VoIP Studio

Administrator Manual

Version: 0.6
Date: 2014-04-25

1. Introduction

General

VoIP Studio is a complete fully featured business class Hosted VoIP PBX systems. It allows you to make and receive phone calls from the Internet and traditional telephone network at the same time. It makes managing your communication easier and helps to reduce operating costs and increase productivity.

Requirements

Your control panel is browser-based. The following are recommended:

- Internet Explorer 7+
- Mozilla Firefox 3.6+
- Google Chrome
- Opera 9+
- Safari 4+

The web control panel is optimized for a screen resolution of minimum 1024 x 786 pixels.

Softphone application requires one of the following operating systems:

- Microsoft Windows XP
- Microsoft Windows Vista
- Microsoft Windows 7
- Linux
- Apple OSX

Intended Audience

This guide is intended for personnel involved in operating hosted VoIP PBX. Readers of this guide should possess the following recommended knowledge and skill sets:

- Basic computer skills
- Familiarity with standard PBX features

Conventions

In this manual, you will find a number of styles of text that distinguish between different kinds of information. Here are some examples of these styles, and an explanation of their meaning.

style	meaning
`www.example.com/login`	Text you need to type into a program.
email field `(1)`	Field shown in a figure. For example: 

Key concepts

VoIP (Voice over IP)

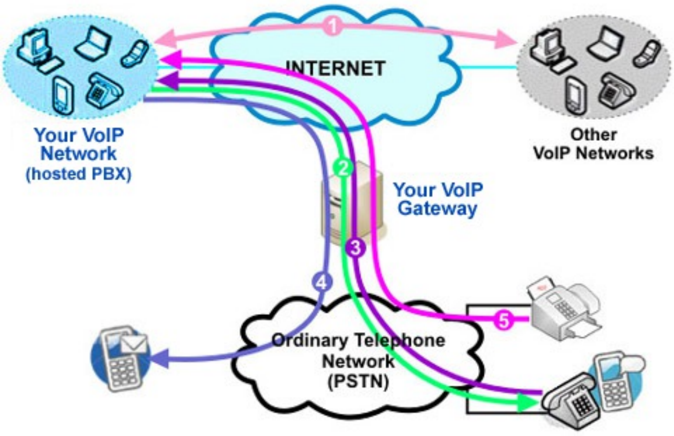


Figure 1.1 How VoIP Hosted PBX works.

VoIP or Internet telephony refers to communications services - voice, facsimile, and/or voice-messaging applications - that are transported via the Internet, rather than the Public Switched Telephone Network (PSTN - traditional telephony). Thanks to VoIP technology you can:

- (1) make and receive free Internet calls to other VoIP networks.
- (2) call any land line or mobile (cell) phone at very low rates.
- (3) receive calls from traditional telephone network.
- (4) send text messages (SMS) to mobiles (cells) world wide at very low rates.
- (5) receive faxes to your email address for free.

Hosted PBX

Hosted PBX is a service provided by us, using equipment located in our premises. This means you don't need to buy or install PBX equipment in your office in order to benefit from advanced PBX features.

Control Panel

This is a web based application which allows to manage all aspects of your hosted PBX system. Create new user accounts, ring group, pickup groups, assign inbound numbers and provision VoIP phones.

Extension

A telephone extension is an internal telephone line attached to a PBX system which allows multiple phones to connect without each phone requiring a separate outside line (inbound number) assigned to it.

Inbound number

Inbound number also called Direct Inward Dialling (DID) in USA, Direct Dial-In (DDI) in Europe, is a range of telephone numbers connected to your PBX, so it can route the call to the desired person or Ring Group (IVR etc.) within the organization.

2. Quick Start

Login

Figure 2.1 Login form.

To login into the control panel navigate your web browser to http://voipstudio.com/login and enter your email address into field `(1)` and password into filed `(2)`. Optionally you can select "Remember me on this computer" checkbox `(3)` to have your email address saved. Finally click Login button `(4)` and once Control Panel is loaded click `Administration (5)` button in the top right corner.

Administrator Interface overview

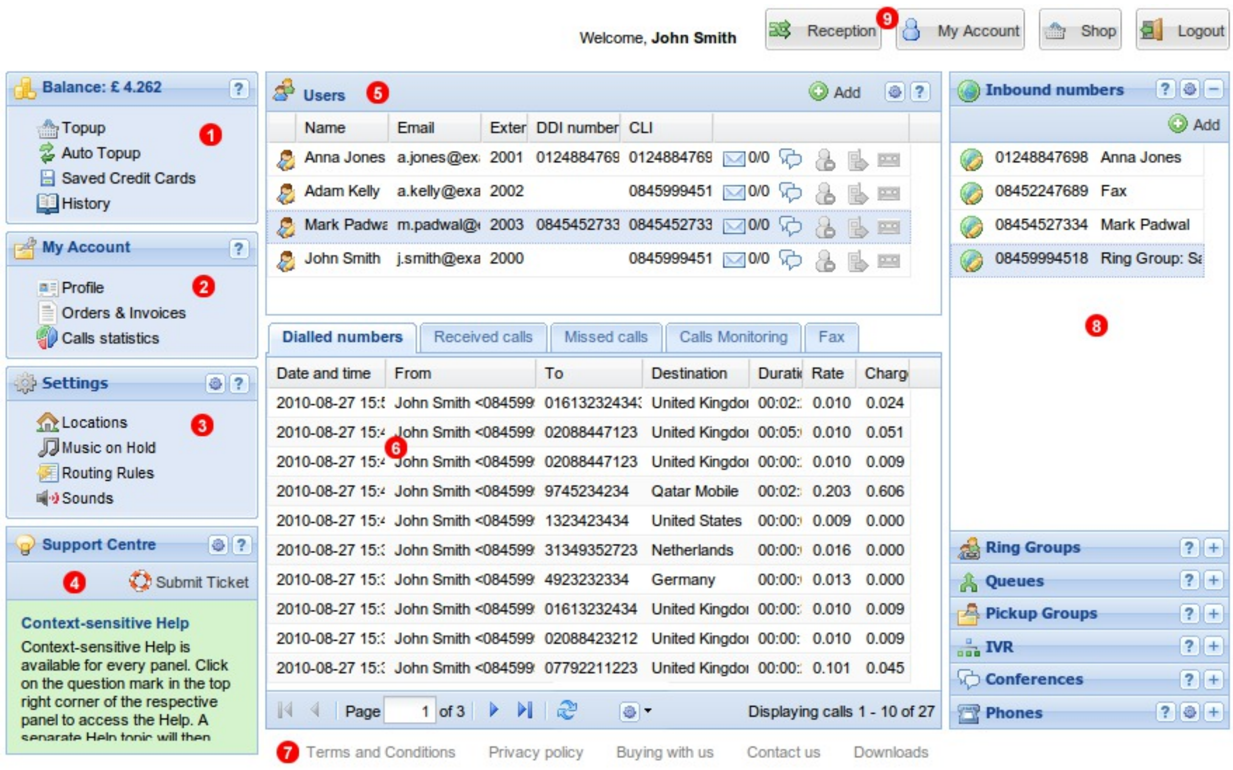


Figure 2.2 Administrator Control Panel.

1. Balance panel - here you can find your current pre-paid balance, account statement, top-up your account and manage saved credit cards.
2. My Account panel - here you can view and edit your company profile, track progress of your orders and obtain invoices.
3. Settings panel - here you can manage your locations and other advanced PBX features
4. Support Centre panel - gives access to context sensitive help. Also here you can submit tickets to our Support Team.
5. Users panel - allows to create new and edit existing user account.
6. Calls History grid - in this panel you can see history of all your calls (including billing details).
7. Footer section - here you will find various links, including the one to download Softphone application.
8. Sidebar panel - here you can manage your Inbound Numbers (DDI) and a number of advanced PBX features like: Queues, Ring and Pickup Groups, IVRs, Virtual Conferences Rooms and VoIP phones.
9. Header buttons - here you will find buttons allowing you to switch to different sections of your Control Panel and to log out.

3. Balance

Figure 3.1 Balance panel.

All our services are pre-paid, so you need to make sure you have enough credit on your account to make outbound calls to chargeable destinations.

Top Up

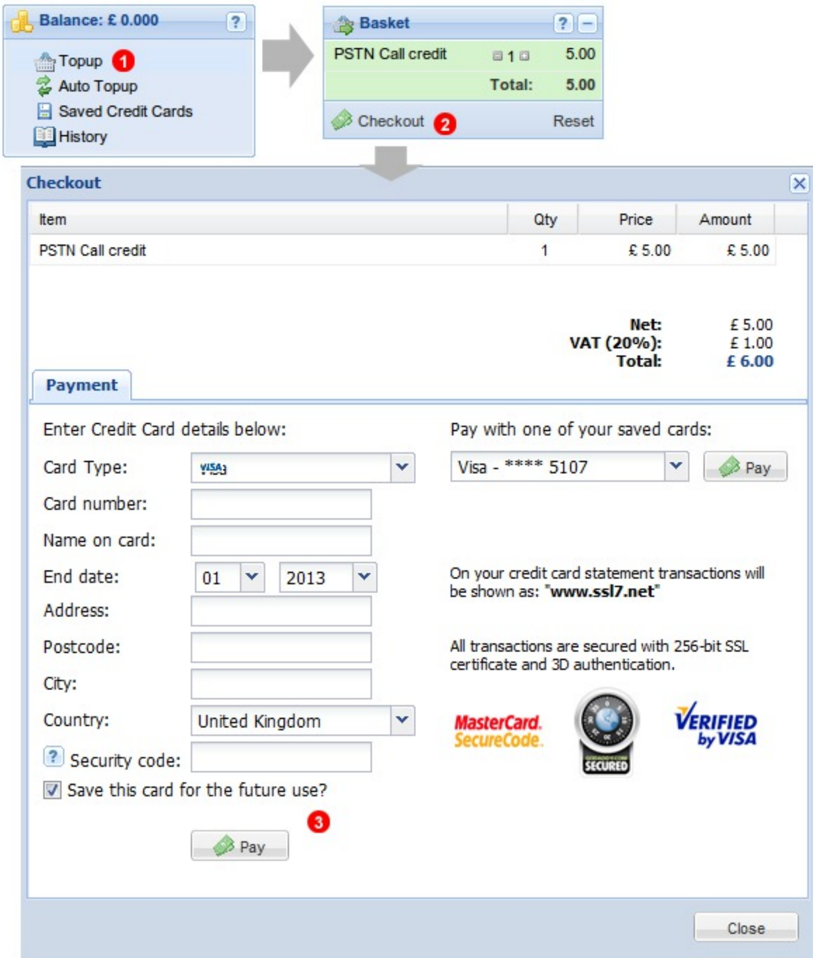


Figure 3.2 Balance Top Up.

To increase balance of your account:

1. Click `Top Up` link in `Balance` panel and PSTN Call credits will be added to your `Basket`. You can use `[+]` and `[-]` buttons to increase or decrease amount of purchased credits.
2. Click `Checkout` button to complete your transaction.
3. Enter your Credit Card details and click `Pay` button.

Please note: before making your first payment please download our Softphone and make a Test Call by dialling 123 - the system will verify if the country where your call originated is the same as billing address of your credit card. If for some reason you need to use a credit card issued in a different country where you are currently located, please open a Support Ticket and mention your location and country of origin of credit card you wish to use.

Auto Top Up

In order to ensure your account is always in credit you can enabled Auto Top Up feature. Once your account reaches pre-defined level, it will be automatically topped up using one of your saved credit cards. This is a convenient way of making sure your account balance never reaches zero.

Activate

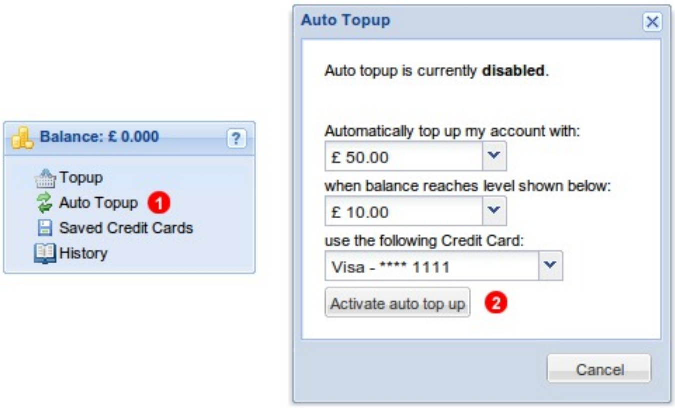


Figure 3.3 Activate auto Top Up.

To activate Auto Top Up feature:

1. Click `Auto Top Up` link in `Balance` panel.
2. Select Top Up amount and desired auto Top Up level. Finally click `Activate auto Top Up` button.

Dectivate

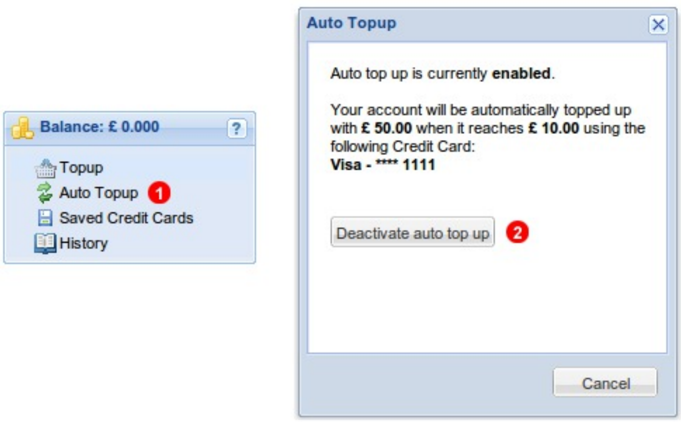


Figure 3.4 Deactivate auto Top Up.

To deactivate Auto Top Up feature:

1. Click `Auto Top Up` link in `Balance` panel.
2. Click `Deactivate auto Top Up` button.

Saved Credit Cards

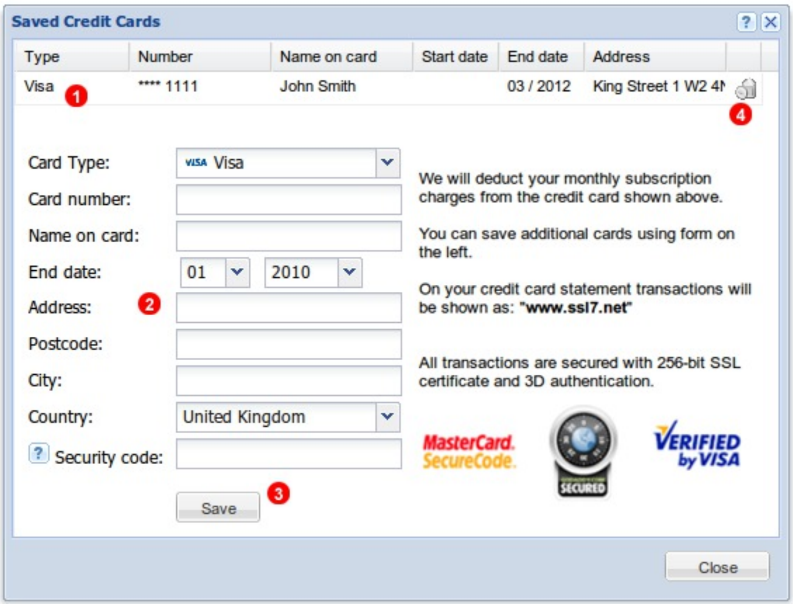
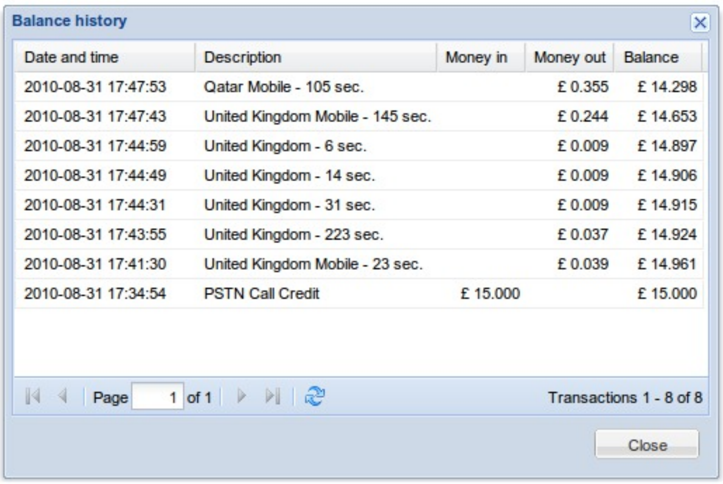


Figure 3.5 Saved Credit Cards window.

In order to pay for monthly subscription fees or use Auto Top Up feature, you need to save details of at least one credit card. To save credit card details, click Saved Credit Cards link in Balance panel. Next fill all fields in the form (2) and click Save button (3). If you no longer wish to use particular credit card for your payments, use delete button (4) to remove it from the system.

History



Balance history

Date and time	Description	Money in	Money out	Balance
2010-08-31 17:47:53	Qatar Mobile - 105 sec.		£ 0.355	£ 14.298
2010-08-31 17:47:43	United Kingdom Mobile - 145 sec.		£ 0.244	£ 14.653
2010-08-31 17:44:59	United Kingdom - 6 sec.		£ 0.009	£ 14.897
2010-08-31 17:44:49	United Kingdom - 14 sec.		£ 0.009	£ 14.906
2010-08-31 17:44:31	United Kingdom - 31 sec.		£ 0.009	£ 14.915
2010-08-31 17:43:55	United Kingdom - 223 sec.		£ 0.037	£ 14.924
2010-08-31 17:41:30	United Kingdom Mobile - 23 sec.		£ 0.039	£ 14.961
2010-08-31 17:34:54	PSTN Call Credit	£ 15.000		£ 15.000

Page 1 of 1   Transactions 1 - 8 of 8

Close

Figure 3.6 Balance history.

To view your account statement click `History` link in `Balance` panel.

4. My Account

Figure 4.1 My account panel.

Profile

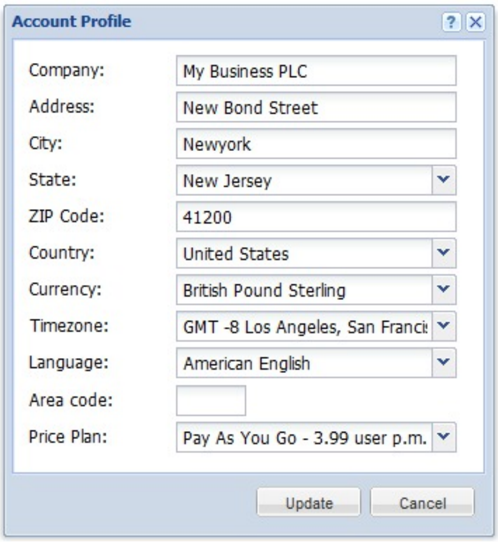


Figure 4.2 Company profile window.

To change your Company address details, click Profile link in My Account panel (see figure 4.1). Please make sure both your address details, preferred currency and VAT number (applicable to European Union countries only) are correct, as this will be used to prepare your invoices. Depending on your location and if VAT number is entered, the following VAT tax rules apply:

- Current VAT rate applies to all domestic / EU customers
- EU Business with a VAT Number will not be charged VAT
- EU Individuals/business without VAT number - current VAT rate applies
- Customers outside EU, no VAT applies

Here you can also change your time zone which will affect how dates and times are displayed in the Control Panel and language of PBX prompts.

Orders and Invoices

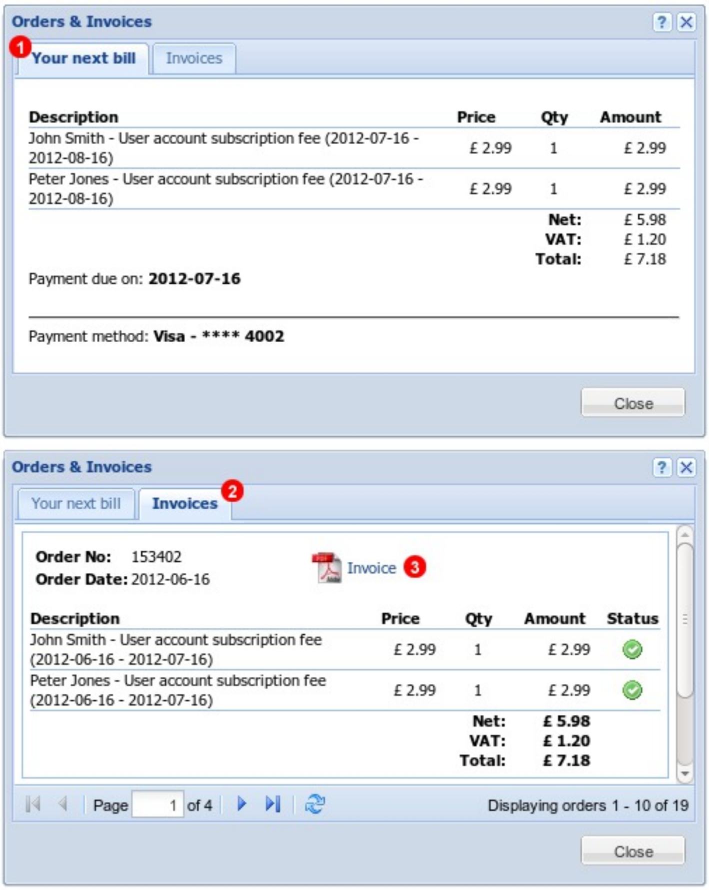


Figure 4.3 Orders and invoices window.

To view or print your invoices click `Orders & Invoices` link located in `My Account` panel (see figure 4.1).

1. Details of your next payment can be found in `You next bill` tab.
2. To view status of your orders click `Invoices` tab.
3. To download an Invoice in PDF format click link `(3)`.

Statistics

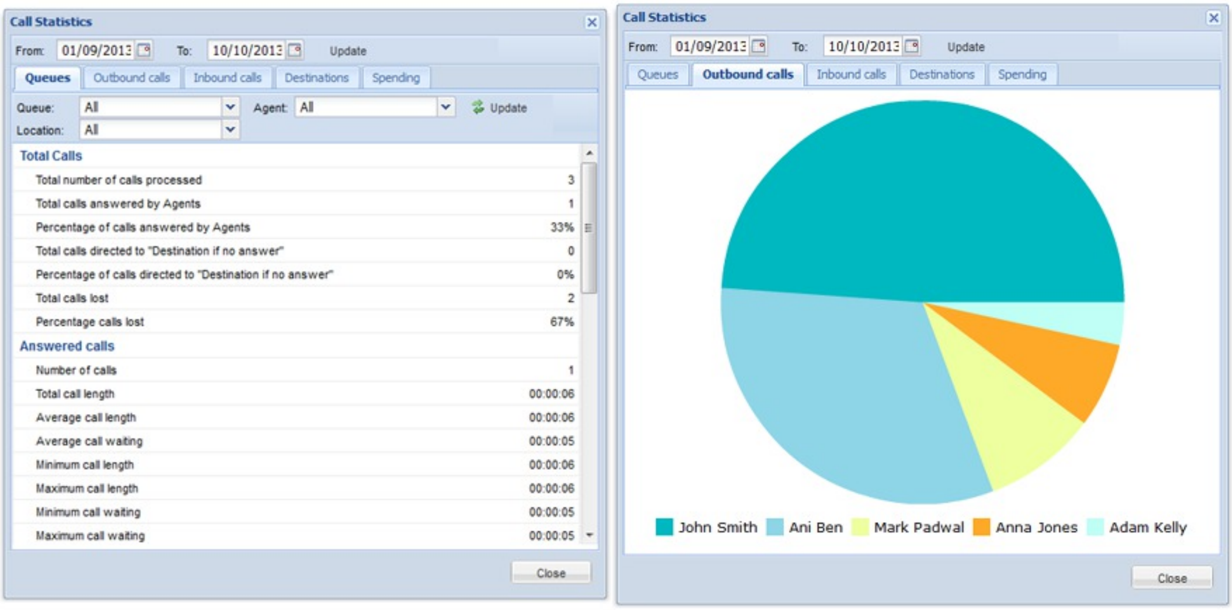


Figure 4.4 Call statistics.

Here you can view statistics of your inbound and outbound calls as well as costs breakdown.

5. Settings

Figure 5.1 Setting panel.

Locations

Figure 5.2 Locations.

Dial plan of each user depends on their geographical location. For example in most European countries prefix 00 for international numbers and single 0 for local numbers is used. In United States prefix 011 for international numbers and no prefix for local numbers is used. By default all your users are assigned to the location (country and time zone) as defined in your Company profile - see Figure 4.2 above. However if some of your users are located in different countries you can define additional location and assign users to them.

To add or modify locations click `Locations` link located in `Settings` panel - see Figure 5.1 above - and follow steps below:

1. Click `Add location` button.
2. Enter your new location address details.
3. Click `Submit` button.

Emergency Services Calling

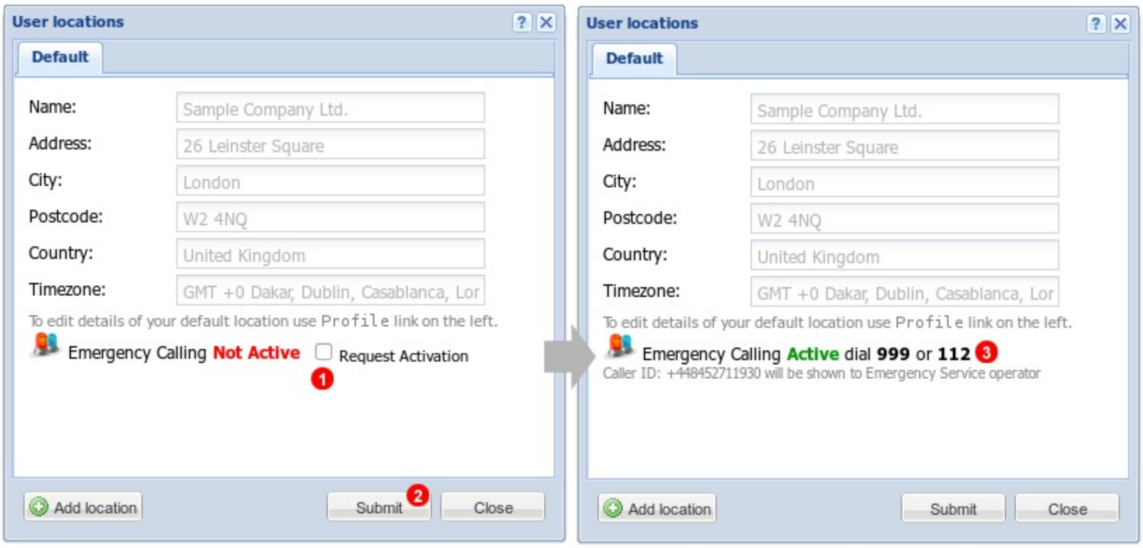


Figure 5.3 Emergency Services calling

In some locations you can enable Emergency Services Calling (Police, Ambulance and Fire Brigade). To enable this feature please ensure your address details are correct as they will be passed to Emergency Centre and used to dispatch Emergency vehicles in case of a "silent call" (when operator is unable to speak with person who called emergency number).

To enable Emergency Calling click `Locations` link located in `Settings` panel - see Figure 5.1 above - and follow steps below:

1. Select `Request Activation` checkbox - see `(4)` in Figure 5.1 above.
2. Click `Update` button.
3. Once you address is successfully validated you will be able to call Emergency Services by dialling 999 in the United States or 999 / 112 in the United Kingdom.

Important: you will not be able to make emergency call in case your Internet connection is down. Please use your mobile (cell) phones in this case.

Music on Hold

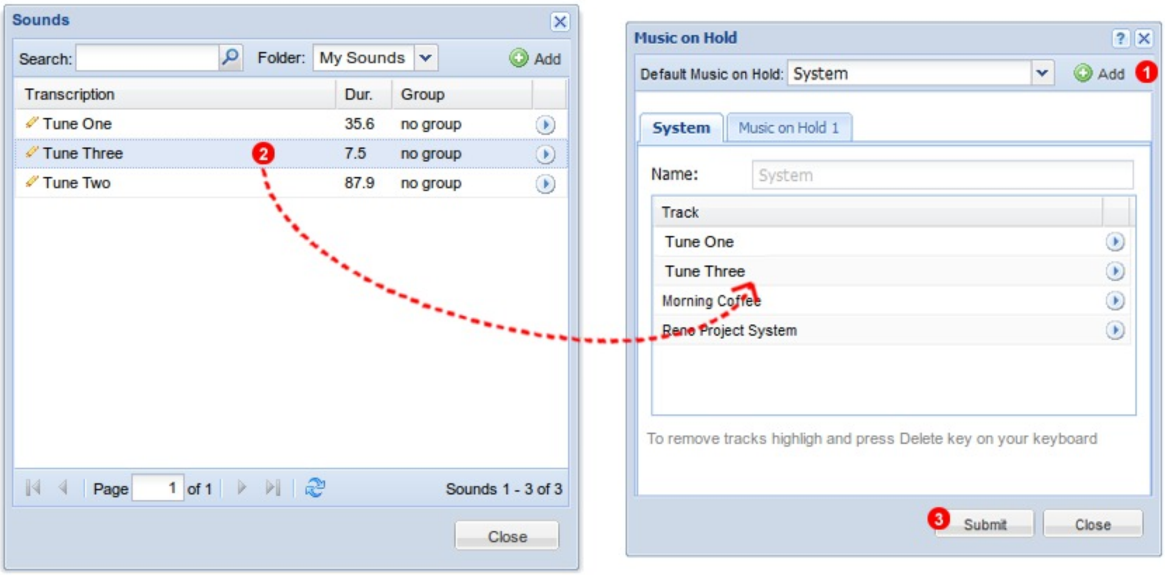


Figure 5.4 Custom Music on Hold.

By default there is one set of music tunes which are being played to callers while they are awaiting for connection. It is possible to create up to 20 custom Music on Hold sets which can include marketing messages and other announcements.

To create custom Music on Hold set click `Music on Hold` link located in `Settings` panel - see Figure 5.1 above - and follow steps below:

1. CLick `Add` button located in top right part of `Music on Hold` window.
2. Drag and drop music tines you want to include in custom Music on Hold set.
3. Click `Submit` button.

Routing Rules

Using Routing Rules it is possible to overwrite default settings for inbound and outbound calls based on pre-defined conditions.

Inbound rules

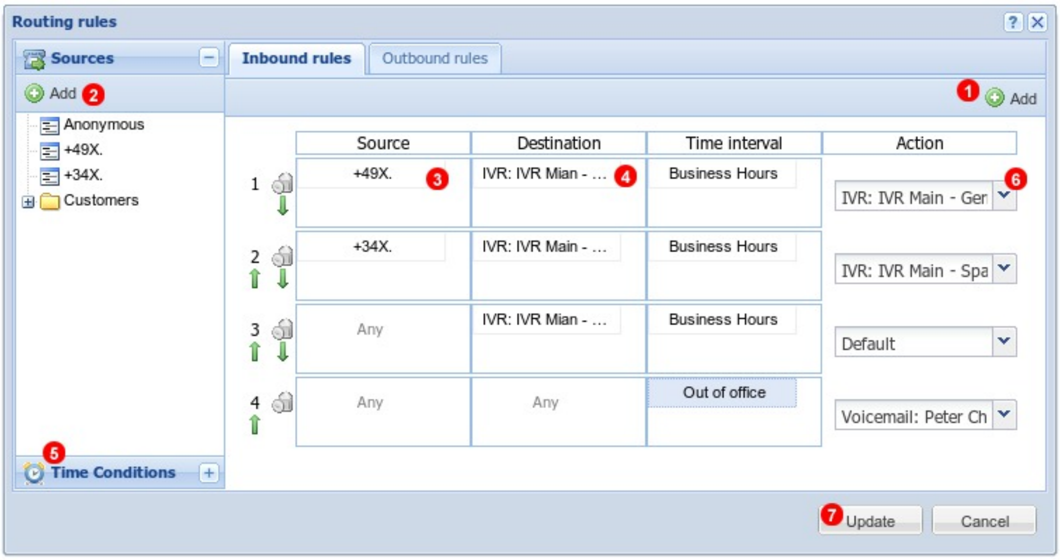


Figure 5.5 Inbound rules.

To define inbound routing rules:

1. Click `Add` button `(1)` to create new `Inbound rule`.
2. Add one or more `Source` numbers and/or patters using `Add` button `(2)`. Characters listed below are interpreted as a pattern rather than a literal:
   - `X` - any digit from 0-9
   - `Z` - any digit from 1-9
   - `N` - any digit from 2-9
   - `[1235-9]` - any digit in the brackets (in this example, 1,2,3,5,6,7,8,9)
   - `X.` - (dot) one or more of X - for example +442X. will match any number starting with 442
3. Drag and drop number and/or patters from `Sources` panel to `Source` column in `Inbound rules` tab.
4. Drag User, Inbound Number, IVR or Queue into `Destination` column.
5. Optionally you can add Time Conditions using panel `(5)`.
6. Set `Action` that PBX should perform if inbound call matches defined conditions using drop down list `(6)`.
7. Click `Update` button to save your rules.

Example

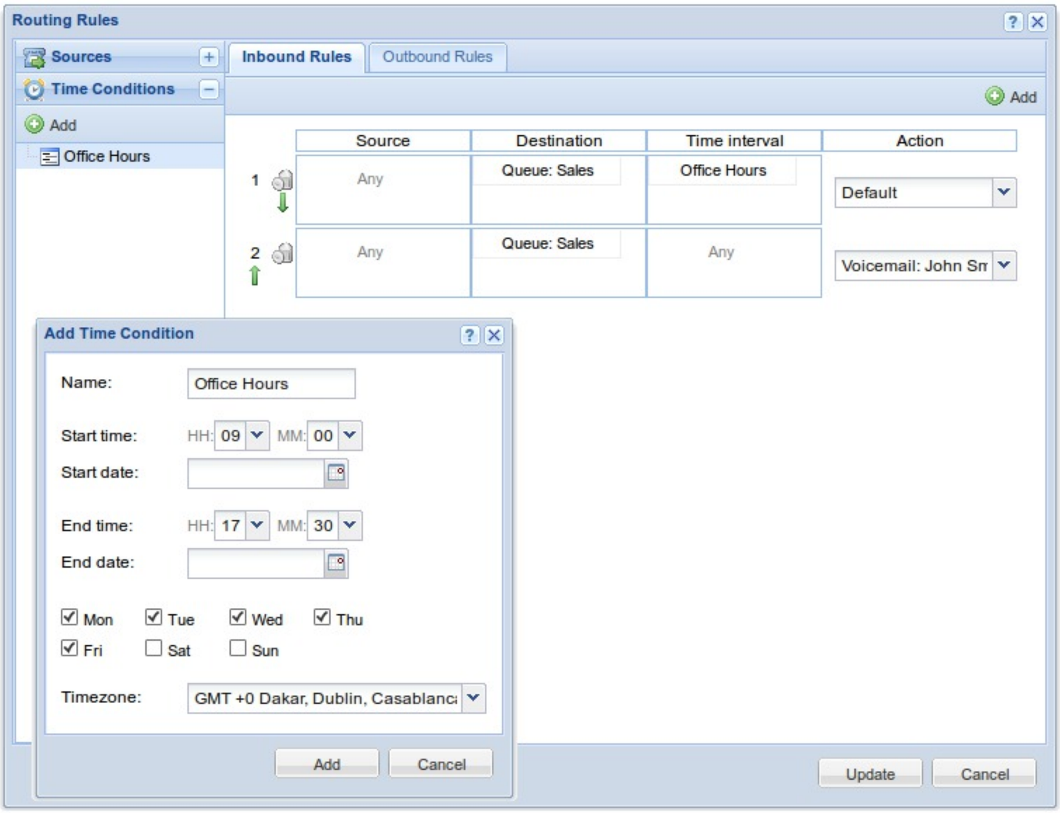


Figure 5.6 Inbound routing rule - example.

Call routing system will try to match all incoming calls according to their source (Caller ID), destination and time interval. In the example above there are two rules defined for "Destination" - "Queue: Sales". Lets analyze the following two scenarios:

Scenario 1. A call from number +1 321 987 323 to "Queue Sales" at 10:45 on Wednesday:

Rule 1 * Source: +1 321 987 323 - matches "Any" ? - Yes * Destination: matches "Queue Sales" ? - Yes * Time interval: 10:45 on Wednesday matches "from 9:00 to 17:30, Mon. - Fri." ? - Yes * Action: connect the call to the "Queue Sales" (Default) and stop execution of routing rules.

Scenario 2. A call from +1 321 987 323 to "Queue Sales" at 6:10 on Wednesday:

Rule 1 * Source: +1 321 987 323 - matches "Any" ? - Yes * Destination: matches "Queue Sales" ? - Yes * Time interval: 10:45 on Wednesday matches "from 9:00 to 17:30, Mon. - Fri." ? - No, skip this rule and continue to the next one

Rule 2 * Source: +1 321 987 323 - matches "Any" ? - Yes * Destination: matches "Queue Sales" ? - Yes * Time interval: 10:45 on Wednesday matches "from 9:00 to 17:30, Mon. - Fri." ? - Yes * Action: route call to John's Voicemail and stop execution of routing rules.

Outbound rules

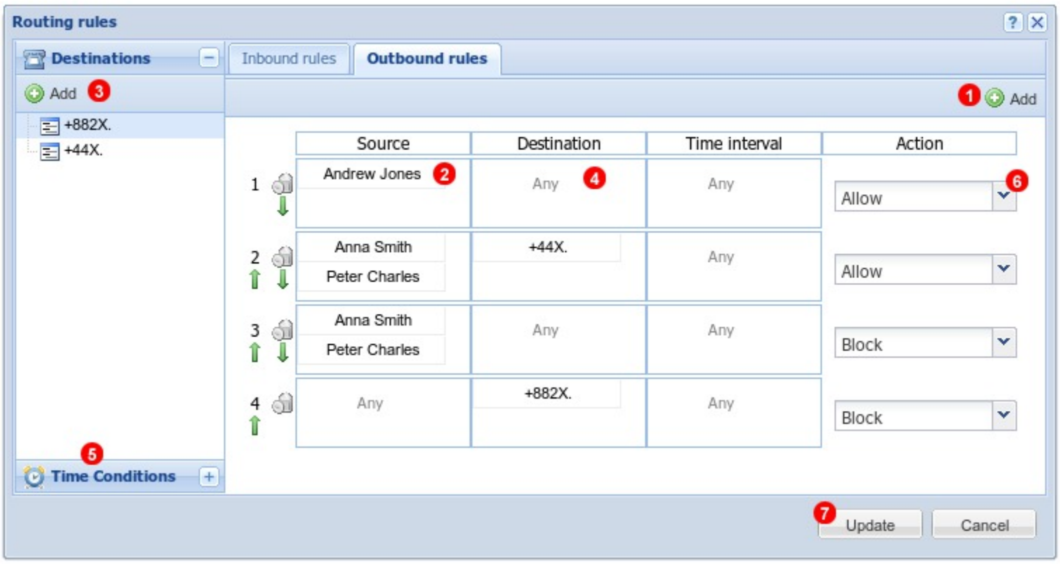


Figure 5.7 Outbound rules.

To define outbound routing rule:

1. Click `Add` button `(1)` to add new rule.
2. Drag and drop users into `Source` column `(2)`.
3. Add one or more `Destination` numbers and/or patters using `Add` button `(3)`. Characters listed below are interpreted as a pattern rather than a literal:
   - `X` - any digit from 0-9
   - `Z` - any digit from 1-9
   - `N` - any digit from 2-9
   - `[1235-9]` - any digit in the brackets (in this example, 1,2,3,5,6,7,8,9)
   - `X.` - (dot) one or more of X - for example +442X. will match any number starting with 442
4. Drag and drop number and/or patters from `Destinations` panel to `Destination` column in `Outbound rules` tab.
5. Optionally you can add Time Conditions using panel `(5)`.
6. Set `Action` that PBX should perform if inbound call matches defined conditions using drop down list `(6)`.
7. Click `Update` button to save your rules.

Sounds

Add sound

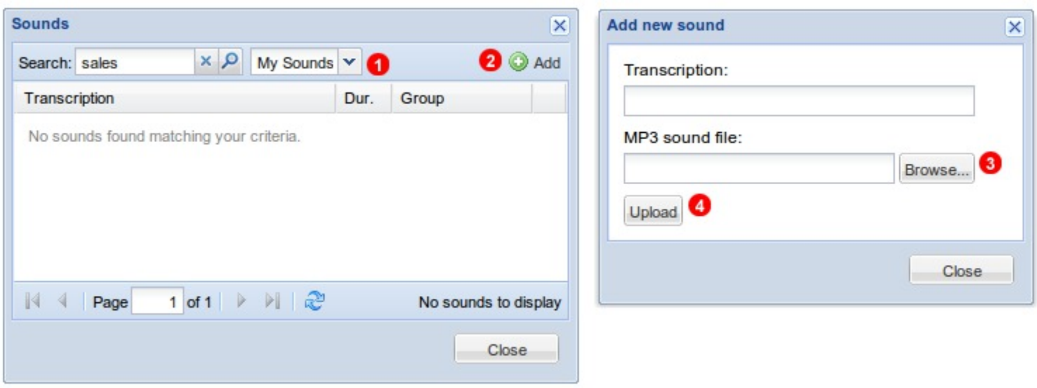


Figure 5.8 Add sound.

Sounds window gives you access to many pre-recorded announcements that can be used to compose your own IVR menus. Here you can upload you own announcements:

1. Select "My Sounds" from the drop down list.
2. Click `Add` button.
3. Select an .mp3 file you want to upload.
4. Check if you want to normalize volume level
5. Click `Upload` button.

Edit sound

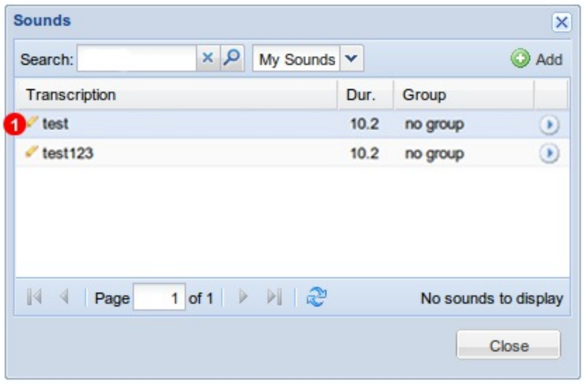


Figure 5.9 Edit sound.

To edit your existing recordings:

1. Click on Edit button of the sound file.
2. You can rename the file from here.
3. You can browse another mp3 and replace the sound file.
4. Check if you want to normalize volume level
5. To delete this file, Click on `Delete`
6. To save your settings, click on `Update` or to cancel, click on `Cancel`.

6. Calls History

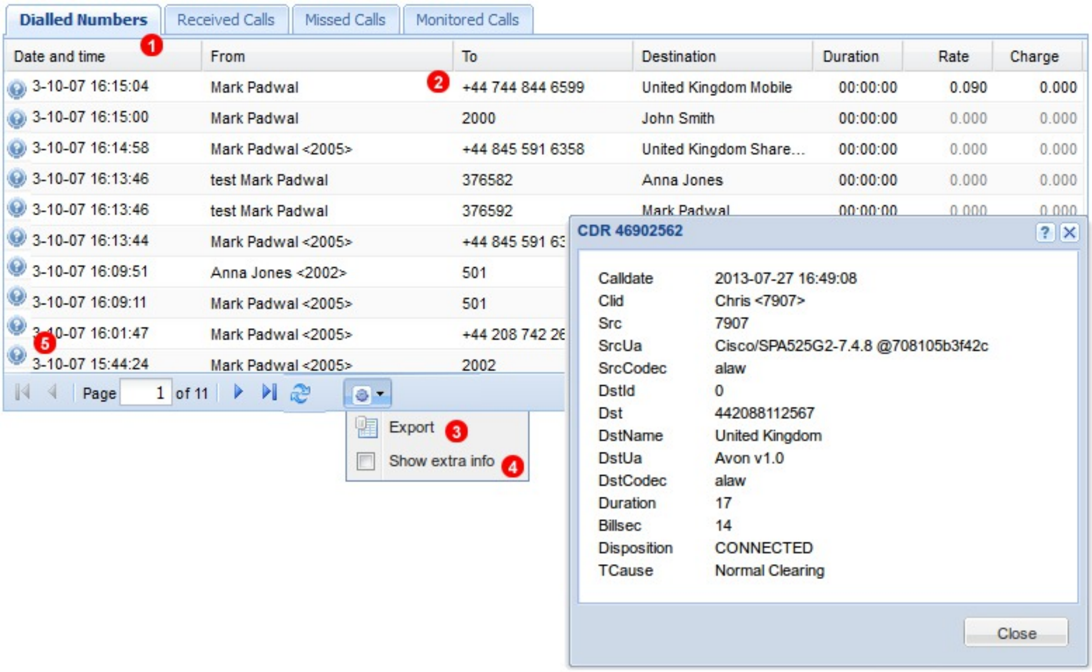


Figure 6.1 Calls History.

In this panel you can view history of all made and received calls. Also here you can obtain detailed billing information about chargeable outbound calls and listen to or download call recordings. To switch between Dialled, Received and Missed Calls use tabs `(1)`. You can use Filters as shown in `(2)` above to search for calls made to/from specific numbers or time ranges.

To view additional information (such as User Agent string and used codecs) select checkbox `(5)` and click icon marked as `(6)` in figure above.

Call Detail Record Export

To export your call details data into Excel file please select `Export` from the context menu as shown in `(3)` above. Please note: export operation is limited to 15 000 records, so you may have to use date range filter to limit number of data exported.

Monitored Calls

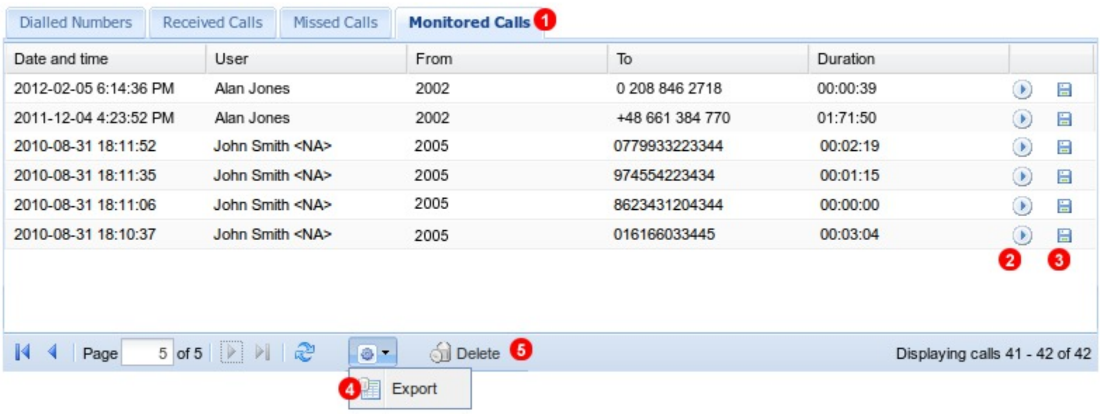


Figure 6.2 Monitored calls.

To view, listen or download your monitored calls (recorded automatically or manually)

1. Click `Monitored Calls` tab.

2. To listen to the recording click `Play` button.
3. To download recording as MP3 file click `Download` button
4. To export the CDR, click on `Export` button
5. To delete, click on `Delete` button

FTP Access

It is also possible to bulk download Monitored Calls MP3 files using secure FTP program. First you need to obtain your FTP login details.

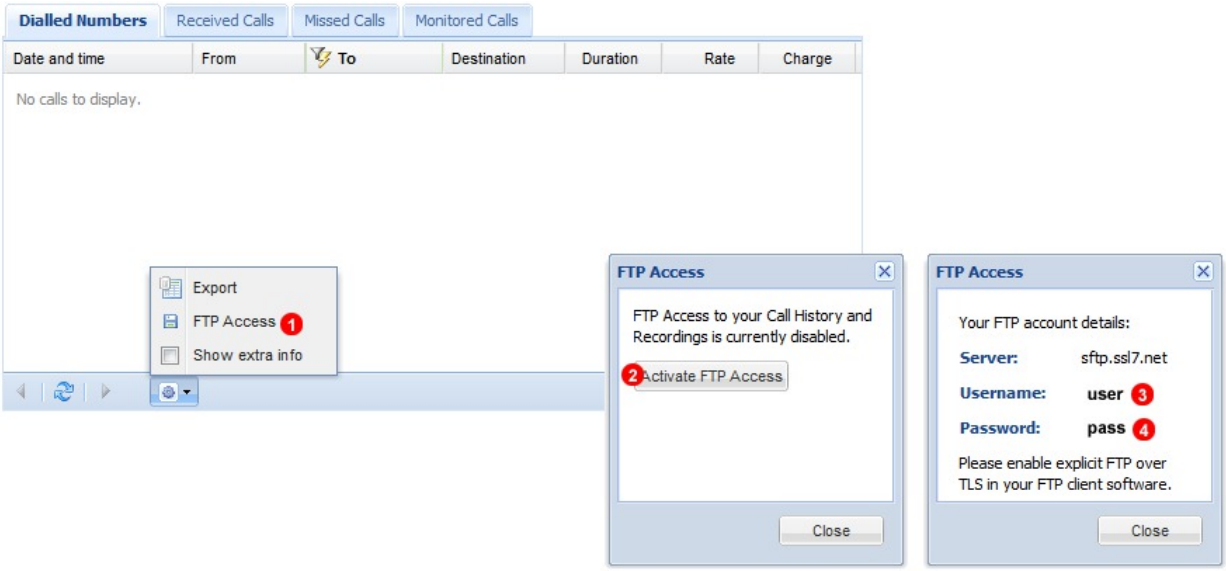


Figure 6.3a Obtaining FTP login details.

1. Click FTP Access
2. Click Activate FTP Access.

Next please follow steps below to configure popular FileZilla http://filezilla-project.org/ FTP client:

1. Select `Site Manager...` from the `File` menu.
2. Enter `sftp.ssl7.net` into `Host` field.
3. Enter `21` into `Port` field.
4. Select `FTP - File Transfer Protocol`.
5. Select `Require explicit FTP over TLS`.
6. Select `Normal` logon type.
7. Enter your username as seen in Figure 6.3a `(3)` into `User` field.
8. Enter your password as seen in Figure 6.3a `(4)` into `Password` field.
9. Click `Connect` button.

Format of recording files as below:

```
YYYY_mm_dd_HH_mm_ss-SIP_USERNAME-CALLER_ID-CALLED_NO-UNIQUEID.mp3
```

Note: recordings are uploaded to FTP site every 24 hours, so you may have to wait until the most recent ones becomes available for bulk download (all recordings are instantly available via web admin panel). Recordings are automatically deleted from FTP site after 30 days.

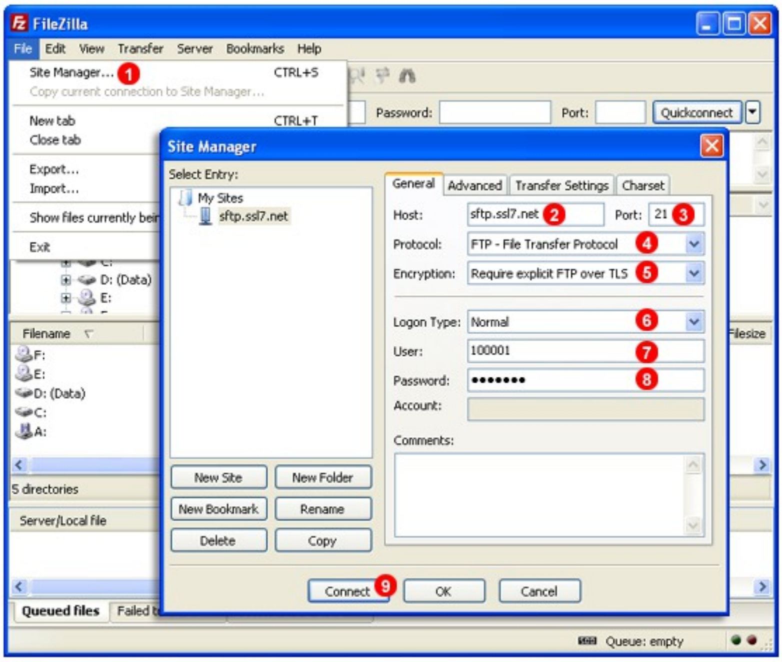


Figure 6.3b Monitored calls - bulk download.

7. Users

Figure 7.1 Users panel.

Here you can view and edit all settings related to your users. You can also import or export list of users from this panel.

Add user

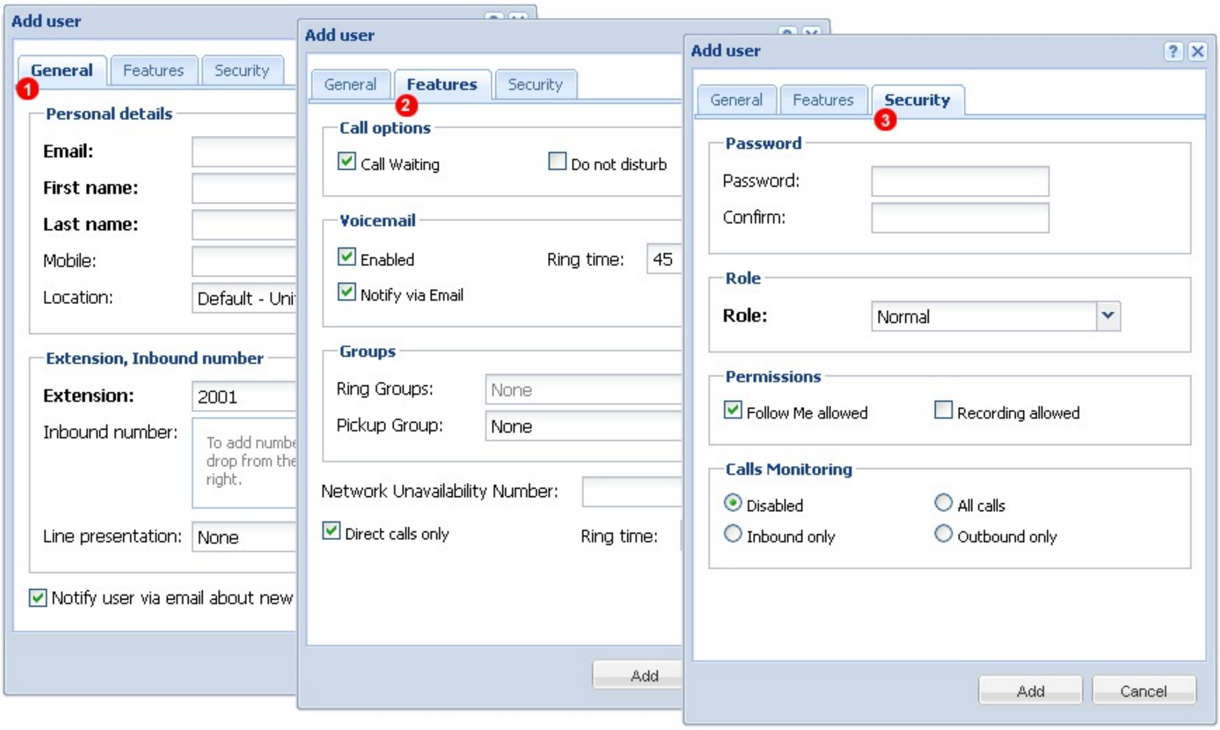


Figure 7.2 Add user.

To add a new user click `Add` button located in top right corner of `Users` panel. The minimum information you need to provide in order to create new user account is: Email, First and Last Name, Extension number. Optionally you can adjust number of additional settings.

If you omit password and confirm password fields, a random password will be automatically generated.

You can assign one of two roles to a new user:

- Administrator - this user will be able to use Administration Control Panel.
- Normal - only Normal User Control Panel available
- Receptions - access to the virtual Reception Console

1. General - In this tab, you need to fill your First & Last Name, Email, Extension. You can drag your Inbound Number here and can set a Line Presentation, Caller ID.
2. Features - You can enable or disable Call Waiting, Do not disturb, Voicemail and set Network Unavailability Number in this tab.
3. Security - In this tab, you can enable or disable Follow Me - call forwarding, Call Monitoring - call recording. Using this tab, you can reset user password for VoIP Studio login.

Edit user

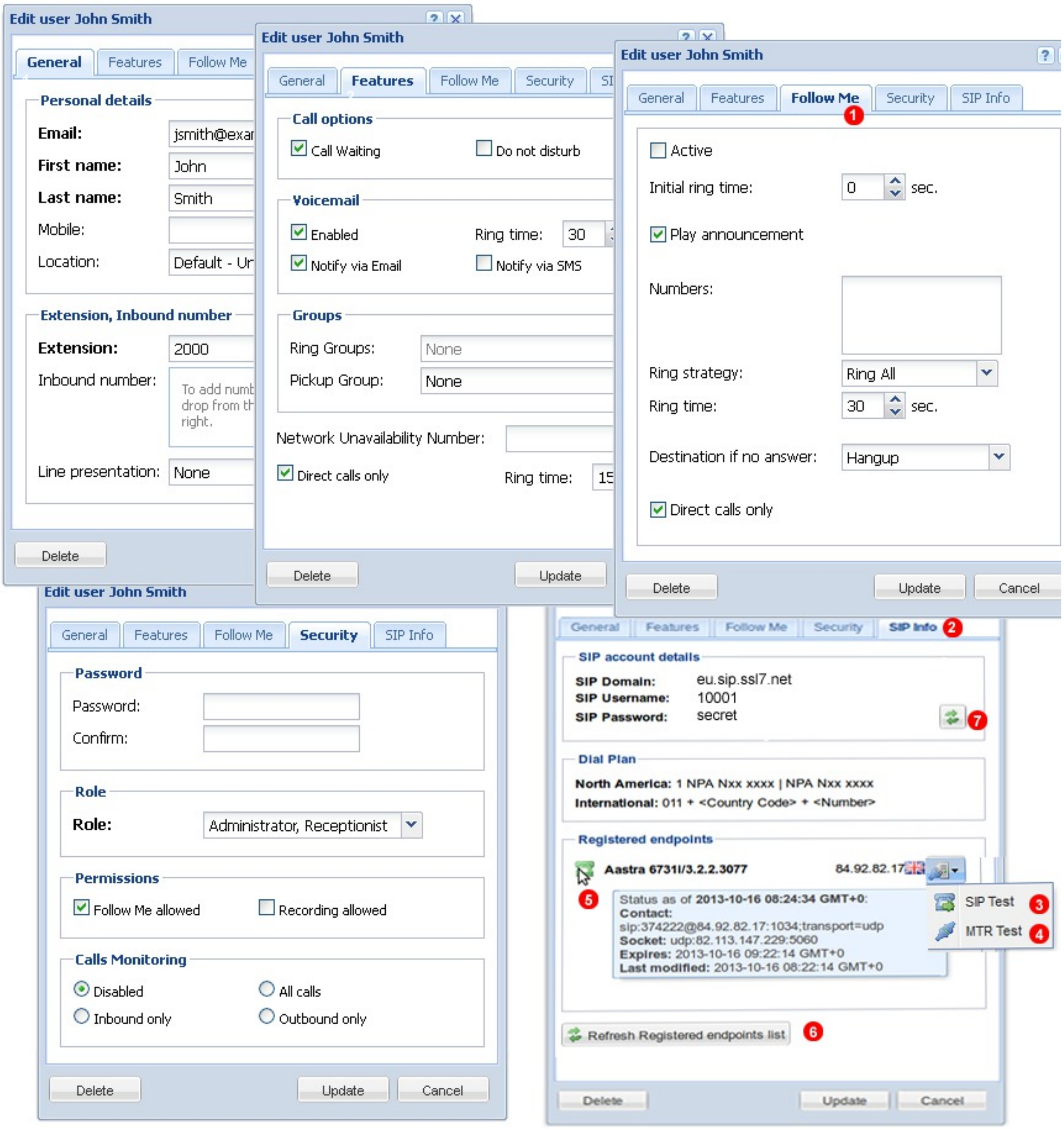


Figure 7.3 Edit user.

To edit user settings click icon next to the user name in `Users` panel - see `(1)` in Figure 7.1 above. 1. General - You can edit Personal details and Extension/Inbound number from this tab. 2. Features - You can edit features available to user such as: Call Waiting, DND, Voicemail, Ring/Pickup Groups. 3. Follow Me - You can activate 'Follow Me' from this tab. 'Initial ring time' allows the incoming call to ring at your phone for selected duration and then gets forwarded to the assigned number. 4. Click edit icon to change your announcements settings: a. Here you can select which announcement should to be played. b. Click play button to listen to currently selected announcement. c. Enter description of the file you want to add. d. Select an .mp3 file you want to upload. e. Click `Upload` button. 5. Security - You can edit security settings such as password, user permissions, call monitoring from this tab. 6. SIP Info - from this tab, you can get your SIP credentials to set up your soft or hard phone manually. You can run a SIP and MTR test from the button 'SIP Test' & 'MTR Test' to check connectivity with VoIP Studio. You can refresh your Phone configurations by clicking on 'Refresh Registered endpoint list' button located at bottom left side of the tab. 7. You can reset your password by clicking the refresh button. Please make sure, if you have reset your password, you need to update that password to all your phones where the user account is set. 8. If you hoover your cursor here, you can check status of your endpoint along with last modified date. 9. SIP Test: This test helps to understand if your SIP endpoints have registered properly. 10. MTR Test: MTR Test checks the connectivity of network. It checks connectivity between endpoint and data centre. 11. By clicking on `Refresh Registered endpoint list`, you can refresh the status of your endpoint.

Import users

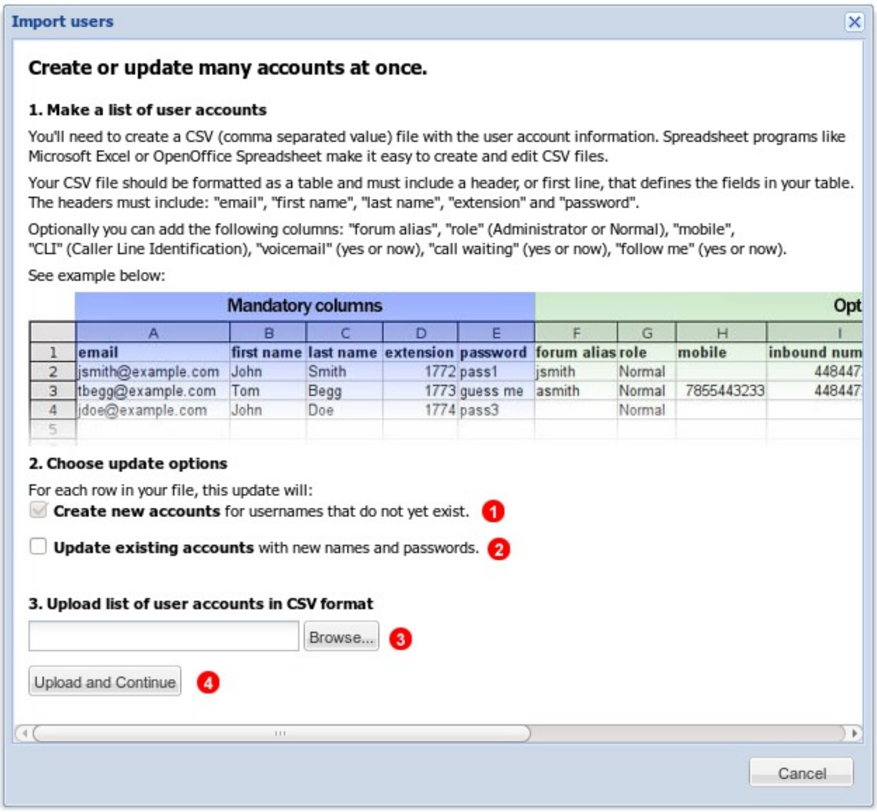


Figure 7.4 User import.

To import users click gear icon in top right corner of `Users` panel - see `(3)` in Figure 7.1 above - and select `Import` from the menu. You will need to create a CSV (comma separated value) file with the user account information. Spreadsheet programs like Microsoft Excel or OpenOffice Spreadsheet make it easy to create and edit CSV files. Your CSV file should be formatted as a table and must include a header, or first line, that defines the fields in your table. The headers must include: "email", "first name", "last name", "extension" and "password". Optionally you can add the following columns: "forum alias", "role" (Administrator or Normal), "mobile","CLI" (Caller Line Identification), "voicemail" (yes or now), "call waiting" (yes or now), "follow me" (yes or now).

Export users

To export your users list to CSV (comma separated value) file click gear icon in top right corner of `Users` panel - see `(3)` in Figure 7.1 above - and select `Export` from the menu.

8. Inbound numbers

Figure 8.1 Inbound numbers panel.

Inbound numbers allow to make calls from traditional telephone network to your hosted VoIP systems. We can assign telephone numbers from more than 4000 cities around the world.

Add inbound number

Figure 8.2 Add inbound number.

To add an Inbound Number:

1. Click `Add` button in the `Inbound numbers` panel.
2. Select country and city in which your telephone number should be located.
3. Enter desired amount of number you want to purchase.
4. Click `Add to basket` button.
5. Click `Checkout` to complete the purchase process.

Edit inbound number

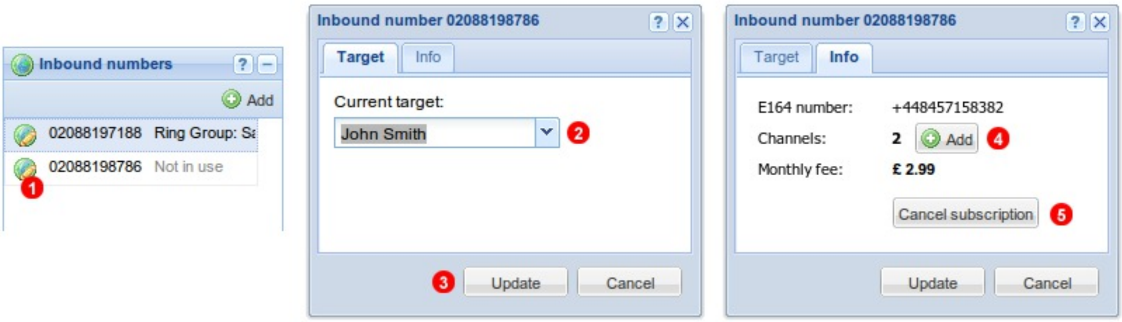


Figure 8.3 Edit inbound number.

To change Inbound Number settings:

1. Click on the icon next to the DDI number.
2. Select number's target which can be a User, Ring Group, IVR, Queue or a virtual Conference Room.
3. Click `Update` button.

To purchase additional channels (that will allow to answer more concurrent calls):

1. Use `Add` button.

If you not longer wish to use a DDI number:

1. Use `Cancel subscription` button.

9. Queues

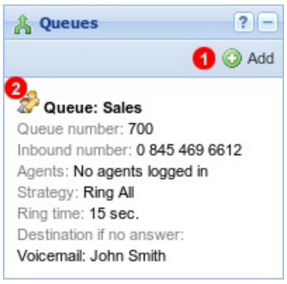


Figure 9.1 Queues panel.

Queues allow to automatically distributes phone calls to a specific group of agents on the first come, first serve basis. Using VoIP Studio Control Panel you can adjust maximum queue times, toggle whether callers are told their queue position and average wait time.

Add Queue

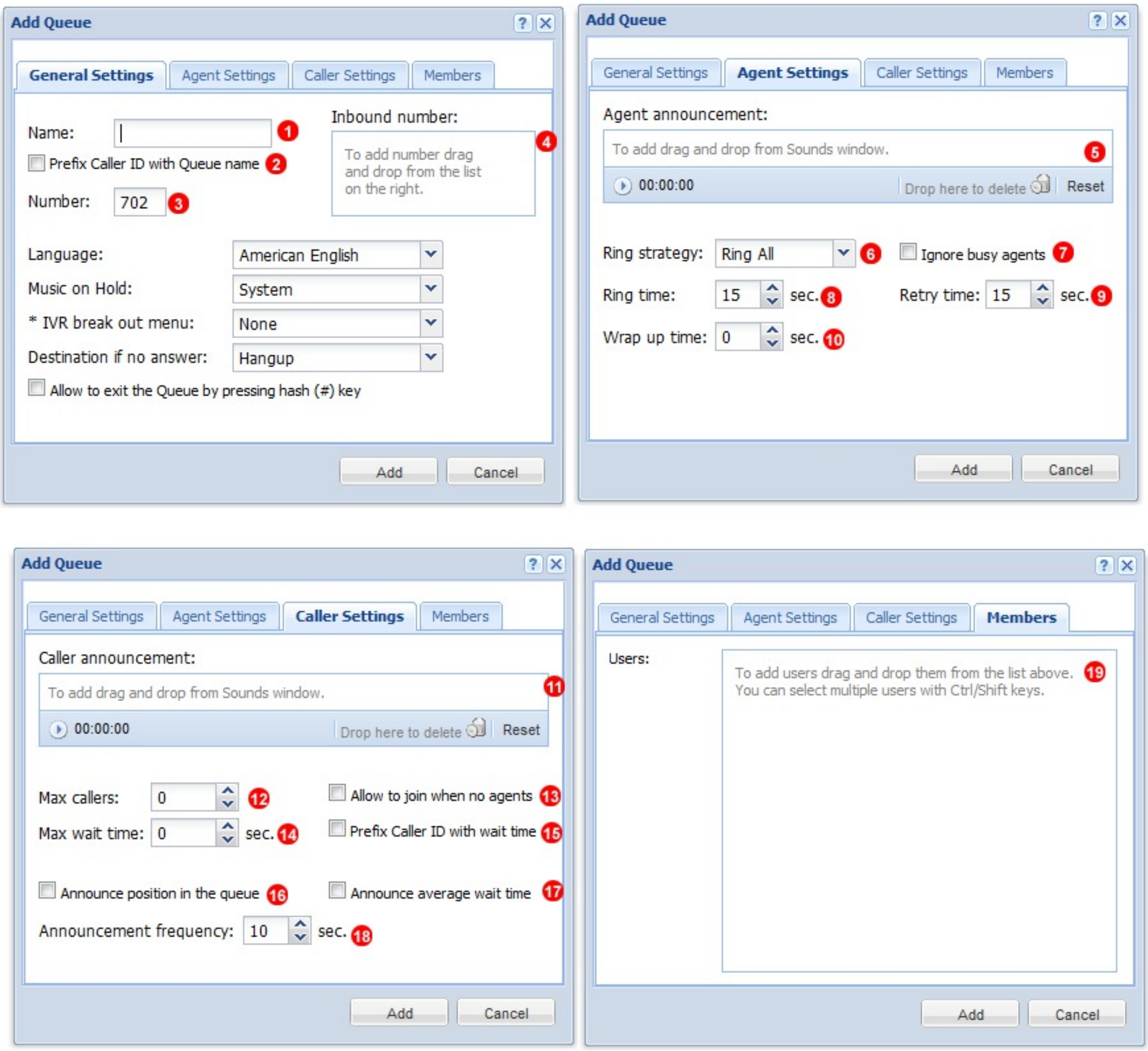


Figure 9.2 Add Queues.

To create a new Queue click `Add` button located in top right part of `Queues` panel - see `(1)` in Figure 9.1 above. Next follow steps below:

General Settings

1. Enter Queue name - it will be used as Caller ID prefix if you select checkbox `(2)` below. Max up to 45 characters long and only letters and numbers allowed.
2. Prefix Caller ID with Queue name - If enabled Caller ID will be prefixed with queue name to help agents identify which queue a call is coming from.
3. Number - Agents will use the following dial codes to login and log out of the queue:
   - Log in - 421# followed by queue number
   - Log out - 422# followed by queue number
4. Inbound number - (DDI) assigned to a queue.
5. Language: British English, American English, Portuguese, Spanish.
6. Music on-hold: Selectable by uploaded file.
7. IVR break out of menu - Allows call to break out of original IVR.
8. Destination if no answer: Allows send call to another queue, IVR or extension voice-mail.
9. Allow to exit the que by pressing #: Allows caller to exit queue.

Agent Settings

1. Here you can set announcement played to the agent prior to connecting a call.
2. Ring Strategy when calling available agents.
3. Ignore busy agents: If enabled the system will not ring agents already on the call.
4. Ring Time: How long to ring each agent before we consider it time out.
5. Wrap up time: After a successful call, how many seconds to wait before sending a potentially free agent another call. Retry time + Wrap up time.

Caller Settings

Requires updating 10. After a successful call, how many seconds to wait before sending a potentially free agent another call. 11. Announcement played to the caller prior to joining the queue. 12. Maximum number of callers allowed in the queue. 13. If enabled allow callers to join the queue even when no agents are currently present. 14. The maximum number of seconds caller can wait in the queue. 15. If enabled Caller ID will be prefixed with total wait time in minutes. 16. Announce position of the caller in the queue. 17. Announce average wait time in the queue. 18. How often to announce position and/or wait time. 19. You can drag and drop users you want to assign in this Queue.

Edit Queue

To edit a Queue click on the icon located on the left side of the Queue name. See `(2)` in Figure 9.1 above.

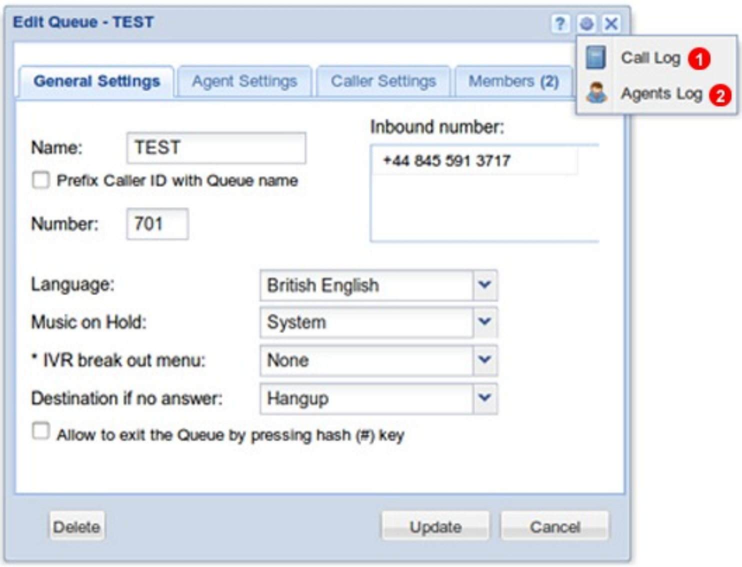


Figure 9.3 Edit Queue Settings.

Click on `Settings` gear located on top right corner of the Edit Queue window.

1. Click `Call Log` to see all logs.

2. Click `Agents Log` to see agent logs.

Call Log

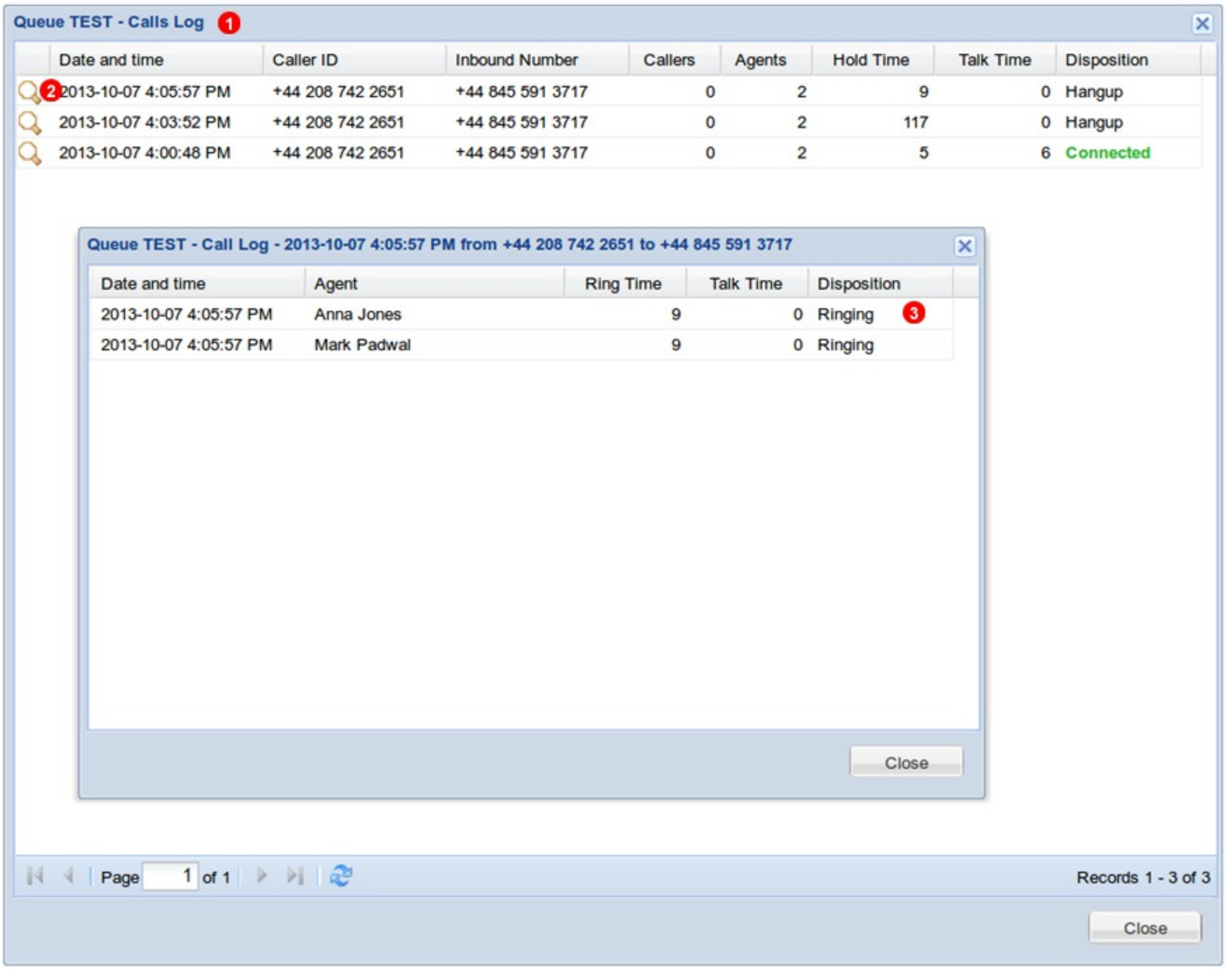


Figure 9.4 Call Log

1. you can obtain Call logs with date & time, Caller ID, Inbound number and status of calls in this window.
2. Click on the `Search` icon to know more details.
3. Here you can see Agent names and other details.

Agents Log

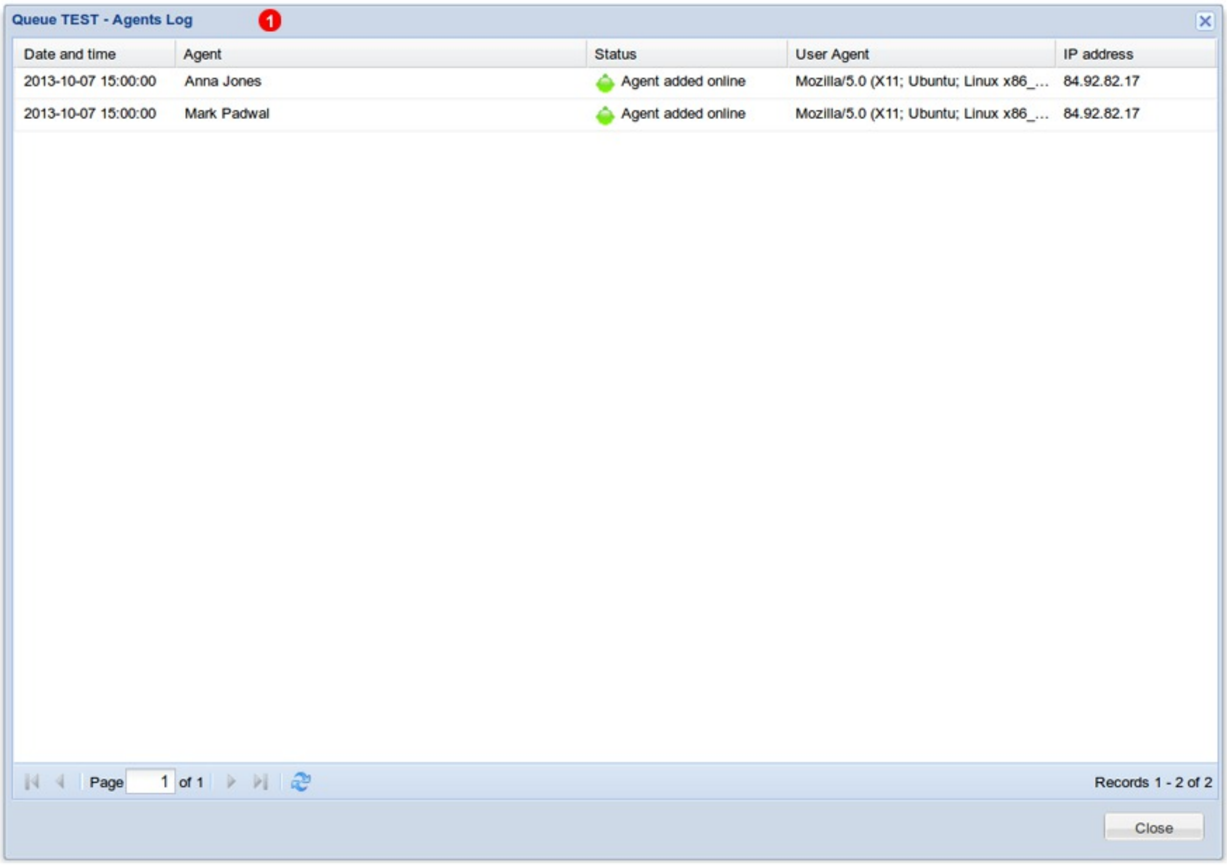


Figure 9.5 Agent Log

In this window, you can see online agents.

10. Ring Groups

Figure 10.1 Ring Groups panel.

Ring Groups allow to link several extensions into a group with a single inbound number (DDI) assigned to it.

Add Ring Group

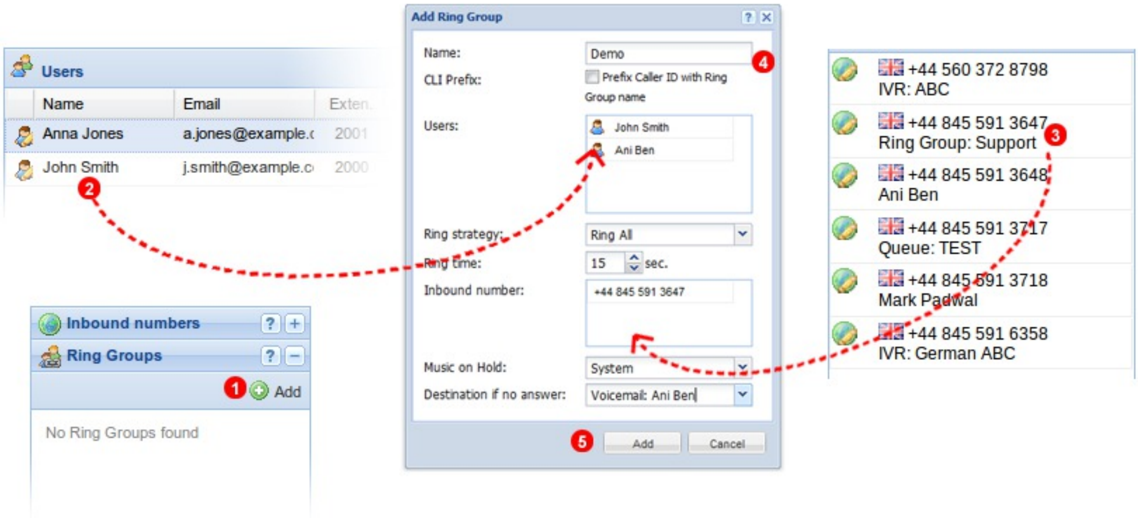


Figure 10.2 Add Ring Group.

To create a new Ring Group:

1. Click Add button located in Ring Groups panel.
2. Drag and drop users you want to assign to this Ring Group.
3. Drag and drop inbound number you wish to assign to this Ring Group.
4. Enter Ring Group name.
5. Click Add button.

Edit Ring Group

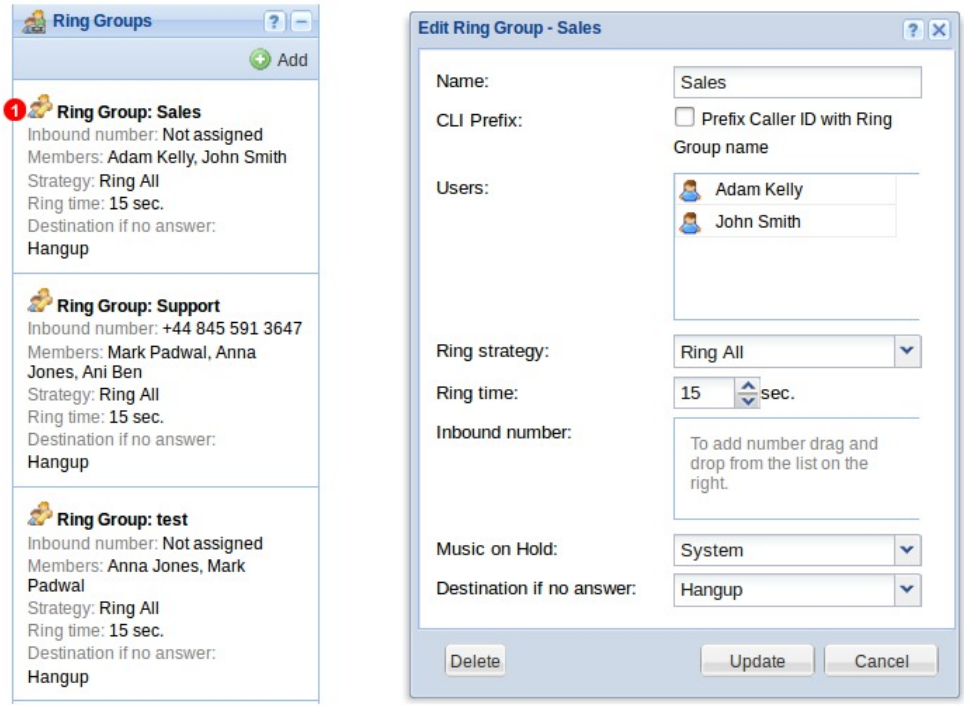


Figure 10.3 Edit Ring Group.

To edit a Ring Group click on the icon located on the left side of the Ring Group name.

11. Pickup Groups

Pickup groups allow users within the same group to answer each other calls. If a colleague's telephone is ringing, one can answer that call by picking up one’s own set and dialing `**` (star key twice), instead of walking to the colleague’s desk.

Add Pickup Group

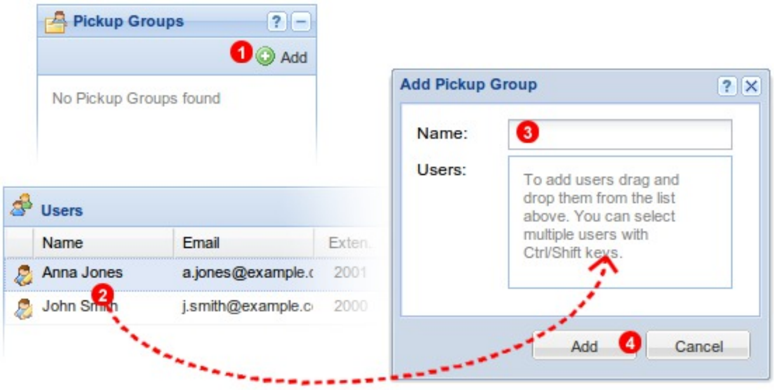


Figure 11.1 Add Pickup Group.

To create a new Pickup Group:

1. Click `Add` button located in the `Pickup Groups` panel.
2. Drag and drop members of your new group from `Users` panel.
3. Enter a pickup group name.
4. Click `Add` button.

Edit Pickup Group

Figure 11.2 Edit Pickup Group.

To edit Pickup Group settings click on the icon `(1)` located on the left to the group name.

12. IVR

Interactive Voice Response system can respond with pre-recorded audio messages to further direct callers on how to proceed. IVR systems can be used to control almost any function where the interface can be broken down into a series of simple menu choices. The use of IVR and voice automation enables a company to improve its customer service and lower costs, due to the fact that callers' queries can be resolved without the cost of a live agent who, in turn, can be directed to deal with specific areas of the service. If the caller does not find the information they need, or require further assistance, the call is then transferred to an agent who can deal with them directly.

Add IVR

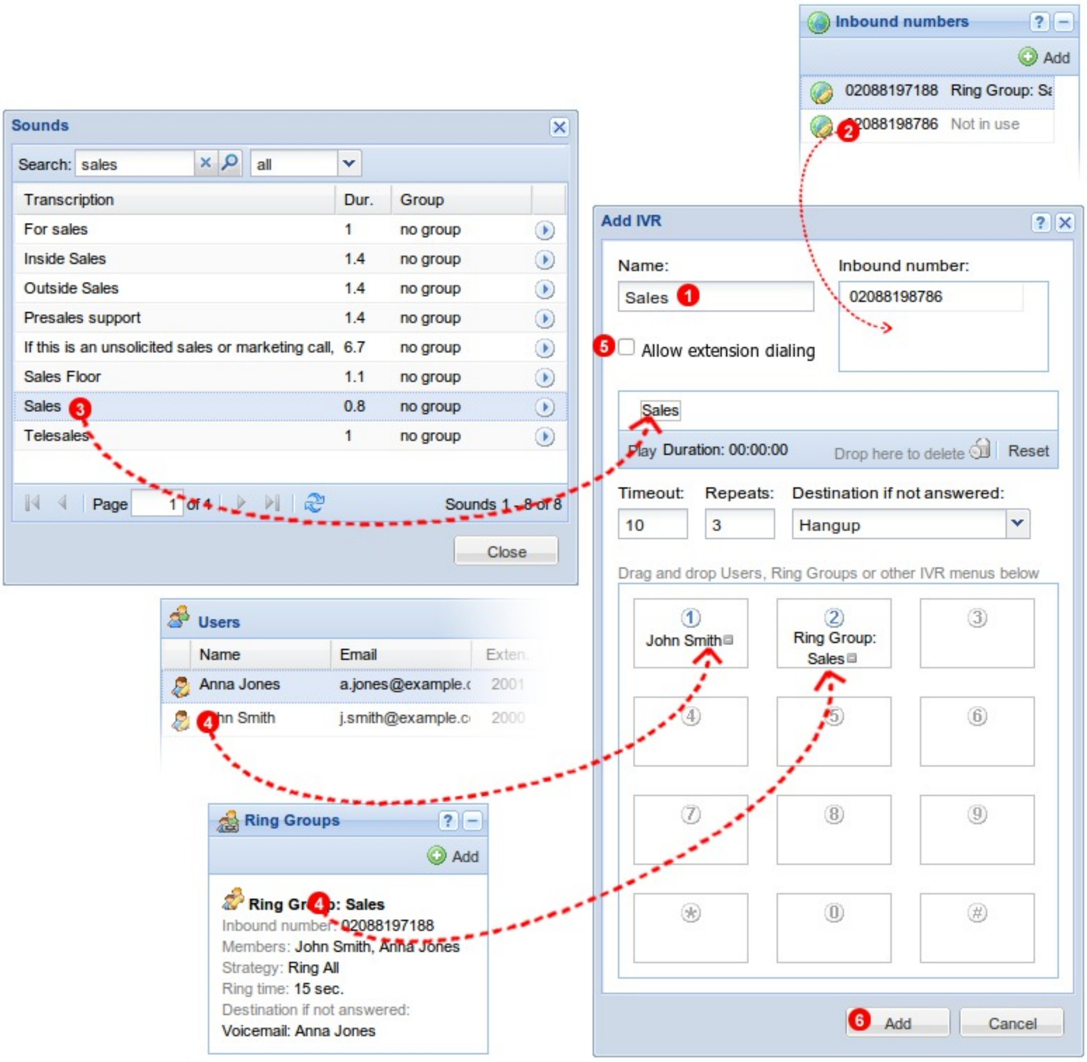


Figure 12.1 Add IVR.

To create a new IVR menu click `Add` button located in `IVR` panel and follow steps below:

1. Enter IVR menu name
2. Drag and drop Inbound Number you wish to assign to this IVR menu.
3. Compose your announcement by dragging and dropping sounds from `Sounds` window. You can also upload you own custom announcement - see Sounds chapter below.
4. Assign targets to keypad digits by dragging and dropping: Users, Ring Group or other IVR menus (by chaining IVRs you can easily create a multi-level menus).
5. Tick 'Allow extension dialling' if you want calls to be passed on user extensions.
6. Finally click `Add` button.

Edit IVR

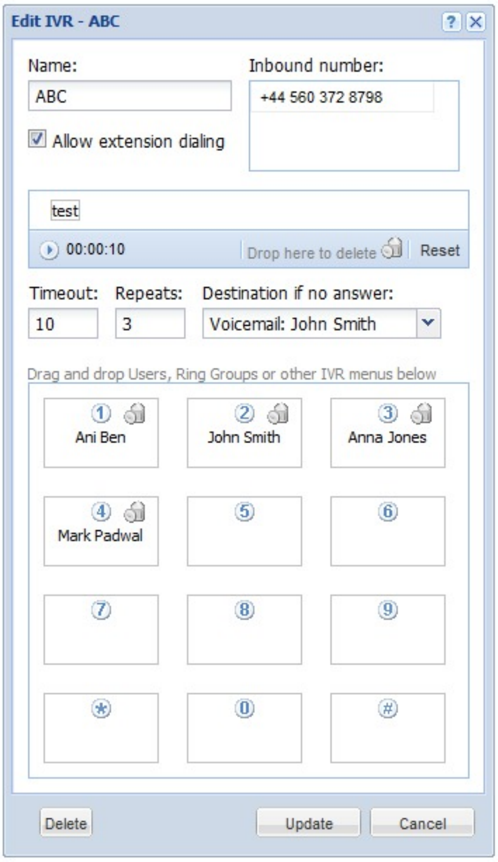


Figure 12.2 Edit IVR.

To edit IVR menu click on the icon located on the left to the menu's name.

13. Conferences

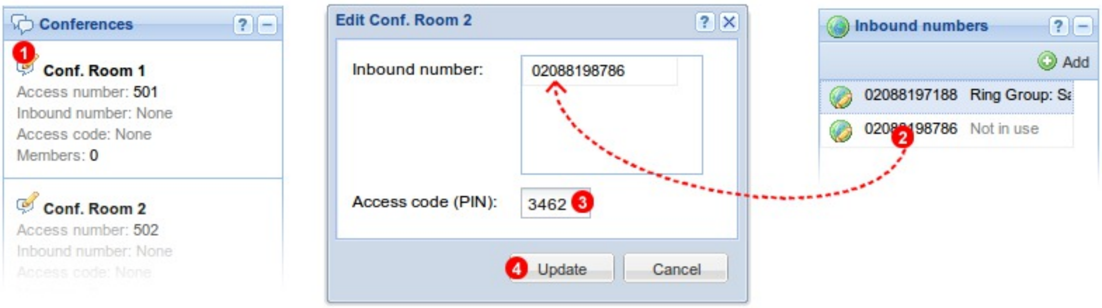


Figure 13.1 Edit conference.

VoIP Studio comes with five Virtual Conferene Rooms. Each user can dial 50N where N is a number of a conference room (from 1 to 5) to join the conference. For example to join Conference Room 1 please dial 501.

You can also assign Inbound Number(s) to your virtual Conference Rooms, so users of PSTN (traditional telephones) can dial into it. To edit a Conference Room settings:

1. Click on the icon located on the left of Conference Room's name.
2. Drag and drop Inbound Number you want to assign to this Conference Room.
3. Enter four digit PIN code that will be required to join this Conference Room.
4. Click `Update` button.

14. Phones

Here you can manage VoIP phones assigned to your account. Once a VoIP phone is registered with your hosted PBX you will be able to easily assign users to particular phone. All phones purchased via your Control Panel (see `Buy` chapter below) will be automatically added to your account. It is also possible to add phones (selected models only) purchased from other vendors.

Buy

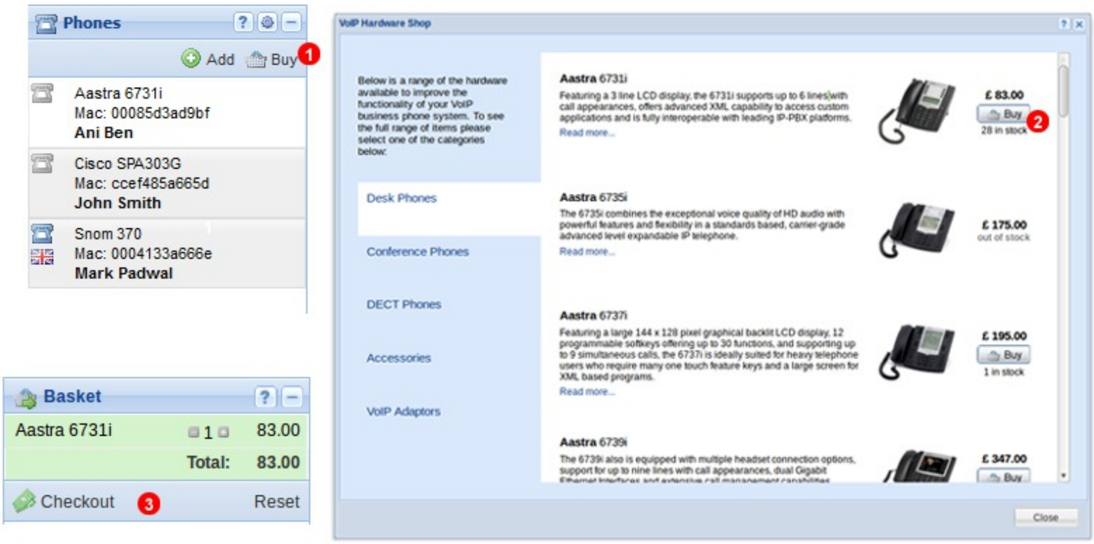


Figure 14.1 Buy phone.

To buy an Internet (VoIP) phone:

1. Click `Buy` button in `Phones` panel or `Shop` located in the header area.
2. Click `Buy` button next to the device you wish to purchase.
3. Click `Checkout` button to complete transaction.

Add

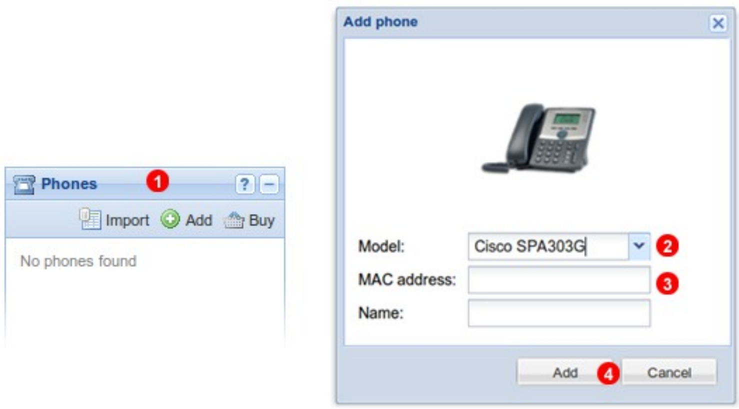


Figure 14.2 Phone add.

To add a phone purchased from the third party vendor:

1. Click `Add` button located in `Phones` panel.
2. Select your device model.

3. Enter your device MAC address - five groups of digits and letters separated by a colon sign for example: `00:1a:4d:5c:25:8d`. Usually printed on the label located at the back of the device.
4. Click `Add` button.

Once a supported devices is added to our system you can obtain auto-provisioning URL, enter it into device settings and manage the phone via your web based control panel.

Edit

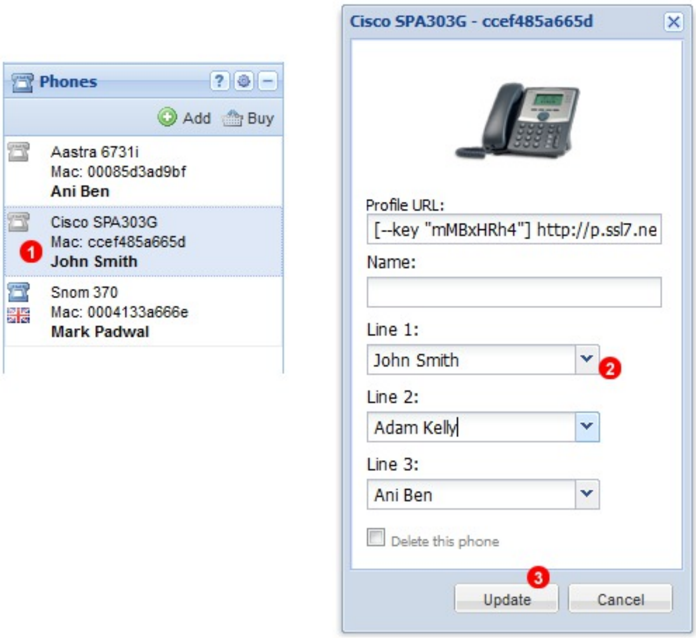


Figure 14.3 Edit phone.

To change user(s) assigned to a particular phone (or phone line) open phone's settings window:

1. Click on the icon located on the left to Phone's name.
2. Select a user you wish to assign to this phone.
3. Click `Update` button.

After a short while device will obtain new configuration profile from VoIP Studio central provisioning server and re-assign phone lines accordingly.

Import

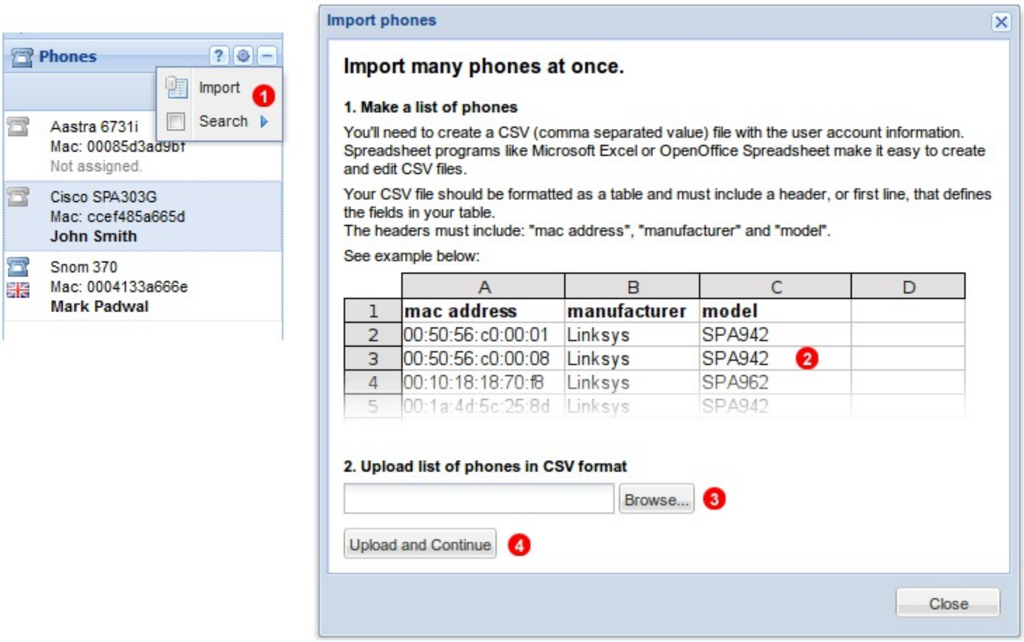


Figure 14.4 Phone import.

To import many phones purchased from a third party vendor at once:

1. Click `Import` button located in `Phones` panel.
2. Prepare a list of phones you want to import as a .CSV (Comma Separated Value) file. Your file must include the following headers: "mac address", "manufacturer" and "model".
3. Select the file you want to upload.
4. Click `Upload and Continue` button.

Factory Reset

If you have not purchased the phone from VoIP Studio, it is always best to perform a factory reset to avoid any misconfiguration.

Aastra

Figure 14.5 Aastra - Factory reset

1. Press the `settings` key of your phone.
2. Scroll to Admin menu.
3. Provide password (by default 22222).
4. Scroll down.
5. `Factory Default`.
6. Press `Enter`.

Cisco PAP2 Adapter

Figure 14.6

1. Dial `****` from your phone.
2. Dial `73738#`.
3. Dial `1`.

Cisco SPAXXX

Figure 14.7 Cisco - Factory reset

1. Press `Settings` button in phone.
2. Scroll down.
3. Press `Factory Reset` [Number 14] and press `enter`.

Polycom

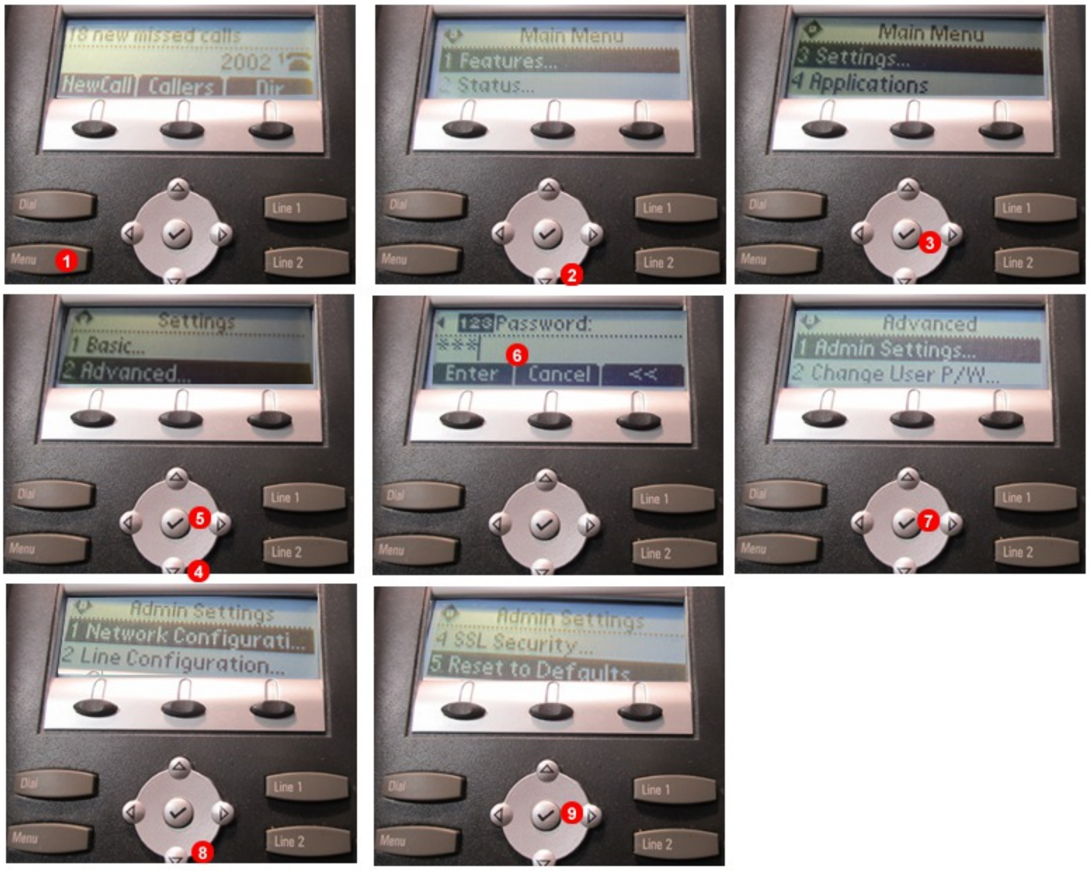


Figure 14.8 Polycom - Factory reset

1. Click on `Menu`.
2. Scroll down to `Settings`.
3. Press `ok`.
4. Scroll down to `Advanced`.
5. Press `ok`.
6. Type password, by default 456.
7. Press `ok` to `Admin Settings`.
8. Scroll down to `Reset to Defaults`.
9. Press `ok`.

SNOM

Figure 14.9 SNOM - Factory reset

Please press: `Volume up + Volume low - #`

Yealink

Figure 14.10 Yealink - Factory reset

1. Press `Settings` button in phone.
2. Scroll down.
3. Press `3` for Settings.
4. Press `2` for Advanced.
5. Type your `Password`.
6. Scroll down.
7. Select `5` for Reset Factory.
8. Press `ok`.

Find phone IP Address

You will need to enter IP Address http://URL where URL stands for IP Address of your phone which can be found below.

Aastra

Figure 14.11 Aastra - Find out phone IP

1. Press `settings` key in the phone.
2. Scroll down the menu and select `Phone Status`.
3. Press `Enter`.

4. Press Enter to `IP&MAC Addresses`.
5. Here you can see `IP&MAC Addresses` of the phone.

Cisco PAP2 Adapter

Figure 14.12 Cisco PAP2 - Find out IP

1. Dial `****`.
2. Dial `110#`.
3. Your IP address would be prompted.

Cisco SPAXXX

Figure 14.13 Cisco - Find out phone IP

1. Press `Settings` button in phone.
2. Scroll down.
3. Select `Network` [Number 9].
4. Find your IP Address here.

Polycom

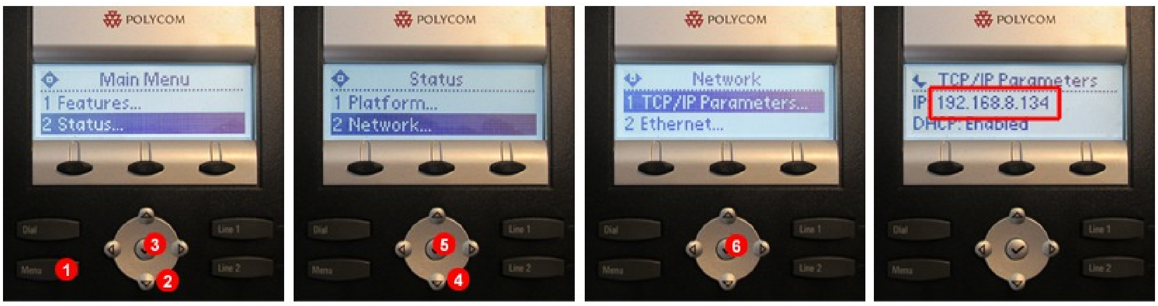


Figure 14.14 Polycom - Find IP address

Please follow steps shown above to find out current IP address assigned to your Polycome phone.

SNOM

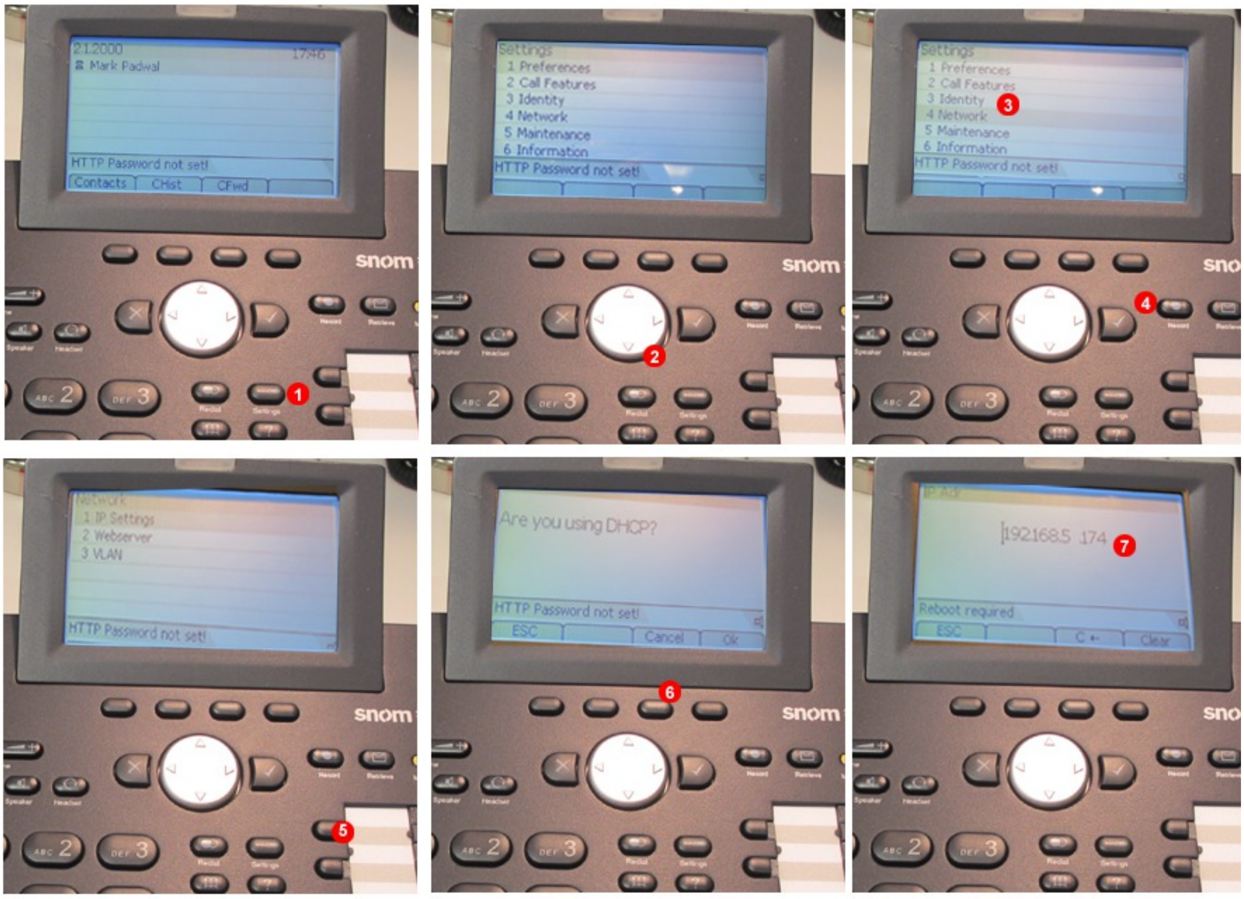


Figure 14.15 SNOM - Find out phone IP

1. Press `Settings` button in phone.
2. Scroll down.
3. Select `Network` [Number 4].
4. Press `enter`.
5. Select `IP settings`.
6. Press `Cancel` for DHCP.
7. Find your IP here.

Yealink

Figure 14.16 Yealink- Find phone IP

1. Press `Settings` button in phone.
2. Press `ok`.
3. Find your IP here.

Auto provisioning

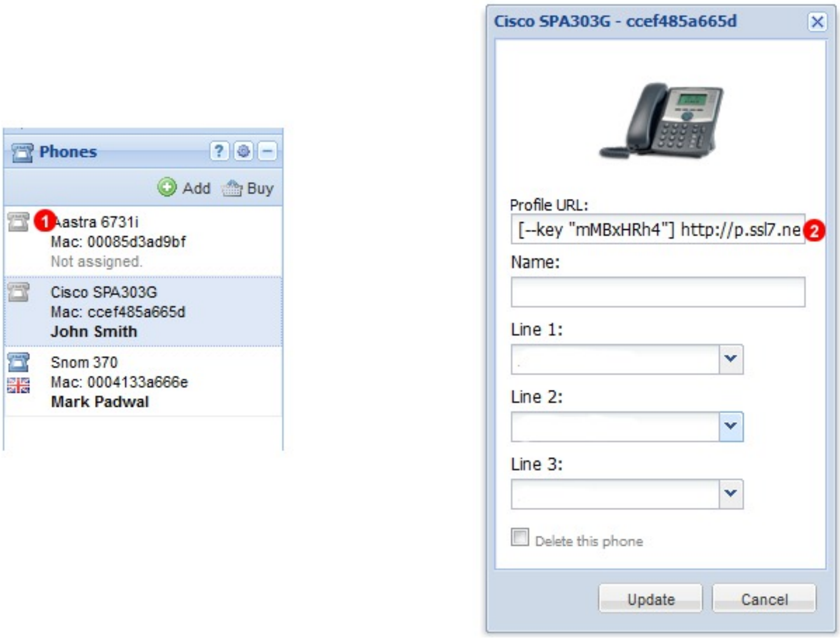


Figure 14.17 Phone Profile URL.

Selected models of VoIP phones can be centrally managed via your VoIP Studio admin panel thanks to auto provisioning feature. Phones purchased from VoIP Studio will be automatically added to your admin panel and set up for auto provisioning. To enable auto provisioning of phones from other vendors you will need to obtain `Profile URL` - see `(2)` in Figure 14.6 above and follow instructions for your device which can be found below.

Aastra

If your phone was purchased from VoIP Studio, it has been already pre configured and there is no need to provision it. Please refer to Phone edit section to assign extensions. If your phone was not purchased from VoIP Studio, please perform a Factory Reset described in Factory Reset section above and Firmware upgrade.

You can obtain your phone IP by following the steps described in Find phone IP section above.

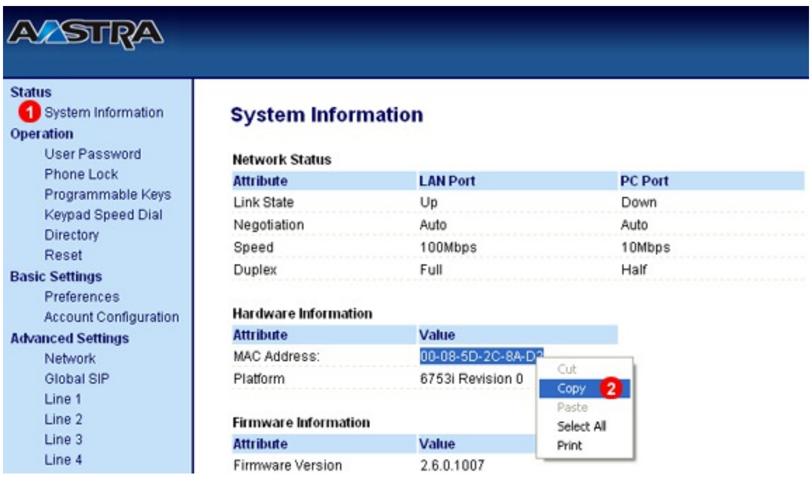


Figure 14.18 Aastra phone auto provisioning - MAC address

To auto provision Aastra phone login into web interface of the device and follow steps below:

1. Click `System information` link in the left sidebar.
2. Copy `MAC Address` into a clipboard.

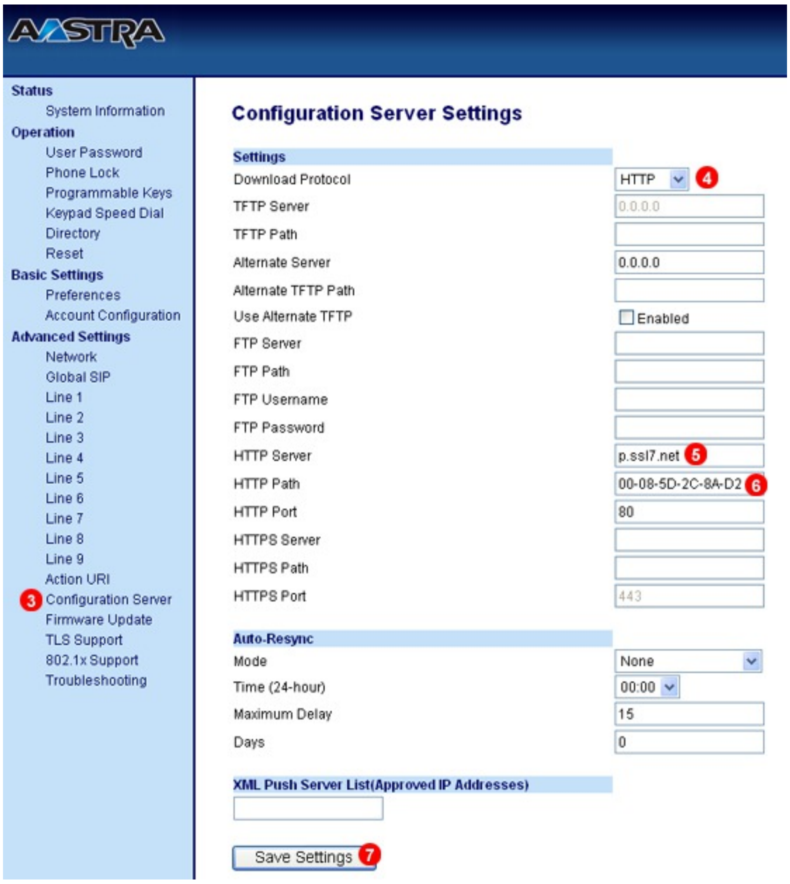


Figure 14.19 Aastra phone auto provisioning - Configuration Server

1. Click `Configuration Server` link in the left sidebar.
2. Select `HTTP` as `Download Protocol`.
3. Enter `p.ssl7.net` into `HTTP Server` field.
4. Paste MAC Address copied in step 2 into `HTTP Path` field.
5. Click `Save Settings` button.

Figure 14.20 Aastra phone auto provisioning - Reset

1. Click `Reset` link in left sidebar.
2. Click `Restart` button.

After a short while your Aastra phone will reboot and obtain configuration data from VoIP Studio provisioning service.

Cisco PAP2 Adapter

If your phone was purchased from VoIP Studio, it has been already pre configured and there is no need to provision it. Please refer to Phone edit section to assign extensions. If your phone was not purchased from VoIP Studio, please perform a Factory Reset described in Factory Reset section above and Firmware upgrade.

You can obtain your phone IP by following the steps described in Find phone IP section above.

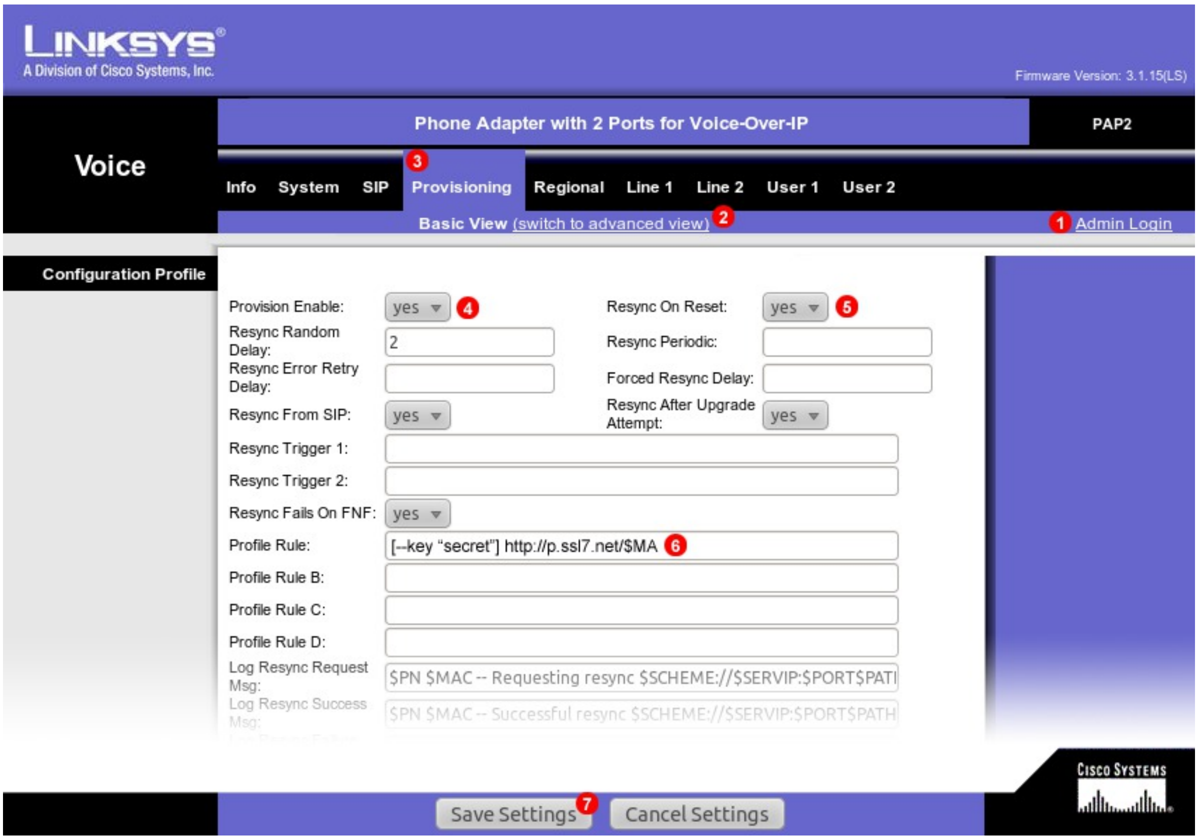


Figure 14.21 Cisco PAP2 Adapter auto provisioning

To auto provision Cisco PAP2 login into web interface of the device and follow steps below:

1. Click `Admin Login` link at the top of the page.
2. Click `Advanced` link at the top right part of the page.
3. Select `Provisioning` tab at the top.
4. Make sure `Provision Enable` is set to `Yes`.
5. Make sure `Resync On Reset` is set to `Yes`.
6. Copy and paste your `Profile URL` (see `(2)` in Figure 14.6 above) into field `(6)`.
7. Click `Save Settings` button.

After a short while your Cisco VoIP adapter will reboot and obtain configuration data from VoIP Studio provisioning service.

Cisco SPAXXX

If your phone was purchased from VoIP Studio, it has been already pre configured and there is no need to provision it. Please refer to Phone edit section to assign extensions. If your phone was not purchased from VoIP Studio, please perform a Factory Reset described in Factory Reset section above and Firmware upgrade.

You can obtain your phone IP by following the steps described in Find phone IP section above.

This instructions apply to all Cisco SPAXXX service devices.

Figure 14.22 Cisco SPA525G auto provisioning

To auto provision Cisco SPAXXX login into web interface of the device and follow steps below:

1. Click `Admin Login` link at the top of the page.
2. Click `Advanced` link at the top right part of the page.
3. Select `Provisioning` tab at the top.
4. Make sure `Provision Enable` is set to `Yes`.
5. Make sure `Resync On Reset` is set to `Yes`.
6. Copy and paste your `Profile URL` - see `(2)` in Figure 13.6 above - into field `(6)`.
7. Click `Submit All Changes` button.

After a short while your Cisco phone will reboot and obtain configuration data from VoIP Studio provisioning service.

Grandstream HandyTone 286

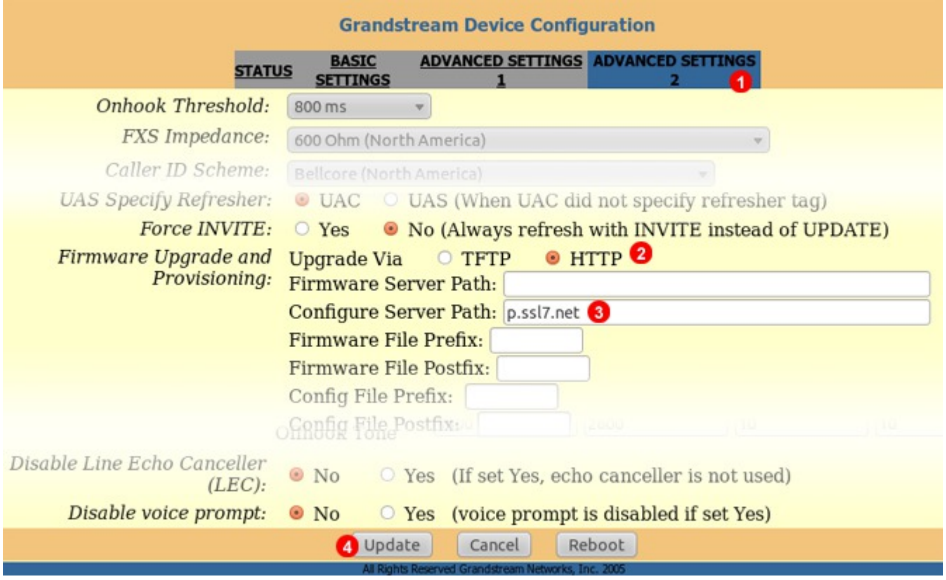


Figure 14.23 Grandstream HandyTone 286 auto provisioning

Firstly you need to find out IP address assigned to your HandyTone 286 device. Pick up the handset and press the button on the HT–286 or dial `***` to use the IVR menu. Next dial `02` and note IP address provided.

To auto provision Grandstream HandyTone 286 login into web interface of the device and follow steps below:

1. Click `Advanced Settings 2` tab at the top of the page.
2. Select `HTTP` provisioning method.
3. Enter `p.ssl7.net` as Configuration Server Path.
4. Click `Update` button at the bottom of the page.

After a short while your Grandstream adapter will reboot and obtain configuration data from VoIP Studio provisioning service.

Snom

If your phone was purchased from VoIP Studio, it has been already pre configured and there is no need to provision it. Please refer to Phone edit section to assign extensions. If your phone was not purchased from VoIP Studio, please perform a Factory Reset described in Factory Reset section above and Firmware upgrade.

You can obtain your phone IP by following the steps described in Find phone IP section above.

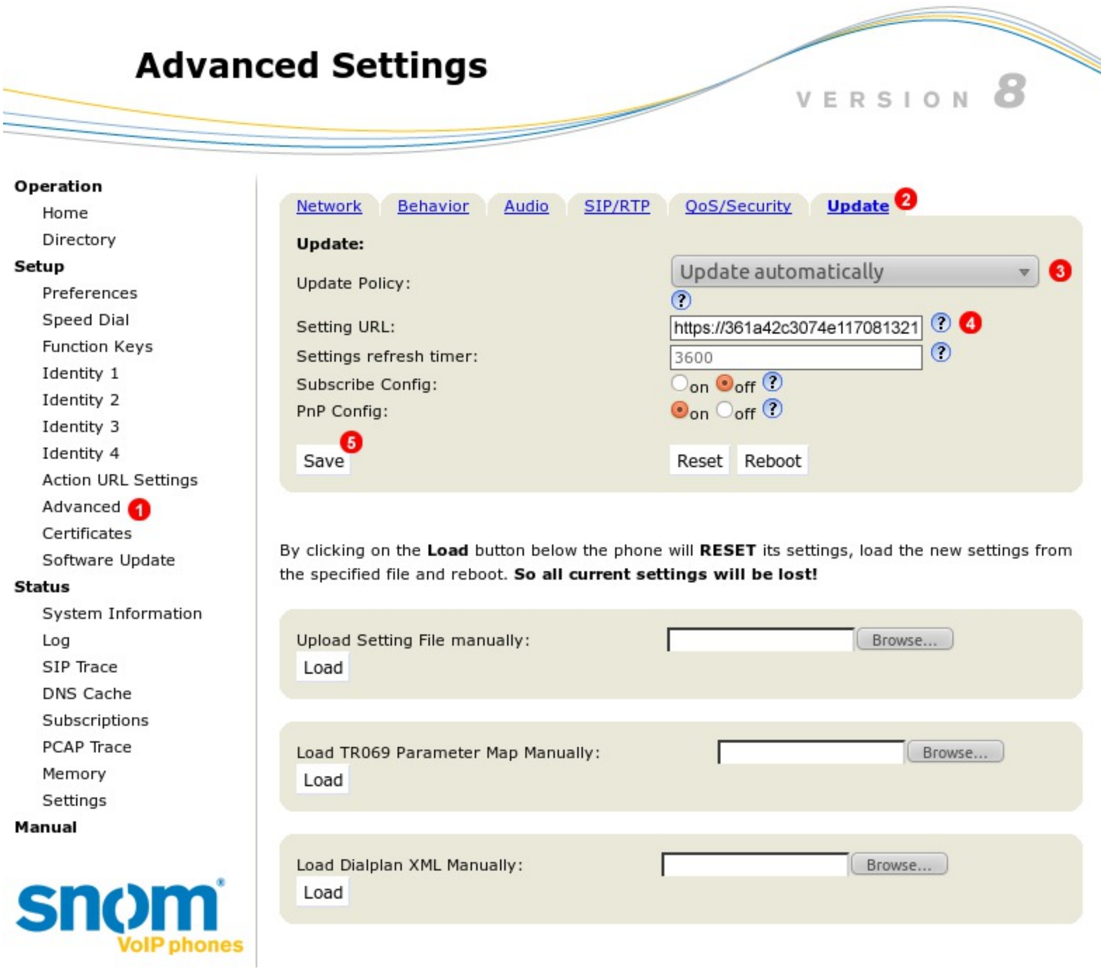


Figure 14.24 Snom phones auto provisioning

To auto provision Snom phone login into web interface of the device and follow steps below:

1. Click `Advanced` link in the left sidebar.
2. Select `Update`.
3. Make sure `Update Policy` is set to `Update automatically`
4. Copy and paste your `Profile URL` (see `(2)` in Figure 13.6 above) into field `(4)`.
5. Click `Save` button.

After a short while your Snom phone will reboot and obtain configuration data from VoIP Studio provisioning service.

Yealink

If your phone was purchased from VoIP Studio, it has been already pre configured and there is no need to provision it. Please refer to Phone edit section to assign extensions. If your phone was not purchased from VoIP Studio, please perform a Factory Reset described in Factory Reset section above and Firmware upgrade.

You can obtain your phone IP by following the steps described in Find phone IP section above.

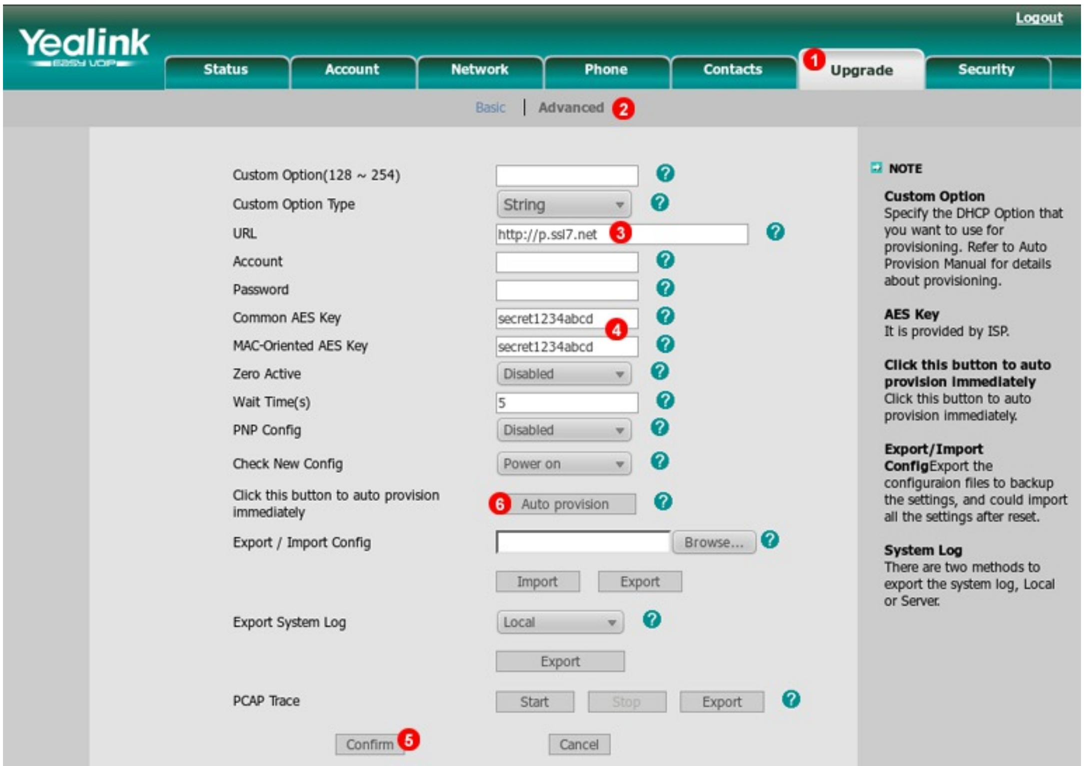


Figure 14.25 Yealink phone auto provisioning

To auto provision Yealink phone login into web interface of the device and follow steps below:

1. Click `Upgrade` tab at the top of the page.
2. Click `Advanced` link at the top.
3. Enter `http://p.ssl7.net` as Configuration Server Path.
4. Enter `AES Key` into `Common AES Key` and `MAC-Oriented AES Key` fields.
5. Click `Confirm` button at the bottom of the page.
6. Click `Auto provision` button.

After a short while your Yealink phone will reboot and obtain configuration data from VoIP Studio provisioning service.

Manual configuration

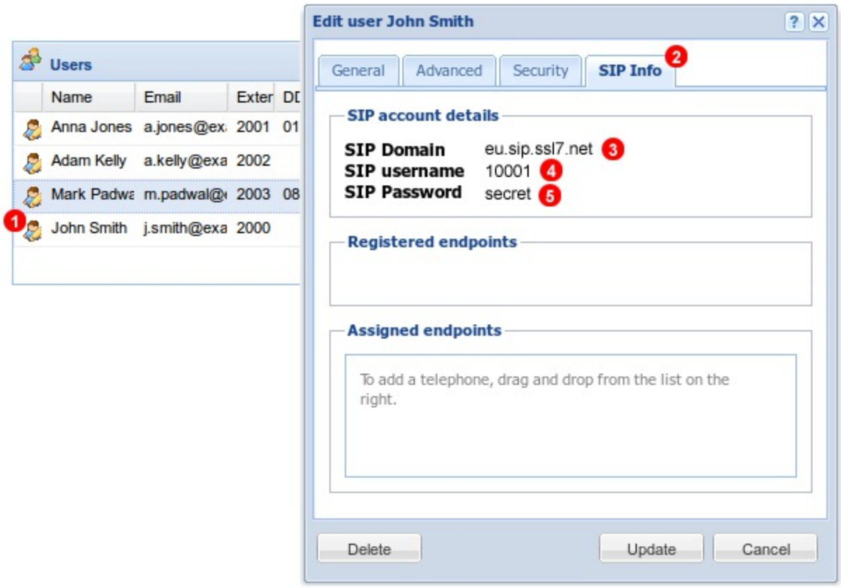


Figure 14.26 SIP account details

Below you will find instructions how to configure popular SIP phones with VoIP Studio service. Before you begin you will need to obtain SIP account details:

1. Click icon next to the name of the user you want to set up phone for.
2. Select `SIP Info` tab.

Use SIP username, password and domain (3,4,5 in the Figure 11.1 above) to configure your device.

Aastra

If your phone was purchased from VoIP Studio, it has been already pre configured and there is no need to provision it. Please refer to Phone edit section to assign extensions. If your phone was not purchased from VoIP Studio, please perform a Factory Reset described in Factory Reset section above and Firmware upgrade.

You can obtain your phone IP by following the steps described in Find phone IP section above.

1. Please upgrade your Firmware from below link-

http://www.aastra.com/document-library.htm?curr_nav=2&curr_fam=Aastra+6730i&prod_id=6167

Steps:

- Login to your phone via web browser
- Firmware Update
- write the exact file name you downloaded with extension [for example 6731i.st]
- Select protocol, Server IP and port as required
- Download Firmware

1. Factory reset:

- Press the settings key of your phone
- scroll to Admin menu
- provide password (by default 22222)
- scroll down
- Factory Default
- Enter

1. Manual Configuration

- Find your phone IP, and log in to your Phone [by pasting the IP on your web browser]

- Configuration Server

- HTTP server [p.ssl7.net]
- HTTP path [MAC Address]]

- save settings

- Line 1

- phone no, caller id, authentication name [SIP user name]
- password [SIP Password]
- Proxy server, back up proxy server, outbound proxy server, back up outbound proxy server, Register server, Back up register server[SIP domain you can view in your web portal]
- all sip port : 5060
- save settings

1. Reboot your phone.

Cisco PAP2

If your phone was purchased from VoIP Studio, it has been already pre configured and there is no need to provision it. Please refer to Phone edit section to assign extensions. If your phone was not purchased from VoIP Studio, please perform a Factory Reset described in Factory Reset section above and Firmware upgrade.

You can obtain your phone IP by following the steps described in Find phone IP section above.

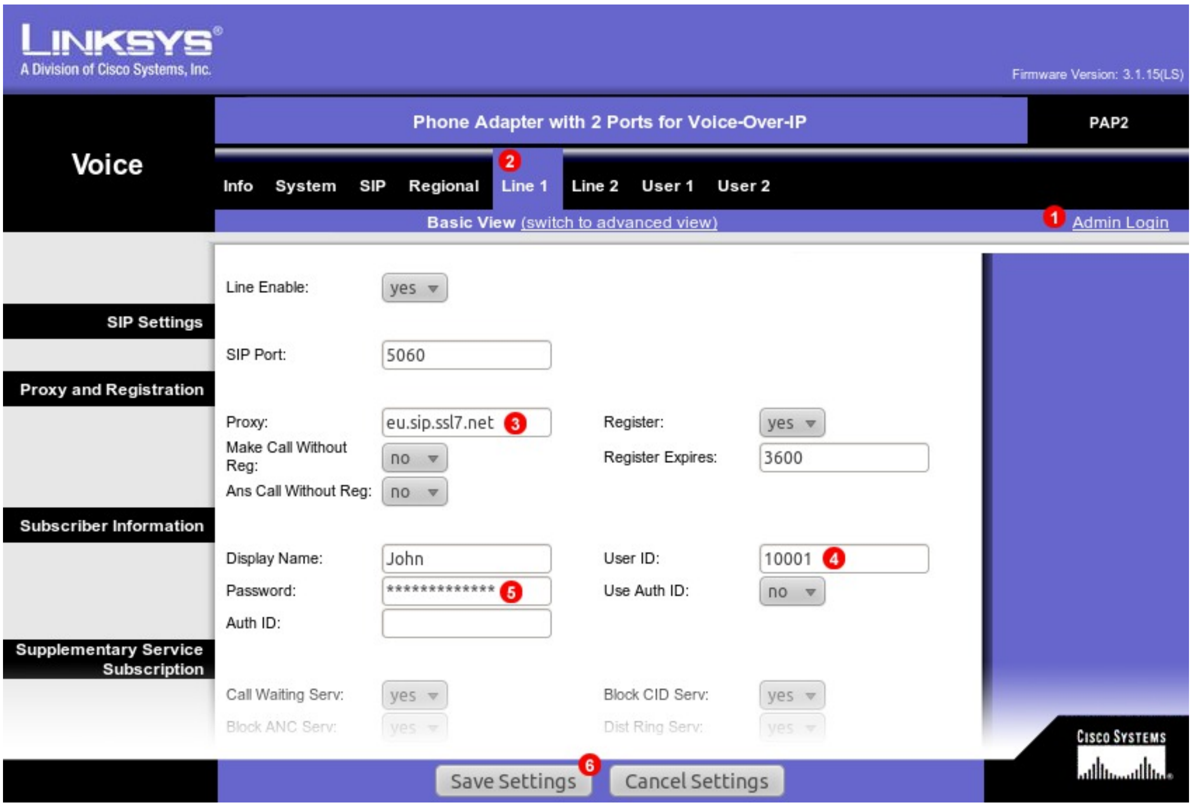


Figure 14.27 Cisco PAP2 configuration

To configure Cisco PAP2 adapter to work with VoIP Studio service login into web interface of the device and follow steps below:

1. Click `Admin` link at the top.
2. Select `Line 1` tab.
3. Enter your SIP domain into field `(3)`.
4. Enter your SIP username into field `(4)`.
5. Enter your SIP password into field `(5)`.
6. Click `Save Settings`.

After a short while Cisco PAP2 adapter will connect to VoIP Studio servers and you will be able to start making and receiving calls.

Cisco SPA525G

If your phone was purchased from VoIP Studio, it has been already pre configured and there is no need to provision it. Please refer to Phone edit section to assign extensions. If your phone was not purchased from VoIP Studio, please perform a Factory Reset described in Factory Reset section above and Firmware upgrade.

You can obtain your phone IP by following the steps described in Find phone IP section above.

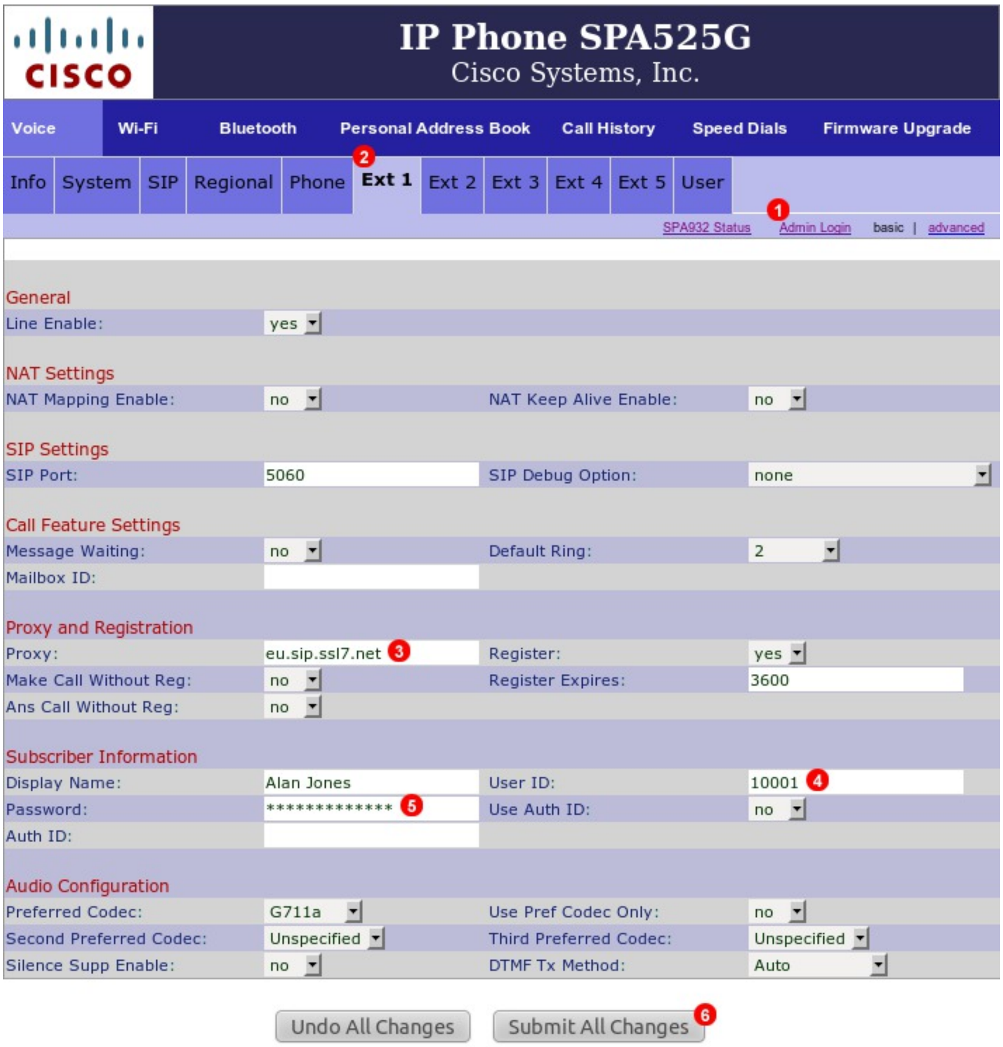


Figure 14.28 Cisco SPA525G configuration

To configure Cisco SPA525G phone to work with VoIP Studio service login into web interface of the device and follow steps below:

1. Click `Admin` link at the top.
2. Select `Ext 1` tab.
3. Enter your SIP domain into field `(3)`.
4. Enter your SIP username into field `(4)`.
5. Enter your SIP password into field `(5)`.
6. Click `Submit All Changes`.

After a short while Cisco SPA525G phone will connect to VoIP Studio servers and you will be able to start making and receiving calls.

Gigaset A580

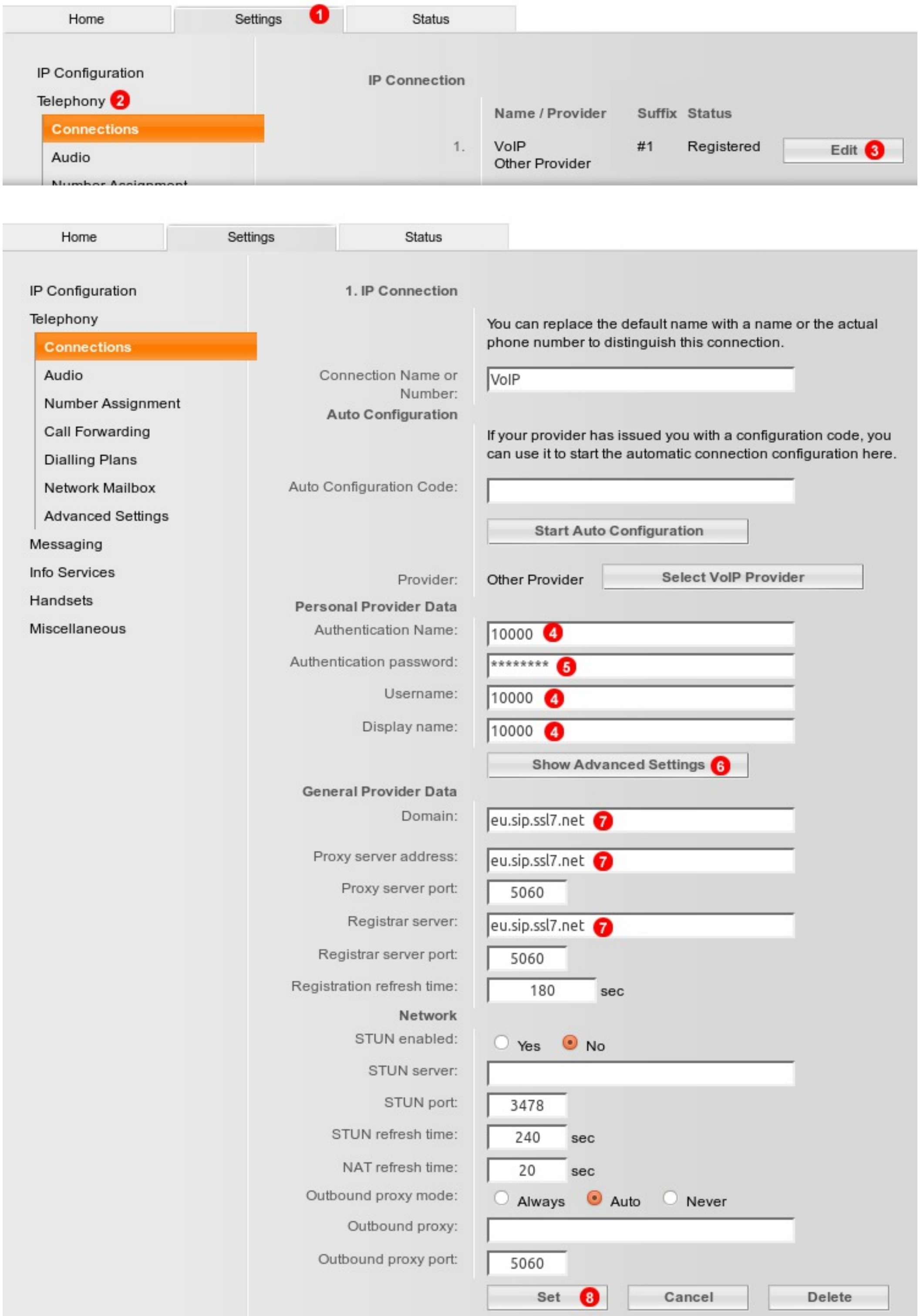


Figure 14.29 Cisco SPA525G configuration

To configure Gigaset A580 phone to work with VoIP Studio service login into web interface of the device and follow steps below:

1. Click `Settings` tab at the top.
2. Select `Telephony` and `Connections` from the menu on the left.
3. Click `Edit` button.
4. Enter your SIP username into fields `(4)`.
5. Enter your SIP password into field `(5)`.
6. Click `Show Advanced Settings` button.
7. Enter your SIP domain into fields `(7)`.

8. Click `Set` button.

After a short while Gigaset A580 phone will connect to VoIP Studio servers and you will be able to start making and receiving calls.

Polycom

If your phone was purchased from VoIP Studio, it has been already pre configured and there is no need to provision it. Please refer to Phone edit section to assign extensions. If your phone was not purchased from VoIP Studio, please perform a Factory Reset described in Factory Reset section above and Firmware upgrade.

You can obtain your phone IP by following the steps described in Find phone IP section above.

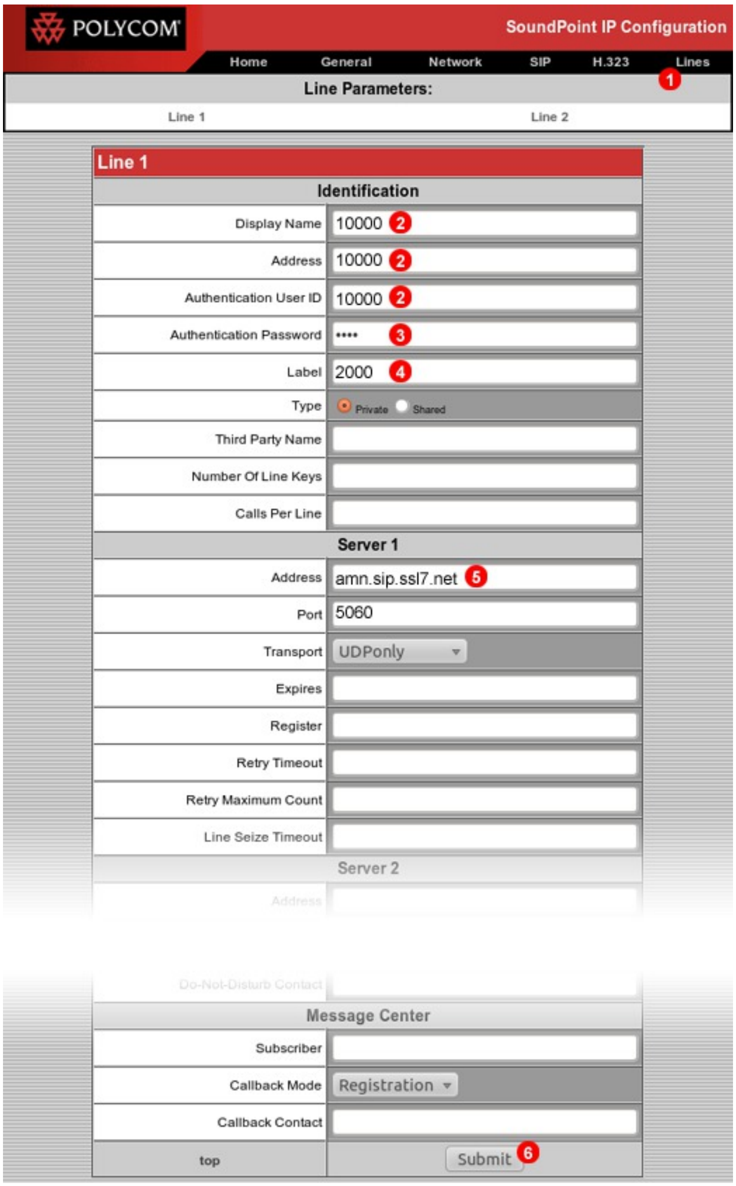


Figure 14.30 Polycom - SIP account settings

To configure Polycom phone to work with VoIP Studio service login into web interface (default username: `Polycom`, default password: `456`) of the device and follow steps below:

1. Click `Lines` link in the top menu.
2. Enter your SIP username into fileds `(2)`.
3. Enter your SIP password into filed `(3)`.
4. Enter your extension into field `(4)`.
5. Enter your SIP domain into field `(5)`.
6. Click `Submit` button `(6)`.

After a short while Polycom phone will connect to VoIP Studio servers and you will be able to start making and receiving calls.

Snom

If your phone was purchased from VoIP Studio, it has been already pre configured and there is no need to provision it. Please refer to Phone edit section to assign extensions. If your phone was not purchased from VoIP Studio, please perform a Factory Reset described in Factory Reset section above and Firmware upgrade.

You can obtain your phone IP by following the steps described in Find phone IP section above.

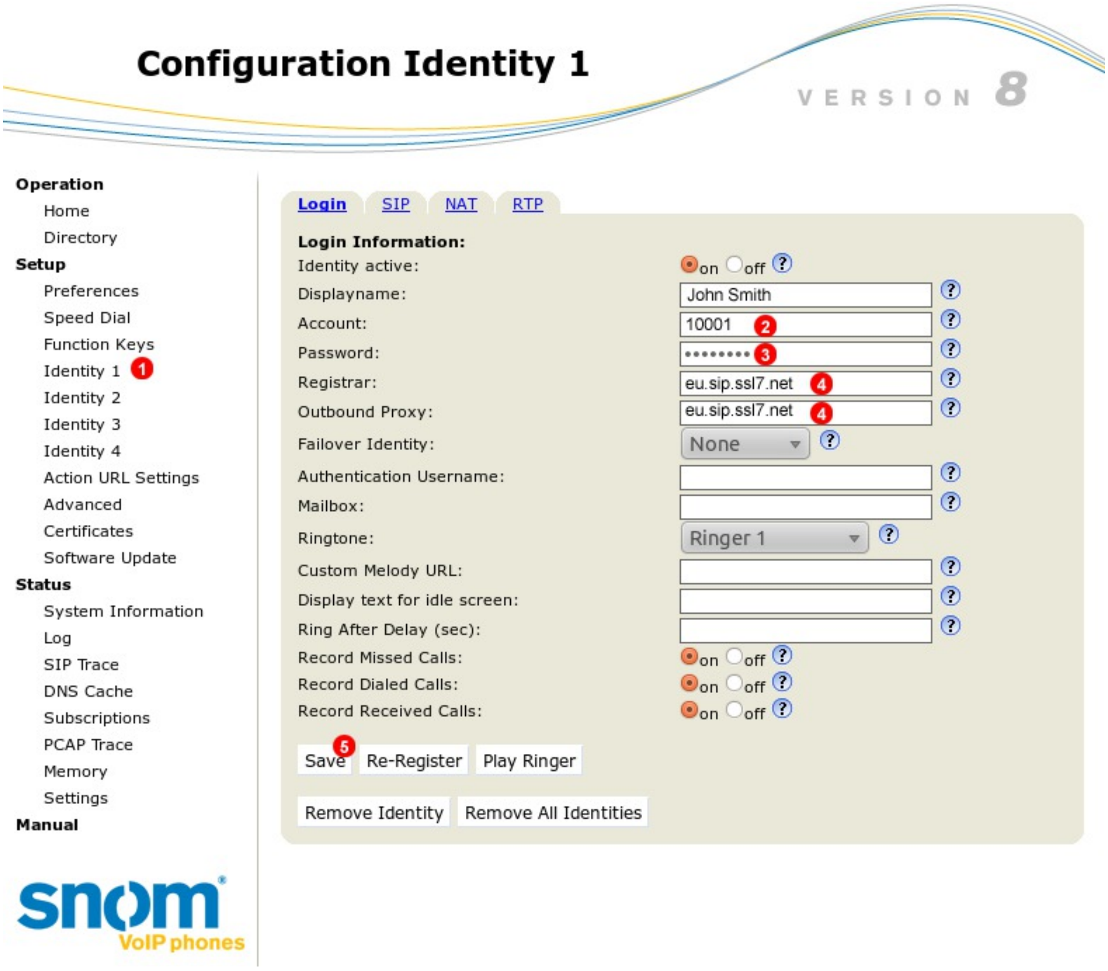


Figure 14.31 Snom manual configuration

To configure Snom phone to work with VoIP Studio service login into web interface of the device and follow steps below:

1. Click `Identity 1` link in the left menu.
2. Enter your SIP username into filed `(2)`.
3. Enter your SIP password into filed `(3)`.
4. Enter your SIP domain into fields `(4)`.
5. Click `Save` button.

After a short while Snom phone will connect to VoIP Studio servers and you will be able to start making and receiving calls.

Yealink

- While doing manual configuration, it is always best to perform a Factory Reset described in Factory Reset section above. You can obtain your phone IP by following the steps described in Find phone IP section above.

To configure Yealink phone to work with VoIP Studio service login into web interface of the device and follow steps below:

1. Upgrade Firmware from below link -

   http://www.yealink.co.uk/downloads/

Follow below Steps:

1. Login to your phone via web browser
2. Upgrade
3. Browse the Firmware you downloaded

4. upgrade

5. Manual Configuration

- Account
- Account Active (ON)
- Register Name, User Name (SIP ID, you can find it in your VoIPdito web portal -> Administrator -> User -> SIP Info)
- Password (SIP Password)
- SIP Server, Outbound proxy server, Backup Outbound proxy server (Domain name you can find it in your VoIPdito web portal)
- port number for these servers 5060
- Enable Outbound Proxy server
- Confirm

1. Reboot your phone.

15. Network configuration

IP Addresses

If your network is protected by firewall which restrict outbound traffic please ensure you allow the following addresses and ports combinations:

North America

- Ports TCP/UDP 5060,5566 (SIP Signaling): 97.107.141.97
- Ports UDP 10000-20000 (RTP Audio): 66.228.45.12, 50.116.53.144, 66.228.44.117, 23.92.17.107

Europe:

- Ports TCP/UDP 5060,5566 (SIP Signaling): 82.113.147.229
- Ports UDP 10000-20000 (RTP Audio): 109.233.112.91, 109.233.119.234, 109.233.112.111

16. Getting Help

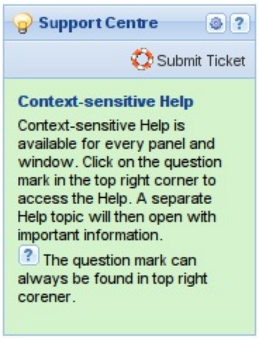


Figure 16.1 Help panel.

Submit ticket

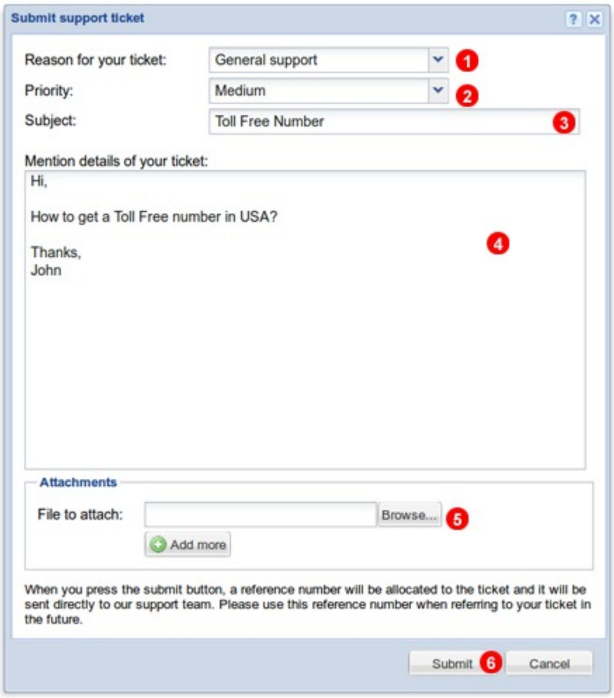


Figure 16.2 Submit support ticket window.

To submit a ticket click "Submit ticket" button located in Help panel - see figure 15.1. Next select your ticket type, from the drop down list (1) and enter details of your problem into text area (2). Finally click Submit button (3).

View your tickets

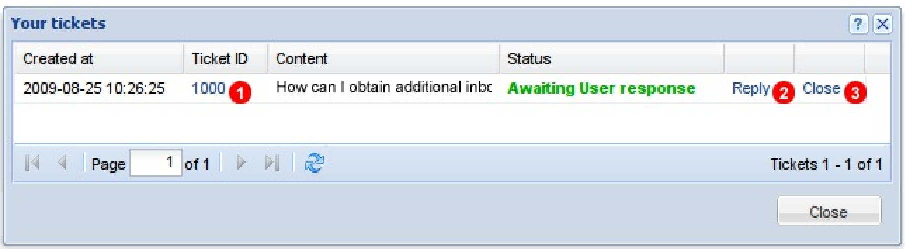


Figure 16.3 Your tickets window.

To view your tickets click gear icon located in top right corner of Help panel. Next select Your tickets from the context menu. To view history of all messages, click ticket ID link `(1)`.

Reply to ticket

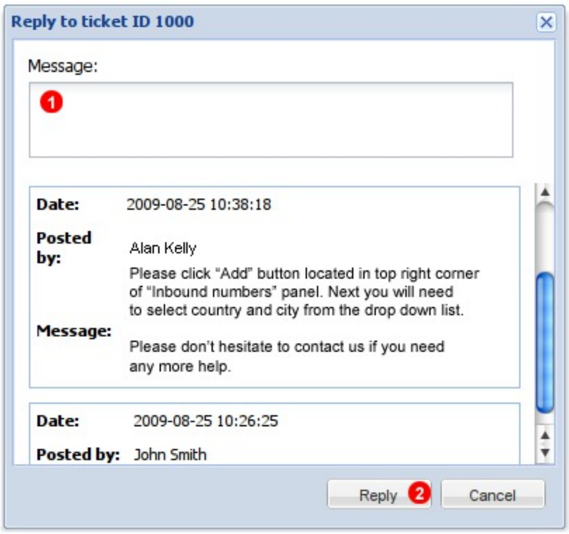


Figure 16.4 Ticket reply window.

To post a new message click Reply link - see `(2)` in figure 16.3. Next enter content of your reply into text area `(1)` and click Reply button `(1)`. Once you are satisfied with the solution provided by support team, you can close your ticket by clicking Close link - see `(3)` in figure 16.3 above.

Remote Support

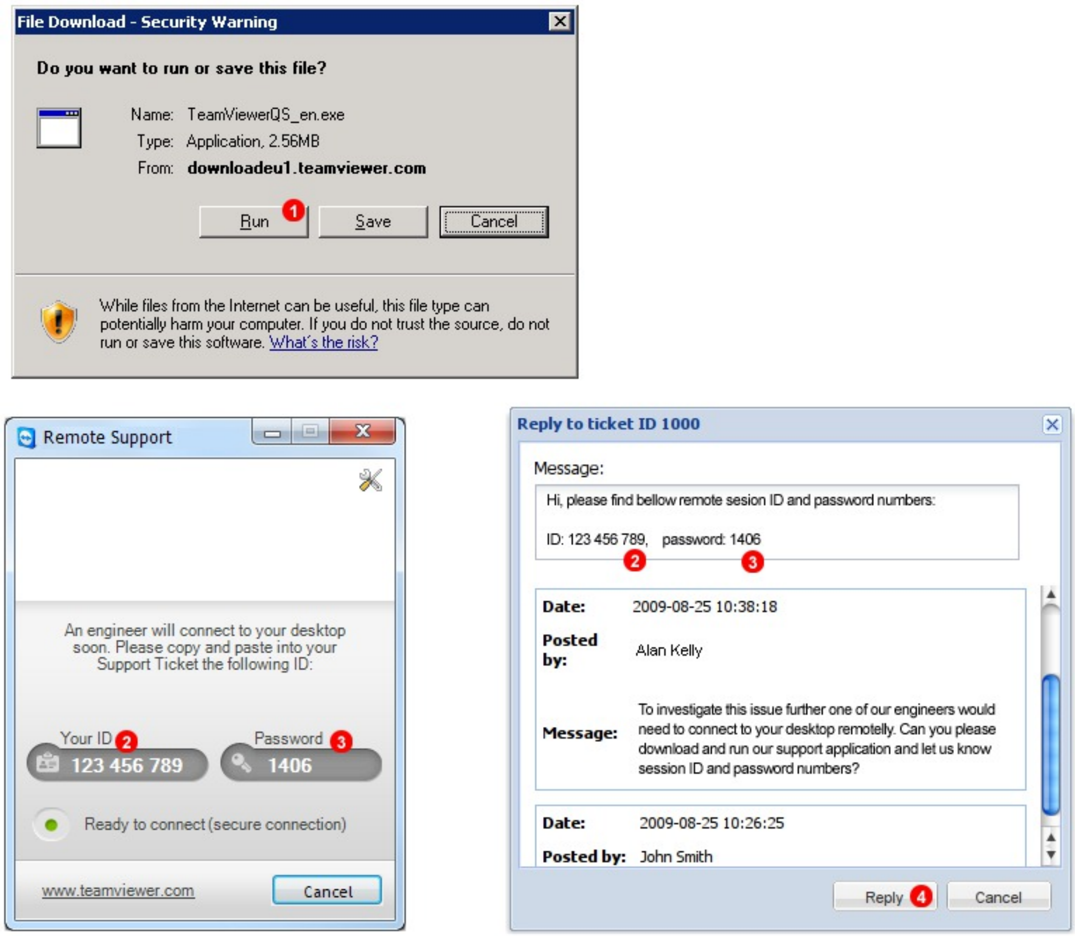


Figure 16.5 Remote Support application.

In some cases a support engineer may need to establish a remote connection with your computer in order to conduct further investigation or troubleshooting of the problem. To allow for that please:

1. Download and run our remote support program using a link provided by a member of our support team.
2. Copy and paste session ID - see `(2)` in figure 16.5 above into Support Ticket.
3. Copy and paste password `(3)` into Support Ticket.
4. Click `Reply` button `(4)`.

After a short while an engineer will establish a connection with your PC.

17. Glossary

Call forwarding

See Follow Me below.

Call pick-up

Call pick-up is a feature used in a telephone system that allows one to answer someone else's telephone call. The "call pick-up" feature is accessed by pressing a preprogrammed button (usually labelled "Pick-Up"), or by pressing a special sequence of buttons on the telephone set

Call transfer

A call transfer is a telecommunications mechanism that enables a user to relocate an existing call to another telephone or attendant console by using the transfer button and dialing the required location. The transferred call is either announced or unannounced.

Call waiting

Call waiting (or catch phone in Japan), in telephony, is a feature on some telephone networks. If a calling party places a call to a called party which is otherwise engaged, and the called party has the call waiting feature enabled, the called party is able to suspend the current telephone call and switch to the new incoming call (Typically, this is done by pushing the flash button), and can then negotiate with the new or the current caller an appropriate time to ring back if the message is important, or to quickly handle a separate incoming call.

DDI

See DID below.

DID

Direct Inward Dialling (DID), also called Direct Dial-In (DDI) in Europe, is a feature offered by telephone companies for use with their customers' PBX systems, whereby the telephone company (telco) allocates a range of numbers all connected to their customer's PBX. As calls are presented to the PBX, the number that the caller dialled is also given, so the PBX can route the call to the desired person or bureau within the organization.

Follow Me

Follow Me (call forwarding or call diverting), in telephony, is a feature on some telephone networks that allows an incoming call to a called party, which would be otherwise unavailable, to be redirected to a mobile telephone or other telephone number where the desired called party is situated. Up to 4 numbers can be added in Follow Me option for call forwarding.

Hosted PBX

A hosted PBX system delivers PBX functionality as a service, available over the Public Service Telephone Network (PSTN) and/or the internet. Hosted PBXs are typically provided by the telephone company, using equipment located in the premises of the telephone company's exchange. This means the customer organization doesn't need to buy or install PBX equipment (generally the service is provided by a lease agreement) and the telephone company can (in some configurations) use the same switching equipment to service multiple PBX hosting accounts.

Instead of buying PBX equipment, users contract for PBX services from a hosted PBX service provider, a particular type of Application Service Provider (ASP). The first hosted PBX service was very feature-rich compared to most premise-based systems of the time. In fact, some PBX functions, such as follow-me calling, appeared in a hosted service before they became available in hardware PBX equipment. Since that introduction, updates and new offerings from several companies have moved feature sets in both directions.

Today, it is possible to get hosted PBX service that includes far more features than were available from the first systems of this class, or to contract with companies that provide less functionality for more simple needs.

IVR

In telephony, interactive voice response, or IVR, is a phone technology that allows a computer to detect voice and touch tones using a normal phone call. The IVR system can respond with pre-recorded or dynamically generated audio to further direct callers on how to proceed. IVR systems can be used to control almost any function where the interface can be broken down into a series of simple menu choices. Once constructed IVR systems generally scale well to handle large call volumes.

Music on Hold

Music on hold (MOH) refers to the business practice of playing recorded music to fill the silence that would be heard by telephone callers who have been placed on hold. It is especially common in situations involving customer service.

PSTN

The public switched telephone network (PSTN) is the network of the world's public circuit-switched telephone networks, in much the same way that the Internet is the network of the world's public IP-based packet-switched networks.

Queue (ACD)

Call centers use an Automatic Call Distributor (ACD) Queues to distribute incoming calls to specific resources (agents) in the center. Queue hold queued calls in First In, First Out order until agents become available. When an agent becomes available, the highest-ranked caller in the queue is delivered to that agent, and everyone else moves up a rank.

SIP

The Session Initiation Protocol (SIP) is a signalling protocol, widely used for setting up and tearing down multimedia communication sessions such as voice and video calls over the Internet. Other feasible application examples include video conferencing, streaming multimedia distribution, instant messaging, presence information and online games.

SMS

Short Message Service (SMS) is a communications protocol allowing the interchange of short text messages between mobile telephone devices.

VoIP

Voice over Internet Protocol (VoIP, IPA: /vɔɪp/) is a protocol optimized for the transmission of voice through the Internet or other packet switched networks. VoIP is often used abstractly to refer to the actual transmission of voice (rather than the protocol implementing it). This latter concept is also referred to as IP telephony, Internet telephony, voice over broadband, broadband telephony, and broadband phone.