VoiceGear

VoiceGear Connect Gateway User Guide – Ver.3.0

1. Overview

Thank you for purchasing VoiceGear Connect, the most advanced integrated Skype-PBX gateway in the world. You can now integrate the power of Skype into your existing office communication infrastructure and unlock the inherent communications freedom and significant long distance call savings as compared to traditional telephone providers.

1.1 Package Contents

Contents of your package depend on the configuration of the gateway you have ordered. In general, all gateways come with a universal power supply which can support both American and European voltage standards (110-220volts) as well as a standard Ethernet cable with RJ-45 connector. In addition, gateways with analog connectivity will include analog RJ-11 type phone wires, while gateways with T1/E1 connectivity will include CAT-5 digital cable(s). Please note that power supply cable comes with North American 3-pin power plug type and you may need to use a suitable plug converter for regional power outlet standard.

1.2 About This Guide

This guide presents a detailed explanation of the gateway's functionality. It is targeted mainly at system administrators for configuration and management of the gateway. However, regular users will find it beneficial to read sections of this guide related to using the Web interface and managing address book contacts. Section 2 – "Getting Started" outlines main configuration steps to get you up and running as soon as possible.

2. Getting Started

Getting started section provides you with a quick overview of steps necessary to get your new VoiceGear Connect gateway running as quickly as possible. Please ensure that your package came with all the contents as described in section 1.1 above, and then proceed with the steps outlined below:

1. Connect the supplied power and network cables to your new VoiceGear Connect gateway.
2. If your VoiceGear Connect gateway features Digital E1/T1 or Analog FXS/FXO connection to the PBX system, please connect the wires as described in section 7.4 – “Analog/Digital PBX Configuration Overview”.
3. Configure networking settings and change system password by reviewing the instructions in section 3 – “Administration Console”.
4. Get familiar with the Web configuration interface by reviewing section 4 – “Web Configuration Interface” in order to understand how to proceed with configuring system settings using the Web interface.
5. In case you are planning to use SIP connectivity to the PBX, you need to configure VoiceGear Connect SIP channel settings as described in section 7.1 – “SIP Channel Configuration”.
6. Before you can start using the VoiceGear connect gateway to make and receive calls, you must register your existing Skype accounts with the gateway. Please follow instructions supplied in section 5 – “Adding Skype Accounts” to accomplish this task.
7. If you are planning to call other Skype accounts using VoiceGear Connect, you will need to add their contact information to the Address Book accessible from the Web interface. Please refer to section 11.1 – “Address Book” to learn more about this functionality.
8. Follow instructions in section 13 – “Purchasing Skype Credits” to purchase Skype credits for making outgoing calls to regular phone numbers.

3. Administration Console

All VoiceGear Connect gateways feature an administration console which allows for basic system configuration and settings. Please follow steps below to configure your system passwords and network settings:

1. Connect a keyboard and a monitor to the gateway and press the power button to start the gateway.

2. When the gateway is fully booted it will prompt you to login into the terminal as show below.

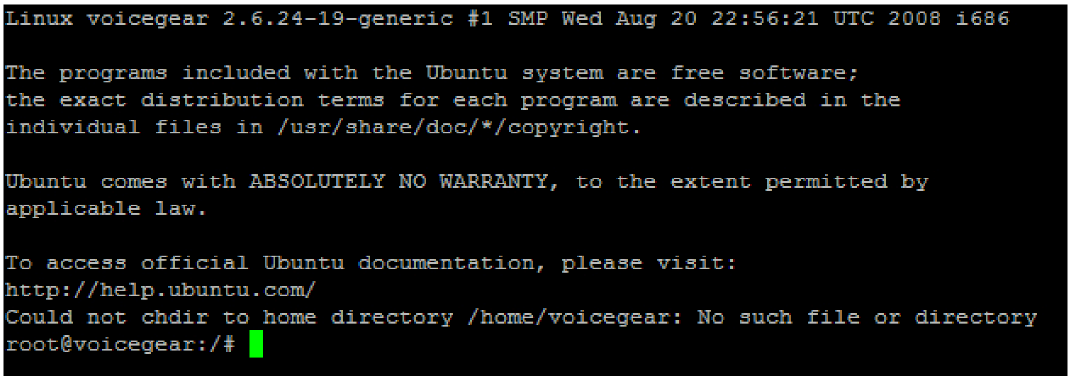


Image 1: VoiceGear Connect terminal

Please Note:
The default terminal username and password set on your new VoiceGear Connect gateway is:
username="root", password="vgcroot123".

3. After logging into the terminal using the default credentials supplied above you should see the VoiceGear Connect terminal console screen.

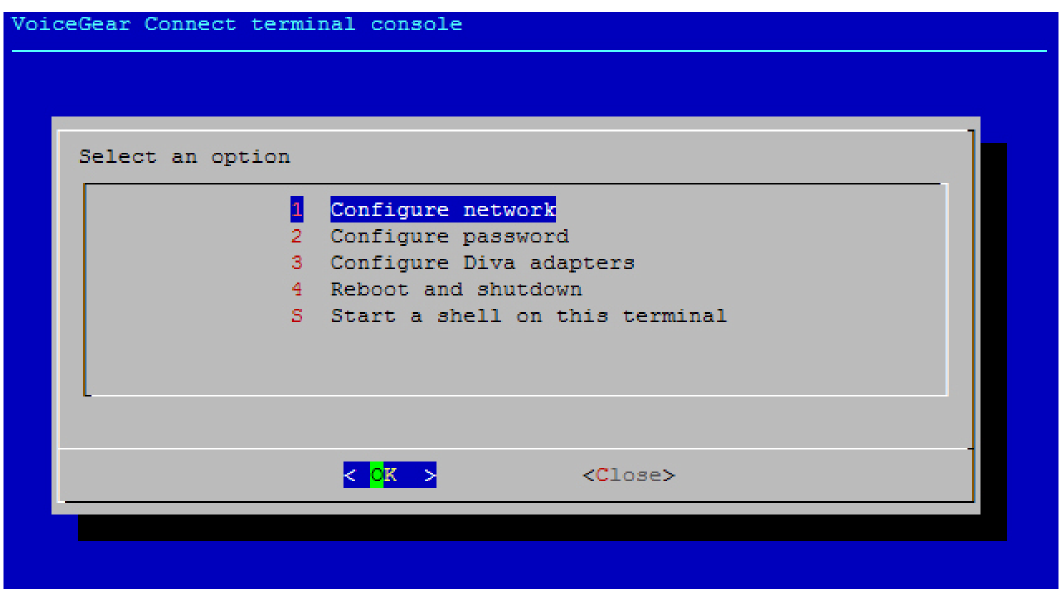


Image 2: VoiceGear Connect terminal console screen

Before you can start using the VoiceGear Connect gateway it must be configured correctly to access your network. All VoiceGear Connect gateways come pre-configured to use DHCP in order to obtain network IP. It is recommended to configure the gateway to use a static IP whenever possible. In case your DHCP server is configured to lease IP addresses infinitely, you can skip the static IP assignment and simply use the automatically assigned IP since it will not change when the gateway is restarted.

4. To assign a static IP to your gateway, please follow the steps below:

    a. Select the "Change IP" option to bring up network settings screen.

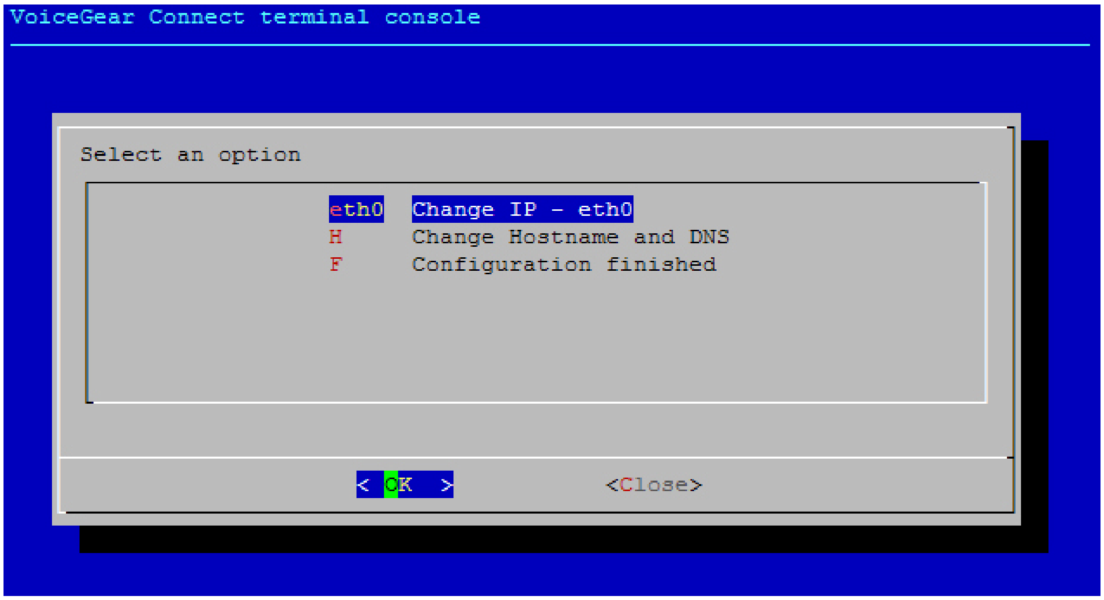


Image 3: VoiceGear Connect network configuration console

    b. Select "No" when prompted to "Configure with DHCP?" in the next screen.

    c. Enter your static IP settings, net mask and gateway addresses when prompted in follow up screens.

    d. Check your settings and select "Yes" when asked to "Please confirm your information" in the final screen.

    e. Once you have updated your static IP settings, the VoiceGear Connect gateway will automatically restart all networking services.

    f. When networking restarts you will be taken back to the main network settings screen.

    g. Select "Change Hostname and DNS" option to setup hostname, domain and DNS settings.

    h. Update all settings as necessary and select "Close" to dismiss this setup screen.

    i. Select "Configuration finished" option from the main network settings screen - this will take you back to the main VoiceGear Connect terminal console screen. Select "Reboot and shutdown" then "Reboot" option to restart the gateway in order for the new network settings to take effect.

j. Once the gateway is up and running again, you should be able to input the **http://x.x.x.x:8080** URL in your web browser (where "x.x.x.x" is the static IP you have assigned and 8080 is the port number) to access the VoiceGear Connect Web configuration interface.

Please Note:
The default Web configuration interface username and password set on your new VoiceGear Connect gateway is: username="admin", password="admin".

5. To change terminal root password or Web interface admin password select "Configure password" option from the main VoiceGear Connect terminal console screen. You will see "Change password" screen as shown below:

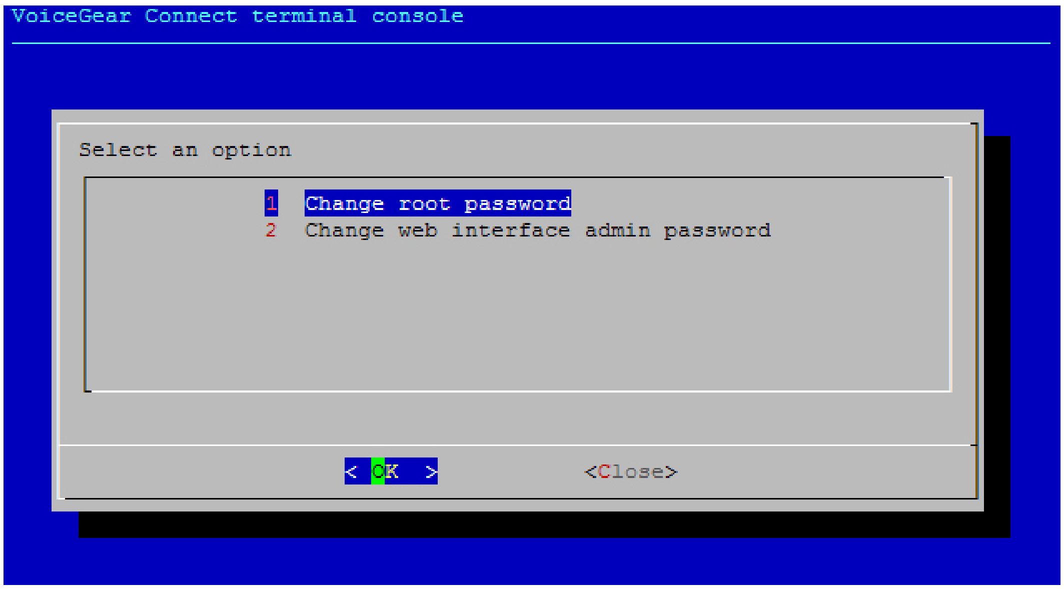


Image 4: VoiceGear Connect change password screen

6. Select "Change root password" option, type in the new password twice and select "OK" to update the root password.

7. Select "Change web interface admin password" option, type in the new password twice and select "OK" to update the Web configuration interface admin password.

3.1 Remote Console Connectivity

Most of the settings on your VoiceGear Connect gateway can be accessed from the web configuration interface; however, functions such as root password change can only be accessed via the console interface. To allow complete remote management, the console can also be accessed remotely via any SSH client such as Putty (http://www.chiark.greenend.org.uk/~sgtatham/putty/) by simply using the IP of the gateway and port 22. Login credentials for the remote console connection are the same as those used for local access. A sample remote console connection is shown below.

Image 5: Remote console connection

4. Web Configuration Interface

All VoiceGear Connect gateways come with a built-in Web based configuration interface which allows controlling virtually every aspect of the system from a Web browser such as Mozilla Firefox or Microsoft Internet Explorer. For a matrix of supported browsers, please refer to the table below.

Browser	Level of Support
Internet Explorer 6	Not Supported
Internet Explorer 7	Supported
Mozilla Firefox 1.1	Not Supported
Mozilla Firefox 2.0	Supported
Mozilla Firefox 3.0	Supported
Apple Safari	Not Supported

To access the Web configuration interface, please open your browser of choice and point it to the following address: http://x.x.x.x:8080 (where "x.x.x.x" is the current IP of the gateway and 8080 is the port number). The resulting screen you will see is shown in the figure below.

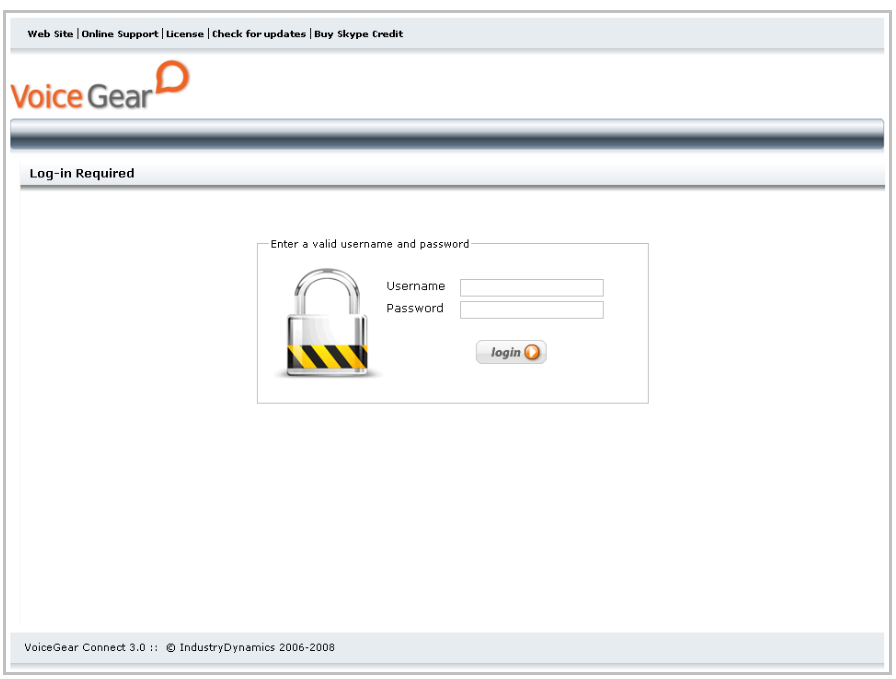


Image 6: Web configuration interface login screen

Default username and password for the Web configuration is set to “admin” and “admin” respectively. It is strongly recommended to change the “admin” user password after the first logon. Please refer to section 10 – “Configuring Users and Groups” for more details on managing users and changing Web configuration interface passwords.

4.1 Web Configuration Interface Layout

The Web configuration interface has been designed to be user friendly, interactive and provide quick response to user requests. Interface layout is designed with visual consistency where all features and information are organized in specific areas located on the same place throughout different pages. This keeps all menus, toolbars, icons and item positioning consistent and makes the system easy to navigate.

The main page interface is divided into 6 major areas. Every area illustrates different functionality and information.

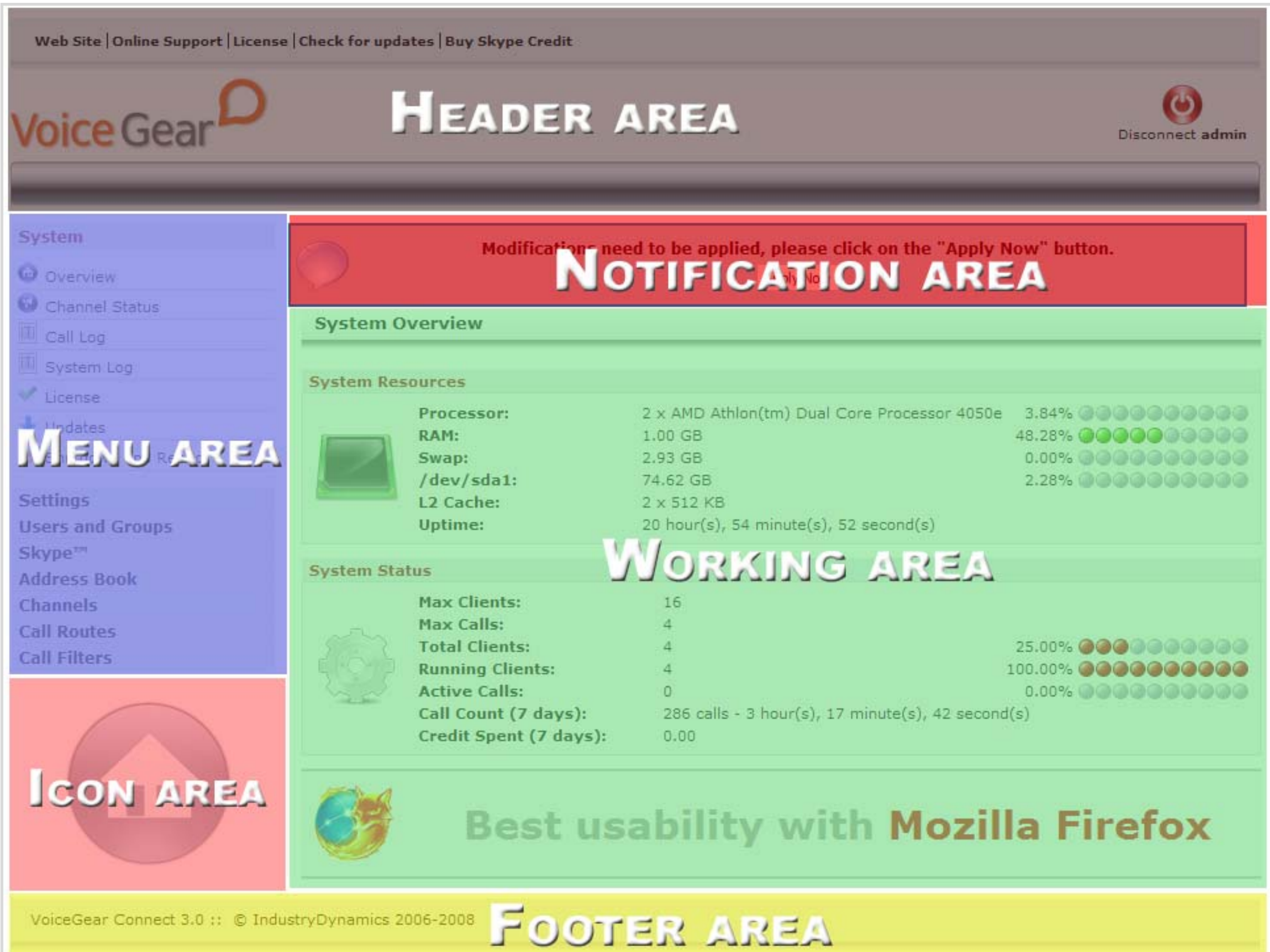


Image 7: Main web configuration interface page layout

Most used areas are “Menu area” and “Working area” as they allow you to browse every section of the system and configure its various aspects.

4.1.1 Header Area

Header contains important information such as product support and licensing links. In addition, the username of the currently logged in user is shown along with an option to disconnect. After disconnecting, you will be taken to the login screen shown in image 6. Information available in the header is detailed in the table below.

Link	Description
Web Site	Connects the user to the IndustryDynamics Web site.
Online Support	Connects the user to the IndustryDynamics support page which allows for creation of support tickets.
License	Allows user to view the licensing terms and EULA.
Check for Updates	Allows user to check for system software updates via the Web.
Buy Skype Credit	Connects system administrator to the Skype Web site where he/she can login to a Skype Business Control Panel in order to centrally manage Skype credit expenditures for employees, purchase credits and coordinate various other features.

4.1.2 Notification Area

Notification area is the space reserved for important real-time system notifications. It is not visible by default but rather slides into view when the system needs to communicate important information to the user. Notifications can contain action buttons to apply system setting changes or restart the system in order for new configuration to take effect. For a sample system notification, please refer to the image below.

Image 8: Sample system notification shown in the notification area

4.1.3 Menu Area

Menu area has been designed with ease of use in mind. It contains the following system categories: System, Settings, Users and Groups, Skype, Address Book, Channels, Call Routes and Call Filters. These categories allow the user navigate through the entire system by selecting items under each category. Clicking on any category will open a list of items underneath it. Clicking on an item, such as “Locale Settings”, within a category will open all available configuration and control options available for that item in the working area.

For example: You can edit all system settings it the “Settings” category. You can configure the “Locale Settings” by clicking on the “Locale Settings” item under the “Settings” category.

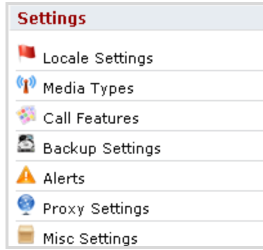


Image 9: Menu area with settings category open

4.1.4 Footer Area

The footer contains copyright information as well as licensing information. Please use the serial number shown in the section for obtaining product support.

4.1.5 Working Area

This area displays all information, settings and buttons to control features of the entire system. Information shown in this area can be changed by clicking on items in the menu area as described in section 4.1.3.

Working area itself is composed of 3 main areas ordered vertically from top to bottom as follows:

- Tips box
- Information boxes
- Toolbar

In addition, several pages shown in the working area can contain item lists.

Tips box

Tips box is a yellow area at the top of each page containing contextual tips with basic information about current section and important notices and warnings about available features. For a sample tips box please refer to the image below.

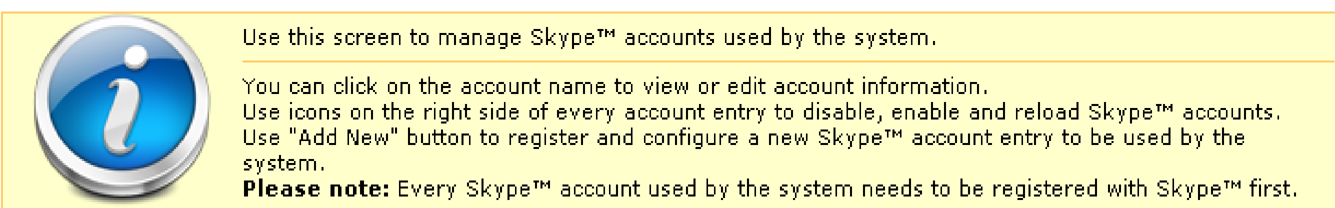


Image 10: A sample tips box

Information boxes

Information boxes are light-blue boxes that display main information and settings that can be edited. Each box can be expanded by clicking on its heading which is highlighted by the blue arrow pointing down. In the image below, “Basic Settings” box is expanded while “File Transfer” and “Account Settings” boxes are not.

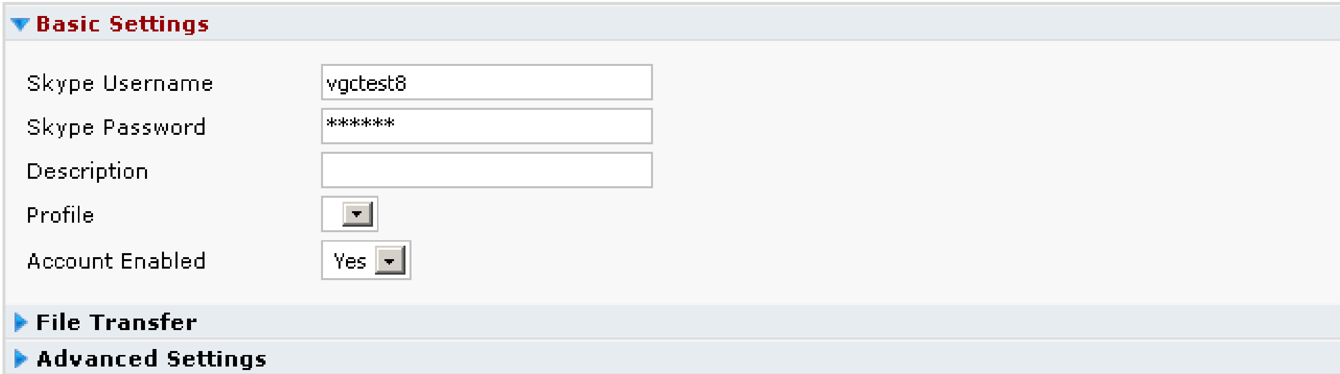
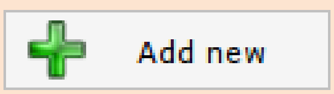
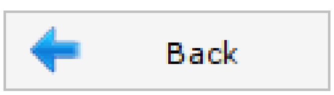
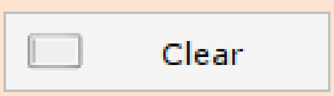
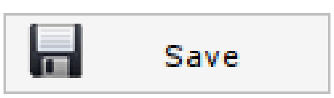


Image 11: Sample information boxes

Toolbar

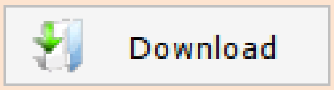
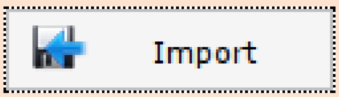
The last area at the bottom of each page contains the main toolbar. The toolbar displays all available operations for the current working area as buttons. In most cases, these operations consist of saving, clearing or deleting settings as well as a standard back operation designed to return you to the previous screen.

Standard toolbar buttons are outlined in the table below. Depending on the current working area view these buttons can vary from page to page.

Button	Action
	**Adds a new item to list** Item type depends on current section. Examples: In Skype>Accounts section adds a new Skype account. In Address Book>Contacts adds a new contact.
	**Goes back to previous page without saving changes**
	**Resets values undoing changes** Restores the last saved values without going back.
	**Saves all changes to the current item** Changes to dropdown menus, text fields and all other settings of current page are saved.
	**Delete current item** Current item is deleted, together with every setting displayed on current page. ***Please note: confirmation is needed before proceeding with deletion*** Examples: In Skype>Accounts section, delete will remove the current Skype account. In Address Book>Contacts delete will remove all information about the current contact.

> **Please Note:**
> It is not possible to use your browser's back and forward buttons to navigate the web configuration interface. If necessary, please use the "Back" button described above to return to a previous screen.

Special toolbar buttons are outlined in the table below. These buttons will appear in the toolbar when certain pages are shown in the working area. They are uncommon and will only be displayed on a handful of pages.

Button	Action
	**Downloads information** Some search results or logs can be downloaded. Example: In System>System Log section, this button downloads all system log entries.
	**Exports lists or tabular data** This button allows the user to transfer lists and other data into a Microsoft Excel document. Default export format is CSV.
	**Imports lists or tabular data** This button allows the user to transfer lists and other data from a Microsoft Excel document into the system. Default import format is CSV.

Item List

Certain information shown in the working area is presented in a list format which is used to provide the best possible access to multiple instances of similar data. One example of this would by a list of Skype accounts associated with the system.

The general layout of every list item from left to right is as follows:

- Edit button to open item and manage details particular to the item
- One or more columns showing properties for each item
- One or more action buttons to control actions for each item

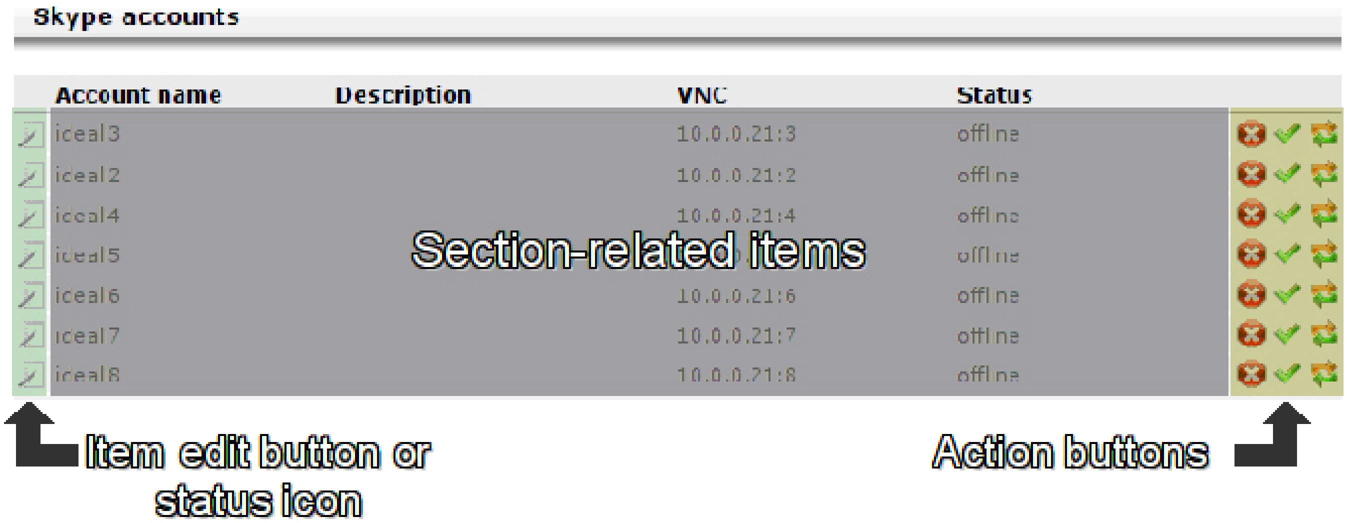


Image 12: Item list layout

Item lists on several pages shown in the working area are drag and drop enabled. Using this feature you can quickly sort items to the desired order. Lists which have drag and drop enabled can be distinguished by hovering over their items. Drag and drop enabled lists will show a “drag” mouse cursor while hovering over their items while standard lists will not. A sample drag and drop enabled item list is shown in the image below.

Action buttons contained within every item list allow you to perform immediate actions for a specific item in a list. They are shown on the right side of an item with an exception for the item edit action button which is shown on the left side of an item. For more information about action buttons, please refer to the table below.

Button	Action
	**Edit button** Opens a details page for the selected item. This page can be used to view or edit item properties.
	**Delete button** Removes an item from a list. ***Please note: confirmation is needed before proceeding with deletion***
	**Toggle On/Off button** The button enables or disables a particular item. The grey icon refers to a currently disabled item; the green one refers to a currently enabled item.
	**Reset/Reload button** Depending on the type of item, this button can both reload configuration and reset all settings related to a selected item to default values.

5. Adding Skype Accounts

Skype accounts are used by VoiceGear Connect gateway as lines to carry voice and DTMF between your PBX and Skype network. Whether you intend to call other Skype accounts or regular phone numbers, you need to register Skype accounts with VoiceGear Connect first.

Please Note:
Only existing Skype accounts can be registered with VoiceGear Connect. Therefore, before adding Skype accounts to VoiceGear Connect gateway, you need to create those accounts using Skype client first. If you already have existing Skype accounts which you want to use with VoiceGear Connect, please skip to section 5.2 below.

Please Note:
Skype accounts are not application users as defined in "Users and Groups" -> "Users" section. Skype accounts are actual Skype accounts registered on Skype network via the Skype client. To create Skype accounts for each line you are planning to use on your gateway, please follow instructions in section 5.1 below.

Each Skype account works as a telephone line available to all VoiceGear Connect users, to a group of users or to a single user depending on configuration. As such, a single Skype account can be used as a shared office line or a direct personal line.

For example, if you want to have 3 Skype lines for 3 of your company's offices (no matter how many people work in each office), you will have to create 3 Skype accounts (eg: company.1, company.2 and company.3) and register them in VoiceGear Connect as described in section 5.2 below.

If you would like to have personal Skype lines for some or all employees, you can register their personal Skype accounts with VoiceGear Connect and configure call routes to have all inbound and outbound calls placed via their personal Skype account (see section 8.1.2).

5.1 Creating new Skype accounts

In case you are not familiar with Skype application, please follow these steps to create new Skype accounts:

1. Download, install and launch the Skype application on your PC. For convenience, you can follow steps in this online tutorial: http://www.skype.com/help/guides/installskype_windows/

2. Create a new Skype account. Follow steps in this online tutorial: http://www.skype.com/help/guides/createskypename_windows/

It is recommended to use the following notation for Skype names for each account: [YourCompanyName].[LineNumber] (For example: industrydynamics.1, industrydynamics.2, etc…)

5.2 Register Skype accounts

The next step is to add your newly created (or existing) Skype accounts into the gateway configuration. To do that, please open the VoiceGear Connect Web configuration interface, login with the "admin" credentials, and navigate to "Skype->Accounts" via the main menu on the left. You will see the page shown below.

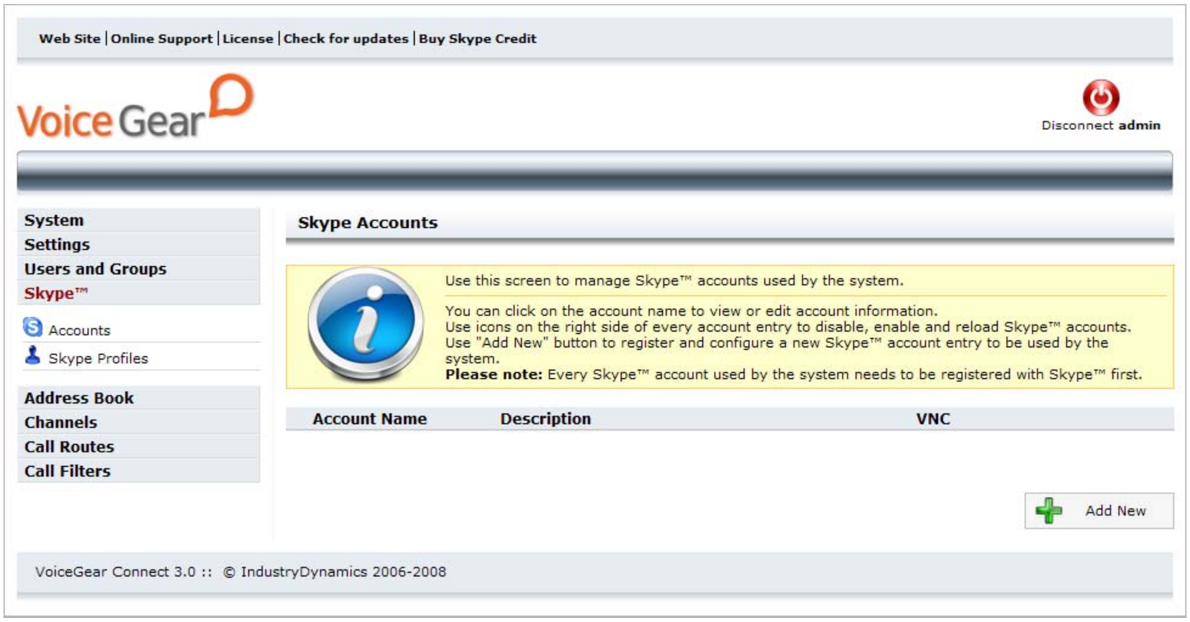


Image 13: Skype accounts screen

Once there, click on the "Add New" button in the bottom right corner of the screen to bring up the "Skype Account Editor: New Skype Account" page shown below.

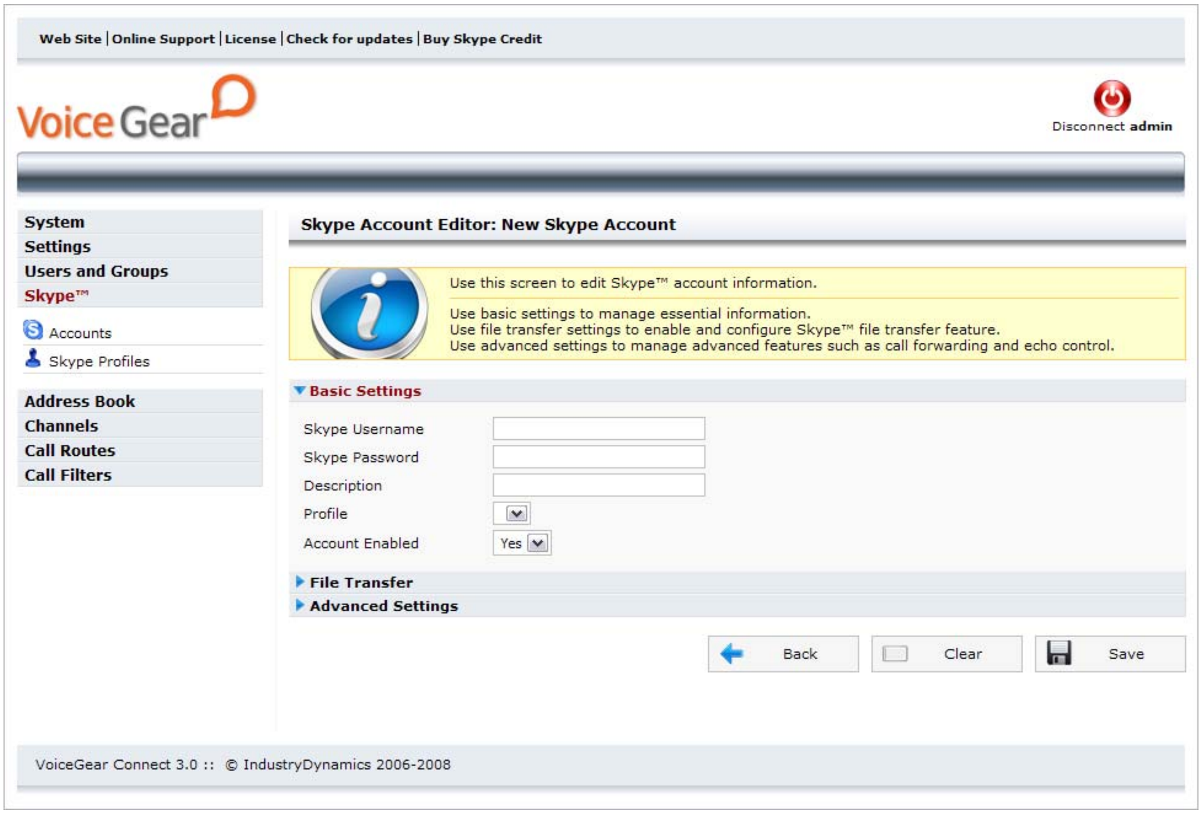


Image 14: Skype account editor screen

After filling in your Skype account username and password and setting the "Account Enabled" option to "Yes", please click the "Save" button. You will be prompted to apply your settings as shown below.

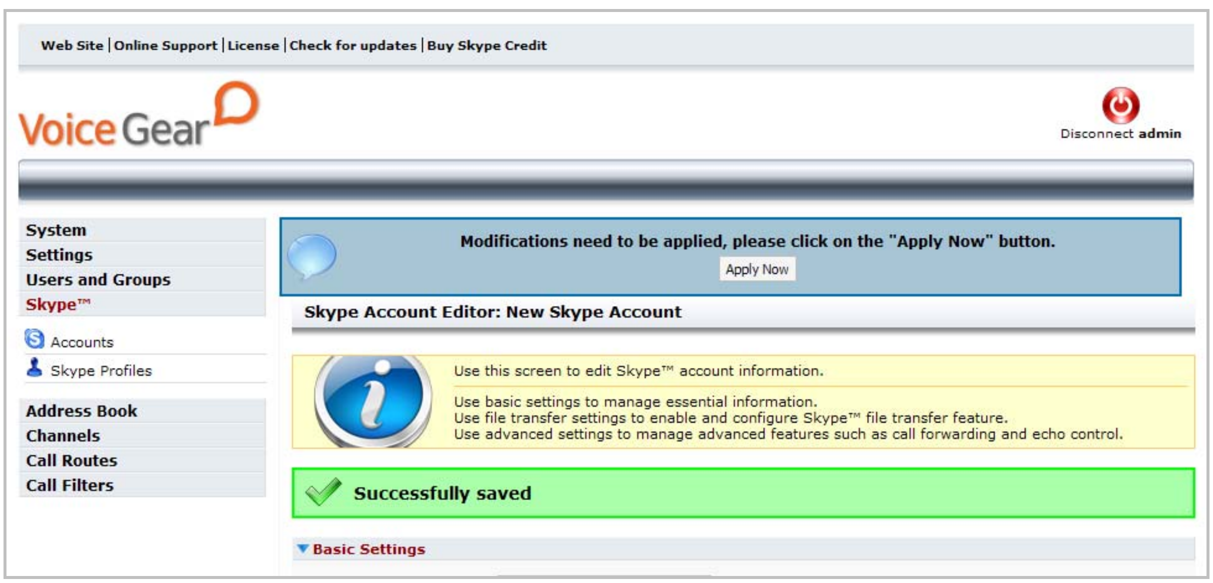
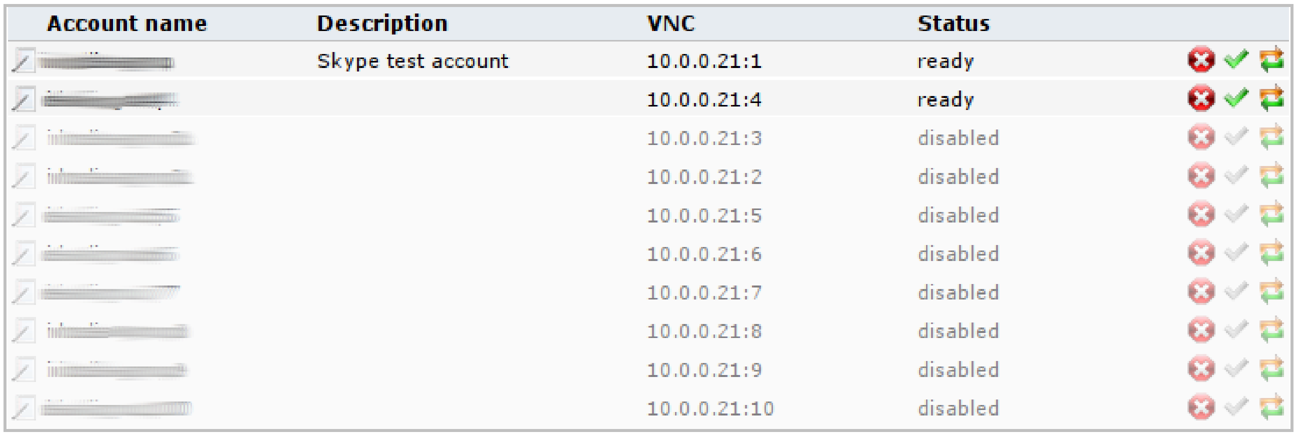


Image 15: Apply Skype account settings

Please click on the “Apply Now” button to confirm your changes and repeat the above process of adding a Skype account for each line of the gateway. All properly activated Skype accounts will show a “ready” Status and will have a green checkmark icon on the right of the account entry.



Account name	Description	VNC	Status
[illegible]	Skype test account	10.0.0.21:1	ready
[illegible]		10.0.0.21:4	ready
[illegible]		10.0.0.21:3	disabled
[illegible]		10.0.0.21:2	disabled
[illegible]		10.0.0.21:5	disabled
[illegible]		10.0.0.21:6	disabled
[illegible]		10.0.0.21:7	disabled
[illegible]		10.0.0.21:8	disabled
[illegible]		10.0.0.21:9	disabled
[illegible]		10.0.0.21:10	disabled

Image 16: Skype accounts table

You can drag and drop Skype account entries in the table above to change their order. The order in which accounts show up in this table does not affect system functionality in any way. You can control account status, by using icons on the right side of every account entry to disable, enable and reload Skype accounts.

5.2.1 Configuring Basic Settings

Basic settings need to be properly configured in order to register the Skype account. Please make sure that Skype username and password are entered correctly. VoiceGear Connect will not be able to use a Skype account with invalid credentials.

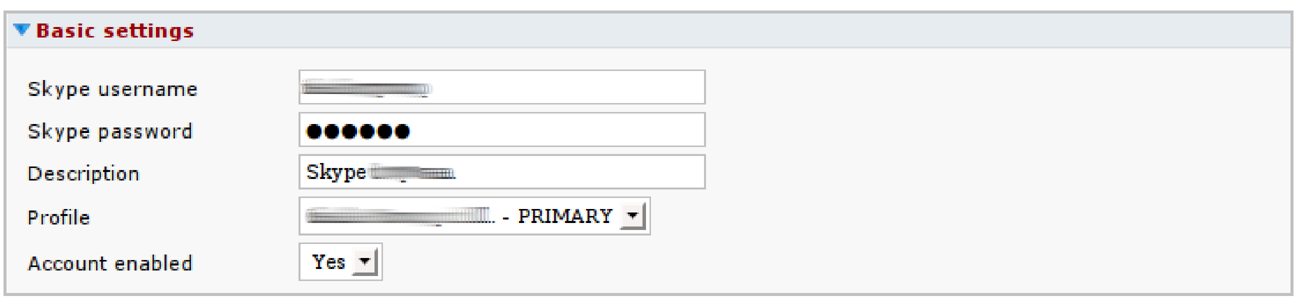


Image 17: Skype account basic settings

Setting	Description
Skype Username	The username of your existing Skype account that you would like to use with VoiceGear Connect. If you do not have an existing Skype account, please refer to section 5.1 above to create a new Skype account first.
Skype Password	The password of your existing Skype account. VoiceGear Connect cannot retrieve a lost or forgotten password and does not allow you to change Skype account password. If you

	forgot your Skype account password, you will have to use a Skype client application installed on your PC to retrieve it. You can also change your Skype account password using the Skype client application.
Description	Generic account description.
Profile	Skype profile assigned to this Skype account. For further details about how to configure and use Skype profiles please see section 5.3 below.
Account Enabled	This setting allows you to enable or disable Skype accounts. By default, this setting is set to “Yes”.

5.2.2 Configuring File Transfer Settings

Every Skype account registered with VoiceGear Connect can be used to receive files from other Skype users. You can take advantage of the settings below to enable or disable file transfer capabilities.

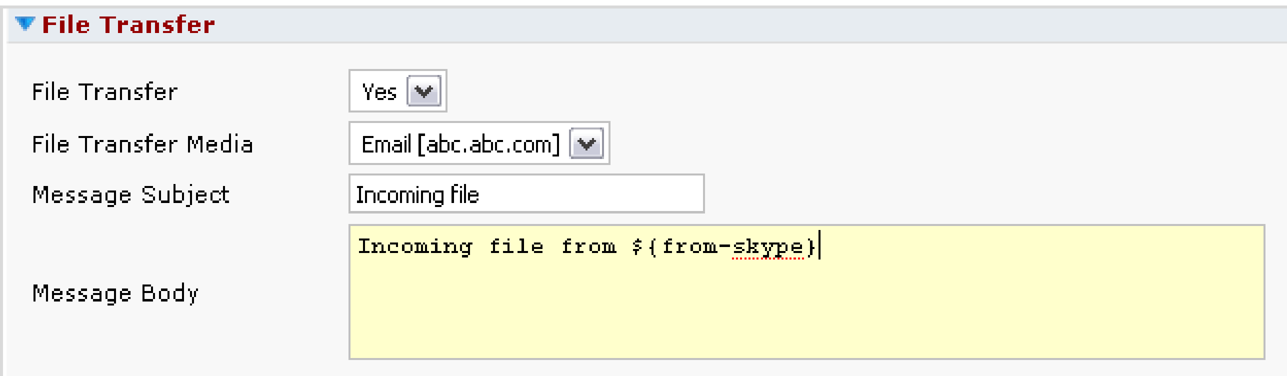


Image 18: Skype file transfer settings

Setting	Description
File Transfer	This setting allows you to enable or disable file transfer capabilities. This setting is set to “No” by default.
File Transfer Media	File Transfer Media is used to specify how every file sent to this Skype account should be transferred from VoiceGear Connect gateway. You must have Media Types created before using this feature. Please refer to section 6.2 below for detailed explanation on creating new Media Types. Existing Media Types are listed in the File Transfer Media drop-down list. Please select one Media Type to use for transferring files incoming to this Skype account.
Message Subject	Specify text to be included in message subject. You can use message variables to customize the text. See section 5.2.2.1 below.
Message Body	Specify text to be included in the message body. You can use message variables to customize the text. See section 5.2.2.1 below.

5.2.2.1 Message Variables

When using Email as File Transfer Media, you can use following system variables to customize Message Subject and Body text.

Each variable can be specified using the following syntax: ${*variablename*}

For a complete list of available variable and related generated strings see the table below:

Variable Name	Generated String
sysname	"VoiceGear Connect"
hostname	<system hostname>
date-time	Current date and time. Date and time is in the following format: YYYY-MM-DD HH:MM:SS Example: String "2008-12-23 15:55:00" means December 23rd 2008 at 15:55
account	Name of Skype account receiving the file
user-name	Name of Skype account sending the file
user-dispname	Display name of Skype account sending the file
file-name	Name of the sent file
file-size	Size of the sent file in bytes

5.2.3 Configuring Advanced Settings

Advanced settings allow you to configure special features for your Skype accounts.

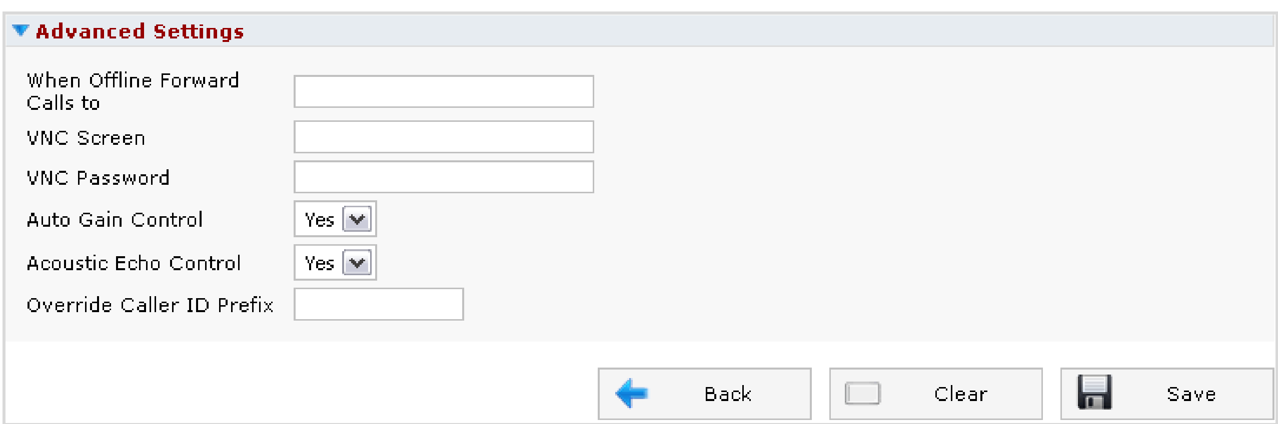


Image 19: Skype account advanced settings

Setting	Description
When Offline Forward Calls to	This setting lets you define another Skype account to which all incoming calls are forwarded in case current account is off-line or disabled.
VNC Screen / VNC Password	The system allows using a standard VNC viewer to connect to individual Skype sessions running on the gateway and view actual Skype clients. VNC Screen and VNC Password settings are used to configure authentication parameters to control access to a VNC session associated with the current Skype account. To connect to the VNC session using a third party VNC client utility, you need to specify the IP address of the VoiceGear Connect gateway, followed by colon sign, followed by the VNC screen set here. As a rule, this capability should only be utilized when the system is in debug mode.
Auto Gain Control	Auto gain control setting enables or disables automatic gain adjustment. With this setting active, the Skype account will continuously adjust voice gains depending on the voice level during the conversation. This setting is set to “Yes” by default.
Acoustic Echo Control	Acoustic echo control setting enables or disables acoustic echo cancellation algorithms built into the Skype client. This setting is set to “Yes” by default.
Override Caller ID Prefix	This setting allows you to override the default caller ID as configured in “Settings” -> “Misc Settings” -> “Caller Identification”.

Now that you have registered Skype accounts to work with VoiceGear Connect gateway, you can use Skype profiles in order to quickly and easily share information about your company with other Skype users.

5.3 Skype Profiles

Skype profile allows you to share your personal, company or generic information with other Skype users. The more information you provide in the profile, the easier it will be for other Skype users to find you on the Skype network. Profile information that you fill in this section is the same information as you fill in the user profile page in the Skype client application.

By creating a Skype profile and assigning multiple Skype accounts to that profile, you can easily control what information is displayed to a remote Skype user for each assigned Skype account.

To create a new Skype profile, please navigate to Skype -> Skype Profiles menu. You will see the page shown below. If you don't have any Skype profiles created, the table below will be empty. If you already have existing Skype profiles created in the system, they will show up in the list below.

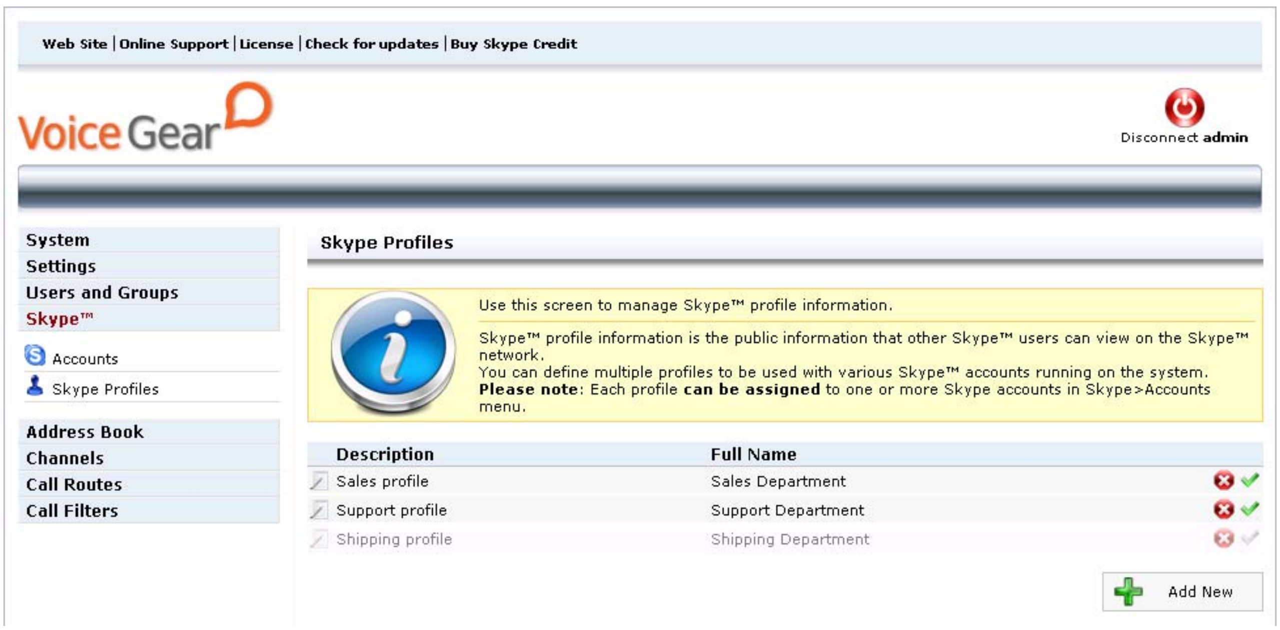


Image 20: Sample list of configured Skype profiles

To add a new Skype profile click on the "Add New" button. You can manage existing Skype profiles using following features:

- Edit an existing Skype profile by clicking on profile "Description" or the editor icon on the left side of the profile entry.
- Enable or disable a profile by clicking on the "checkmark" icon on the right side of every profile entry.
- Delete a profile by clicking on the "red cross" icon on the right side of every profile entry.
- Sort existing Skype profiles by dragging and dropping entries up and down in the table.

5.3.1 Profile Basic Settings

After clicking on the “Add New” button in the Skype profiles page you will see the “Skype Profile Editor” page shown below.

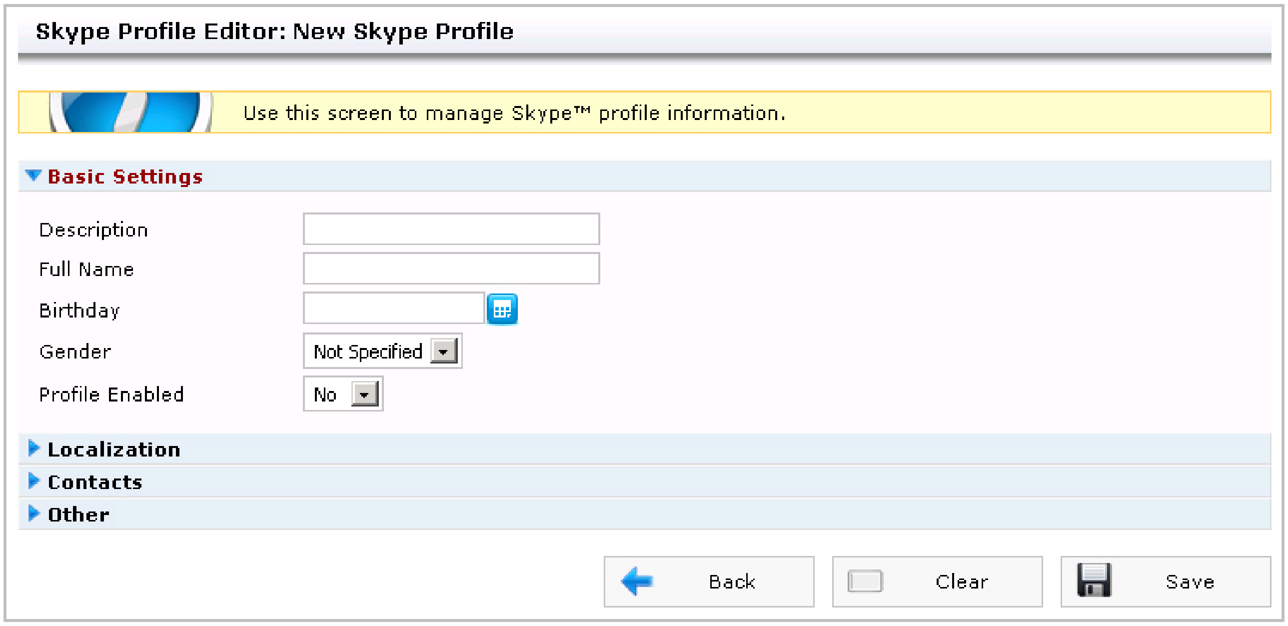


Image 21: Basic profile settings

Basic settings allow you to configure essential information about the user. When creating a new Skype profile you should provide a meaningful name in the Description field. This way, it will be easily identifiable in other areas of the VoiceGear Connect Web interface.

Setting	Description
Description	Allows you to specify a descriptive profile name to easily identify this profile in various places in the VoiceGear Connect Web interface.
Full Name	Allows you to specify the full name in the profile.
Birthday	Allows you to specify the birthday in the profile. This field is optional.
Gender	Allows you to specify the gender in the profile. This field is optional.
Profile Enabled	This setting allows you to enable or disable Skype profile. By default, this setting is set to “No”.

5.3.2 Profile Localization Settings

Localization settings allow you to define details about geographical location and spoken language. This information will be displayed to other Skype users and can be used to assist your customers with choice of language and best time to reach you depending on your spoken language and time zone settings.

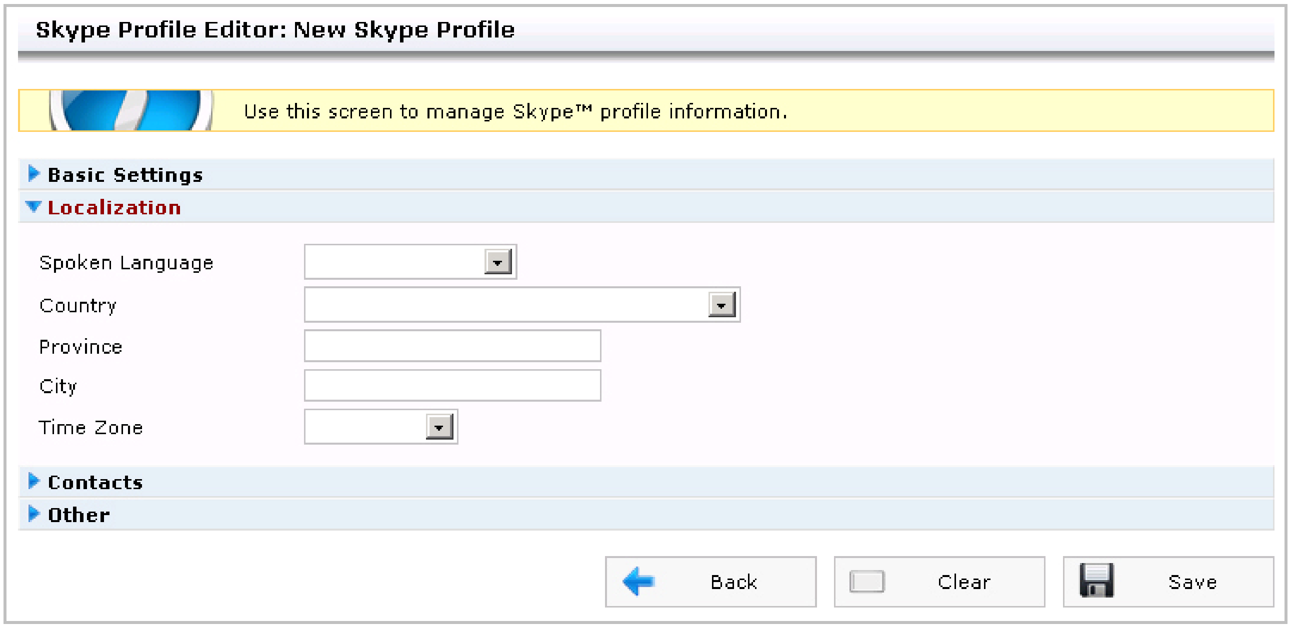


Image 22: Profile localization details

Setting	Description
Spoken Language	Spoken language setting is used to specify your preferred language to speak. This feature is particularly useful when you have bi-lingual teams accepting incoming Skype calls.
Country	Country setting is used to specify your country of residence or business location.
Province	Province setting is used to specify your province of residence or business location.
City	City setting is used to specify your city of residence or business location.
Time Zone	Time zone is used to specify the time zone in your location.

5.3.3 Profile Contacts Settings

Profile contact settings allow you to define contact information pertaining to this profile. This information will be displayed to other Skype users and can be used to contact you.

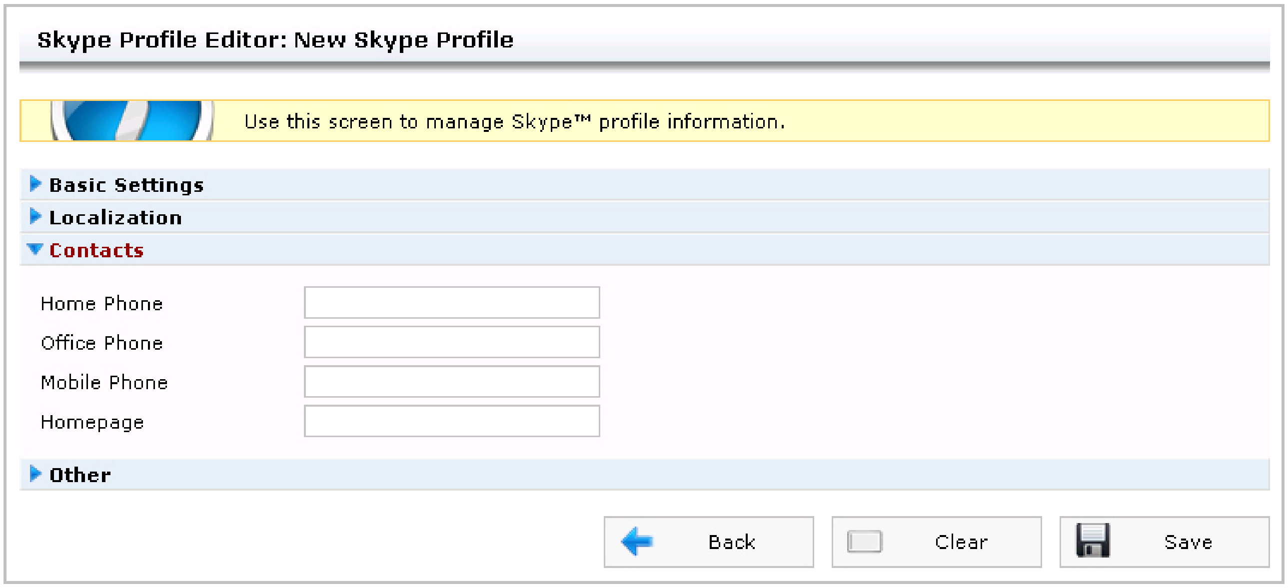


Image 23: Profile contact details

Setting	Description
Home Phone	Home phone setting is used to specify your home or alternative phone number that you can be reached at when outside of the office.
Office Phone	Office phone setting is used to specify your main office phone number to be reached at.
Mobile Phone	Mobile phone setting is used to specify your mobile or alternative phone number that you can be reached at when outside of the office.
Homepage	Homepage setting can be used to provide a generic URL to you company website or specific URL pertaining to your office location or department.

5.3.4 Profile Other Settings

Profile other settings allow you to configure some descriptive information about the user.

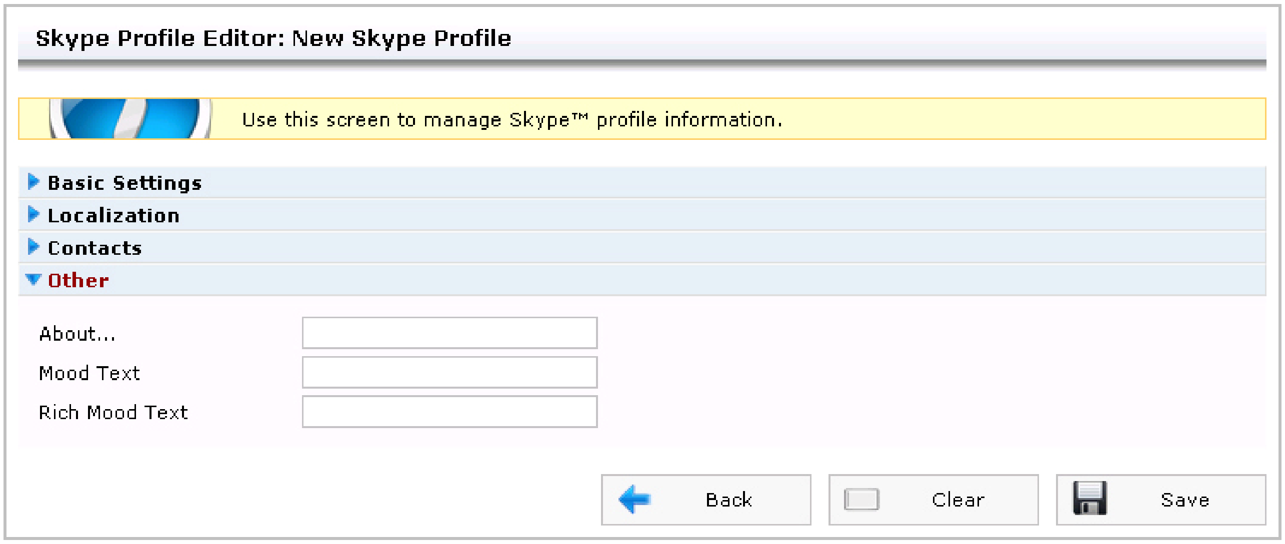


Image 24: Profile other details

Setting	Description
About…	About setting can be used to specify some descriptive information about the user.
Mood Text	Mood text can be used to specify some descriptive information relevant at the moment. It can be changed as frequently as your "mood" changes.
Rich Mood Text	Mood text in rich text format.

You can assign multiple configured Skype accounts to a single Skype profile in order to display, for example, same location and contact information for all accounts. If you want to display different profile information for each Skype account, you need to create separate Skype profiles and assign them to individual Skype accounts using Skype -> Accounts menu.

6. System Settings

This section outlines and describes settings that affect the entire system. Typically, system settings are set once during the installation of the gateway and, as a rule, don't need to be adjusted after unless the gateway is moved or a certain feature needs to be activated or deactivated.

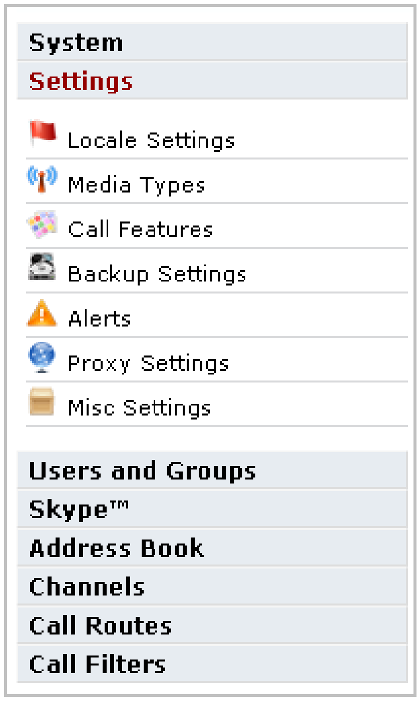


Image 25: System settings menu

System settings can be found by selecting "Settings" in the left menu bar of the VoiceGear Connect Web Interface. They are divided into several subsections as follows:

- Locale Settings
- Media Types
- Call Features
- Backup Settings
- Alerts
- Proxy Settings
- Misc Settings

6.1 Locale Settings

Locale settings describe the geographical location of the gateway and only need to be defined once unless the gateway is physically moved.

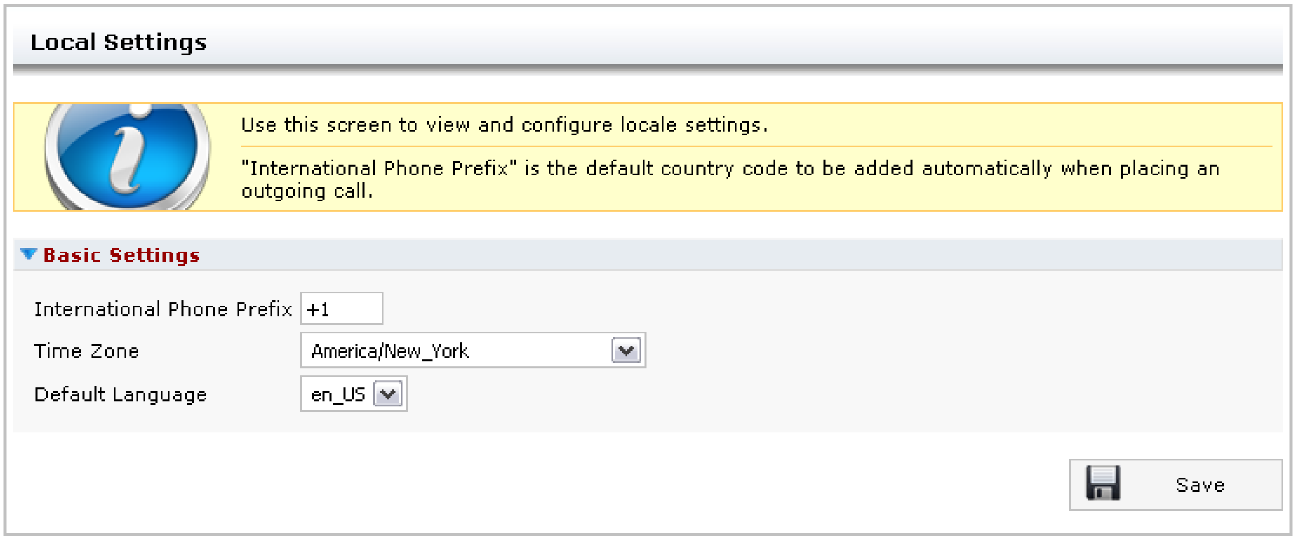


Image 26: Locale settings

Details for each setting are outlined in the table below:

Setting	Description
International Phone Prefix	International phone prefix is used by the system to simplify SkypeOut calling. By default, SkypeOut service requires the international phone prefix to be entered for every call, even if it appears to be a local call. VoiceGear Connect will automatically prefix the value of this setting to any SkypeOut call where it is not explicitly entered by the caller. The proper value of this setting depends on the location of the gateway. For North American locations, the value of “+1” should be used for this setting.
Time Zone	Time zone is used for system internal clock synchronization and should be set to the closest entry to the physical location of the gateway.
Default Language	Default language setting is used to configure the language used for the web interface as well as VoiceGear Connect console. Out of the box, it is set to English.

Please Note:
A caller can override the international phone prefix by explicitly dialing it from a touch tone telephone. As an example, a user can dial 416 123 4567 which would result in the system dialing +1 416 123 4567 if the prefix is set to “+1”. Alternatively, a user can explicitly dial +1 416 123 4567 in which case the system will not add the default prefix.

6.2 Media Type Settings

Media types define transports that can be used to notify various users of the system status and can be used by system-wide backup and notification services. There are two available media type protocols to choose from which affect the way notifications are delivered to a given destination:

- Email
- FTP

The system supports unlimited number of media types which can be used by various notification services. The screen below shows 2 configured media types, one using email protocol, the other using FTP. Initially, there are no predefined media types. A new media type can be added by clicking on the “Add New” button.

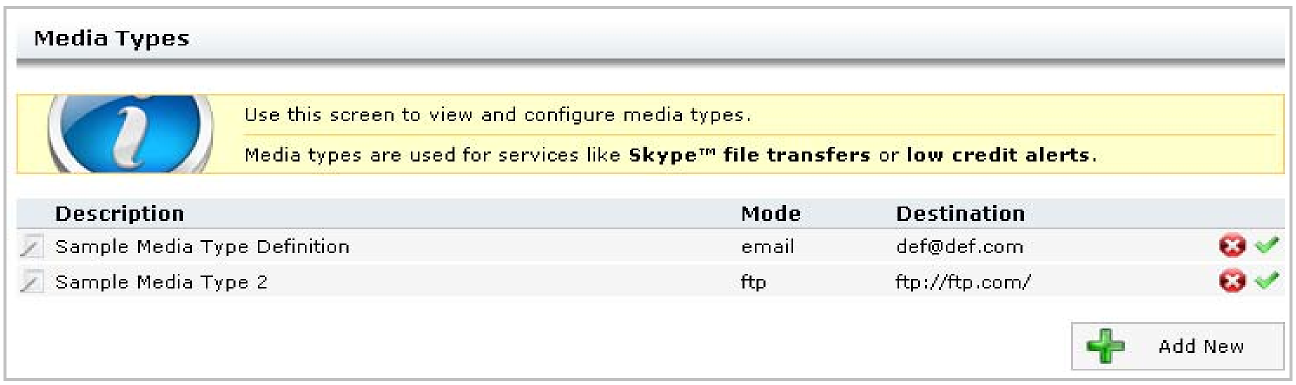


Image 27: List of configured media types

6.2.1 Configuring Media Type using Email

Media type configured using email protocol can be used to send automated emails to a predefined party which contain notification data as an attachment.

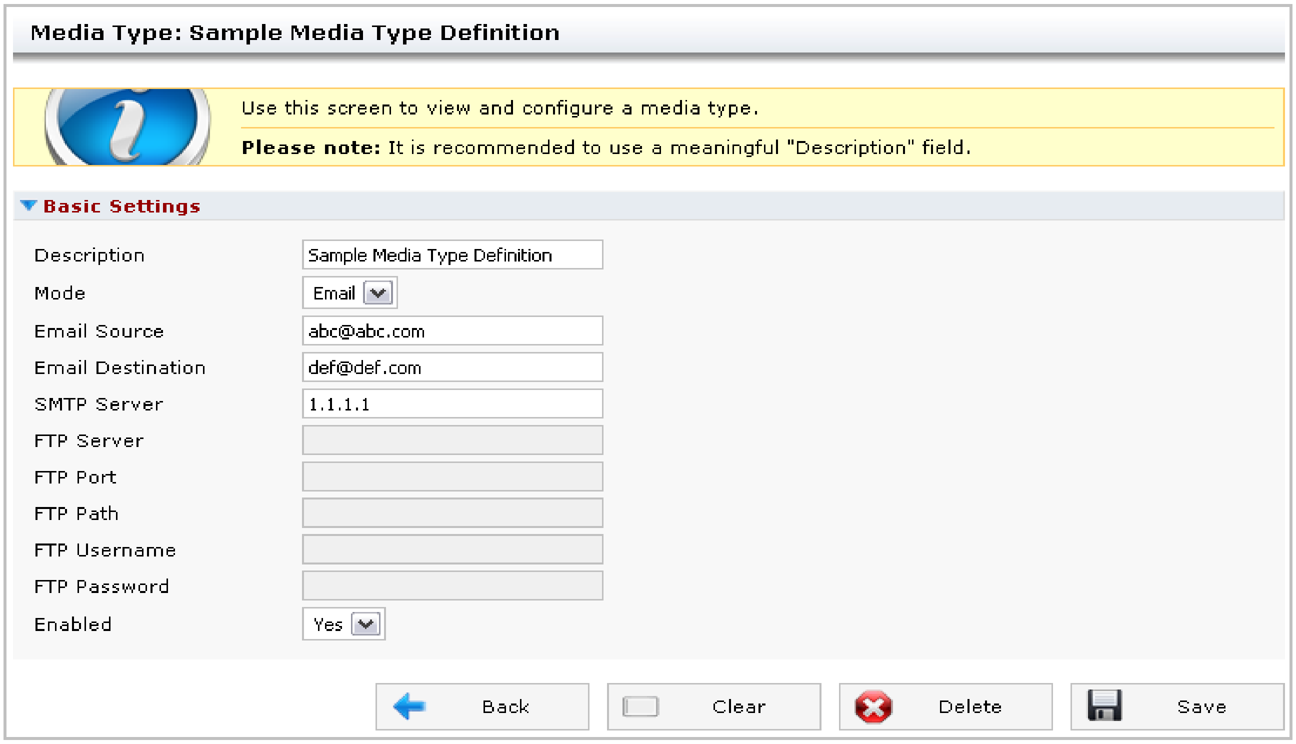


Image 28: Email media type configuration

To configure a media type using email, click on the "Add New" button and select "via email" option from the mode pick list that appears. Required settings to configure a media type using email are detailed in the table below.

Setting	Description
Description	A meaningful name describing the media type. This value will be displayed in the media type list once at least one media type has been configured.
Email Source	The recipient will receive emails from and email address defined in this field.
Email Destination	A valid destination email address is specified in this field. Invalid email address value will result in errors during notification delivery process.
SMTP to Use	A valid IP address or a fully qualified domain name of an outgoing (SMTP) server is specified in this field. Invalid value will prevent email notifications from being sent.

6.2.2 Configuring Media Type using FTP

Media types configured using FTP can be used to send notifications as well as system backups to a predefined FTP server.

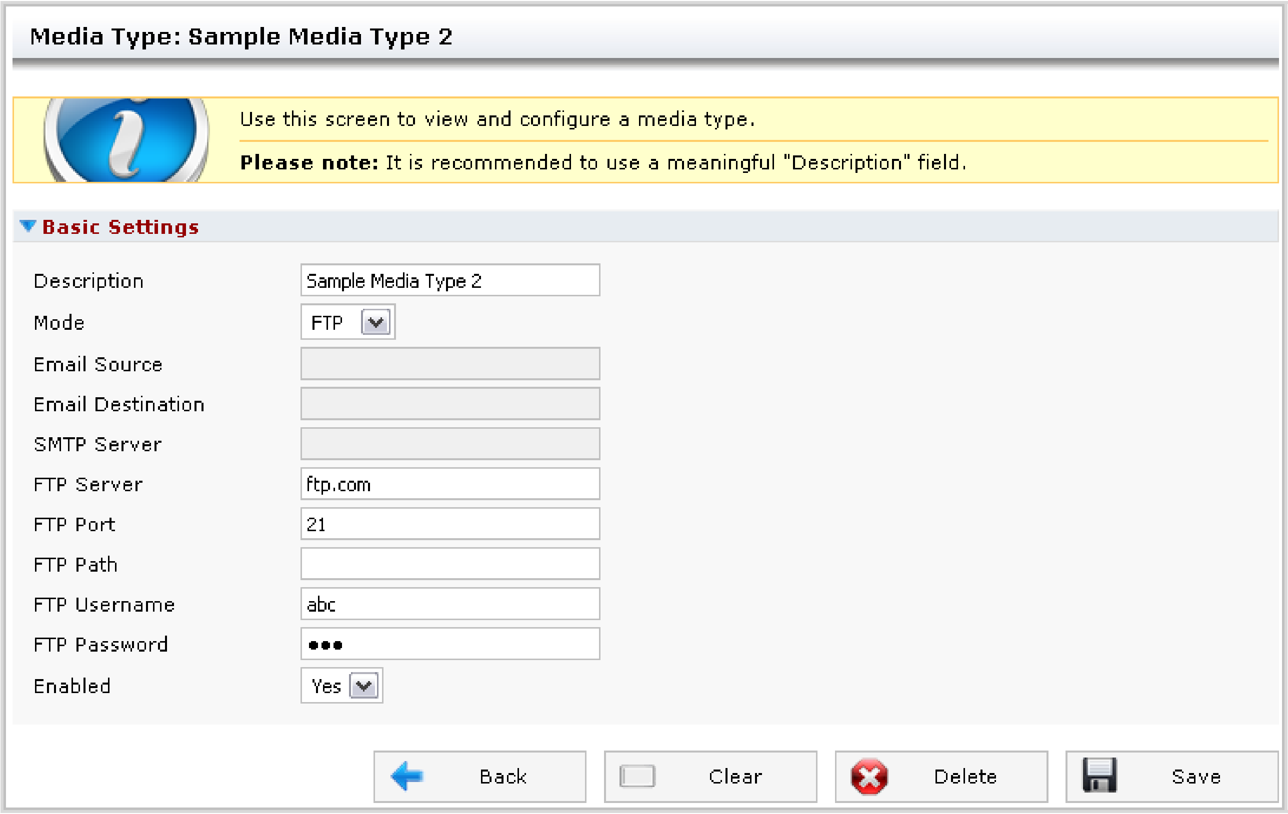


Image 29: FTP media type configuration

To configure a media type using FTP, click on the "Add New" button and select "via FTP" option from the mode pick list that appears. Required settings to configure a media type using FTP are detailed in the table below.

Setting	Description
Description	A meaningful name describing the media type. This value will be displayed in the media type list once at least one media type has been configured.
FTP Server	A valid IP address or a fully qualified domain name of an FTP server is specified in this field. Invalid value will prevent notifications from being sent.
FTP Port	A TCP port on which to contact the FTP server.
FTP Path	Directory path on the FTP server to upload notification files to. To use the root path of the server, please specify "." in this field.
FTP Username	A valid FTP username with "write" permissions used to login into the remote FTP server.
FTP Password	A matching FTP password used to login into the remote FTP server.

6.3 Call Features

Call features allow users to interact with the gateway using a touch tone phone. Certain call features can be enabled or disabled by a user using special key combinations defined in this section both before and during a call.

6.3.1 Feature Configuration Before a Call

To configure call features accepted by the system from a touch tone phone before the call is placed, please select the “Before the Call” section using the web interface. User input prior to placing a call is limited by the system to user authentication functions. As an example, a user can pick up a handset connected to the gateway and authenticate using a special sequence composed from authentication prefix, numeric username and numeric password. Once an authentication is performed, the system can utilize rules defined for the authenticated user to block or allow calls as well as log user activity.

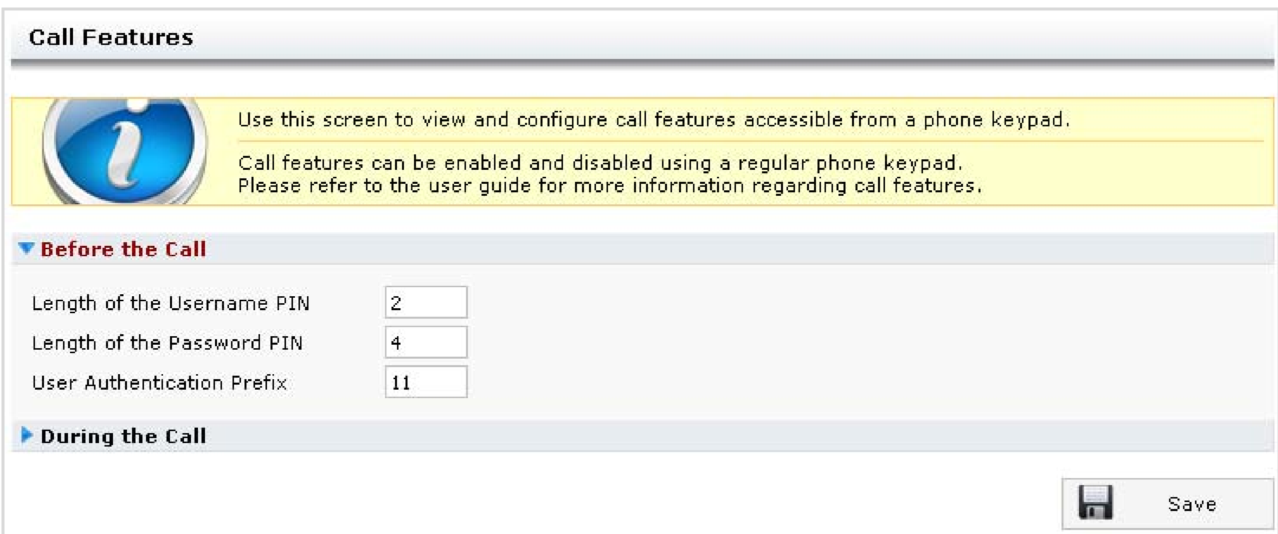


Image 30: Parameters for user authentication before a call

Settings that control user authentication format from a touch tone phone are defined in this section and detailed in the table below.

Setting	Description
Length of Username PIN	A number of digits that define a user’s username PIN in the authentication sequence. Please note that the actual username PIN can be defined on a per user basis via “Users and Groups”-> “Users” menu.
Length of Username Password	A number of digits that define a user’s password PIN in the authentication sequence.
User Authentication Prefix	A specific number sequence that signifies the start of user authentication prior to making a call. When the system detects this number sequence, it will compare subsequent number input against defined users and attempt to authenticate a match. Should no

	match be found, the system will assume the number input is a SkypeOut phone number and proceed with dialing.

Please Note:
If no authentication sequence is entered or if an authentication sequence cannot be matched against the user database, the system will assume the user to be anonymous. For example, if user authentication prefix is set to 11, username PIN length set to 2 and username password PIN length set to 3, a user can login using the following sequence: 11 12 123. Should this sequence match an existing user, an authentication will succeed. Should this sequence not match an existing user, authentication will fail and the system will assume an anonymous user for the next call.

6.3.2 Feature Configuration During a Call

A greater number call features are available to a touch tone phone user when the call is already in progress. A user can choose to convert a call into a conference or even logout and login with as a different user with a series of simple touch tone numeric commands.

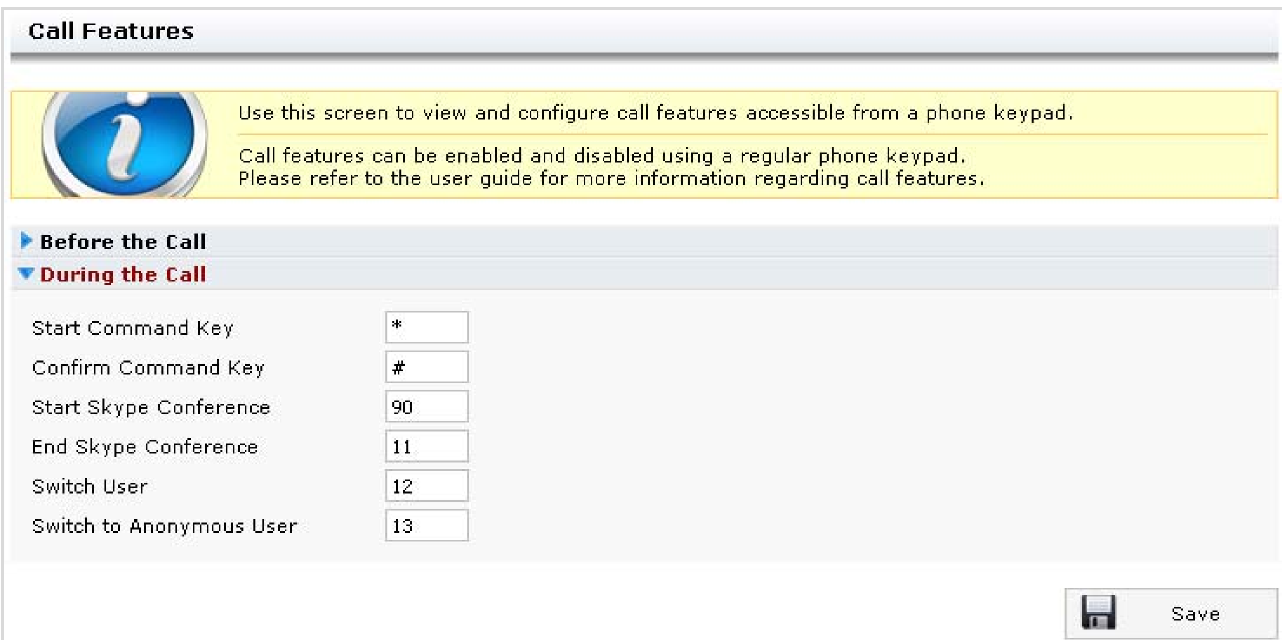


Image 31: Commands for users to use during a call

Detailed settings for features available during the call are shown in the table below.

Setting	Description
Start Command Key	Start command key indicates the start of command input. All commands that invoke features while a call is in progress must start from this key.
Confirm Command Key	Confirm command key indicates the end of command input. All commands that invoke features while a call is in progress must end with this key in order to get processed by the

	system.
Start Skype Conference	Command that converts the call currently in progress into a Skype conference.
End Skype Conference	Command that ends a Skype conference if there is one already in progress. All remote parties in the conference at the moment of confirmation of this command will be automatically dropped.
Switch User	Command that switches the currently logged-in user to a different user provided a new authentication sequence is entered that matches a valid user.
Switch to Anonymous User	Command that logs out a currently logged in user.

Please Note:
Every command sequence that activates a feature has to start from the start command key and finish with confirm command key in order to be properly recognized by the system.

Please Note:
To activate a Skype conference, it is necessary to enter the start command key, followed by start Skype command key combination, followed by the Skype shortcut or a phone number of a remote party to be included in a conference, followed by the confirm command key. For example, to initiate a Skype conference with a remote party at 416 123 4567, it is necessary to enter *904161234567# from a touch tone phone if start command key is defined as *, confirm command key is defined as #, and start Skype conference sequence is defined as 90.

6.4 Backup Settings

Backup settings allow system administrators to control the way system configuration and call logs are backed up in case of system failure. Backup service uses predefined system media types to be able to send backed-up data to a remote location and needs at least one media type to be created in order to function. Media type configuration is detailed in section 6.2.

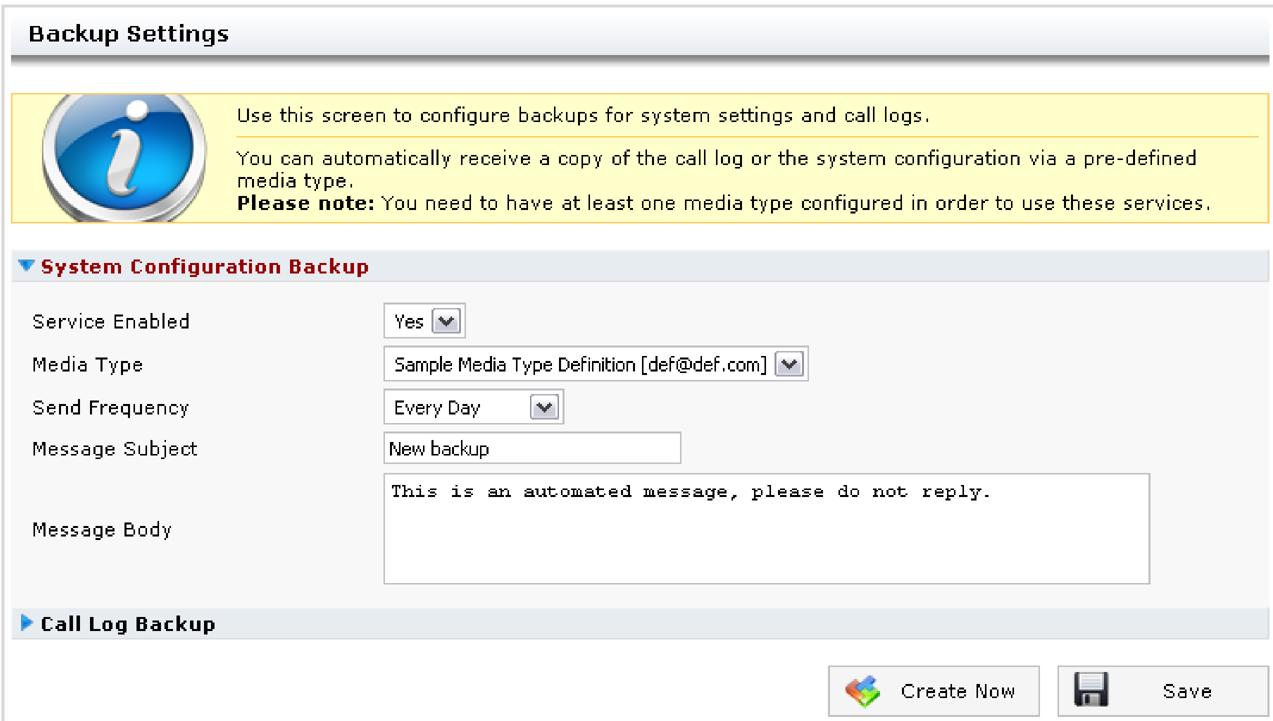


Image 32: System configuration backup settings

To create a backup, enable the backup service by setting “Service Enabled” pick list to “Yes”, select the media type to use and select the backup frequency. To run the backup using the newly entered information, push the “Create Now” button. Otherwise, push “Save” button to store the backup information in system. Periodic backups will run on selected intervals starting from the time backup information has been saved.

When backup is compiled, the system automatically collects all system settings defined in the web user interface.

To copy call logs for book keeping or accounting purposes, select the “Call Log backup” section, select the appropriate media type and send frequency and push the Save button. To save bandwidth, call logs can be compressed using gzip compression compatible with WinZip application.

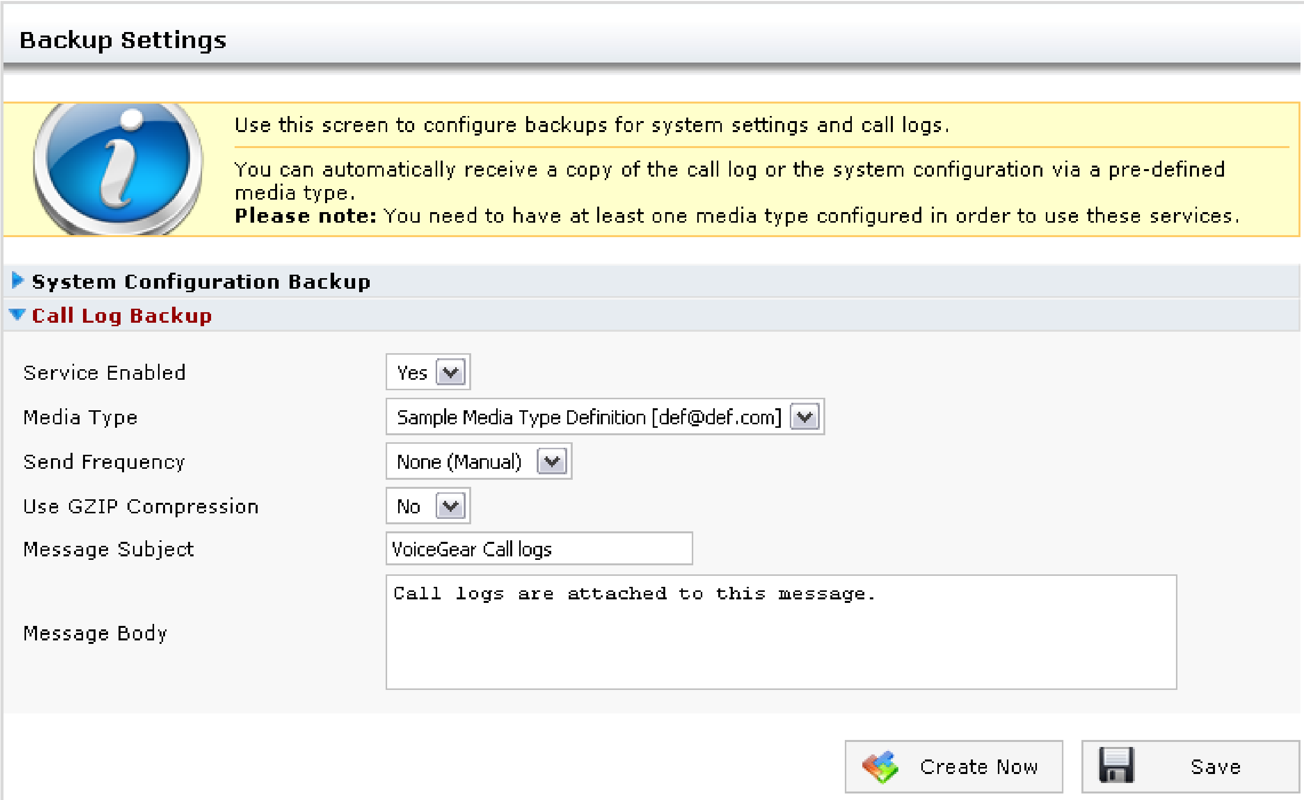


Image 33: Call log backup settings

Both system setting and call log backup services can be configured to send email messages with custom message subject and body fields provided a media type selected is using an email protocol for delivery. It is possible to specify certain system variables for both message subject and body. Available system variables are listed in the table below.

Variable Name	Description
sysname	Outputs "VoiceGear Connect"
hostname	Outputs the current network hostname.
date-time	Outputs the current date and time based on the internal system clock.

Please Note:
Variables can be specified for email message subject and body using the following syntax: ${variable}. For example, to output the current host name for the gateway, it is necessary to specify ${hostname}.

6.5 Alert Settings

The system can be configured to notify system administrators when Skype accounts registered with the gateway are running low on Skype credit. Enabling this service can prevent line downtime.

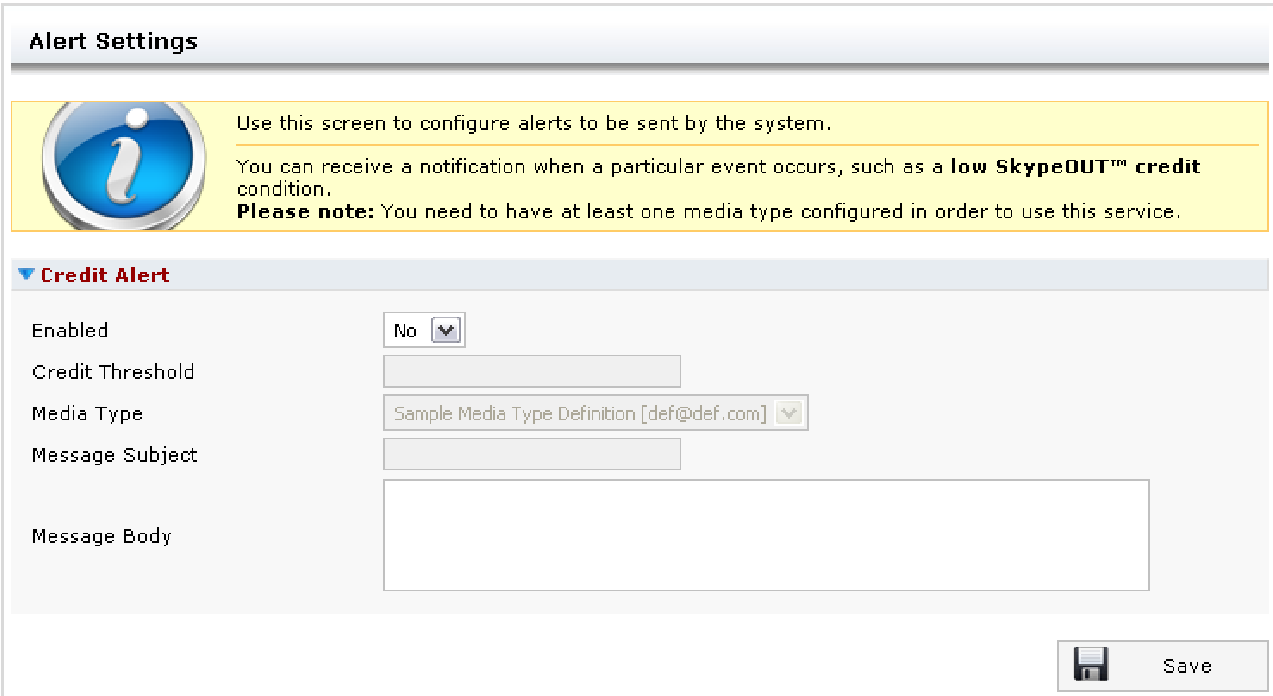


Image 34: Low Skype credit alert settings

To configure low credit alerts, at least one media type has to be defined in the system. Media type configuration is detailed in section 6.2. For more details on low credit alert settings please refer to the table below.

Setting	Description
Enabled	Enabled setting indicates whether the low credit alert service is enabled. When using SkypeOut services without a Skype unlimited option it is recommended to keep this setting turned on.
Credit Threshold	Credit threshold setting indicates the lowest credit level for any Skype account registered with the system before the low credit alert is triggered. The currency value for this setting is the default currency value for the country the gateway is located in.
Message Transport	The media type name to use for low credit alert notifications.
Message Subject	Low credit alert notification email message subject.
Message Body	Low credit alert notification email message body.

Low credit alert service can be configured to send email messages with custom message subject and body fields. It is possible to specify certain system variables for both message subject and body. Available system variables are listed in the table below.

Variable Name	Description
sysname	Outputs "VoiceGear Connect".
hostname	Outputs the current network hostname.
date-time	Outputs the current date and time based on the internal system clock.
account	Outputs the name of the Skype account that has crossed the low credit threshold.
balance	Outputs the Skype credit balance remaining on that account.
balance-currency	Outputs the Skype credit currency so the Skype account running low on credit.
threshold	Outputs the current low credit alert notification threshold configured in the system.

Please Note:
Variables can be specified for email message subject and body using the following syntax: ${variable}. For example, to output the current host name for the gateway, it is necessary to specify ${hostname}.

6.6 Proxy Settings

If your network is set up to use a proxy to access the Internet, it is possible to configure the system to utilize this proxy for all registered Skype accounts. It is recommended to avoid using proxy settings whenever possible in order to maximize gateway performance.

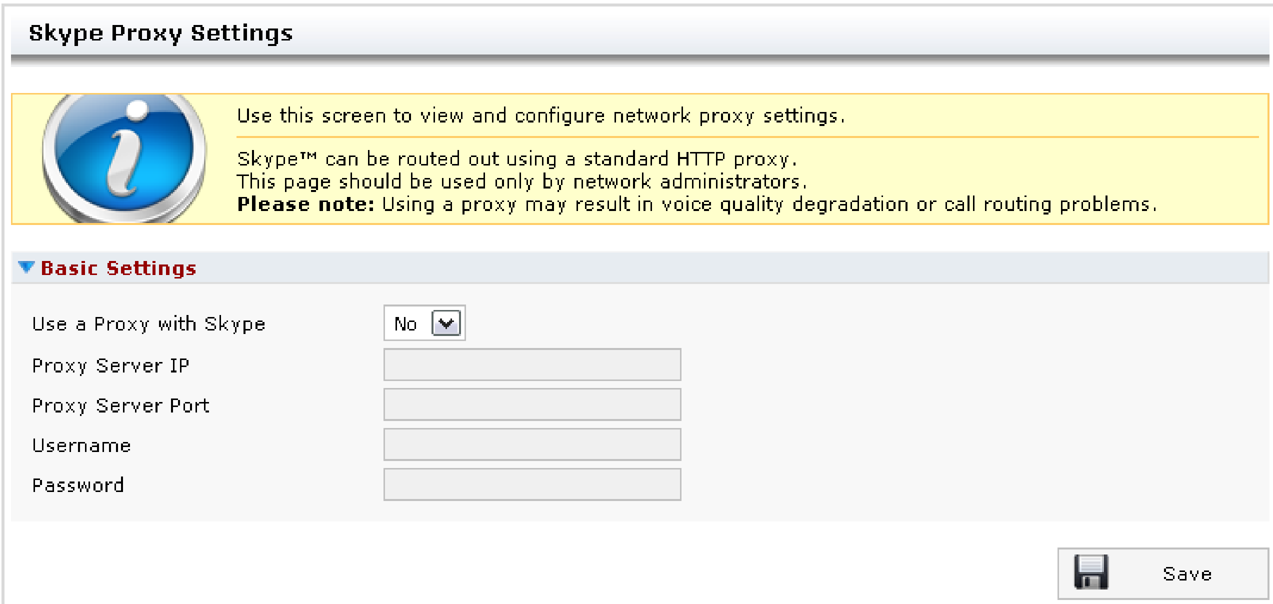


Image 35: Proxy server configuration

Detailed proxy settings are shown in the table below.

Setting	Description
Use Proxy with Skype	This setting allows to set proxy service status. If it is set to “Yes”, proxy service is enabled and all Skype accounts registered with the system are using proxy settings defined in this section to access the Internet.
Proxy Server IP	IP or a fully qualified domain name of the proxy server.
Proxy Server Port	Port of the proxy server.
Username	A username to use to authenticate with a proxy server.
Password	A matching password to use to authenticate with a proxy server.

6.7 Misc Settings

Misc settings section contains a collection of system-wide settings that are not essential for proper system operation. A high level list of misc setting categories is shown below.

- Address Book Preferences
- Caller Identification Options
- Skype Authorization Policy
- Skype Chat Response Policy
- Time Synchronization Preferences

6.7.1 Address Book Preferences

It is possible to configure the system to setup and maintain an automatic address book for all incoming Skype callers. If this ability is enabled, the will system automatically detect incoming caller Skype registration details such as name, location, etc. and create appropriate address book entries in sequence starting from a shortcut number defined in the "Starting Shortcut Number…" field.

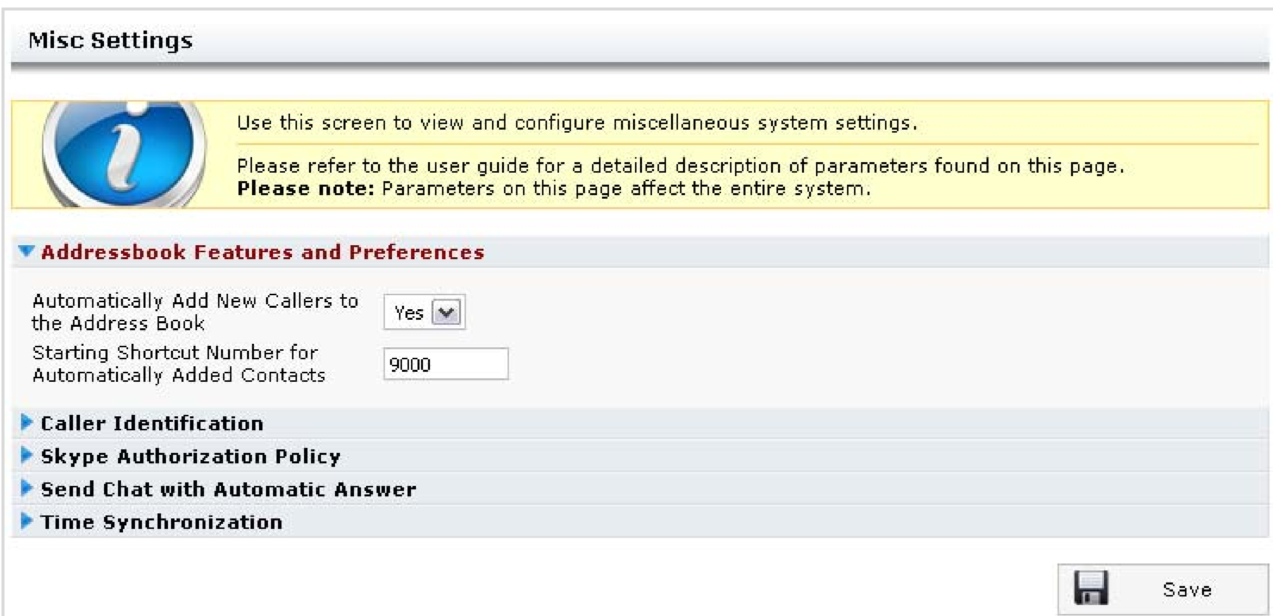


Image 36: Automatic address book settings

To enable this ability, set the "Automatically Add New Callers to the Address Book" to yes; set the starting number for automatically added contacts, and click on the "Save" button.

Please Note:
Disabling this ability does not delete any automatically added contacts already present in the address book.

6.7.2 Caller Identification Options

Caller identification settings available in this section allow users to set default, system-wide policy for passing caller ID to the PBX system integrated with VoiceGear Connect. This policy can be overridden for each address book contact or Skype account registered with the system.

Image 37: Caller ID options

For more details on available settings please refer to the table below.

Setting	Description
Default Caller ID	This setting allows selecting a default caller ID passed to the PBX system for incoming calls. The default value for this setting is set as "Shortcut number" in which case the system will pass a numeric address book shortcut for all incoming callers.
Default Caller Name	This setting allows selecting a default caller name passed to the PBX system for incoming calls. Analog PBX systems do not support extended caller name values within the caller ID.
Default Caller Prefix	This setting allows users to add an alphanumeric prefix to any caller ID passed to the PBX system. This setting can be used to distinguish incoming Skype calls from all other incoming calls.

Please Note:
New SIP PBX systems are able to support alphanumeric characters in the caller ID, while older analog or hybrid systems are not. Please refer to your PBX system configuration guide to determine whether your system supports alphanumeric caller ID.

Please Note:
If your PBX system supports alphanumeric caller ID, it is possible to configure default caller prefix as "Skype" and Default Caller ID to pass "Skype username". In this case, an incoming caller ID shown on an office phone will be shown as "Skype: incoming caller name".

6.7.3 Skype Authorization Policy

Skype authorization policy settings allow setting the default system behavior for handling Skype authorization requests sent to any of the registered Skype accounts. The system can be set to automatically grant an authorization request allowing a remote Skype user to add one of the Skype accounts registered with the system to their contact list. Alternatively, the system can be configured to deny such a request.

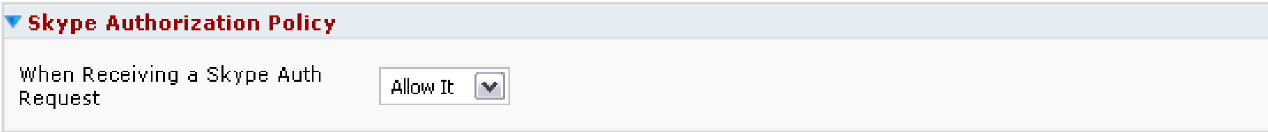


Image 38: Skype authorization request policy settings

6.7.4 Skype Chat Response Policy

Skype chat response policy settings allow setting the default system behavior for handling Skype chat requests sent to any of the registered Skype accounts. A custom message can be set to be automatically used by the system to answer any incoming chat requests from remote Skype users.

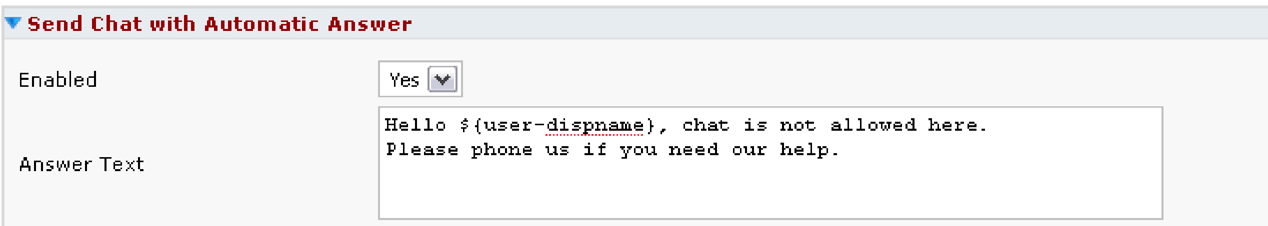


Image 39: Skype chat response settings

6.7.5 Time Synchronization Preferences

Time synchronization preferences allow setting the default time synchronization behavior for internal system clock. It is possible to enable time synchronization in which case the system will automatically synchronize internal clock with a NTP server using the specified URL. Alternatively, time synchronization can be turned off.

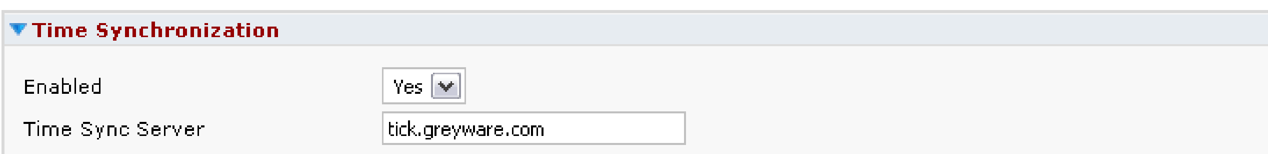


Image 40: Time synchronization settings

Please Note:
It is highly recommended not to turn off time synchronization.

7. Channel Configuration

Channel configuration section allows you to configure the PBX interface that will be used by the VoiceGear Connect gateway to pass incoming or outgoing calls. There are several integration options available which cover majority of PBX systems available on the market today. Depending on your gateway configuration, it is possible to utilize SIP, analog (FXS/FXO), or digital (ISDN PRI) connectivity to the PBX system.

A detailed list of available connectivity options is shown below.

Channel Designator	Connection Description
Analog	This connection is either extension side (FXO) or trunk side (FXS) or mixed analog connection which plugs into to the PBX via RJ-11 connectors or bare wires.
Digital	This connection is a digital type connection and can be set to either T1 mode (24 lines) or E1 mode (30 lines). As a rule, T1 connectivity is used in North America and Japan while E1 is used elsewhere in the world including Europe and Asia.
SIP	This is a VoIP connection using G.711 A-law/µ-law codec for integrating with SIP or hybrid PBX systems including Asterisk.

Please Note:
As a rule, a single interface is used to integrate VoiceGear Connect with a PBX system be it SIP, analog or digital.

Please Note:
Any changes made to a working channel configuration will result in a channel reset which drops all calls currently in progress.

7.1 SIP Channel Configuration

SIP channel configuration settings allow you to configure PBX connectivity settings using SIP protocol. VoiceGear Connect utilizes G.711 A-law/µ-law codecs over SIP 2 protocol. To configure SIP integration to your PBX, the following general approach has to be followed:

1. Locate SIP channel configuration screen by selecting Channels->SIP from the main menu.
2. Configure basic and advanced SIP settings to values required by your PBX system. Default settings will work fine with most PBX systems.
3. Setup a SIP trunk by clicking the “Create SIP trunk” button and point it to the IP or a fully qualified domain name of your PBX system.
4. Create a SIP trunk on your PBX and point it to the IP or a fully qualified domain name of the VoiceGear Connect gateway.
5. Create incoming and outgoing routes to be able to access the newly created VoiceGear Connect trunk.

Please Note:
Please consult your PBX user manual for more information on how to perform the PBX-side setup. In most cases, configuring a PBX system to work with VoiceGear Connect via SIP is the same as configuring it to work with an external SIP provider.

7.1.1 Basic SIP Settings

Basic SIP settings allow you to configure basic parameters applicable to the SIP server component running on the VoiceGear Connect gateway.

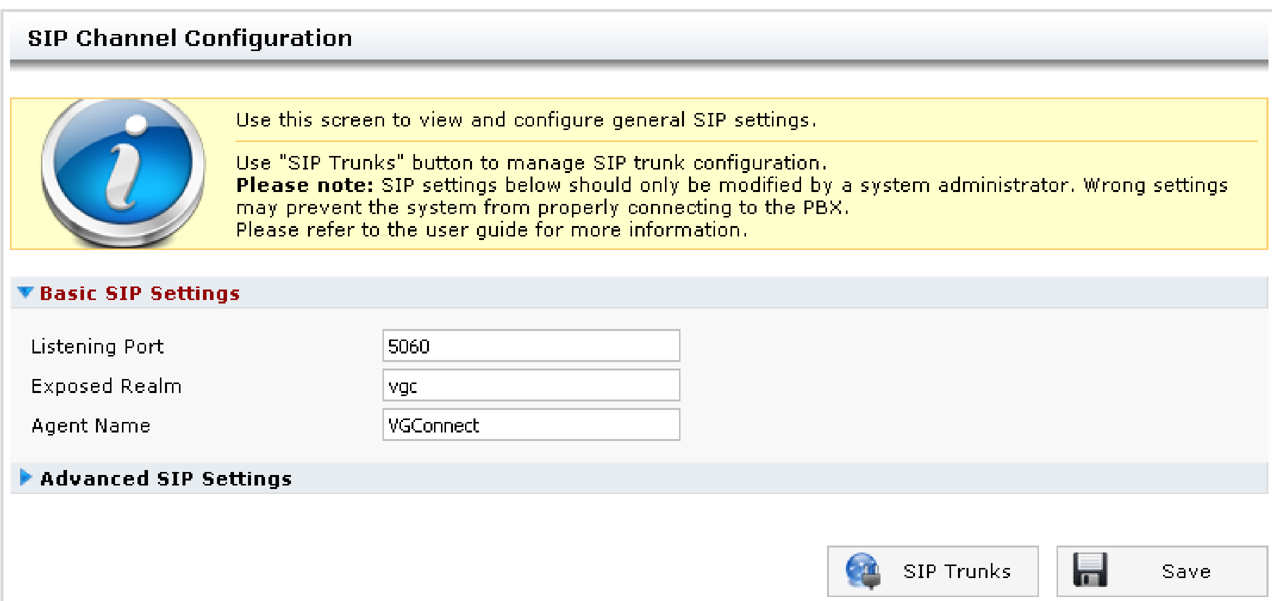


Image 41: SIP server settings

For more details on basic SIP settings please refer to the table below.

Setting	Description
Listening Port	This setting allows you to change the network (UDP) port the VoiceGear Connect is using to listen for new calls from the PBX. The default value of 5060 is used in a vast majority of PBX systems and is usually defined on the PBX side during SIP trunk configuration. As a rule, port value specified on the PBX side as well as in this section has to match.
Exposed Realm	The name of the realm VoiceGear Connect exposes to the PBX system. In most cases, this value does not need to change.
Agent Name	The SIP agent name VoiceGear Connect exposes to the PBX system. In most cases, this value does not need to change.

7.1.2 Advanced SIP Settings

Advanced SIP settings allow you to configure detailed parameters of your SIP connection.

Please Note:
Advanced settings should only be changed by a system administrator with a very good understanding of SIP protocol. In most cases, default settings are sufficient to configure a functional SIP channel to the PBX system.

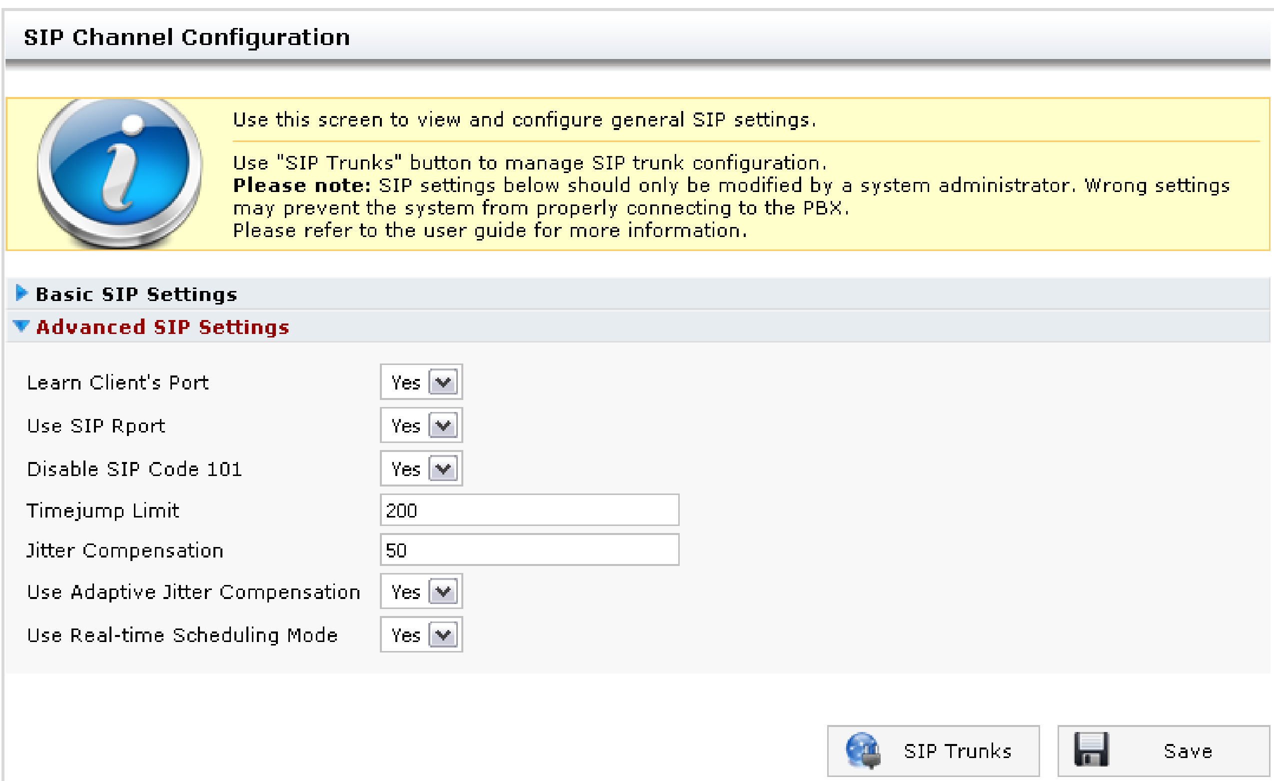


Image 42: SIP server advanced settings

7.1.3 SIP Trunks

Out of the box, VoiceGear Connect gateway with SIP connectivity can communicate with multiple PBX systems or SIP clients. Connection details to each PBX system or SIP client are contained within a single SIP trunk.

To access the SIP trunk configuration, please select “Channels->SIP” from the main menu and click on the “SIP Trunks” button. The interface will show a list of existing trunks (if any) and provide an option to create a new trunk via the “Add New” button.

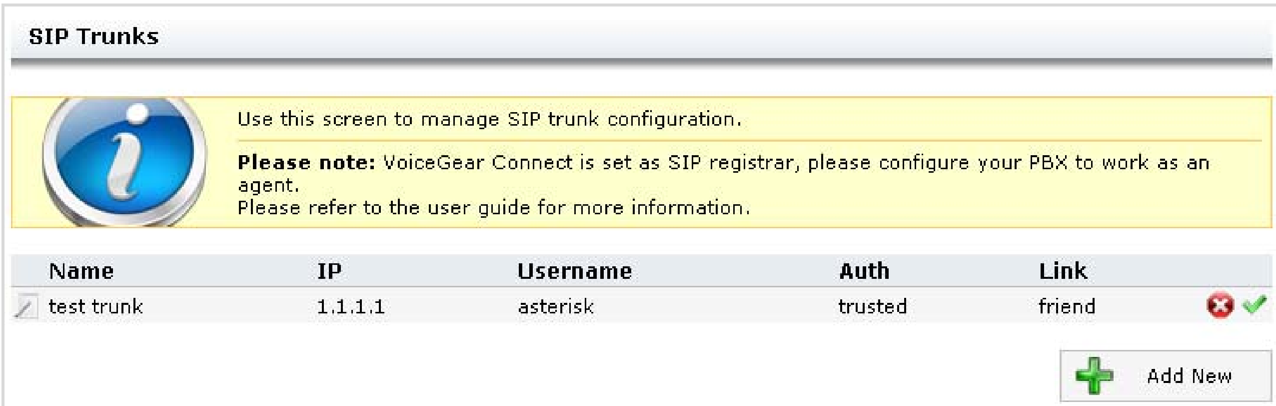


Image 43: List of configured SIP trunks

To create a new SIP trunk, push the “Add New” button to show the new trunk configuration screen. Every SIP trunk configuration screen also features sections for basic and advanced settings.

To edit an existing SIP trunk, locate the desired trunk in the list of available trunks and click on the name of the trunk.

7.1.3.1 Basic SIP Trunk Settings

Basic SIP trunk settings allow you to define specific PBX system connection options as well as authentication information for making calls.

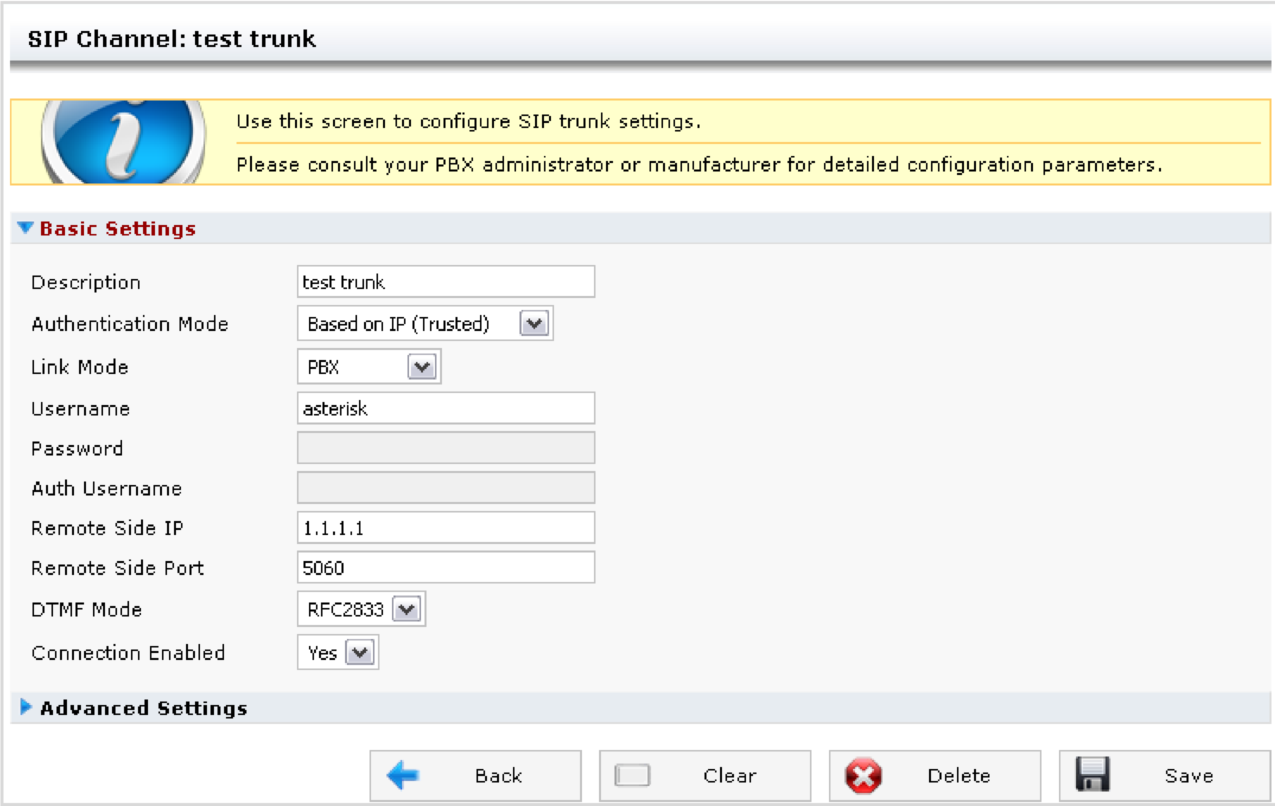


Image 44: Basic SIP trunk configuration settings

Detailed list of basic SIP trunk settings is shown below.

Setting	Description
Description	A meaningful name to be used to distinguish between multiple defined SIP trunks.
Authentication Mode	This setting allows configuring a security level to use between VoiceGear Connect gateway and the PBX system. When set to “trusted”, VoiceGear Connect assumes no security is enabled and uses the specified remote side IP setting to access the PBX system. When set to “authentication”, VoiceGear Connect will also supply a defined username and password for authentication prior to any call. In this case, authentication credentials must much those set in the PBX configuration.
Link Mode	This setting tells the gateway whether it is connecting to a PBX system or a single SIP client like a soft phone.
Username	Username used by the system in case authentication mode is not set to “trusted”.
Password	Password used by the system in case authentication mode is not set to “trusted”.
Auth Username	SIP auth username used by the system in case authentication mode is not set to “trusted”.

Remote Side IP	An IP or a fully qualified domain name of the PBX system required for trusted authentication.
Remote Side Port	A network port (UDP) the PBX system uses to listen for incoming calls.
DTMF Mode	This setting specifies the method of passing DTMF tones (touch tone key presses) from the PBX to the VoiceGear Connect gateway. Available modes include INFO and RFC2833 and must match DTMF mode configured on the remote PBX side.
Connection Enabled	Specifies if the trunk is active.

7.1.3.2 Advanced SIP Trunk Settings

Advanced SIP trunk settings are used to fine tune PBX connectivity and in most cases do not need to be changed.

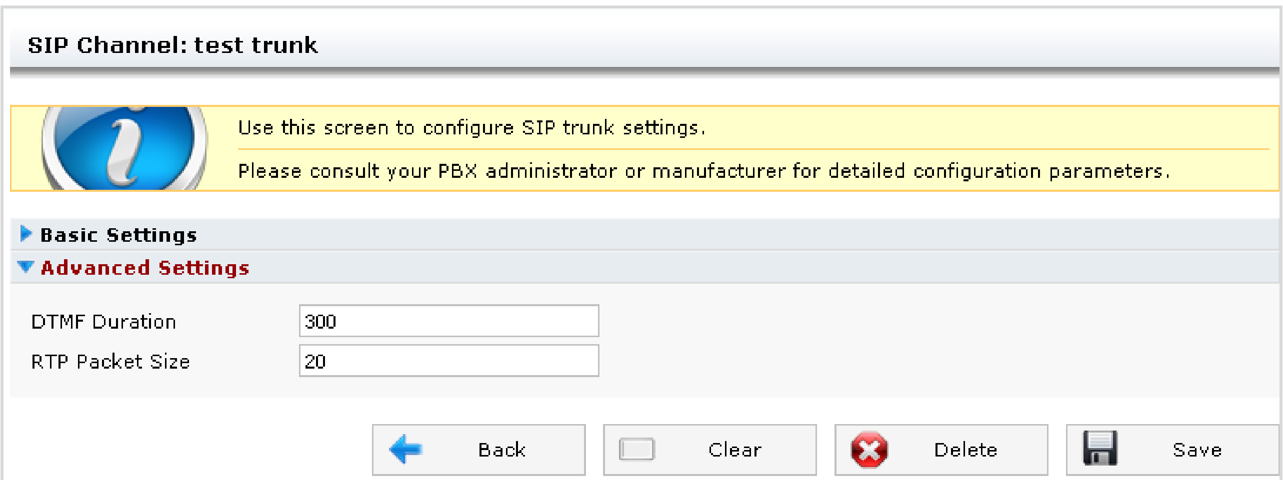


Image 45: Advanced SIP trunk configuration settings

7.2 Analog Channel Configuration

Depending on the VoiceGear Connect model, your gateway can come with pre-installed analog connectivity option. To make sure this is the case, please navigate to “Channels->Analog/Digital” screen via the main menu and make sure the list of installed hardware is showing at least one “Analogic” type device. With this connectivity option, it is necessary to connect analog RJ-11 cables (supplied) from the back of the gateway into your PBX system.

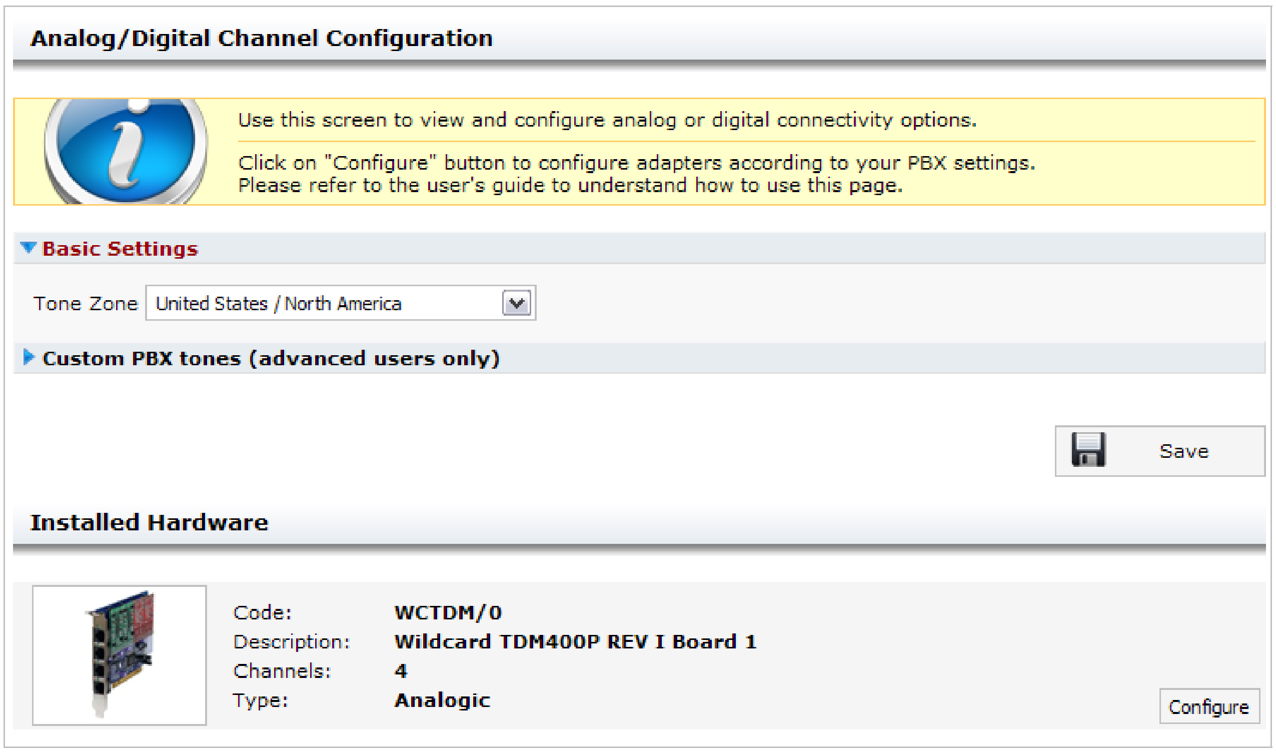


Image 46: Analog/digital channel configuration

Once analog hardware is detected, an overview is shown on the “Channels->Analog/Digital” page accessible via the main menu. The overview shows unique hardware code, description, number of supported channels as well as hardware type. The analog hardware comes pre-configured out of the box on all VoiceGear Connect gateways; however, it is possible to change the default configuration by clicking the “Configure” button.

7.2.1 Basic Tone Settings

VoiceGear Connect is a universal gateway designed to be compatible with various PBX systems and dial tone zone standards used around the world. In order to make sure that VoiceGear Connect is able to correctly identify your local dial tone, it’s necessary to set the tone zone setting. You can select the appropriate tone zone setting from the Tone Zone drop-down box.

Most of the world tone zones have a predefined setting in this list. However, if your tone zone is not listed here, or you require custom settings please select “Custom tones (advanced users)” entry from the list. This choice will enable you to define custom dial tone settings in the “Custom PBX tones” section.

7.2.2 Custom PBX Tones

Custom PBX tones should be used only by advanced users who have a good understanding of tone definition syntax as well as how tones are generated and used by various phone systems. If you want to learn more about this feature, please review this online reference:

http://www.voip-info.org/tiki-index.php?page=Asterisk+config+indications.conf

Once you understand the syntax used for defining tones, you can create your own custom tones using the settings shown below. Click “Save” button to apply your new tone settings, then click “Apply Now” button in the notification area when you are asked to apply the new modifications.

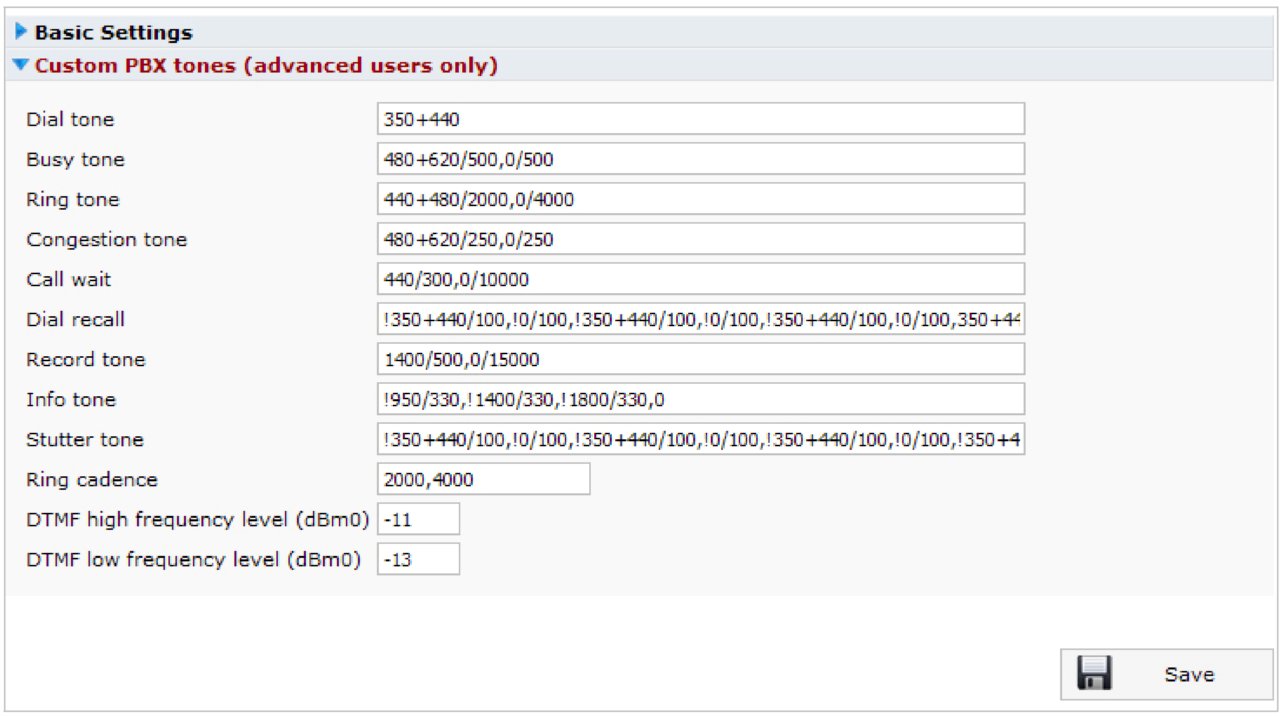


Image 47: Custom PBX tones configuration

Now you can continue configuring your analog hardware. Clicking on the “Configure” button brings up a configuration screen for the specific hardware. Analog hardware settings are divided into basic and advanced categories.

7.2.3 Basic Configuration

Basic analog hardware configuration allows users to tweak analog hardware behavior and slightly adjust voice quality to better fit the PBX system configuration.

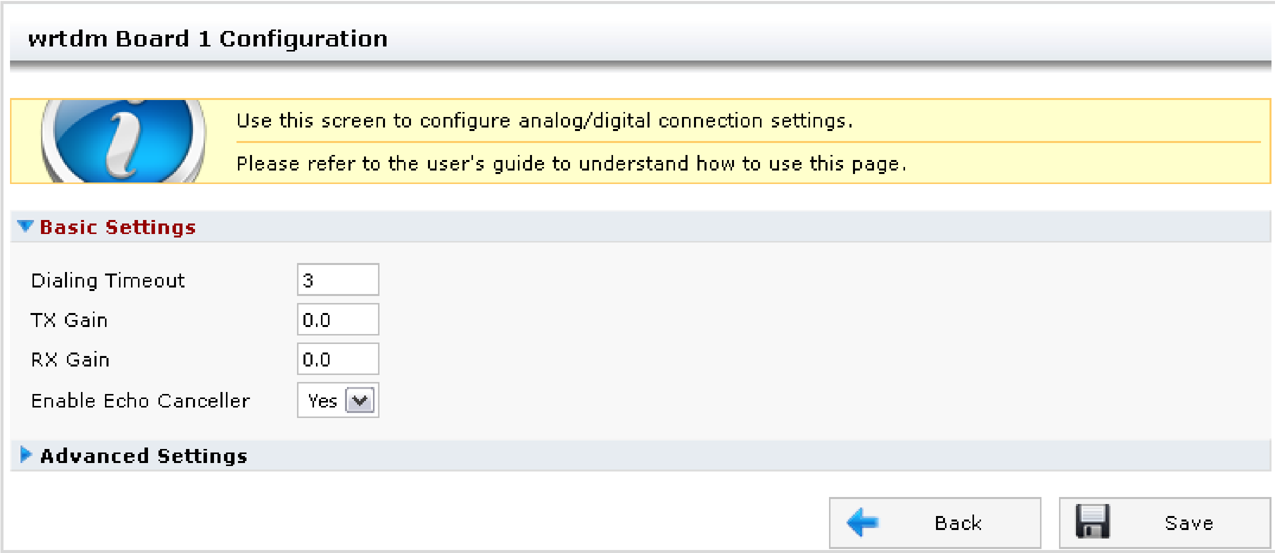


Image 48: Basic analog hardware settings

Details for each setting are shown in the table below.

Setting	Description
Dialing Timeout	This setting configures the maximum amount of time in seconds between touch tone key presses the system waits before assuming the phone number or Skype shortcut entry is complete. This setting can be safely adjusted based on user preference.
Tx gain	Transmission gain setting to adjust the transmit voice volume. Each increment can increase or decrease voice level by 3db.
Rx gain	Receive end gain setting to adjust the receive voice volume. Each increment can increase or decrease voice level by 3db. In case remote party voice echoes back through the line, this setting can be adjusted to a negative value between -3 and -5 to address the issue.
Enable Echo Canceller	This setting enables built-in software echo-cancellation algorithms used to improve voice quality. If lines between PBX and VoiceGear Connect have excessive echo or static, this setting should be enabled.

7.2.4 Advanced Configuration

Advanced analog hardware configuration allows users to profoundly change the behavior of the installed analog hardware. It is not recommended to adjust these settings without having an advanced understanding of the system.

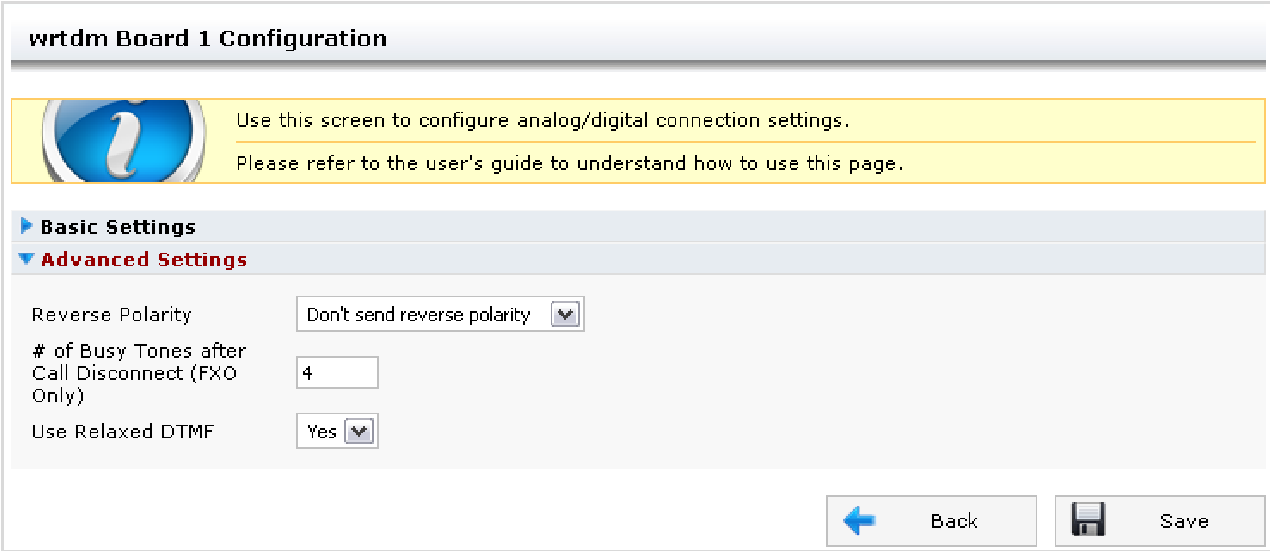


Image 49: Advanced analog hardware settings

Details for each setting are shown in the table below.

Setting	Description
Reverse Polarity	This setting allows configuring system behavior for sending reverse polarity to the PBX. It is possible not to send reverse polarity or to send reverse polarity during a call or after a call. Sending reverse polarity after a call can improve the number of seconds it takes to free up a line on certain PBX systems.
Number of Busy Tones after Call Disconnect	In extension-side analog configurations (see section 7.4), this settings configures the number of busy tones to wait before assuming the line is disconnected.
Use Relaxed DTMF	This setting enables or disables the relaxed DTMF option which changes the way touch tone key presses are interpreted by the system. Should you experience any problems with DTMF entry, this setting can be adjusted to improve DTMF recognition. As a rule, VoiceGear Connect systems with advanced hardware echo cancellation require this setting to be turned on, while those systems without advanced hardware echo cancellation require this setting to be turned off.

7.3 Digital Channel Configuration

Depending on the VoiceGear Connect model, your gateway can come with pre-installed digital connectivity option. To make sure this is the case, please navigate to “Channels->Analog/Digital” screen via the main menu and make sure the list of installed hardware is showing at least one “E1” or “T1” type device. With this connectivity option, it is necessary to connect analog RJ-45 cable(s) (supplied) from the back of the gateway into your PBX system.

Once digital hardware is detected, an overview is shown on the “Channels->Analog/Digital” page accessible via the main menu. The overview shows unique hardware code, description, number of supported channels as well as hardware type. The digital hardware comes pre-configured out of the box on all VoiceGear Connect gateways; however, it is possible to change the default configuration by clicking on the “Configure” button.

Clicking on the “Configure” button brings up a configuration screen for the specific hardware. For digital hardware, settings are divided into basic and advanced categories.

7.3.1 Basic Configuration

Basic digital hardware configuration allows users to tweak digital hardware behavior and slightly adjust voice quality to better fit the PBX system configuration.

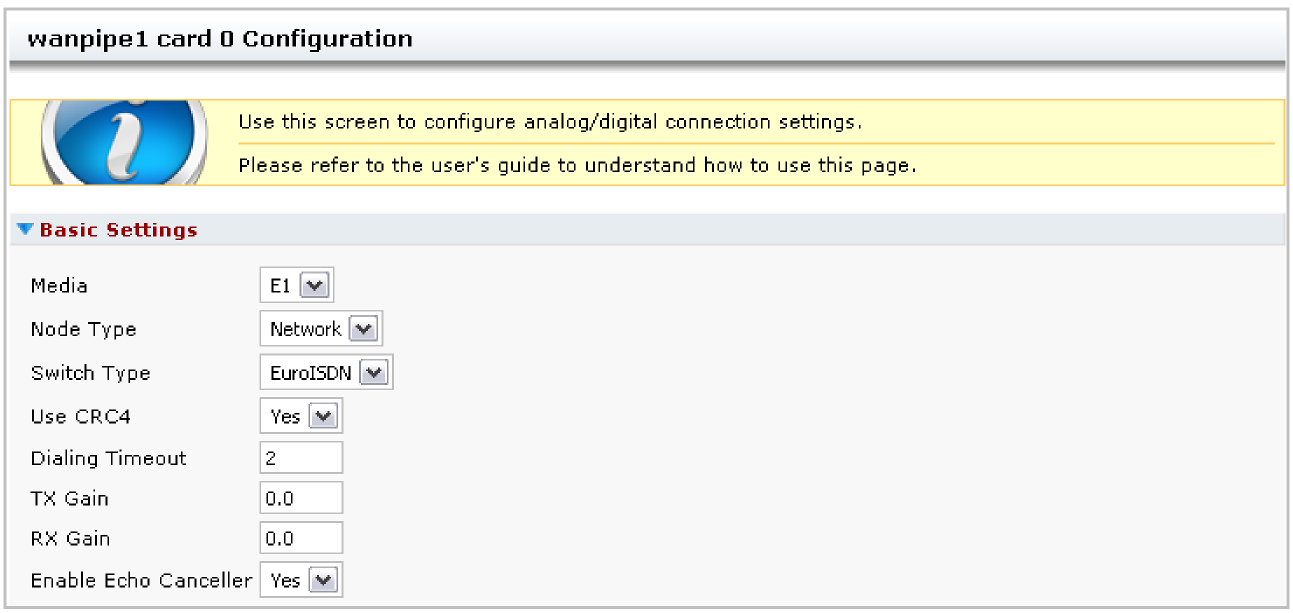


Image 50: Basic digital hardware settings

Details for each setting are shown in the table below.

Setting	Description
Media	This setting configures the connection media which can be set to either “E1” or “T1”. As a rule, T1 connectivity is used in North America and Japan while E1 is used elsewhere in the

	world including Europe and Asia.
Node Type	This setting allows configuring the way VoiceGear Connect treats the PBX connection. Available choices include “Network” and “CPE”.
Switch Type	This setting allows configuring the protocol VoiceGear Connect uses to communicate to the PBX system. As a rule, the value used in North America is “National”, however the following options are also available and can be configured to match your PBX settings: “EuroISDN”, “DMS100”, “Lucent5e”, “Att4ees”, “NI1”, “gr303eoc”, “gr303tmc”, and “Q.SIG”.
Use CRC4	This setting controls turns cyclic redundancy check algorithms used to check for errors in transmitting data between VoiceGear Connect and your PBX on or off.
Dialing Timeout	This setting configures the maximum amount of time in seconds between touch tone key presses the system waits before assuming the phone number or Skype shortcut entry is complete. This setting can be safely adjusted based on user preference.
Tx gain	Transmission gain setting to adjust the transmit voice volume. Each increment can increase or decrease voice level by 3db.
Rx gain	Receive end gain setting to adjust the receive voice volume. Each increment can increase or decrease voice level by 3db. In case remote party voice echoes back through the line, this setting can be adjusted to a negative value between -3 and -5 to address the issue.
Enable Echo Canceller	This setting enables built-in software echo-cancellation algorithms used to improve voice quality. If lines between PBX and VoiceGear Connect have excessive echo or static, this setting should be enabled.

7.3.2 Advanced Configuration

Advanced digital hardware configuration allows users to profoundly change the behavior of the installed analog hardware. It is not recommended to adjust these settings without advanced understanding of the system.

Image 51: Advanced digital hardware settings

Details for each setting are shown in the table below.

Setting	Description
Codec Layer	This setting allows configuring the codec VoiceGear Connect uses to communicate with the PBX system.
Overlap Dial	This setting allows configuring the inter-digit timer. Setting it to “Yes” enables VoiceGear Connect to wait up to 2 seconds between every digit entered which allows destination numbers to be specified partially to the PBX in order to complete the first stage of the call routing path.
Sync Source Priority	This setting allows controlling synchronization source priority.
LBO	This setting allows configuring cable line build out for the digital connection. It is adjusted based on the cable type and length and in most cases should be set to 0 unless a very long cable is used for PBX integration.
Use Yellow Alarm	This setting configures whether VoiceGear Connect can set yellow alarm on the PRI line.
Use Relaxed DTMF	This setting enables or disables the relaxed DTMF option which changes the way touch tone key presses are interpreted by the system. Should you experience any problems with DTMF entry, this setting can be adjusted to improve DTMF recognition. As a rule, VoiceGear Connect systems with advanced hardware echo cancellation require this setting to be turned on, while those systems without advanced hardware echo cancellation require this setting to be turned off.

7.4 Analog / Digital PBX Configuration Overview

Even though specific PBX configuration for analog/digital connectivity to the VoiceGear Connect gateway varies heavily on the type of your PBX, there are two main connection approaches for gateway integration: extension side or trunk side.

7.4.1 Extension Side Connectivity

With extension side PBX integration, your gateway is connected to the PBX as if it was a regular phone (or set of phones). This is achieved by connecting analog FXO ports on the gateway to analog FXS ports on the PBX. With this setup, each of your Skype lines is essentially a PBX extension and has an extension number. In order to be able to detect which line is free and offer it to potential users who need to dial out with Skype, you need to setup a hunt group to go through all of the extensions connected to Skype lines and find the one that is not occupied. Most PBX systems offer the extension hunting functionality, however this is an extra configuration step.

7.4.2 Trunk Side Connectivity

With trunk side PBX integration, your gateway is connected to the PBX via analog or digital lines which are treated as external phone lines. The only setting required on the PBX is a prefix assignment or a dial plan to be able to connect to Skype lines by, for example, dialing 8 before the actual number or Skype shortcut. Any incoming calls from the gateway can easily be routed to your existing auto attendant or operator. In order to utilize this connectivity, your gateway must come equipped with analog FXS or digital lines.

8. Configuring Call Routes

Call routing is a comprehensive system for routing inbound and outbound calls. You can configure routes in many different ways to suite your business needs.

Please Note:
Default configuration on VoiceGear Connect is set to allow all inbound and outbound calls without restrictions or forwarding. Please make sure you understand how call routing is handled before making changes. Improper routing may result in calls being sent to invalid destination or handled incorrectly.

8.1 Inbound Call Routing

In order to manage your inbound call routes, please navigate to Call Routes -> Inbound Calls menu. You will see the screen shown below. Initially, the route list has one default route called "Incoming calls any/any". This default route allows the system to accept an incoming call to any running Skype account and direct the call to any available line connected to the PBX.

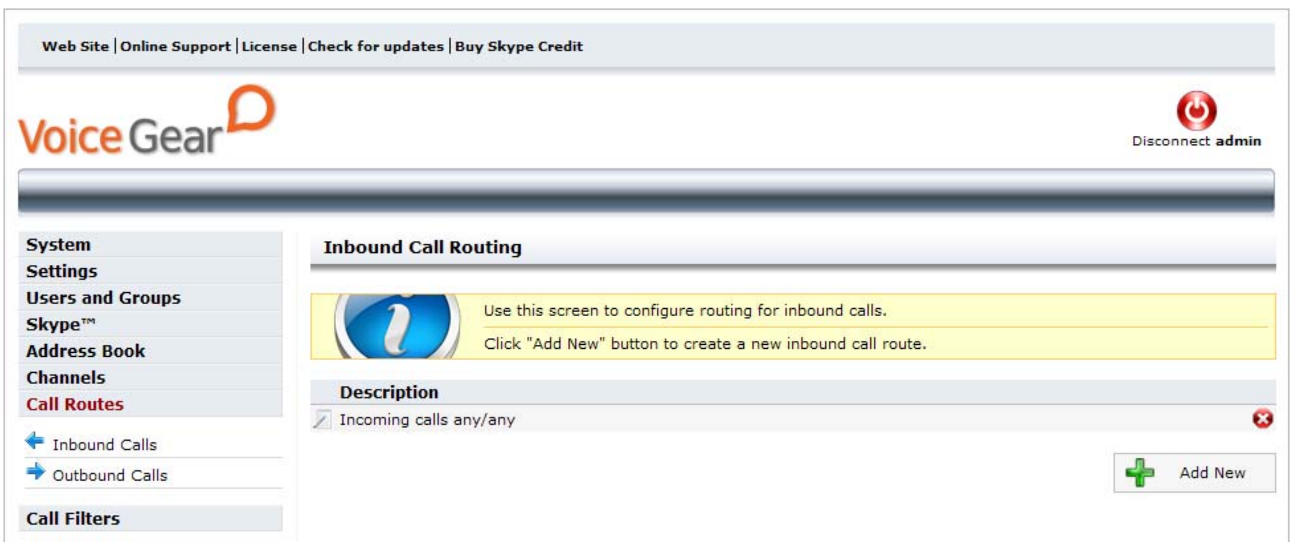


Image 52: Inbound call routing screen

If you are not planning to modify this default call routing behavior, you can skip this section completely. However, if you have special business requirements for handling incoming calls, you can use the Inbound Call Routing feature to achieve the following behavior:

- Incoming calls can rollover between Skype accounts defined in the same route. This feature is activated when the original Skype account for which there is an incoming call is busy (no timeout value is required, the call will rollover immediately). This allows you to provide a single Skype name to handle all incoming calls. If the first Skype account in the route is busy, the call will rollover to the next available Skype account until all accounts are occupied.

- "If the Route is Full Forward Calls to" option can be used to forward incoming calls to a Skype account defined in a different route or a SkypeOut phone number. This feature is activated when all Skype accounts defined in the route are busy.

- Calls to Skype accounts defined within one route will be directed to the first destination number in the list. If there are multiple destination numbers defined in that route and the first destination is busy, the call will be automatically directed to the second destination defined in this route, and so on.

- Incoming calls to any Skype account defined in one route will not rollover to a different Skype account defined in another route. For example, you can configure separate incoming call routes for French and English support teams and have this feature ensure that English calls don't rollover to French agents and vice versa.

To create a custom call route for incoming calls, click the "Add New" button on the bottom right-hand side. You will see the screen shown below.

Routes for inbound calls allow you to manage inbound calls to one or more of your Skype accounts and route them to particular trunks. The screen below consists of two sections: basic settings and route settings.

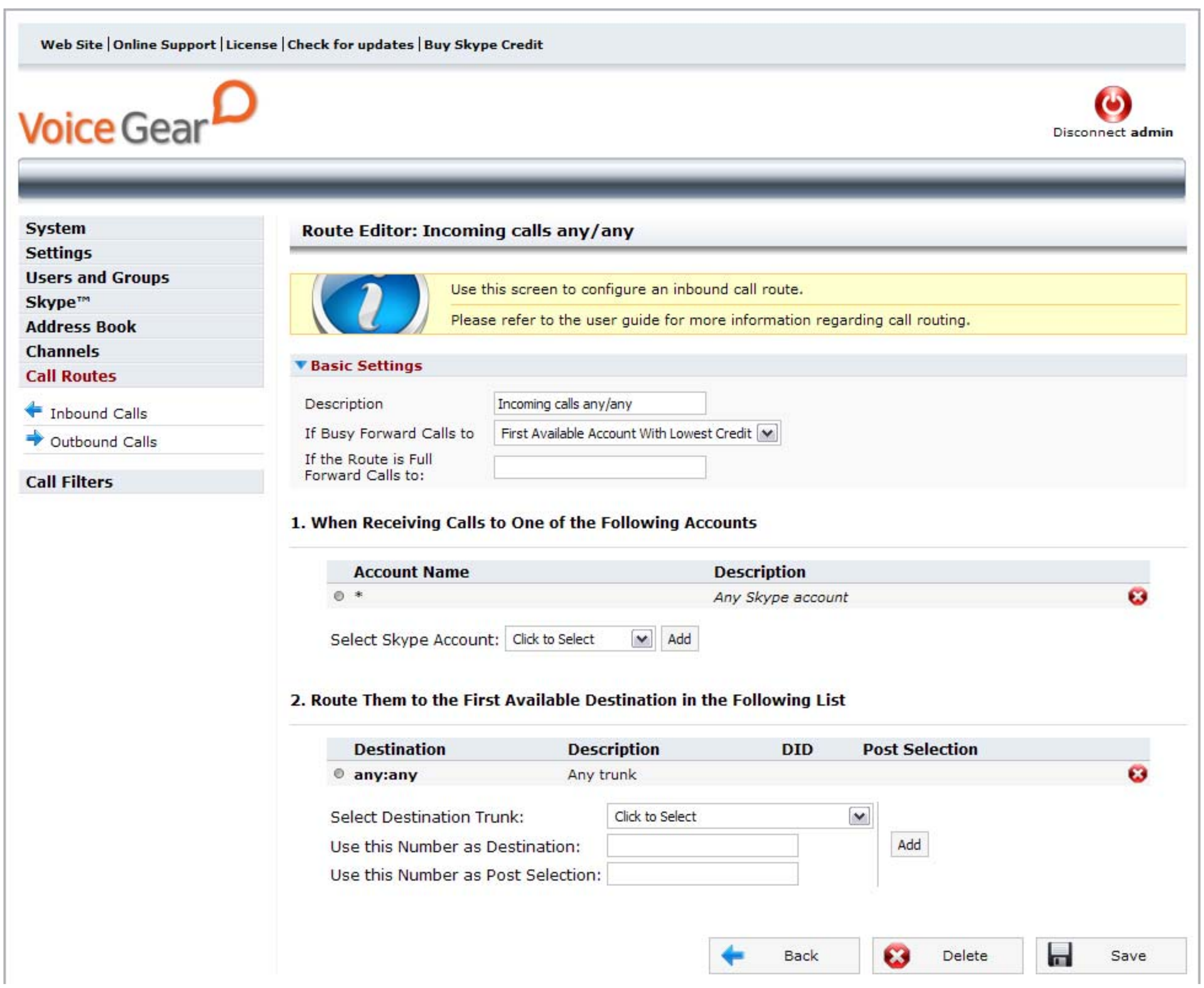


Image 53: New inbound call route screen

8.1.1 Basic Settings

Basic settings allow you to identify the call route by giving it a full description.

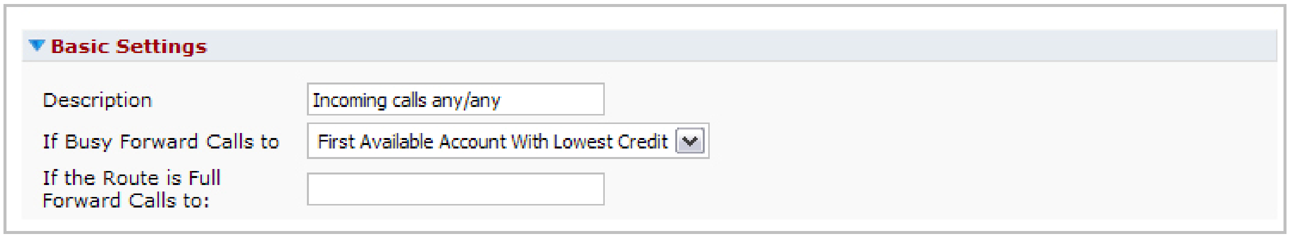


Image 54: Inbound call route basic settings

Setting	Description
Description	Description setting is used to identify the call route by giving it a descriptive name.
If Busy Forward Call to	If Busy Forward Call to setting is used to define forwarding rule which is used in case when there is an incoming call to a Skype account which is currently busy. You can choose between “First Available Account” in the list of accounts managed by the current route or “First Available Account With Lowest Credit” options. The last option allows you to keep accounts with higher credit free to place outgoing calls by occupying those with low credit for incoming calls first.
If the Route is Full Forward Calls to	If the Route is Full Forward Calls to setting is used when all Skype accounts defined in the route are busy. You can specify a Skype account name or other Skype destination (including SkypeOut) to use as default fallback in case the route is full. This function is useful when using multiple VoiceGear Connect gateways in a stacking mode.

8.1.2 Route Settings

There are two parts to defining a new route. The first part called “When Receiving Calls to One of the Following Accounts” allows you to select the Skype account (registered in VoiceGear Connect under Skype -> Accounts) which will be managed by this route. Adding a Skype account to this list means that this route will be applied each time a call is placed to the selected account.

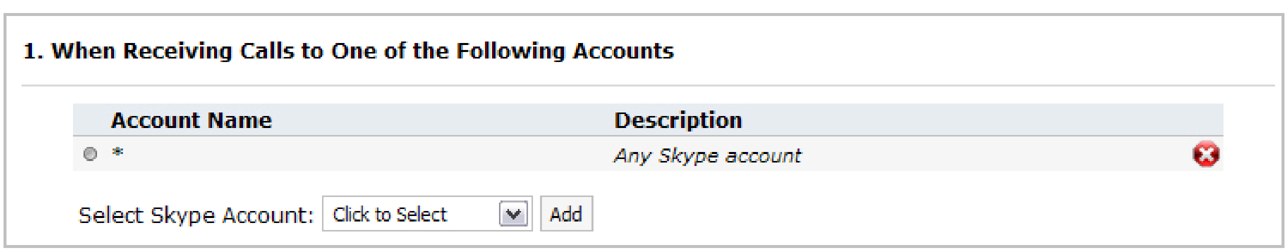


Image 55: Define a list of Skype accounts to manage in current route

Setting	Description
Select Skype Account	Select Skype Account setting allows you to specify a list of Skype accounts to be managed by this call route. Whenever there is an incoming call to one of the Skype accounts in this list, it will be directed to the destination trunk as indicated in the second part of this route. If the original Skype account is busy the incoming call will automatically rollover to the next available account in this list.

Once you have selected the desired Skype account click “Add” button to add this Skype account to the list.

Please Note:
To manage ALL Skype accounts you can select “Any Skype account” option in the list.

The second part called “Route Them to the First Available Destination in the Following List” allows you to define a list of PBX extensions from any of the trunks connected to VoiceGear where the call should be routed. The order of destinations defined in this list is important because call forwarding will be attempted in the specified order until a call is answered by one of the destination extensions.

Image 56: List of routing destinations: trunk, extension and post selection parameters

Please Note:
To manage ALL destinations trunks you can select “Any trunk” option in the list.

Setting	Description
Select Destination Trunk	Destination trunk setting is used to select the destination trunk where an incoming call should be directed. If you are using an analog connection to the PBX, this list will have all analog channels (1 through n) which are connected to PBX lines. If you are using SIP PBX then you will have a list of all predefined SIP trunk (Channels -> SIP -> SIP Trunks menu) connected to the PBX. This gives you a fine control over the destination of each call.
Use this Number as Destination	Destination number setting is used to forward incoming calls to an extension using direct inward dialing method. Destination number can be used if your PBX supports “direct

	inward dialing" and the extension has a DID number assigned. The Destination Number must be specified for all FXO channels on VoiceGear Connect plugged in as extensions on the PBX.
Use this Number as Post Selection	Post selection setting defines a number which VoiceGear Connect will pass to the PBX as DTMF tones sequence. Post selection setting can be used to forward a call to a specific extension if your PBX doesn't support "direct inward dialing" but supports DISA or IVR options.

Every destination route is defined as a combination of trunk name selected from a drop-down list of all configured channels and a destination or post selection number. Once you have configured all parameters in this section click on the "Add" button to add the destination to the list.

Order of destinations defined in this list is important because incoming calls will always be directed to the first available destination. If there are multiple destinations defined in this list and the first destination is busy, the call will be automatically directed to the second destination and so on until an available destination is found.

If you are satisfied with your incoming route settings click the "Save" button to add the new route to the Inbound Call Routing table. To remove an existing call route from the Inbound Call Routing table click on the "Delete" action button on the right of each call route entry.

Please Note:
Order of routes defined in the call routing table is essential: routes are processed in the order they appear in the list. You can order existing call routes by dragging and dropping entries up and down in the call routing table.

Section 5 above mentioned that VoiceGear Connect can run Skype accounts in two modes: shared office line or a direct personal line. If you wish to have all calls to a particular Skype account forwarded to a specific office extension (for example, you can use your own Skype account and have all incoming calls to this account forwarded to your office extension) you can create a new Incoming call route with the following settings:

- Enter "Incoming calls to John's extension" in the description field of the basic settings section.
- Select your personal Skype account from a list of all Skype accounts running in VoiceGear Connect (you must have this Skype account registered first using Skype -> Accounts menu) in the first part called "When Receiving Calls to One of the Following Accounts".
- Select a destination trunk (or leave as "Any trunk") and enter your office extension number in the "Use this Number as Destination" field in the second part called "Route Them to the First Available Destination in the Following List"
- Click "Save" button to create the new incoming call route. From now on, any call incoming to your personal Skype account will be directed to your office extension.

If you want to have incoming calls sent directly to your PBX auto-attendant, please leave “Use this Number as Destination” field empty. This configuration will allow VoiceGear Connect to send an incoming call directly to the default route setup on your PBX. From here, the PBX needs to be configured to handle an incoming call from the VoiceGear Connect trunk.

Please Note:

- Some PBX systems require a call pattern to be configured to accept incoming calls from an incoming trunk. It is recommended configuring the call pattern to accept all incoming calls from the VoiceGear Connect trunk.
- Some VoIP PBX systems may require an explicit setting in the “Use this Number as Destination” field in order to route incoming calls to the auto-attendant or IVR. In this case, try putting “s” (stands for PBX service user) or the user name specified in the PBX settings for the trunk connected to the VoiceGear Connect gateway.

8.2 Outbound Call Routing

Routes for outbound calls allow you to manage outbound calls placed from one or more trunks and route them via Skype accounts. In order to manage your outbound call routes, please navigate to Call Routes -> Outbound Calls menu. You will see the screen shown below. Initially, the route list has one default route called “From any to any”. This default route allows the system to accept an outgoing call from any PBX trunk connected to VoiceGear gateway and send it out via any available Skype account.

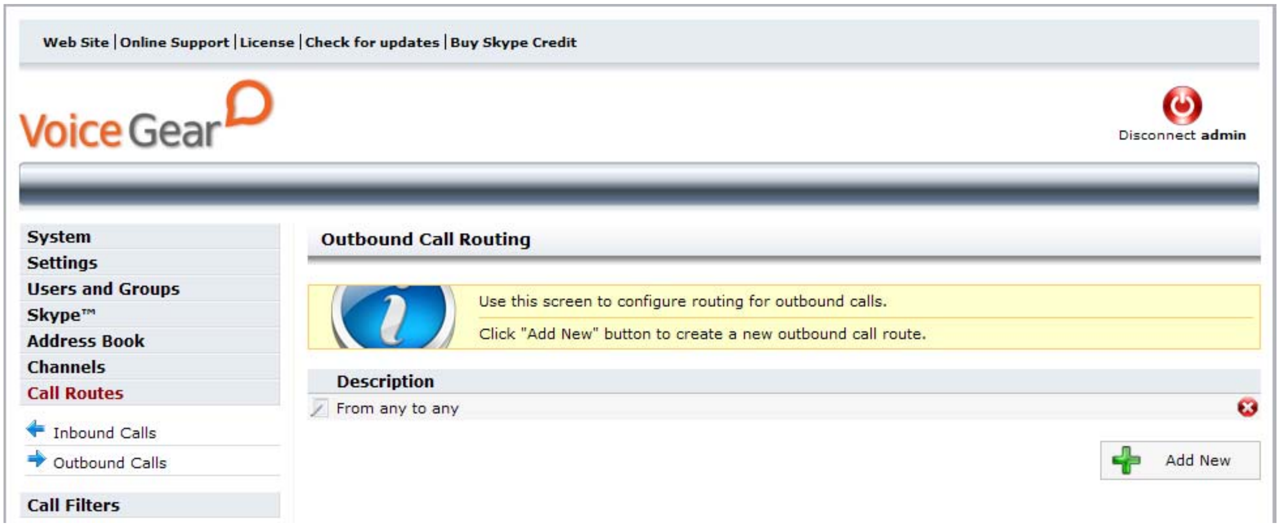


Image 57: Outbound call routing screen

If you are not planning to modify this default call routing behavior, you can skip this section completely. However, if you have special business requirements for managing outgoing calls, you can use the Outbound Call Routing feature to achieve the following behavior:

- Outgoing calls from a particular PBX trunk can be routed via any Skype account defined in the current route. If the first account is busy, VoiceGear Connect will attempt to place the call from the next Skype account defined in this route and so on until it finds an available account. This is useful if you would like to dedicate or limit certain Skype accounts for departmental use. For example, you can have Sales department use Skype accounts mycompany.1 and mycompany.2, and Service department use Skype accounts mycompany.3 and mycompany.4.

- You can dedicate a Skype account for making outgoing calls from a particular office extension. This is useful if you would like to make outgoing calls from your office extension using your own Skype account registered with VoiceGear Connect.

To create a custom call route for outgoing calls, click on the “Add New” button on the bottom right-hand side of the screen. You will see the screen shown below.

Routes for outbound calls allow you to manage outgoing calls from particular PBX trunks and route them out via specified Skype accounts. The screen below consists of two sections: basic settings and route settings.

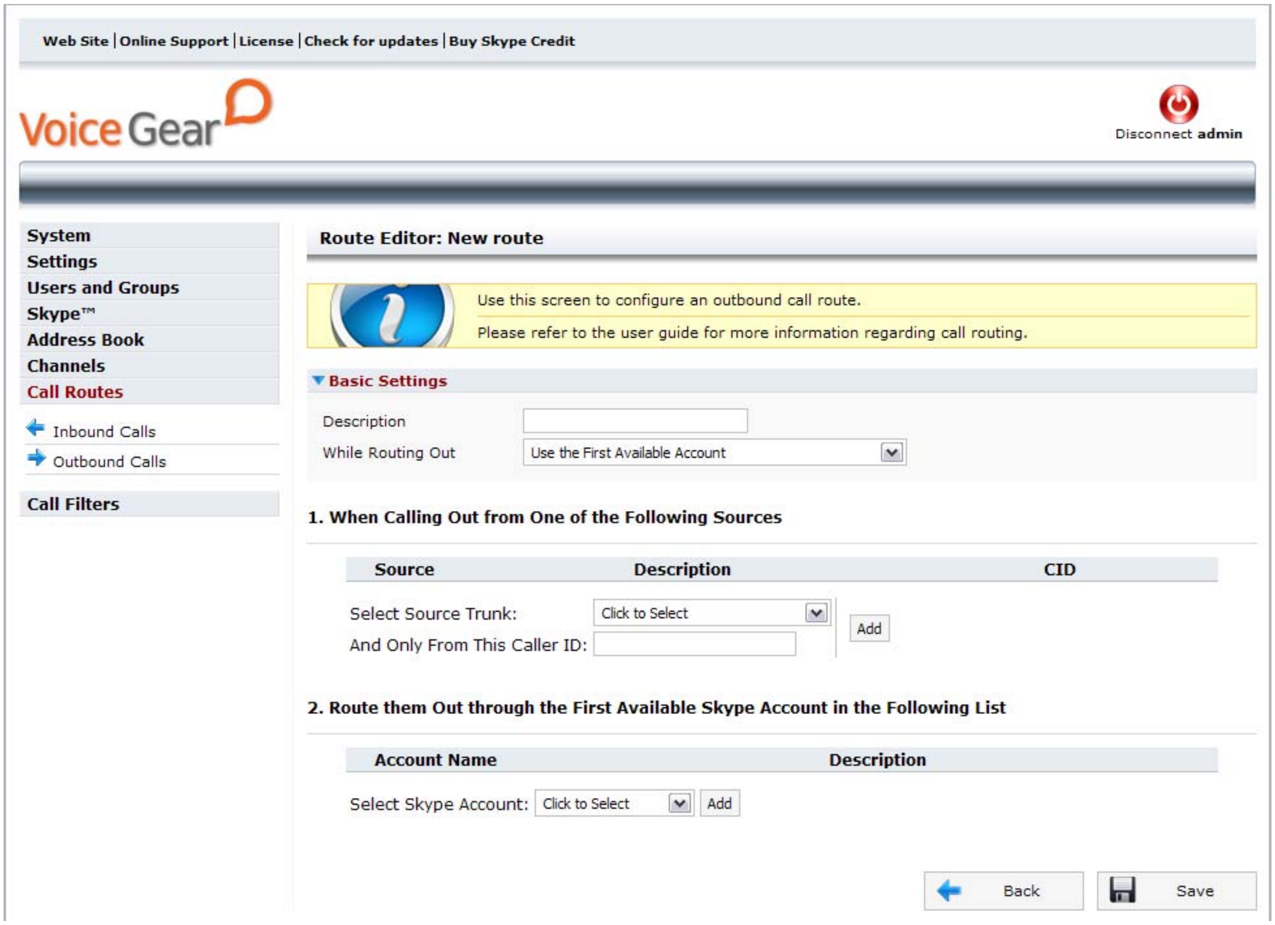


Image 58: New outbound call route screen

8.2.1 Basic Settings

Basic settings allow you to identify the call route by giving it a full description.

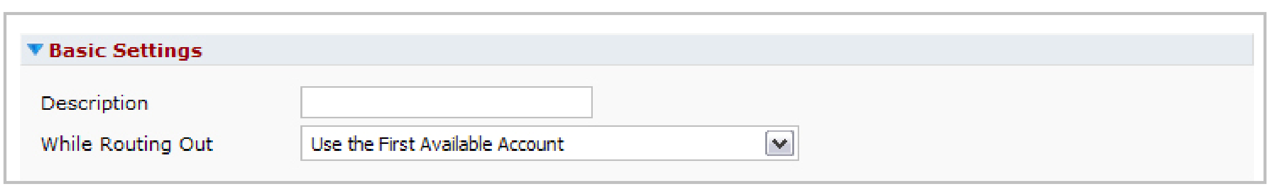


Image 59: Outbound call route basic settings

Setting	Description
Description	Description setting is used to identify the call route by giving it a descriptive name.
While Routing Out	While routing out setting allows you to select between "Use the First Available Account" and "Use the First Available Account With the Highest Credit" options. The last option

	allows you to place outgoing calls from Skype accounts with highest available credit.

8.2.2 Route Settings

There are two parts to defining a new route. The first part called "When Calling Out from One of the Following Sources" allows you to select the source trunks which will be managed by this route. Adding a trunk to this list means that this route will be applied each time a call is placed from the selected PBX trunk.

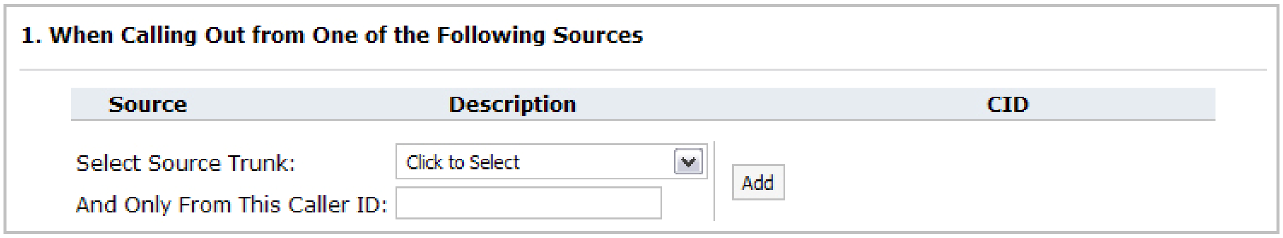


Image 60: Sources of calls to route via Skype accounts

Setting	Description
Select Source Trunk	Source trunk setting allows you to select the source trunk where an outgoing call is originating from. If you are using an analog connection to the PBX, this list will have all analog channels (1 through n) which are connected to PBX lines. If you are using SIP PBX then you will have a list of all predefined SIP trunk (Channels -> SIP -> SIP Trunks menu) connected to the PBX. This gives you a fine control over the origin of every call.
And Only From This Caller ID	Caller ID setting allows you to filter calls from a particular caller ID of the selected trunk. For example, you can specify the extension number of the PBX where the call is originating from.

Once you have selected the desired source trunk and entered an optional Caller ID press the "Add" button to add a new Call Source entry to this list.

Please Note:
To manage ALL source trunks you can select "Any trunk" option from the drop-down box.

The second part called "Route them Out through the First Available Skype Account in the Following List" allows you to define a list of Skype accounts which will be used for placing the call. The order of Skype accounts defined in this list is important because call forwarding will be attempted in the specified order until a call is answered by one of the destination extensions.

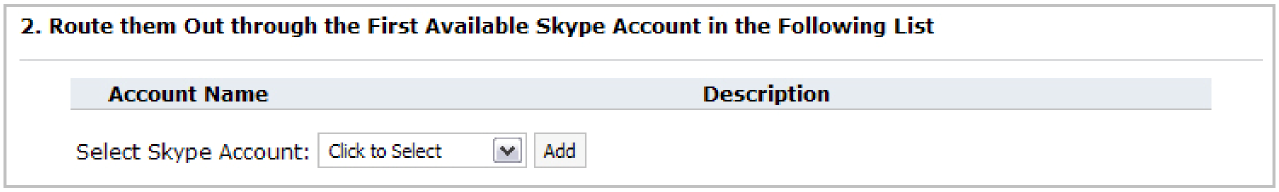


Image 61: Define a list of Skype accounts for routing outgoing calls through

Setting	Description
Select Skype Account	Select Skype Account setting allows you to specify a list of Skype accounts to be used for placing outgoing calls. An outgoing call will be placed using the first available Skype account in this list. If the first account is busy, the system will attempt placing the call from the second Skype account in the list and so on until it finds an available Skype account.

After selecting the desired Skype account from the drop-down box press “Add” button to add it to the list. The order of Skype accounts in this list will determine the order in which VoiceGear Connect will use the accounts for placing outgoing calls. You can sort Skype accounts in this list simply by dragging them up and down.

If you are satisfied with your incoming route settings click the “Save” button to add the new route to the inbound call routing table.

9. Configuring Call Filters

VoiceGear Connect features a comprehensive Call Filtering system which allows the user to create rules for managing restrictions on inbound and outbound calls. You can create multiple cascading rules which will take effect depending on various conditions specified for each rule. Filter rules are executed in the order in which they are specified, thus it is important to ensure correct rule sorting order. You can rearrange the order in which rules are executed by dragging and dropping each rule up and down in the list.

Each filter can contain one or more expressions that are used to match against the source or destination caller ID as a condition. When a condition is matched, the related policy is applied. Conditions can contain letters, numbers and special regular expressions, which allow you to match anything you need to filter your calls.

Regular expressions are python-based. If you want to know more about python regular expressions syntax and usage we recommend you to read official on-line Python Regular Expression reference at http://docs.python.org/lib/module-re.html .

9.1 Inbound Call Filter

In order to manage your inbound call filters, please navigate to Call Filters -> Inbound Calls menu. You should see the screen shown below.

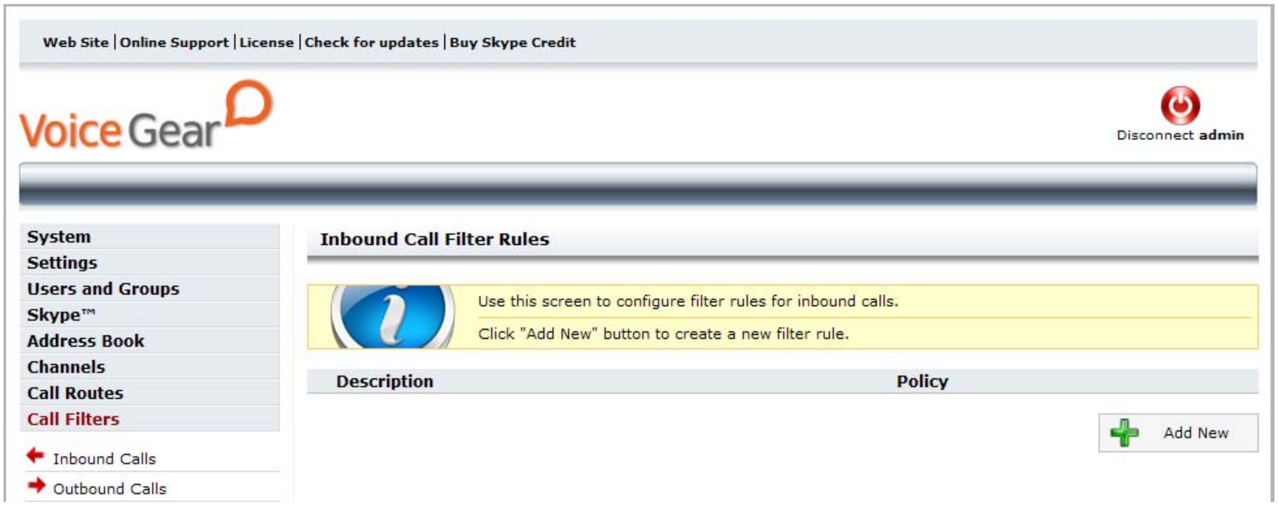


Image 62: Inbound call filters

By default, there are no rules defined in this list. VoiceGear Connect gateway is configured to accept all incoming calls without any restrictions. If you are not planning to modify this default behavior, you can skip this section completely. However, if you have special business requirements for restricting incoming calls, you can create a new Inbound Call Filter by clicking on the “Add New” button in the bottom right corner of the screen.

You will see a new “Filter Rule: New Rule” screen as shown below.

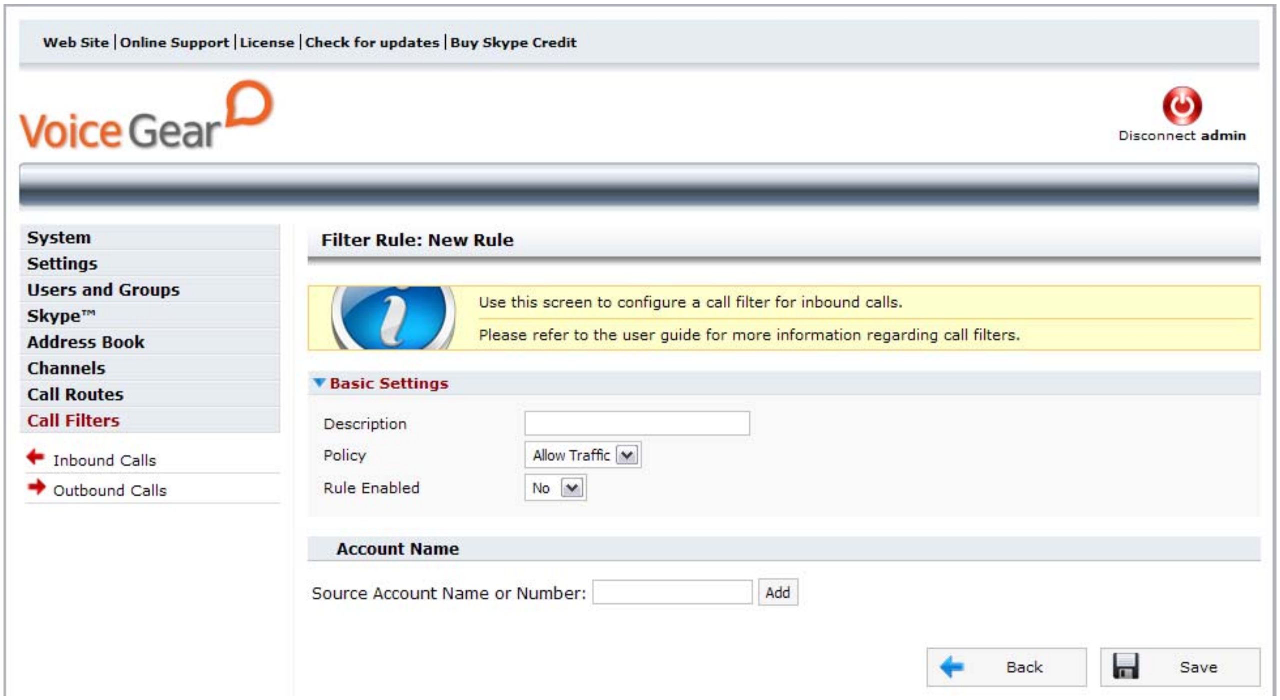


Image 63: New inbound call filter screen

9.1.1 Basic Settings

Basic settings section allows you to define a rule description, set the policy and enable/disable the filter.

Setting	Description
Description	Description setting is used to identify the call filter by giving it a descriptive name.
Policy	Policy setting is used to define call traffic policy. When “Allow Traffic” is selected, all incoming calls matching the specified name or number pattern are allowed. When “Deny Traffic” is selected, all incoming calls matching the specified name or number pattern are denied.
Rule Enabled	Rule Enabled setting allows you to enable/disable the call rule.

9.1.2 Filter Settings

Filter settings section allows you to define the actual expression to be used for filtering incoming calls. To add a new call filter, you must define a new expression in the “Source Account Name or Number” field and click “Add” button.

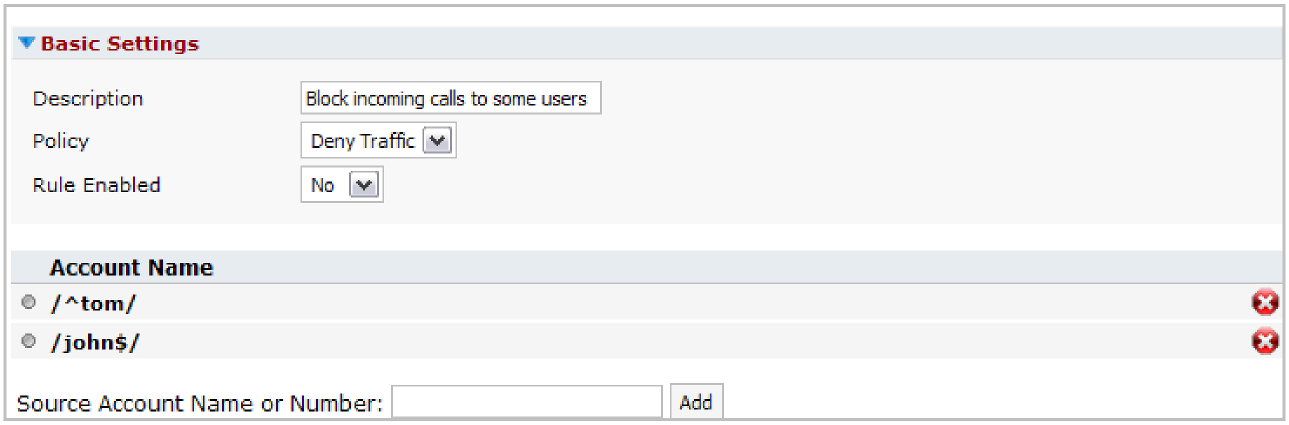


Image 64: Basic settings using a simple regular expression

Filter expressions are based on regular expression grammar which can be used to define any pattern to match against the caller ID of an incoming call. To learn more about regular expression grammar we recommend you to read the official on-line Python Regular Expressions reference at http://docs.python.org/lib/module-re.html .

In the example above we have defined two filter expressions using characters and special pattern symbols as explained in the Python Regular Expressions reference. According to this filter rule all incoming calls from Skype accounts starting with the name “john” or ending with the name “tom” will be denied.

Once you are satisfied with your call filter settings click on the “Save” button to add the new filter rule to the call filter table as shown below. It’s recommended to provide a detailed description for each call filter in order to be able to easily identify the rule in this table.

Image 65: Call filter list

When a new filter rule is added to the Call Filter table, it is disabled by default (you will see the rule entry being grayed out). You can enable the rule by clicking on the green checkmark icon on the right of each entry. You can also disable the rule by clicking the green checkmark icon again. When a rule is disabled you will see the table entry grayed out again. To remove an existing expression click on the “Delete” action button on the right of each filter rule.

Please Note:
Order of call filter rules defined in the call filter table is essential: filters are processed in the order they appear in the list. You can order existing call filters by dragging and dropping entries up and down in the call filter table.

9.2 Outbound Call Filter

In order to manage your outbound call filters, please navigate to Call Filters -> Outbound Calls menu. You will see the screen shown below.

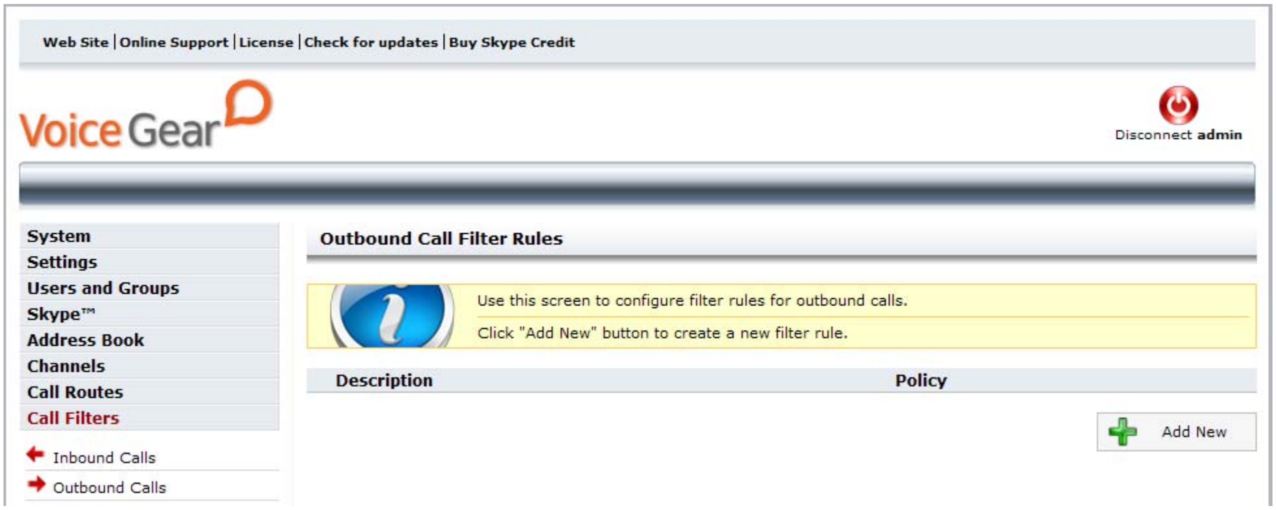


Image 66: Outbound call filters

By default, there are no rules defined in this list. VoiceGear Connect gateway is configured to make outgoing calls without any restrictions. If you are not planning to modify this default behavior, you can skip this section completely. However, if you have special business requirements for restricting outgoing calls, you can create a new Outbound Call Filter by clicking “Add New” button in the bottom right corner of the screen.

You will see a new “Filter Rule: New Rule” screen as shown below.

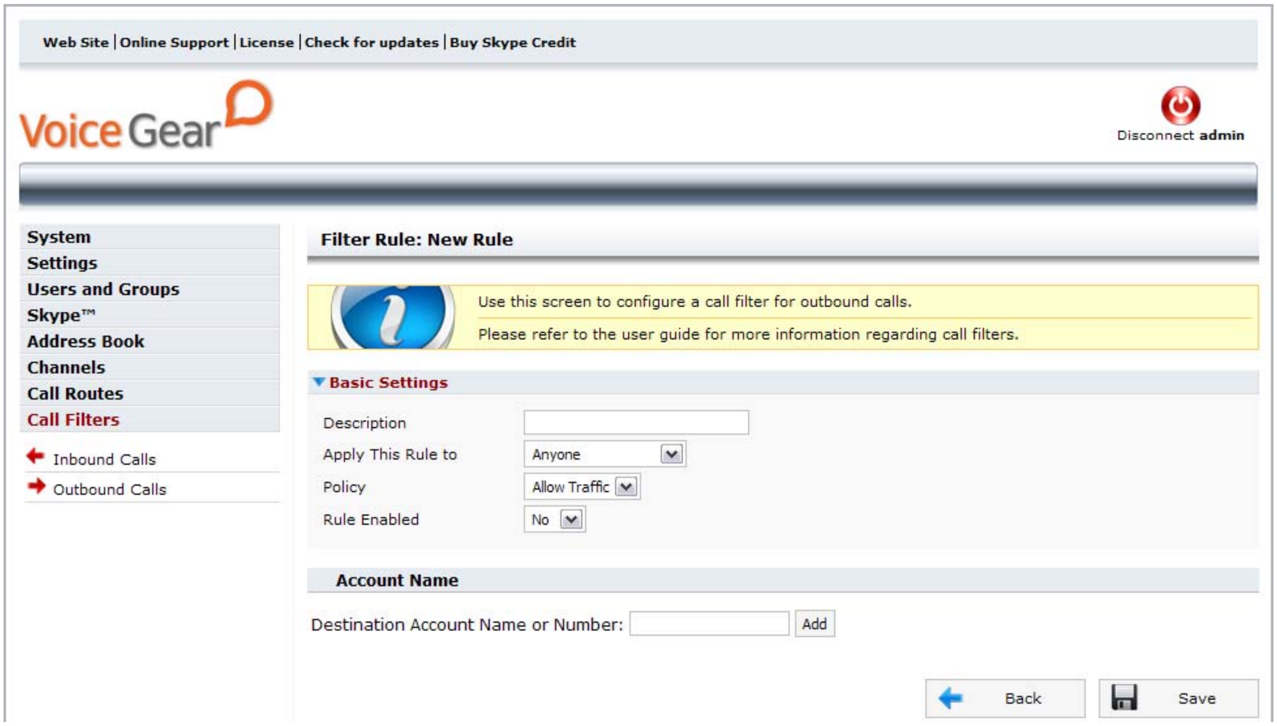


Image 67: New outbound call filter screen

9.2.1 Basic Settings

Both inbound and outbound call filters have the same Basic and Filter settings which are used to define a new call filter rule. Outbound call filter has an additional field in the Basic settings section called "Apply This Rule to" which can be used to define users and groups to apply the filter to.

Setting	Description
Apply This Rule to	Apply This Rule to setting allows you to define users and groups to apply this filter to. You can select one of the available options: • Anyone - Policy applies to every user • Any logged in user - Policy applies only to authenticated users • Group name - Policy applies only to specified group • User name - Policy applies only to specified user

9.2.2 Filter Settings

Similarly to Inbound Calls, Filter settings section allows you to define the actual expression to be used for filtering outgoing calls. To add a new call filter, you must define a new expression in the "Destination Account Name or Number" field and click on the "Add" button.

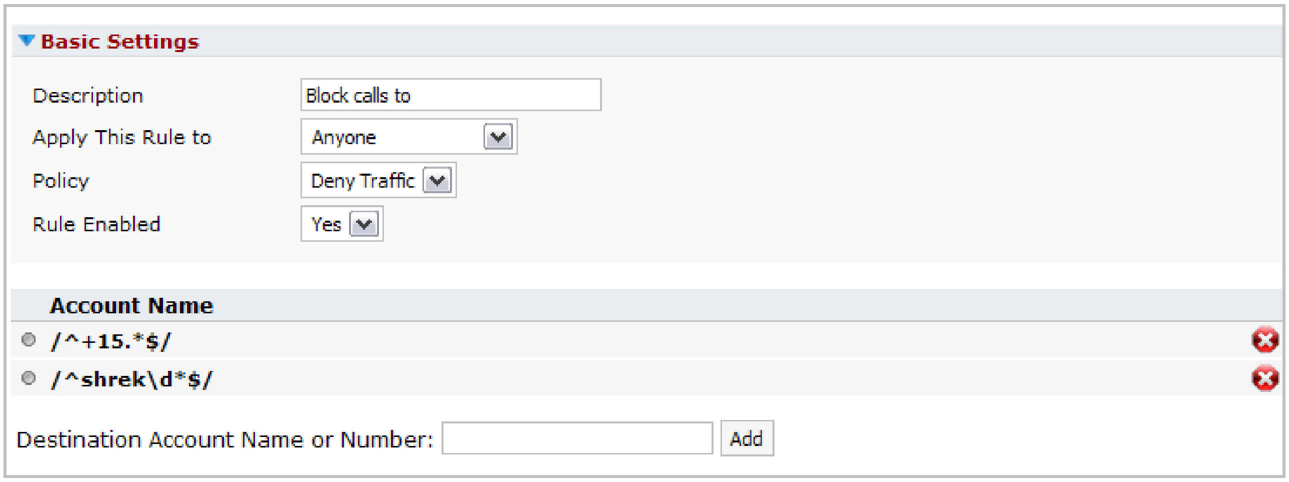


Image 68: Filter settings and regular expression matching example

In the example shown above, we have defined two filter expressions which are used to match against the destination caller ID. According to this filter rule, all calls placed to phone numbers starting with +15 and Skype account names starting with the name "shrek" and followed by zero or more digits [0-9] will be denied. For example, calls to +1 (515) 978-1234 or to Skype accounts with user name "shrek", "shrek1", "shrek2", etc. will be denied for all callers. Alternatively, we could define a specific user or a group of users to restricting outgoing calls using the "Apply This Rule to" field.

10. Configuring Users and Groups

Users and groups configuration allows system administrator to manage VoiceGear Connect users and their permissions. Settings in this section allow to create and manage user roles and permissions as well as to organize users into groups.

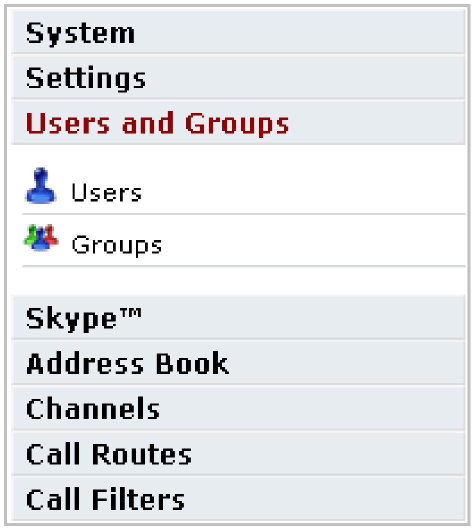


Image 69: Users and groups menu

In basic terms, a user is anyone who wants to use VoiceGear Connect gateway via a phone or optionally access the system via a web interface. A group is an organizational term used to manage distinct user accounts that share public address books and call filter rules.

Please Note:
User and Groups configuration is only available to system administrators.

10.1 Configuring Users

To configure users, please navigate to the user configuration screen via the main menu by selecting "Users and Groups->Users". The system will show all users already registered as well as allow new users to be created.

To create a new user, please push the "Add New" button. To modify an existing user, please click on a desired username. User configuration settings are divided into basic and advanced categories outlined in sub-sections below.

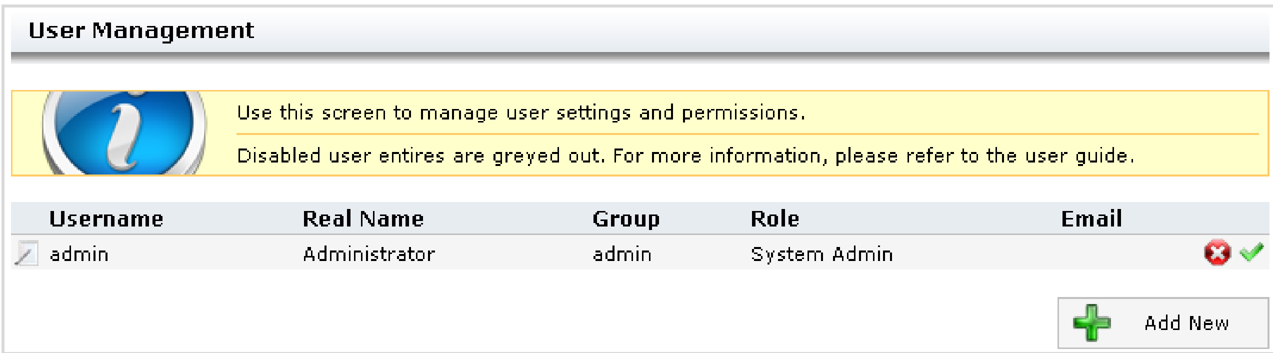


Image 70: User management

10.1.1 Basic Settings

Basic user settings allow system administrator to configure personal user information as well as individual authentication settings.

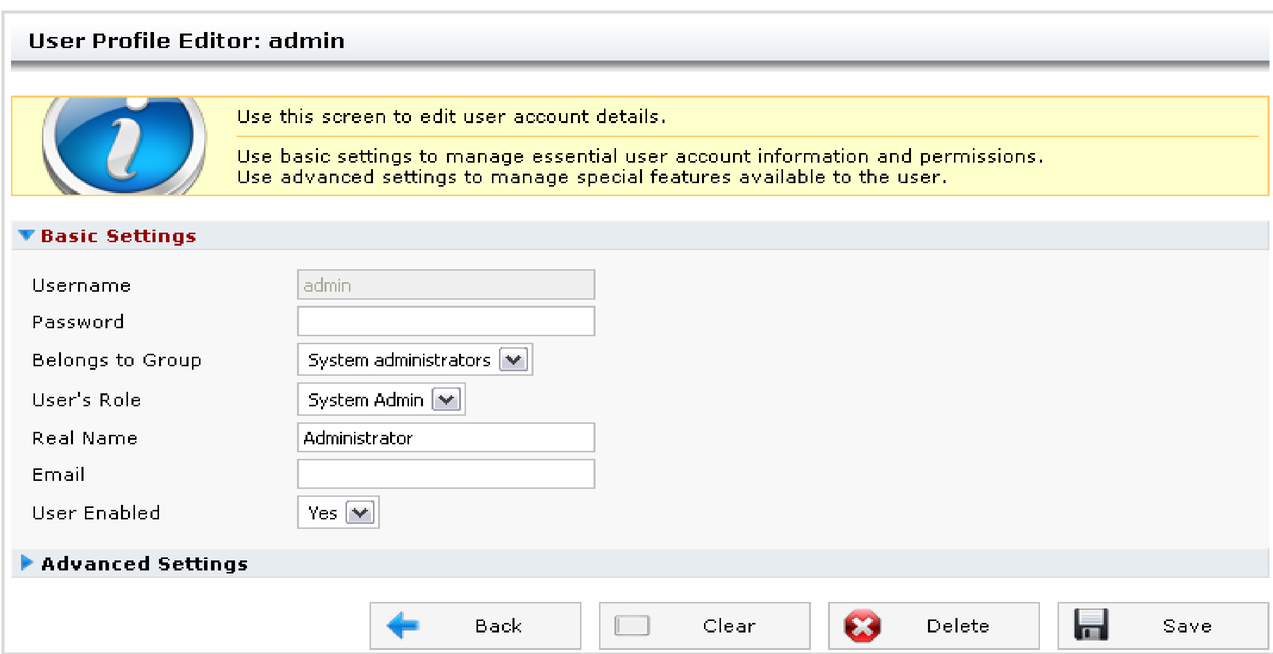


Image 71: User basic settings

Details on individual settings are shown in the table below.

Setting	Description
Username	The username a user has to specify in order to login into the VoiceGear Connect web configuration interface.
Password	The matching password a user has to specify in order to login into the VoiceGear Connect web configuration interface.
Belongs to Group	The name of the group the user belongs to. Please note that group membership does affect user permissions.
User's Role	The administrative role of the user. This setting affects user permissions based on the following three levels: regular user, group admin, and system admin. Regular user can make calls using VoiceGear Connect but is unable to change system settings, public address books or user permission. Group user can make calls using VoiceGear Connect but is unable to change system settings and can only modify public address books belonging to his/her group. System admin can perform all available operations and modify all system settings.
Real Name	User's first and last names.
Email	User's email address.
User Enabled	This setting indicates whether a particular user account is enabled or disabled.

10.1.2 Advanced Settings

Advanced user settings are primarily used to configure phone authentication details for each user.

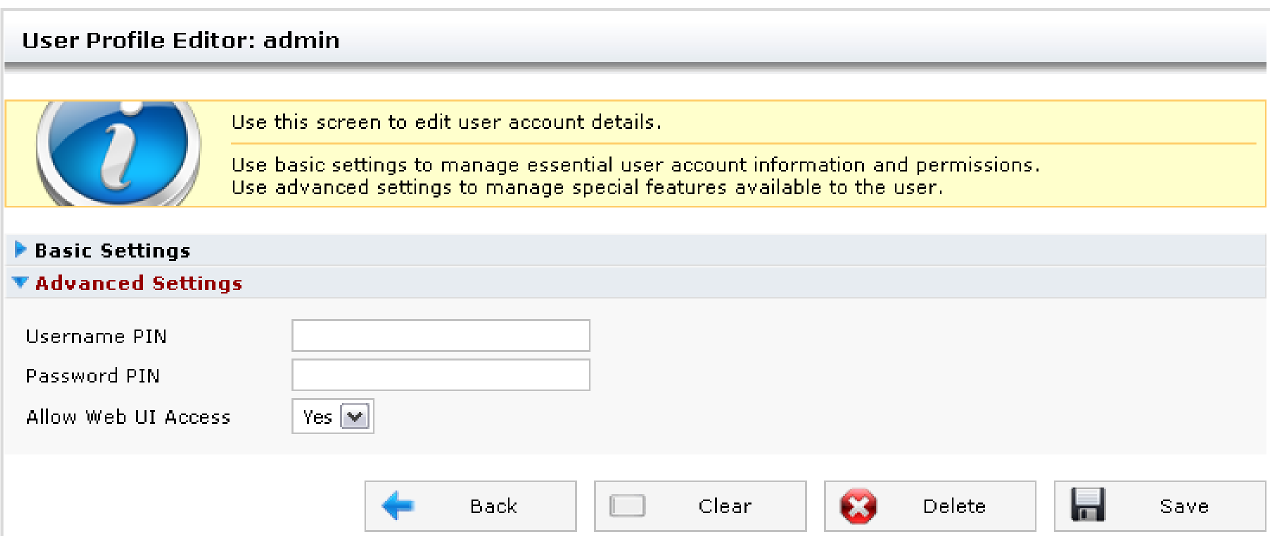


Image 72: User advanced settings

Details on individual settings are shown in the table below.

Setting	Description
Username PIN	The numeric PIN username that can be used during user authentication via a touch tone phone.
Password PIN	The numeric PIN password that can be used during user authentication via a touch tone phone.
Allow Web UI Access	This setting controls user access level to the Web configuration interface. Turning it on allows a particular user to login via the Web configuration interface. Turning it off – disallows it.

10.2 Configuring Groups

To configure groups, please navigate to the group configuration screen via the main menu by selecting "Users and Groups->Groups". The system will show all groups already registered as well as allow new groups to be created.

To create a new group, please push the "Add New" button. To modify an existing group, please click on a desired group name. Group configuration settings include a single basic settings category.

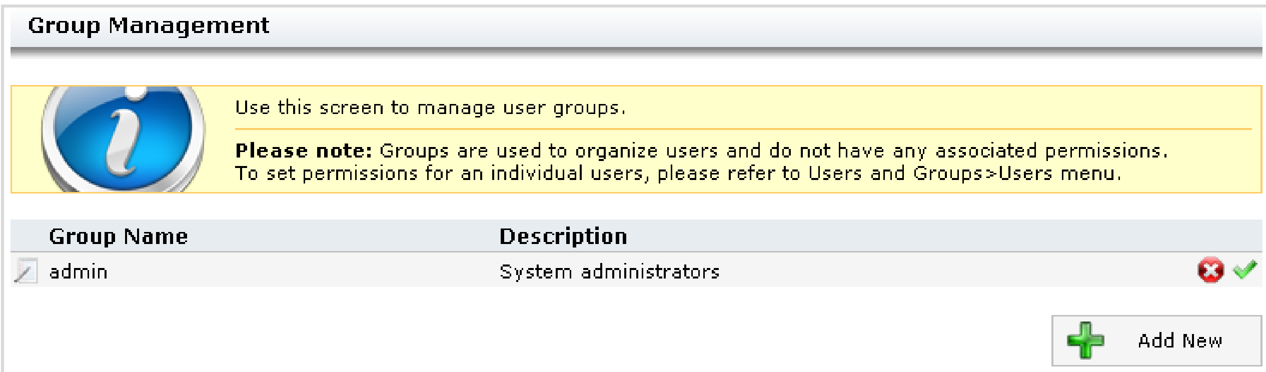


Image 73: Group management

10.2.1 Basic Settings

Basic group settings allow system administrator to configure personal group definitions to be able to better organize individual user accounts.

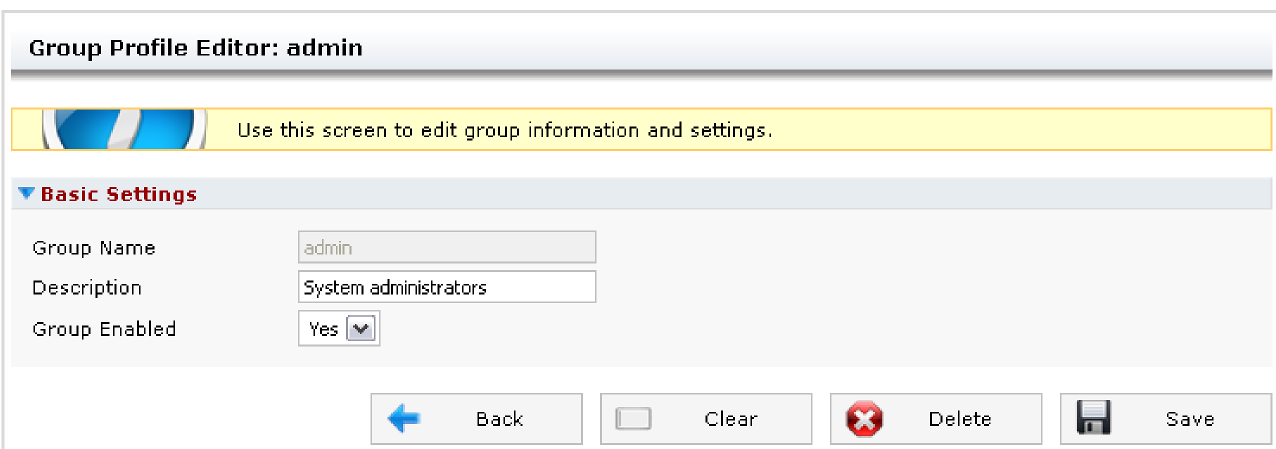


Image 74: Group basic settings

Details on individual settings are shown in the table below.

Setting	Description
Group Name	A meaningful name which can be used to identify a particular group.
Description	Detailed information to complement the group name setting.
Group Enabled	This setting controls the current group state. Each group can be enabled or disabled. Disabled groups will not be available for user assignment. At the same time, all users belonging to the disabled group will also be disabled.

Please Note:
A user group is a purely organizational term which has no influence on user account permissions.

11. Managing Address Book

Address book section provides an overview of the VoiceGear Connect contact management system. Address book can be used by every user registered with the system to manage their Skype contacts and contains three different address book types as shown in the table below. Each address book type carries with it different permission levels. To access the address book interface, click on the "Address Book" heading in the navigation menu shown in the menu area.

Type	Description
System address book	Company-wide address book containing contacts available for all users.
Group address book	Address book containing contacts available for a certain groups of users.
User address book	Address book containing contacts available for an individual user.

Every **user** can read and modify contacts in his/her own user address book, but can only read system and group address books.

Every **group admin user** can read and modify contacts in group address book as well as in all address books of users belonging to his/her own group. He/ she can also read the contacts in the system address book. However, group admin users cannot access address books belonging to other groups or private address books owned by users of other groups.

The **system admin user** can read and modify contacts in all address books.

For a complete matrix of address book permissions, please refer to the table below.

Address book	User Permissions	Group Admin Permissions	Admin Permissions
System address book	read	read	read/write
Group address book	read	read/write	read/write
Another group's address book	X	X	read/write
User address book	read/write	read/write	read/write
Another user's address book within the same group	X	read/write	read/write
Another user's address book not within the same group	X	X	read/write

11.1 Contacts

Contacts screen allows you to search for and manage contacts belonging to various address books. To access it, please click the “Contacts” link under the “Address Book” heading in the navigation menu shown in the menu area. For reference, a sample contacts list is shown below.

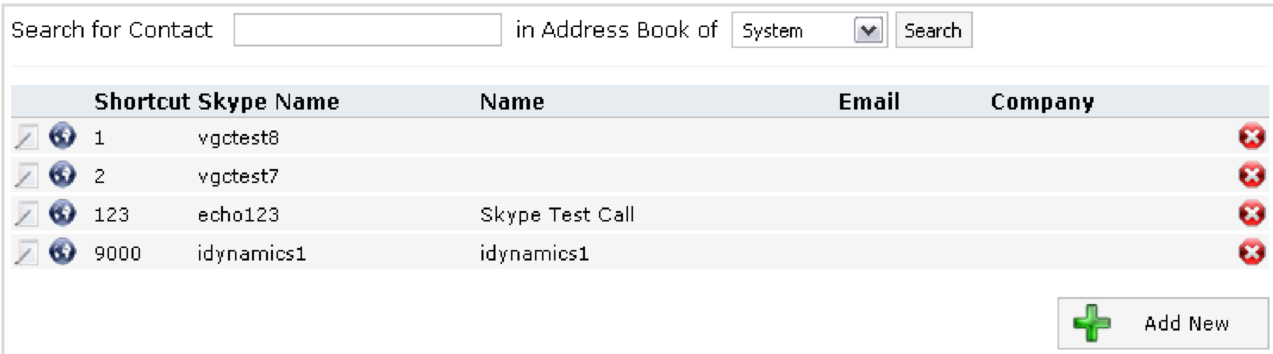


Image 75: Address book contact list

Contact list shows different icons for each contact to quickly identify the type of address book the contact has been saved in. For more details, please refer to the table below.

Icon	Address Book Type
	System address book.
	Group address book.
	User address book.

Provided that you have enough permission to do so, you can edit any contact shown in the contact list by clicking on the edit button of the desired contact. You can also add contacts to the list by clicking on the “Add New” button. Doing so will open the contact details form.

11.1.1 Basic Settings

Basic settings form allows you to define essential contact information necessary for making calls to and identifying the contact in the system.

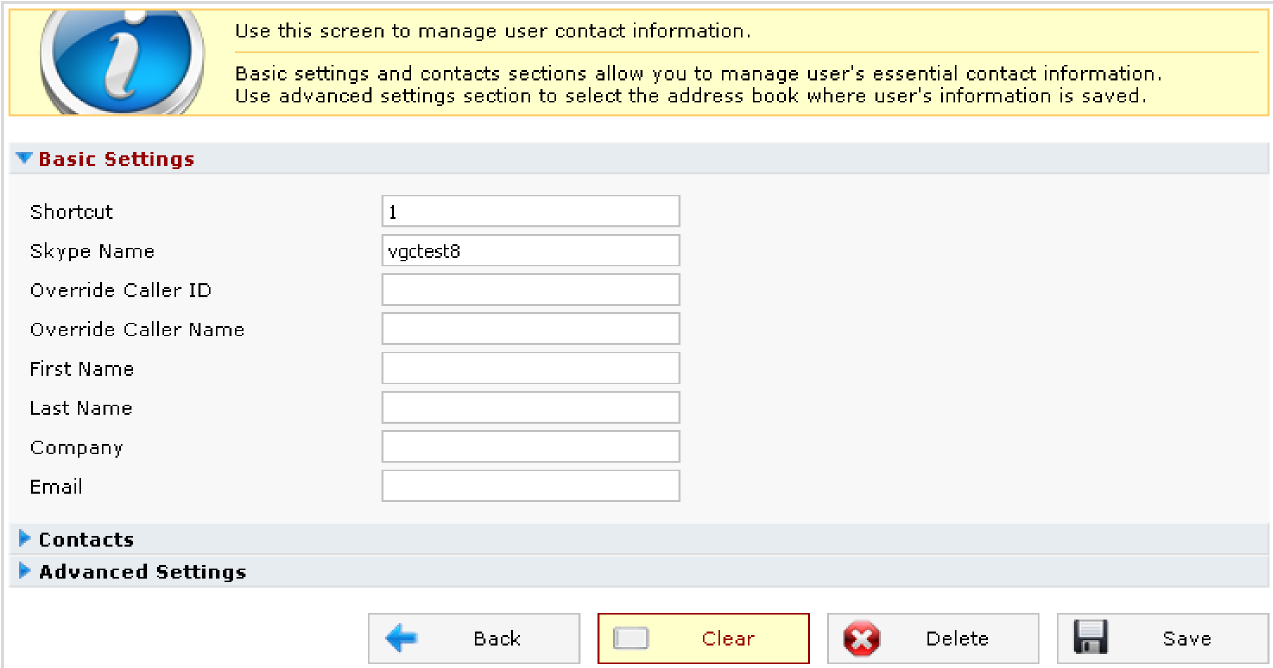


Image 76: Basic information of contacts in the address book

Only fields required for any contact are defined in this form and include "Shortcut" and "Skype Name". For more details on available fields, please refer to the table below.

Setting	Description
Shortcut	A numeric value associated with every contact to facilitate dialing that contact from a touch tone phone. For example, if a contact with Skype name defined as "echo123" has shortcut "123", you can dial "123" from a touch tone phone integrated with VoiceGear Connect to make a Skype call out to that contact.
Skype Name	Valid name of the Skype account associated with a given contact.
Override Caller ID	Caller ID setting passed to the PBX system for all incoming calls from a given contact. This value overrides the default setting configured in "Settings>Misc Settings>Caller Identification".
Override Caller Name	Caller name setting passed to the PBX system for all incoming calls from a given contact. This value overrides the default setting configured in "Settings>Misc Settings>Caller Identification". This setting is only applicable when VoiceGear Connect is integrated with a SIP PBX system.
First Name	First name of the contact.
Last Name	Last name of the contact.

Company	Contact's company name.
Email	Email address of the contact.

Please Note:
Certain older PBX systems might not support caller ID.

11.1.2 Contact Details

Contact details form allows you to define additional information about each address book contact such as mobile phone, website address, etc.

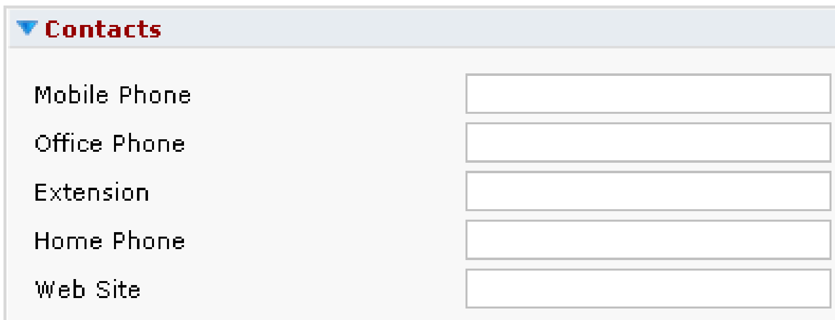


Image 77: Contact details

For more details on available fields, please refer to the table below.

Setting	Description
Mobile Phone	The mobile phone number of the contact.
Office Phone	The office phone number of the contact.
Extension	The office extension of the contact.
Home Phone	The home phone number of the contact.
Web Site	The web site URL of the contact.

Please Note:
Contact phone numbers are defined for informational purposes only. When dialing out to a given contact, the system will always use the Skype name of that contact.

11.1.3 Advanced Settings

Advanced settings form allows you to choose which address book a given contact will be saved in. For more information, please refer to the image below.

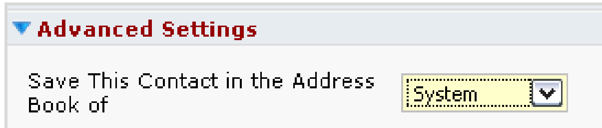


Image 78: Contact advanced settings

11.2 Export Contacts

The system allows exporting the content stored in all address books. In order to bring up the contact export screen in the working area, please use the "Address Book -> Export Contacts" link from the main menu. The resulting screen is shown in the image below.

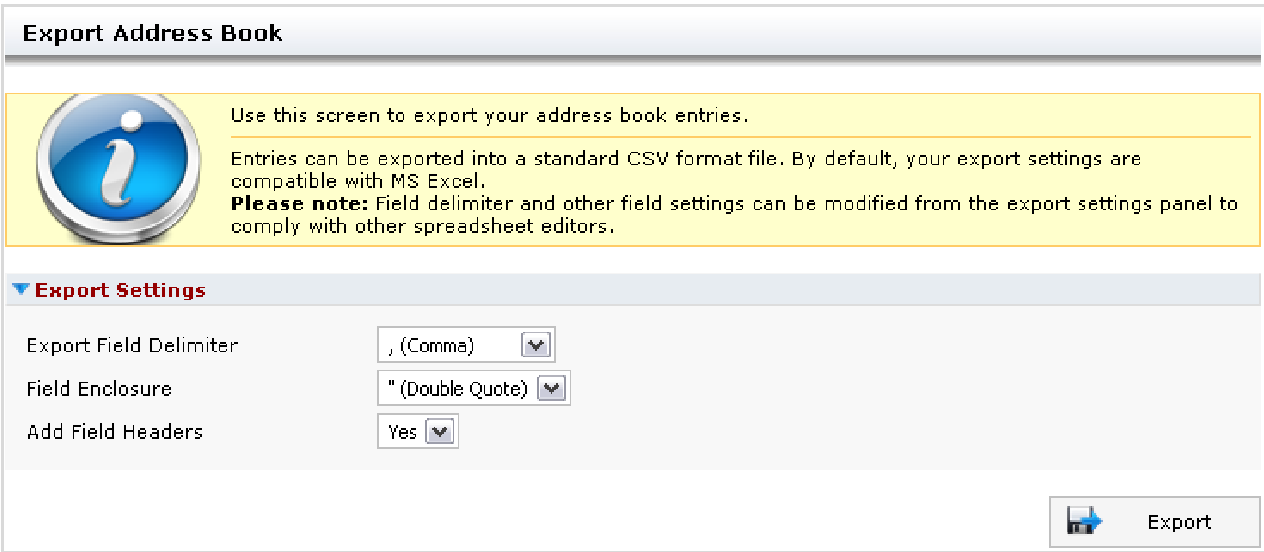


Image 79: Contact export settings

To export contacts, please select field delimiter, enclosure settings and click on the "Export" button. By default the system exports in a CSV format tailored for Microsoft Excel, however, delimiter and enclosure settings can be adjusted to be better recognized by other spreadsheet applications. For more details on available fields, please refer to the table below.

Setting	Description
Export Field Delimiter	The delimiter character used to separate individual contact fields.
Field Enclosure	The character that is used to enclose individual contact fields.
Add Field Headers	An option that allows including or excluding column headers containing contact field names.

Please Note:
Contacts can only be exported by the system administrator.

11.3 Import Contacts

The system allows importing address book contacts from a CSV file originally generated by the contact export function. Contact import can facilitate system migration or bulk contact edit where existing contacts can be exported and edited in Microsoft Excel and imported back into the system. In order to bring up the contact import screen in the working area, please use the “Address Book -> Import Contacts” link from the main menu. The resulting screen is shown in the image below.

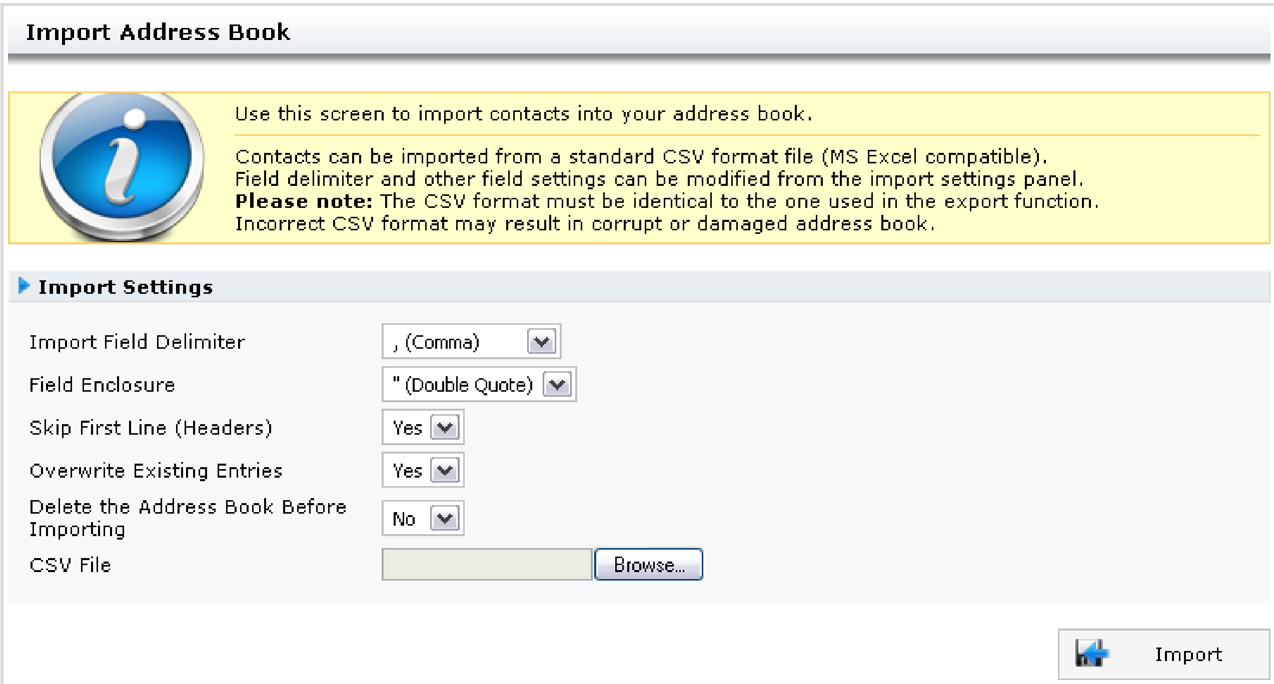


Image 80: Contact import settings

To import contacts, please select applicable import settings and push the “Import” button. For more details on available fields, please refer to the table below.

Setting	Description
Import Field Delimiter	The delimiter character used to separate individual contact fields in the CSV file to be imported.
Field Enclosure	The character that is used to enclose individual contact fields.
Skip First Line (Headers)	An option that lets the system know how to treat the first line in the CSV file to be imported. If the CSV file already contains field headers, please set this option to “Yes” in order to prevent field headers from being imported as contacts.
Overwrite Existing Entries	Setting this option to “Yes” will make the system overwrite existing contacts with the

	same shortcut as the ones in the CSV file to be imported.
Delete Address Book Before Importing	Setting this option to "Yes" will make the system permanently delete all existing contacts prior to importing new ones.
CSV File	The location of the CSV file to be imported on a local file system.

Please Note:
Contacts can only be imported by the system administrator.

Please Note:
Importing an invalid contacts file can cause loss of address book data. It is strongly recommended to import only those files that have been generated by the contact export function.

12. Legal Information

THIS VOICEGEAR GATEWAY PRODUCT (HEREAFTER REFFERRED TO AS “GATEWAY”) HAS BEEN CONFIGURED TO INTERACT WITH THE SKYPE COMMUNICATIONS SERVICE (HEREAFTER REFFERED TO AS “SKYPE”) AS OF THE DATE OF SHIPMENT. INDUSTRYDYNAMICS WILL HAVE NO LIABILITY WHATSOEVER IF ANY FEATURE OR FUNCTIONALITY OF THE GATEWAY FAILS TO PERFORM IN CONFORMANCE TO THE PUBLISHED SPECIFICATIONS BECAUSE OF ANY CHANGES MADE BY SKYPE TO ITS APPLICATION, PLANS OR SERVICES. BY TURNING ON THE GATEWAY YOU ACCEPT THE CONTENT OF THIS DISCLAIMER. IF THIS DISCLAIMER IS NOT ACCEPTABLE TO YOU, PLEASE RETURN THE GATEWAY FOR A REFUND IN ITS ORIGINAL PACKAGING INCLUDING ALL ORIGINAL ACCESSORIES TO THE POINT OF PURCHASE WITHIN TEN (10) DAYS OF DATE OF PURCHASE.

13. Purchasing SkypeOut Credit

In order to be able to make cheap calls to landline or mobile numbers anywhere in the world at Skype rates, you need to purchase SkypeOut credit. You can easily do this from the VoiceGear Connect web configuration interface by simply clicking on the “Buy Skype Credit” link at the top of the page as shown in the following screenshot. We recommend using this link to make sure your credit is properly applied.

Image 81: Buy Skype credit link

The “Buy Skype Credit” link will open a new window to the Skype business control panel which can be used to buy bulk SkypeOut credit and assign portions to all of your Skype lines. It can be used to centrally manage all of your existing Skype accounts. Best of all, Skype business control panel is completely free to register for and use. The page shown after clicking on the “Buy Skype Credit” link is shown below. All you need to do is click on the “Features” link to get started.

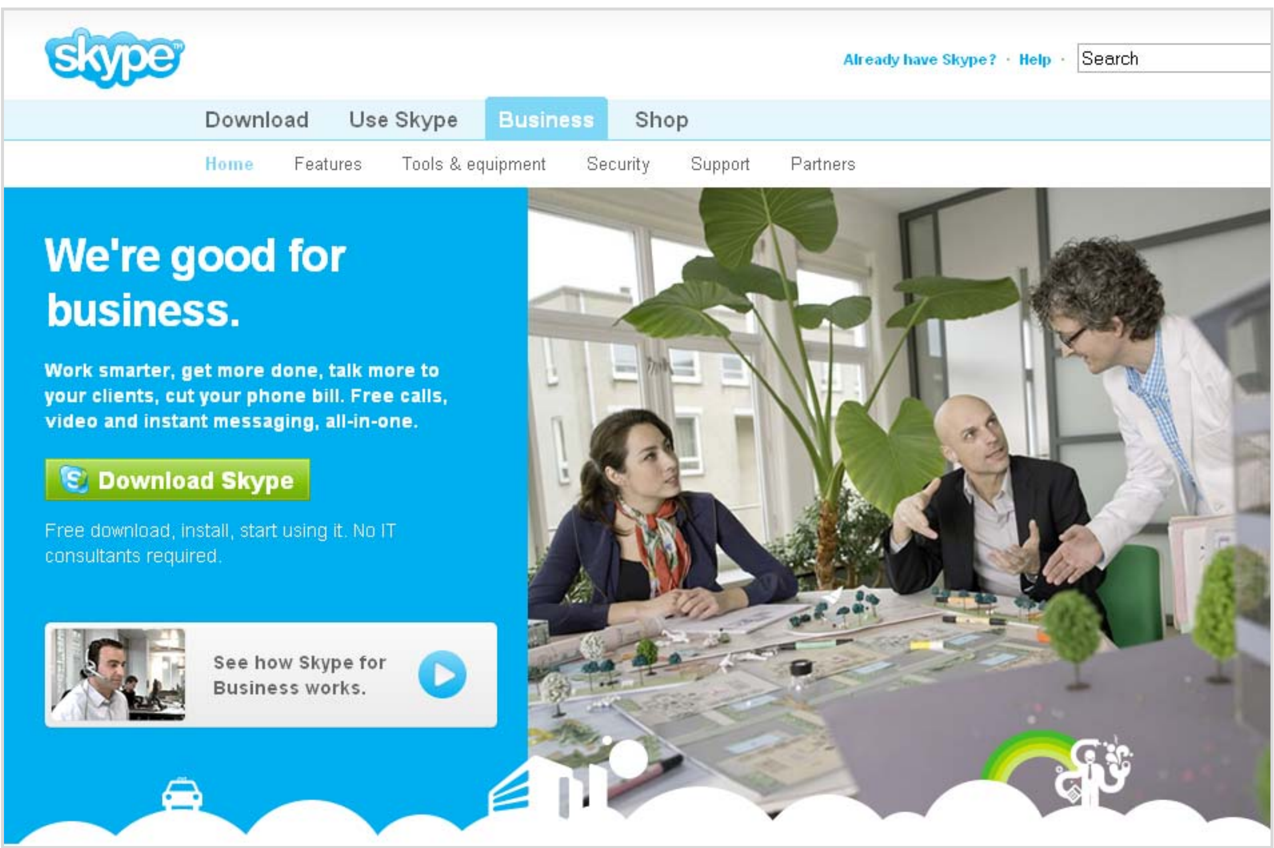


Image 82: Skype business panel

14. FAQ

Question	Answer
I hear my own voice as echo during calls through the VoiceGear Connect gateway when using analog PBX connectivity.	Please set the Tx gain value between -5 and -10 under the “Channels->Analog/Digital” form accessible via the navigation menu of the web configuration interface. To access the Tx gain setting, please click on the “Configure” button under “Installed Hardware”. Alternatively, please opt for the advanced echo cancellation option on your gateways.
I cannot access the web configuration interface of my VoiceGear Connect.	Please make sure the gateway is properly connected to the network by logging into the administration console interface and verifying values under the “Configure Network” option.
My Skype lines are not accessible through the office phone.	Please make sure the gateway is properly connected to the PBX and the Internet and all Skype lines are currently active. This can be done by navigating to “System->Channel Status” screen using the navigation menu in the web configuration interface. If Skype lines are not enabled, you can enable them from the Skype Accounts page.
My SIP PBX cannot integrate with the VoiceGear Connect gateway.	Please make sure your PBX SIP trunk is properly configured using G.711 A-law or µ-law codec, 20ms packet size and no silence suppression. On the gateway side, please make sure the PBX IP is correct and SIP authentication settings match those used on your PBX.
Can I export my call logs to Excel for further analysis?	All system call logs can be exported via the “System->Call Log” option in the navigation menu. Automatic recurring call log transfer can be configured via “System->Backup Settings” screen.