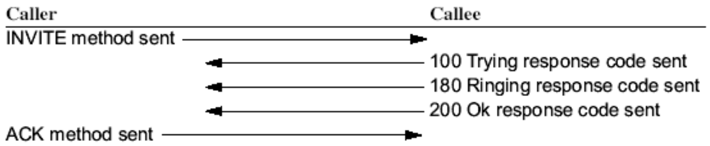


Figure 56. Call setup

Call teardown

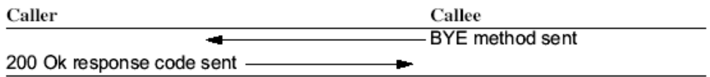


Figure 57. Call teardown

Successful blind transfer

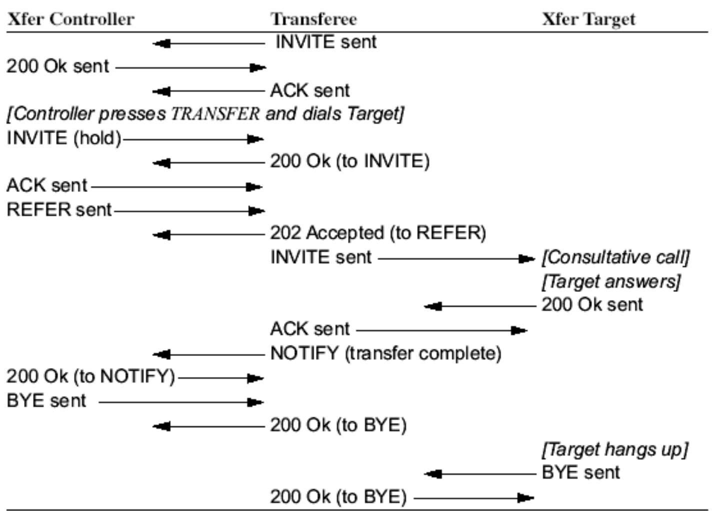


Figure 58. Successful blind transfer

Successful consultative transfer

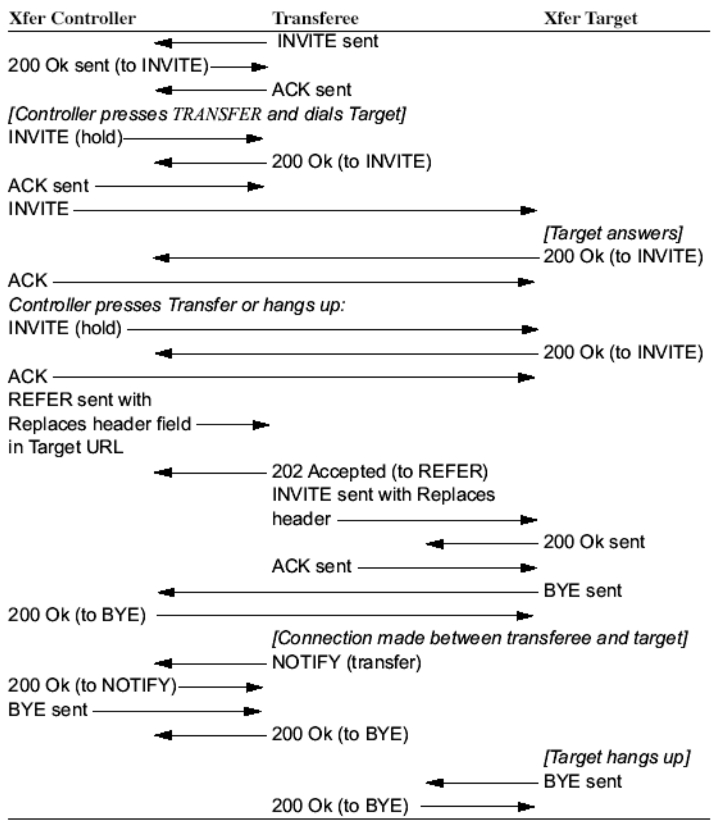


Figure 59. Successful consultative transfer

SIP URLs

Instead of only being assigned to specific devices the way that telephone numbers traditionally are assigned, you can assign SIP URLs to both a specific device and to the individual users who participate in SIP sessions. As a result, when a phone call (or other interactive session) is made to a SIP address, it can be routed either to a specific location, or to the appropriate individual, regardless of a change in physical location or IP telephone device.

The formats of some sample SIP URLs might appear as:

```
sip:123@example.com
sip:4444@10.1.1.123
<sip:freds@sip.example.net:5070>;Q=1.0
"Fred Smith"<sip:fsmith@sip.example.net:5070>
```

The From and To fields in every SIP message header contain the SIP URLs of the session's originator and target.

SIP Transactions

A SIP transaction consists of a set of related SIP messages: usually, a request such as an INVITE, zero or more provisional responses (1xx response code), and a final response (2xx or greater). For example, the set of messages sent by a callee phone during call setup to indicate trying, ringing, and Ok make up a SIP transaction. See page 90 for an example of this message flow.

The message header To, From, Call-ID, and Cseq fields have the same values in every message in a SIP transaction.

SIP Sessions

SIP sessions encompass all messages sent between two SIP endpoints. That is, all SIP messages sent between two phones, beginning with call setup and ending with call teardown, make up a SIP session.

The headers of all messages involved in a SIP session have the same values in the To, From, and Call-ID fields. However, the addresses in the To and From fields switch to reflect the endpoint that originated the message.

SIP Calls

A SIP call consists of one or more SIP sessions. An example of a call that encompasses multiple sessions is a conference call.

All SIP messages in a call will have the same values in all Call-ID message header fields. However, a user can set an alias as the Caller ID (see "Caller ID" below) for SIP INVITE requests.

Caller ID

A user can set an alias for outbound calls so that the alias will appear as the Caller ID on the recipient's phone. To set an alias for the caller ID, modify the From header in the SIP INVITE request.

An external number can also be specified on a gateway so that all outbound calls through the gateway will have the same caller ID. This only affects calls that do not have a user alias specified.

Caller ID can be defined per user, on a user group level, or per gateway. Caller ID can also be blocked on a per user or per gateway basis (CLIR).

System Components

Components in a SIP system send and respond to the messages that set up, establish, and terminate sessions.

Table 11. SIP System Components

Component	Description
UAC (User Agent Client)	An application that initiates a request and sends it in a SIP message.
UAS (User Agent Server)	An application that uses a SIP message to respond to a request; accepts, redirects, or refuses sessions.
Proxy server	An intermediary program that accepts a SIP message, optionally performs services, and then passes the message on. Acts as both a UAC and a UAS.
Redirect server	An intermediary program that accepts a SIP message and then returns a response to the sender. A Redirect server may or may not perform services.
Registry server	A server that accepts SIP messages from, and registers the current location of, a user agent. Maintains a database of addresses for user agents.
Location server	A server that provides information to Redirect and Proxy servers about the possible locations of a session target.

Note SIP components are abstractions. They do not have a one-to-one correspondence to specific VoIP devices.

Generally,

- User agents are the peers in the VoIP peer-to-peer communication model.
- Proxy servers monitor sessions and provide services.
- Redirect, Registry, and Location servers support user mobility by tracking the location of, and redirecting messages to, session targets.

Appendix B Firewalls and NAT

Chapter contents

Working with Firewalls and NAT

For phones to make calls to parties on the other side of a firewall, you configure both the firewall and the phone.

- If your firewall is packet-based, you configure both the firewall and the phone to identify the ports that allow incoming VoIP traffic (SIP, RTP, and RTCP packets) to pass through it.
- If your firewall uses NAT (Network Address Translation) and is packet-based, you configure both the firewall and the phone to identify the firewall's external or Internet IP address in addition to identifying the ports for incoming VoIP traffic. See page 98.
- A proxy-based firewall must use a SIP-specific proxy. See page 99 for tips to help you set up phones in your installation.

Configure the firewall

This section provides an overview of the tasks that you will complete for your packet-based firewall when you prepare to use phones. Refer to the documentation provided with your firewall software for instructions.

Recording the external IP address

While you are working with the server or router that provides your firewall services, determine and record its external or Internet IP address for reference during firewall/phone configuration. This address may be identified as the WAN IP address, or with another label.

Opening VoIP ports

On your firewall, you define the ports to open for incoming SIP, RTP, and RTCP traffic.

- The SIP (Session Initiation Protocol) port is used for call control: setting up and tearing down calls. For SIP packets, you define a single port. The well known port number for SIP is 5060.
- The RTP (Real-time Transport Protocol) port receives the audio for a call, and the RTCP (Real-time Control Protocol) port receives the control and media statistics stream. Two consecutively numbered ports are required per call to receive these packet streams. The default value for the first port is 8766.
- To allow a phone user to place calls on hold or make conference calls, four pairs (eight ports) are recommended. At a minimum, two ports are needed to support a single connection.

If your firewall has NAT, see page 98 for additional information.

Configure a Phone

Once you have identified the external IP address and opened ports for the incoming VoIP traffic on your packet-based firewall, you configure the phone to use those same values. Use the phone's browser-based interface to:

- Identify the external IP address
- Set the SIP port
- Set the RTP/RTCP ports

Identifying the external IP address

PHONESET_EXTERNAL_IP_ADDRESS

The "Host address outside NAT firewall" device setting applies to all phones that make calls through a firewall. You supply an IP address or host name that is outside of the firewall as the value for this parameter.
For example:

209.251.66.16

In a configuration file, this address appears as follows:

PHONESET_EXTERNAL_IP_ADDRESS: 209.251.66.16

The phone includes this IP address or host name in the SIP messages it sends to other SIP user agents to indicate that this is the address to which SIP, RTP, and RTCP packets should be sent.

Setting the SIP port

To define the SIP port, you set two different device settings to the same value. Both the port for inbound SIP TCP messages and the port for inbound SIP UDP messages must be set to the phone's assigned SIP port number.

Setting the RTP/RTCP ports

The **Starting port for RTP/RTCP packets** device setting identifies the first in a pair of consecutively numbered ports. RTP and RTCP use these ports to receive audio media and control information for each concurrent connection. The starting port defaults to 8766. You do not need to explicitly set the next port.

> **Note** You can open several pairs of consecutively numbered ports on your firewall to support more than one concurrent call. If so, you still define only the starting RTP/RTCP port on the phone with this parameter.

Work with a Firewall with NAT

If you are using multiple phones behind a firewall with NAT, and if that firewall has only a single external IP address, you must open a unique set of external SIP and RTP/RTCP port/address pairs for each phone that makes calls through the firewall.

When you set up these ports for your firewall, you associate the phone's IP address with each one. This process is sometimes called port mapping; that is, mapping a port on the external or public side of the firewall to a specific port and device IP address inside the firewall.

Note The RTP ports specified for a phone must be consecutive.

However, when you have multiple phones behind the firewall, you use different ports for each one. For example, open unique SIP port 5060 for phone A, port 5061 for phone B, and 5062 for phone C. Then open a range of eight unique, consecutive RTP/RTCP ports for each phone: 8000 to 8007 for phone A, 8008 to 8015 for phone B, 8016 to 8023 for phone C.

The configuration for your firewall may look something like this:

IP address forphone A: 192.168.0.3
phone B: 192.168.0.4
phone C: 192.168.0.5

You must also make sure that each phone's **Port for inbound SIP TCP messages, the Starting port for inbound SIP UDP messages**, and **the Starting port for RTP/RTCP packets** reflect these values:

Table 12. Parameter settings for certain phones

Parameter Setting:	Phone A 192.168.0.3	Phone B 192.168.0.4	Phone C 192.168.0.5
SIP_TCP_PORT	5060	5061	5062
SIP_UDP_PORT	5060	5061	5062
PHONESET_RTP_PORT_START	8000	8008	8016

Work with a Proxy Firewall

For phones to work in an environment with a proxy-based firewall, the firewall must have a SIP-specific proxy implemented. Refer to the documentation provided by your firewall vendor for instructions on how to configure its SIP features.

Configuring phones for a SIP firewall proxy

For phones to work with a SIP firewall proxy, you are likely to need to set the `SIP_PROXY_SERVER` parameter **Route all outbound SIP messages through proxy** to the internal IP address of your SIP firewall proxy.

Depending on the requirements of the SIP firewall proxy that you use, you may also need to configure your phones by setting one or more of the following:

- **HTTP proxy host name** and **HTTP proxy port number**
  PHONESET_HTTP_PROXY_HOST and PHONESET_HTTP_PROXY_PORT
- **Host address outside NAT firewall**
  PHONESET_EXTERNAL_IP_ADDRESS
- **Port for inbound SIP TCP messages** and **Port for inbound SIP UDP messages**
  SIP_TCP_PORT and SIP_UDP_PORT
- **Starting port for RTP/RTCP packets**
  PHONESET_RTP_PORT_START

Using HTTP proxy settings

Phones use the HTTP protocol to download software upgrades and application .jar files. If these HTTP transactions have destinations on a remote server and must go through a proxied firewall, you may need to set values for the **HTTP proxy host name** and **HTTP proxy port number** to allow the phones to originate the HTTP transactions. Examples for these parameters follow.

PHONESET_HTTP_PROXY_HOST: **HTTP proxy host name:** httpproxy.Patton.com

PHONESET_HTTP_PROXY_PORT: **HTTP proxy port number:** 8080

Appendix C **Configuration APIs**

Chapter contents

SIPxNano Configuration APIs

This guide introduces the features of Patton's SIPxNano Configuration APIs. This collection of APIs (Application Program Interfaces) contains tools for importing data from an existing data store, such as an LDAP directory, into a SIPxNano installation. These tools help programmers:

- Migrate data to a new SIPxNano installation by importing a large number of user and device records
- Periodically update SIPxNano with new and revised user and device data as an ongoing system activity

Install the APIs

Patton delivers the files for SIPxNano Configuration APIS in a single compressed zip file. Extracting the contents of this file creates a *** directory on your PC or workstation.

To install the APIs, you extract the contents of the zip file.

Core classes and operations

An introduction to the core Java classes and WSDL operations that make up the SIPxNano Configuration APIs follows.

Note You can invoke operations from Java using the supplied Java classes. Alternatively, you can access the API from other languages as long as the access conforms to the WSDL.

Add User

Creates a new user in SIPxNano from your external source data. To add a user, you use either:

- createUser operation in the UserService. See the WSDL user-service.xml file
- AddUserCommand class in the com.Patton.pds.applications. loader.commands package

Table 13. Core Operations: Add User

WSDL Parameter Java Method	Description	Required?
userId setId()	Valid strings contain only: ¥ a-z ¥ A-Z ¥ Numbers ¥ Under score(_) Must be less than 80 character length. Cannot be a duplicate of an ex user ID or extension or in the reserved extension pool.	Yes
PIN setPIN()	Valid strings contain only dig to 9.	No
firstName setFirstName()	Valid strings contain Unicode ters between 0000 and 007F and 00FF. Must be less than 20 character length.	No
lastName setLastName()	Valid strings contain Unicode ters between 0000 and 007F and 00FF. Must be less than 30 character length.	No
extension setExtension()	Valid strings contain only dig to 9, hyphen (-) characters, c Must be less than 30 character length. Cannot be a duplicate of an ex user ID or extension or in the reserved extension pool.	No
alias setAlias()	Must be less than 256 characte length.	No

Table 13. Core Operations: Add User

WSDL Parameter Java Method	Description	Required?
userGroupName setUserGroupName()	Identifies the fully-qualifiec for the user. You must specify plete group hierarchy, includi user group names of both the i ate parent group and all ances groups, relative to the root c tem. The ancestry is delimited with ters. A leading '/' should not For example, to create a new u the sales user group, which ir belongs to the (top-level) wes user group, supply west coast (without the quotation marks).	Yes

Edit User

Modifies data for an existing SIPxNano user to match your external source data. To edit a user, you use either:

- editUser operation in the UserService. See the WSDL user-service.xml file
- EditUserCommand class in the com.Patton.pds.applications. loader.commands package

Note If a value is "null", the value of that field will be set to null in SIPxNano.

Table 14. Core Operations: Edit User

WSDL Parameter Java Method	Description	Required?
existingUserId setExistingUserId()	Must reference an existing User ID. See "Add User" on page 102.	Yes
newUserId setNewUserId()	A new user ID for the existing user. See "Add User" on page 102.	No
PIN setPIN()	See "Add User" on page 102.	No
firstName setFirstName()	See "Add User" on page 102.	No
lastName setLastName()	See "Add User" on page 102.	No
extension setExtension()	See "Add User" on page 102.	No
alias setAlias()	See "Add User" on page 102.	No
userGroupName setUserGroupName()	See "Add User" on page 102.	Yes

Delete User

Removes an existing user from SIPxNano. To delete a user, you use either:

- deleteUser operation in the UserService. See the WSDL user-service.xml file
- DeleteUserCommand class in the com.Patton.pds.applications. loader.commands package

Table 15. Core Operations: Delete User

WSDL Parameter Java Method	Description	Required?
existingUserId setExistingUserId()	Must reference an existing User ID. See "Add User" on page 102.	Yes

Add Device

Creates a new device in SIPxNano from your external source data. To add a device, you use either:

- createDevice operation in the DeviceService. See the WSDL device-service.xml file
- AddDeviceCommand class in the com.Patton.pds.applications.loader.commands package

Table 16. Core Operations: Add Device

WSDL Parameter Java Method	Description	Required?
serialNumber setSerialNumber()	Valid strings contain only: a-f A-F 0-9 Must be 12 characters in length. Cannot be a duplicate of an existing serial number. For Patton phones, find the serial number on the bottom of an xpressa phone or in the key-config (license key) file of an instant xpressa softphone.	Yes

Table 16. Core Operations: Add Device

WSDL Parameter Java Method	Description	Required?
deviceType setDeviceType()	Valid strings are: xpressa_strongarm_vxworks (for Patton xpressa phones) ixpressa_x86_win32 (for Patton instant xpressa softphones) 7960 (for Cisco 7960G IP phones) 7940 (for Cisco 7940G IP phones)	Yes
name setName()	Must be less than 60 characters in length.	Yes
deviceGroupName setDeviceGroupName	Sets the fully-qualified device group of the device. You must specify the complete group hierarchy, including the device group names of both the immediate parent group and all ancestor groups, relative to the root of the system. Ancestry is delimited with '/' characters. A leading '/' should not be used. For example, to create a new device within the sales device group, which in turn belongs to the (top-level) west coast device group, supply “west coast/sales” (without the quotation marks).	Yes
description setDescription	Must be less than 60 characters in length.	No

Edit Device

Modifies data for an existing SIPxNano device to match your external source data. To edit a device, you use either:

- editDevice operation in the DeviceService. See the WSDL device-service.xml file
- EditDeviceCommand class in the com.Patton.pds.applications. loader.commands package

If a value is "null", the value of that field will be set to null in SIPxNano.

Note You cannot change the device type for a SIPxNano device. Instead, you must delete the current device, add a new device with the different device type, and then assign the device to a user.

Table 17. Core Operations: Edit Device

WSDL Parameter Java Method	Description	Required?
existingSerialNumber setExistingSerialNumber()	See "Add Device" on page 104.	Yes
newSerialNumber setNewSerialNumber()	See "Add Device" on page 104.	Yes
name setName()	See "Add Device" on page 104.	No
deviceGroupName setDeviceGroupName()	See "Add Device" on page 104.	No
description setDescription()	See "Add Device" on page 104.	No

Delete Device

Removes an existing device from SIPxNano. To delete a device, you use either:

- deleteDevice operation in the DeviceService. See the WSDL device-service.xml file
- DeleteDeviceCommand class in the com.Patton.pds.applications. loader.commands package

Table 18. Core Operations: Delete Device

WSDL Parameter Java Method	Description	Required?
serialNumber setSerialNumber()	Must be an existing device's serial number. See "Add Device" on page 104.	Yes

Assign Device to a User

Creates an association between a user and a device in SIPxNano. An association between a user and a device allows SIPxNano to route incoming and outgoing calls properly. To assign a device to a user, you use either:

- assignDevice operation in the UserService. See the WSDL user-service.xml file
- AssignDeviceCommand class in the com.Patton.pds.applications. loader.commands package

Table 19. Core Operations: Assign Device to a User

WSDL Parameter Java Method	Description	Required?
userId setUserId()	Must be the user ID of an existing user. See "Add User" on page 102.	Yes
deviceSerialNumber setDeviceSerialNumber()	Must be an existing device's serial number. See "Add Device" on page 104.	Yes

Unassign Device from a User

Severs the association between a device and its associated user. To unassign a device from a user, you use either:

- unassignDevice operation in the UserService. See the WSDL user-service.xml file
- UnassignDeviceCommand class in the com.Patton.pds. applications.loader.commands package

No user ID is needed as a device can only be assigned to one user.

Table 20. Core Operations: Unassign Device from a User

WSDL Parameter Java Method	Description	Required?
deviceSerialNumber setDeviceSerialNumber()	Must be an existing device's serial number.	Yes

Note There is no need to unassign before you delete a user or a device.

Resync data sets

Rebuilds all of the export data sets in SIPxNano, including:

- alias.xml
- authexception.xml
- credential.xml
- extension.xml
- permission.xml

A cumulative resynchronization should be performed after all other SIPxNano operations are complete. To resynchronize all data sets, you use either:

- rebuildDataSets operation in the DataSetService. See the WSDL dataset-service.xml file
- ResyncAllDatasetsCommand class in the com.Patton.pds.applications.loader.commands package

Send all profiles

Sends all user, device, and application profiles to all phones managed by SIPxNano. You should send all profiles after you add users or devices to SIPxNano, or after assigning devices to users. To send all profiles, you use either:

- sendAllProfiles operation in the SendProfilesService. See the WSDL send-profiles-service.xml file
- SendAllProfilesCommand class in the com.Patton.pds. applications.loader.commands package

Send all profiles to a specified user

Sends user, device, and application profiles to all phones assigned as devices to a specified SIPxNano user. To send profiles to a user's phones, you use either:

- sendProfilesForUser operation in the SendProfilesService. See the WSDL send-profiles-service.xml file
- SendUsersProfilesCommand class in the com.Patton.pds. applications.loader.commands package

Table 21. Core Operations: Send all profiles to a specified user

WSDL Parameter Java Method	Description	Required?
userId setId()	Must be an existing SIPxNano user ID.	Yes

Preparing source data

The SIPxNano Configuration APIs import character-delimited ASCII rows from a flat file into SIPxNano. The test value will be an op-code for the operation that the Loader is to perform on this record.

Sample code

The SimpleClient class in the com.Patton.pds.applications package demonstrates how to use LoaderCommand objects to send requests to SIPxNano. A sample Java application follows.

```
///////////////////////////////////////////////////////////////////////////
// Public Methods
////
    public static void *main(String[] args) {
        if(args.length != 3){
            System.err.println("Usage: <hostname> <use TLS (true|false)> <superadmin-password>");
        }

        try {
            // create the specific LoaderCommand object. In this case we
            // are going to add a new user to the system.
            AddUserCommand addUserCmd =
                    new AddUserCommand( args[0], // hostname
                                        Boolean.valueOf(args[1]).booleanValue(), // user TLS
                                        args[2], // superadmin password
                                        null); // no 'raw' record.

            // set the various AddUserCommand properties
            addUserCmd.setId("jsmith");
            addUserCmd.setPIN("1234");
            addUserCmd.setFirstName("John");
            addUserCmd.setLastName("Smith");
            addUserCmd.setUserGroupName("default");

            // execute the command.
            addUserCmd.execute();
        }
        catch(LoaderException e) {
            e.printStackTrace();
        }

    }


///////////////////////////////////////////////////////////////////////////
// Implementation Methods
////


///////////////////////////////////////////////////////////////////////////
// Nested / Inner classes
////


///////////////////////////////////////////////////////////////////////////
// Native Method Declarations
////

}
```

Appendix D **Compliance information**

Chapter contents

Compliance

EMC

- FCC Part 15, Class A
- EN55022, Class A
- EN55024

Safety

- IEC/EN 60950-1

Radio and TV interference

This equipment generates and uses radio frequency energy, and if not installed and used properly—that is, in strict accordance with the manufacturer's instructions—may cause interference to radio and television reception. This equipment has been tested and found to comply with the limits for a Class A computing device in accordance with the specifications in Subpart B of Part 15 of FCC rules, which are designed to provide reasonable protection from such interference in a commercial installation. However, there is no guarantee that interference will not occur in a particular installation. If the equipment causes interference to radio or television reception, which can be determined by disconnecting the cables, try to correct the interference by one or more of the following measures: moving the computing equipment away from the receiver, re-orienting the receiving antenna, and/or plugging the receiving equipment into a different AC outlet (such that the computing equipment and receiver are on different branches).

CE Declaration of Conformity

We certify that the apparatus described above conforms to the requirements of Council Directive 89/336/EEC, as amended by Directives 92/31/EEC and 93/68/EEC on the approximation of the laws of the member states relating to electromagnetic compatibility; and Council Directive 2006/95/EC on the approximation of the laws of the member states relating to electrical equipment designed for use within certain voltage limits.

The safety advice in the documentation accompanying this product shall be obeyed. The conformity to the above directive is indicated by the CE sign on the device.

Authorized European Representative

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