IceWarp Unified Communications

VoIP Service Reference

Version 11.4

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Contents

VoIP Service

NOTE: In IceWarp Server, the registrar, proxy and redirect servers are integrated to the software, no further software is required.

Legend

Icon	Description
	Warning – very important!
	Note or tip – good to know.
NOTE: Areas...	*Note within a table.*
▸ Figure 4	Figure link – click the link to reveal the figure. Click it again to close it. (Works only in the ***CHM*** format.)

Registered Trademarks

About

IceWarp Server VoIP service implements SIP. The SIP (Session Initiation Protocol) is designed to allow devices, both software and hardware, to establish a communication session.

The VoIP service in IceWarp Server is actually a SIP domain which is to be defined within IceWarp Server as a domain or domain alias. This domain must have a valid DNS "A" record.

The four basic components of a SIP session are:

SIP User Agents

These are the end-user devices.

These can be software devices, running on PCs, PDAs, cell phones, etc. or they can be SIP-enabled network devices such as SIP-phones, or even, via SIP gateways, ordinary telephony devices.

A SIP call is initiated by a user agent client and responded to by user agent server.

SIP Registrar Servers

These are databases containing the location of all user agents within a domain. There servers retrieve and send IP addresses and other information at the request of a SIP proxy server.

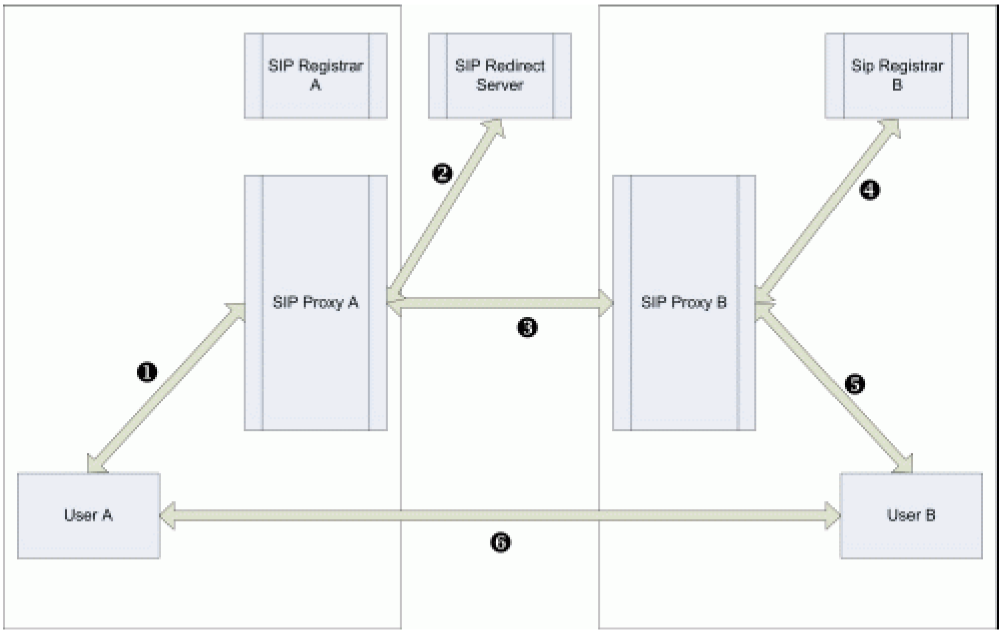
SIP Proxy Server

A SIP proxy server accepts session requests from a user agent and queries a SIP registrar for the recipient's address. Then it forwards the session invitation directly to the user agent if it is in the same domain or to another proxy server in the case the user agent is in another domain.

SIP Redirect Servers

These allow proxy servers to locate other, external proxy servers (rather like a DNS for SIP).

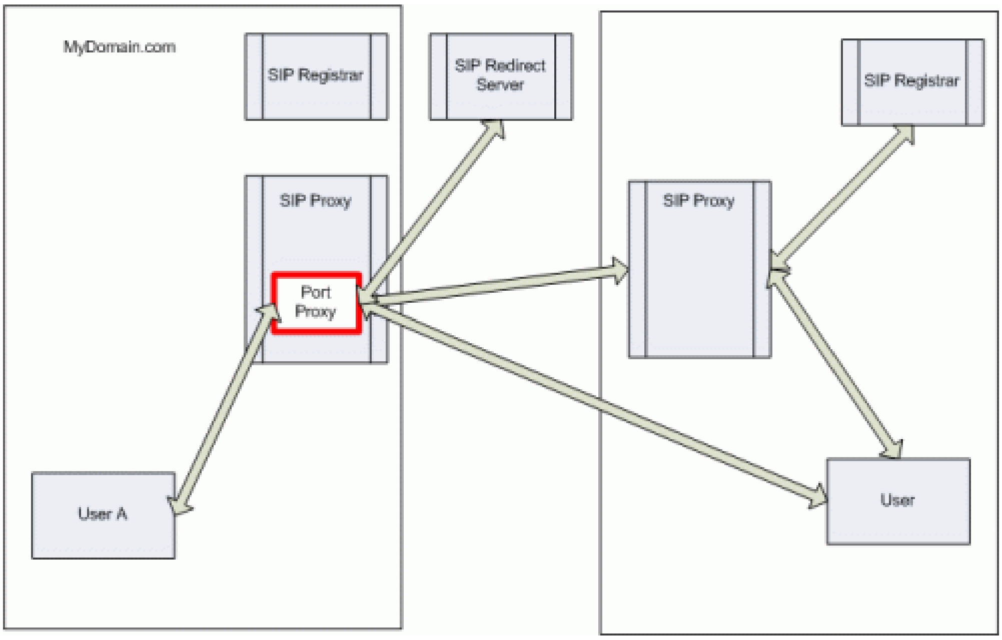
The following diagrams and examples should help explain the structure and process of placing a SIP call.



1. User A places a call to User B (User B is in a domain external to User A's domain). This request is picked up by SIP proxy A (arrow 1).
2. SIP proxy A determines that User B is outside its domain so asks a SIP redirect server where "User B of domain B" can be found (arrow 2).
3. SIP redirect server responds with the address for SIP server B (arrow 2).
4. SIP proxy A sends the call request to SIP server B (arrow 3).
5. SIP server B requests the location of User B from SIP registrar B (arrow 4).
6. SIP registrar B responds with User B's location (arrow 4).
7. SIP proxy B contacts User B's device (arrow 5).
8. User B accepts the call.
9. User B's device tells SIP proxy B (arrow 5).
10. SIP proxy B tells SIP proxy A (arrow 3).
11. SIP proxy A tells User A's device (arrow 1).
12. Channel is established (arrow 6).

If you have multiple users behind a firewall or router then you will probably need to enable NAT Traversal on the SIP server (see the**SIP – Advanced** tab).

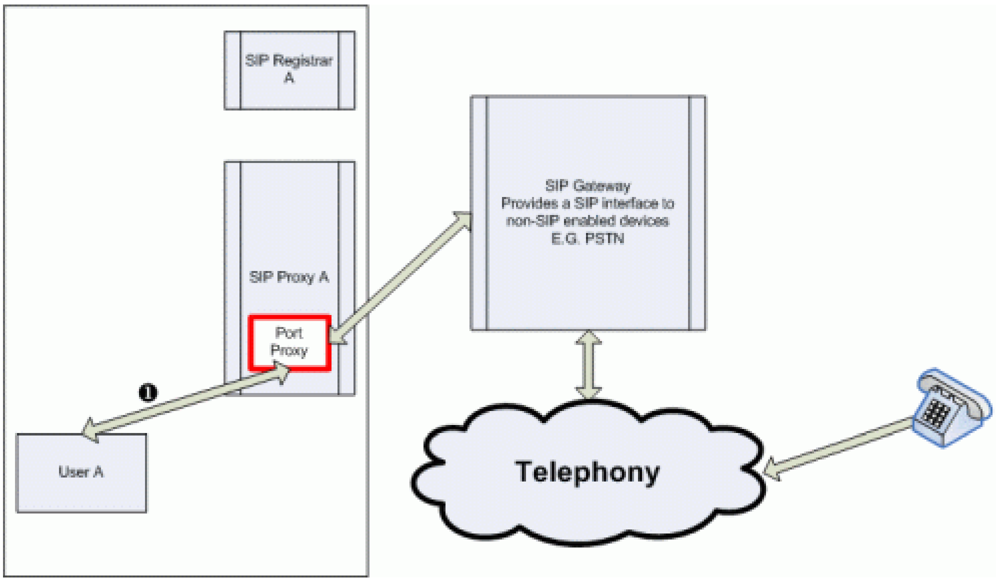
The following diagram shows a call using NAT traversal and proxy ports.



The basic functionality is the same except that all communication outside of the domain is done via a proxy port.

One proxy port is opened for each user communicating outside the domain.

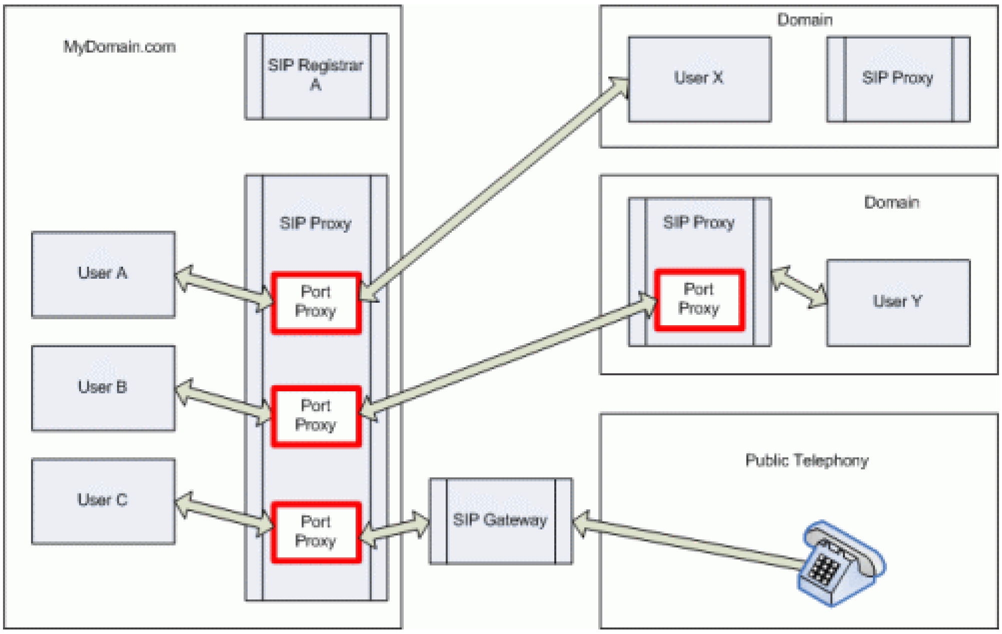
A SIP gateway is a service provided that allows you to connect to non-SIP devices, such as the public telephone network. These services usually have to be paid for.



The initiation of the call is the same up to the point where the SIP gateway is reached.

The Big Picture

The SIP server allows you to offer a complete voice communication solution to your users.



Reference

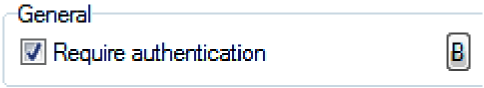
The VoIP process flow is rather complicated, but from the priority point of view we can simplify it and describe an incoming call in these routing phases:

1. The dial plan is processed first.
2. VoIP trunks next.
3. Location service next (registered VoIP users).
4. Extensions.

If you have your extension linked to an email account or a list of members and there are no preceding rules for the number, the extension will route the call accordingly.

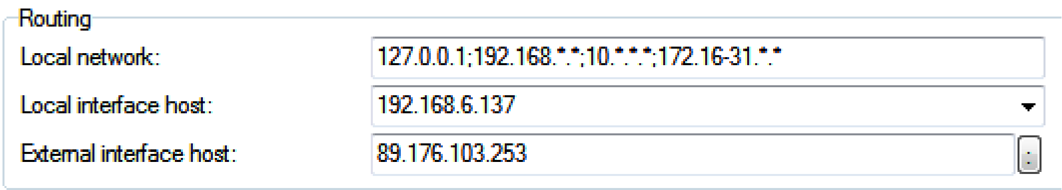
So not only you can define the available extensions and groups in your system, but you can also control to whom they are linked to automatically.

General



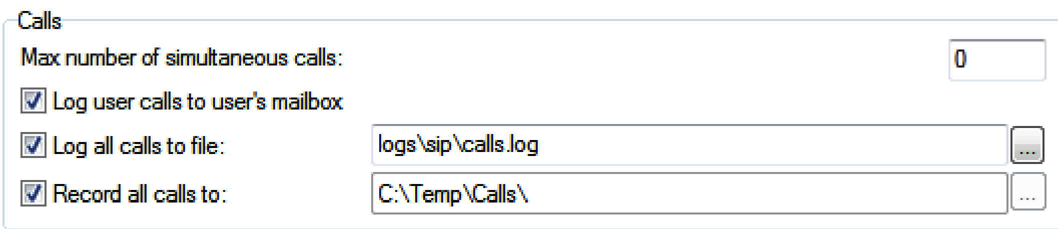
Field	Description
Require authentication	Check this box if you do not wish to allow anonymous users access to the SIP service. If you select this option you can use the ***B*** button to edit a bypass file, allowing IP ranges, users and domains anonymous access.

*NOTE: Access mode to the service can be set on both domain and user levels. See the appropriate places (**[domain] – Policies, [user] – Policies**).*



Field	Description
Local network	Here you need to specify all the local IP addresses that this SIP server should be available for. Separate them by semicolons.
Local interface host	Specify the local IP address of the SIP server here.
External interface host	Specify the local IP address of any external interface (probably your router or firewall)

NOTE: Incorrect routing information is the biggest cause of problems with SIP communications. Make sure you set this up correctly.

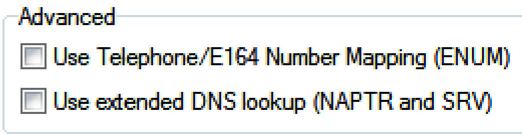


Field	Description
Max number of simultaneous calls	Once the maximum number of simultaneous calls is reached any further attempted calls will be rejected by the service. This can be useful if you want to limit the bandwidth that is used by the SIP server. A typical voice call is around 8kB/s.
Log user calls to user mailbox	Check this box to have SIP calls logged to the mailbox of the user who made the call.

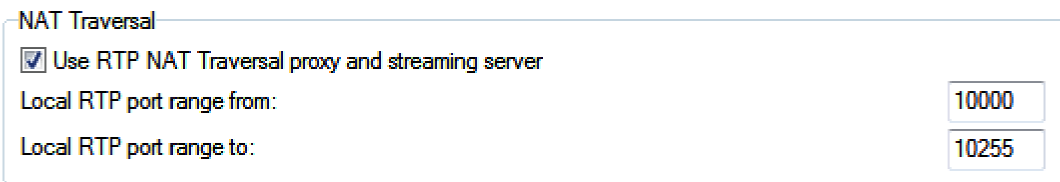
	*NOTE: This option **must** be checked for the REDIAL to work.*
Log all calls to file	Check this box and specify a fully qualified path to the file where a log of ALL calls is to be stored. Note that ***yyyy, mm*** and ***dd*** can be used in the directory name, For example: *<InstallDirectory>\SIPyyyymmdd\sip.log* Click the "..." button to open a usual browser.
Record all calls to	Check this box and specify a fully qualified path where ALL calls should be recorded in the MP3 format.

Advanced

The **Advanced** tab allows you to specify how the SIP server will perform NAT translation, and whether the server will use a parent SIP proxy.

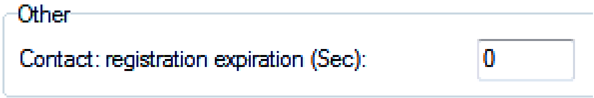


Field	Description
Use Telephone/E164 Number Mapping (ENUM)	Check this box to enable your users to place calls to standard telephone numbers. The ENUM system allows a standard telephone number to be dialed from a SIP client. The SIP server will check for a NAPTR DNS record based on the number dialed (the **164.arpa** server is tried out, if record not found, then the **e164.org** one.) *NOTE: The option mentioned above has higher priority than SIP gateways (if used.)*
Use extended DNS lookup (NAPTR and SRV)	Select this option IceWarp Server to check for SRV and NAPTR DNS records to determine the hostname for a SIP server. IceWarp Server checks for a NAPTR DNS record first and , if not found, it will check for an SRV DNS record. *Syntax and examples of SRV DNS records:* ***_sip._udp.<Domain>. 86400 IN SRV 0 0 5600 sip.<Host>.*** ***_sip._tcp.<Domain>. 86400 IN SRV 0 0 5600 sip.<Host>.*** ***_sip._tls.<Domain>. 86400 IN SRV 0 0 5601 sip.<Host>.*** *_sip._udp.icewarpdemo.com. 86400 IN SRV 0 0 5600 sip.icewarpdemo.com.* *_sip._tcp.icewarpdemo.com. 86400 IN SRV 0 0 5600 sip.icewarpdemo.com.* *_sip._tls.icewarpdemo.com. 86400 IN SRV 0 0 5601 sip.icewarpdemo.com.* *For more details, refer to **http://www.ietf.org/rfc/rfc3263.txt**.*

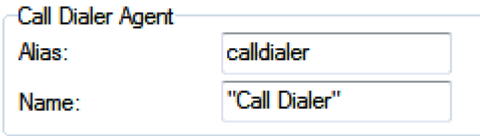


Field	Description
Use RTP NAT Traversal proxy and streaming server	Tick the box if you want to use NAT Traversal feature. Use API and API console to manage proxy and streaming server settings. (Use the ***rtp*** string to filter API variables.) *NOTE: NAT Traversal has a limitation of concurrent calls. The **c_system_services_sip_rtpmax** API variable sets the number of RTP streams. The number of concurrent calls is 1/2 of this value.* *Remember: The higher number of streams, the higher use of resources.*
Local RTP port range from	You need to specify the ports to be used as proxies by the SIP server. You should specify the start of the port range to be used.

Local RTP port range to	Specify the last port of the range to be used. *NOTE: The port range specified here must be open in your router/firewall setup.*

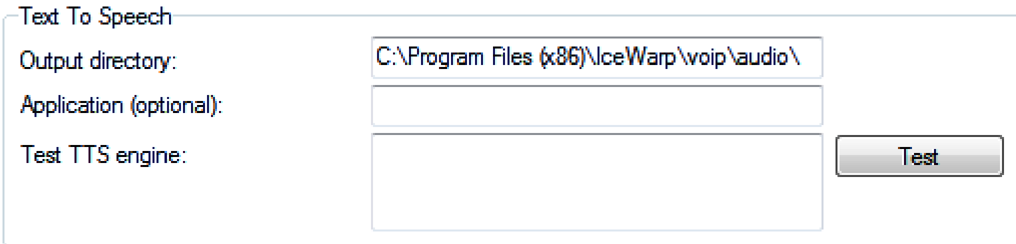


Field	Description
Contact: registration expiration (Sec)	Specify a value to tell clients that they should re-register with the server at the interval specified. This can be very useful to keep the client/server connection alive.



Field	Description
Alias	Enter a call dialer alias.
Name	Enter a call dialer name. When initiating a SIP call, a server rings the caller's phone first. This name is shown on the phone display.

Text to Speech



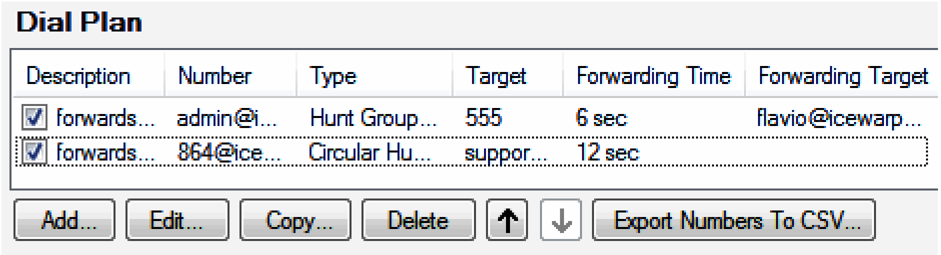
Field	Description
Output directory	Specify the directory where the generated audio files are to be saved. Use an absolute path.
Application (optional)	Absolute path to a text-to-speech application. If not specified, the default Windows text-to-speech API is used. (With its default settings – can be changed via Control Panel.) You may want to use another application. In this case you have to be able to "tell" this application from what text file to read input and where to save the output. Fill in this application in this field – use with a mask specifying how to call. E. g. ***d:\icewarp\TSSpeech.exe /Wav /Hidden "%s" "%s"*** The first ***%s*** is replaced by a text file containing the text to convert. The second ***%s*** contains a path to an output file.
Test TTS engine /Test	Fill in any text and click the ***Test*** button to hear the output. *NOTE: It does not work for all types of actions. E.g. text to speech does not work in*

	the IVR – Test.

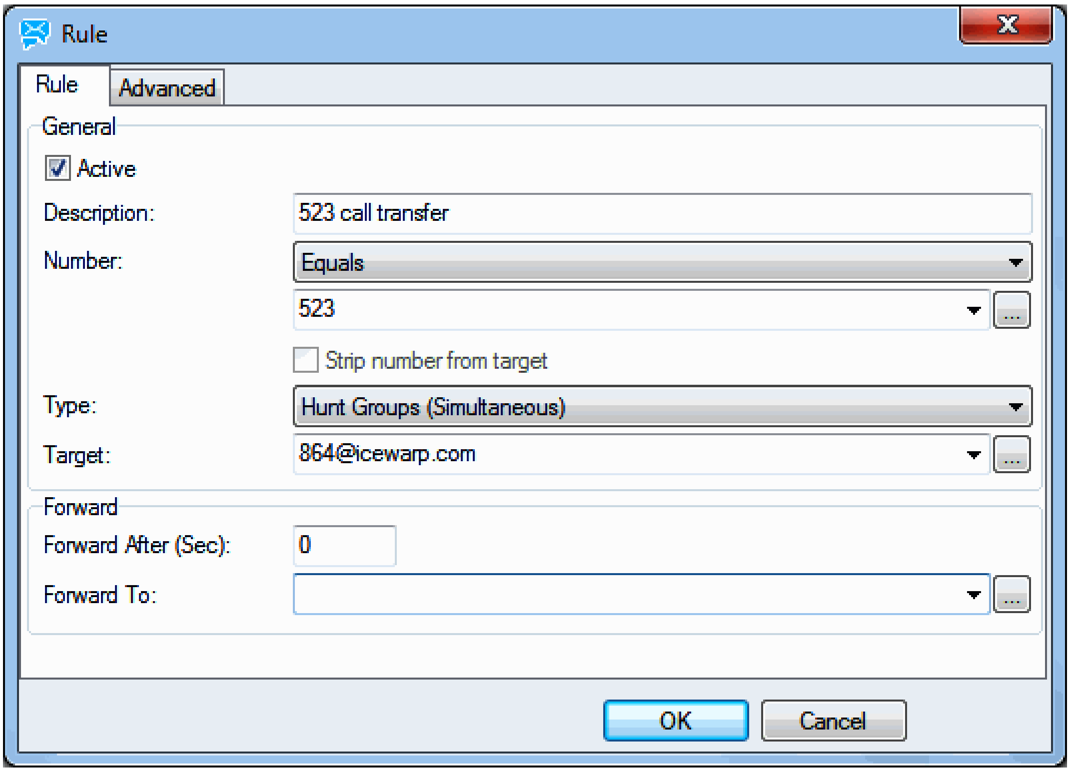
Dial Plan

The **Dial Plan** tab is a general call forwarding, dial plan, hunt group, circular hunting, address book, away, etc. place.

You can define your own numbers. Regexes can be used.

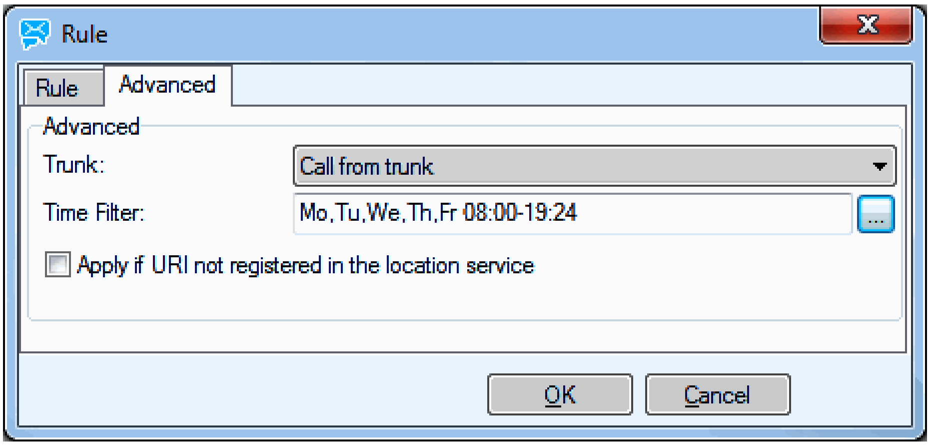


Button	Description
Add	Click the button to add a new dial plan rule. The **Rule** dialog opens.
Edit	Select a rule from the list and click the button to edit this rule. The **Rule** dialog opens.
Copy	Select a rule from the list and click the button. The **Rule** dialog opens, here you can change some features. Click ***OK*** to create a new similar rule.
Delete	Select a rule and click the button to remove this rule.
Arrows	Select a rule and use the buttons to change a rule position in the list. *NOTE: The order of dial plans in the list is important! If a number of some dial plan is called, dial plans higher in the list are omitted even in the case the used dial plan refers to some of these dial plans. See the* ***Dial Plan Examples*** *section.*
Export Numbers To CSV	Click the button if you want to export descriptions and numbers of all dial plans. A standard **Save As** dialog lets you to save numbers as a ***.csv*** file (the ***.txt*** format is also possible).



Field	Description
Active	Tick the box to activate this rule. (If not ticked, the rule does not apply.)
Description	Write some descriptive text.
Number `first field`	Select from the list: ▪ Equals ▪ Matches (RegEx) – for RegEx use ▪ Starts with ▪ Ends with ▪ Contains Option names are self-explanatory.
Number `second field`	This is the number we use to match the rule. This field may contain multiple numbers separated by semicolons (if non-regex only). It may contain a domain. If a domain is not specified any domain will be matched.
Strip number from target	Can be used with the ***Starts with***, ***Ends with*** options. Example: Say, non-local numbers are dialed with 0 (zero) at the beginning. If this box is ticked, ***Starts with*** selected and 0 entered into the ***Number*** field, this 0 is removed and the correct number is dialed "outside".
Type	▪ ***Hunt Groups*** – the default hunt group mode. You can forward the call to many concurrent targets. Target phones will all ring simultaneously. ▪ ***Circular Hunting*** – similar as ***Hunt Groups*** but target phones ring via the ***Round Robin*** mechanism. Only one target rings at a time and they are changed for each new call. You can also combine ***Hunt Groups*** and ***Circular Hunting*** if ***Hunt Groups*** is defined on the upper level. ▪ ***Set Call Forwarding*** – used for former ***Special Numbers***, target defines what happens. If blank, the forwarding is deleted, if set, the forwarding is created if set to something non existent, an away mode is created. ▪ ***Redial Last Number*** – re-dials the last received or dialed number. The ***Log user calls to user mailbox*** option must be active. ▪ ***Call Pickup*** – you can define a rule that allows any member of some group to pickup a call that is not answered by another group member (his/her phone still ringing). The member willing to answer just has to dial a number defined within this rule.
Target	Define the list of targets separated by semicolons where to direct the call to. Target can be a system group, VoIP group, system account or another dial plan number. This means you can use as many combinations as you want, forward calls to whole domains, VoIP defined groups and even following dial plan rules. Simply create a rule to forward to number which is defined below. The target will contain the number and following rules will take place. Rules always win above local accounts. This means you can override any account receiving their calls (this is how call forwarding is done on the user level – it just creates a new rule for that user). You can use a drop down to select from a list of extensions and groups defined on the **Extensions/Groups** tab. *NOTE: When the * (asterisk) is in the **Target** field for **Call Pickup**, all users can use this dialing rule. When the field is empty, the **Call Pickup** service is just allowed to users within the same VoIP group.*
Forwarding After (Sec)	Define a time (in seconds) after what a call is redirected to recipients defined in the ***Forwarded Target*** field. Forwarding takes place in several cases:

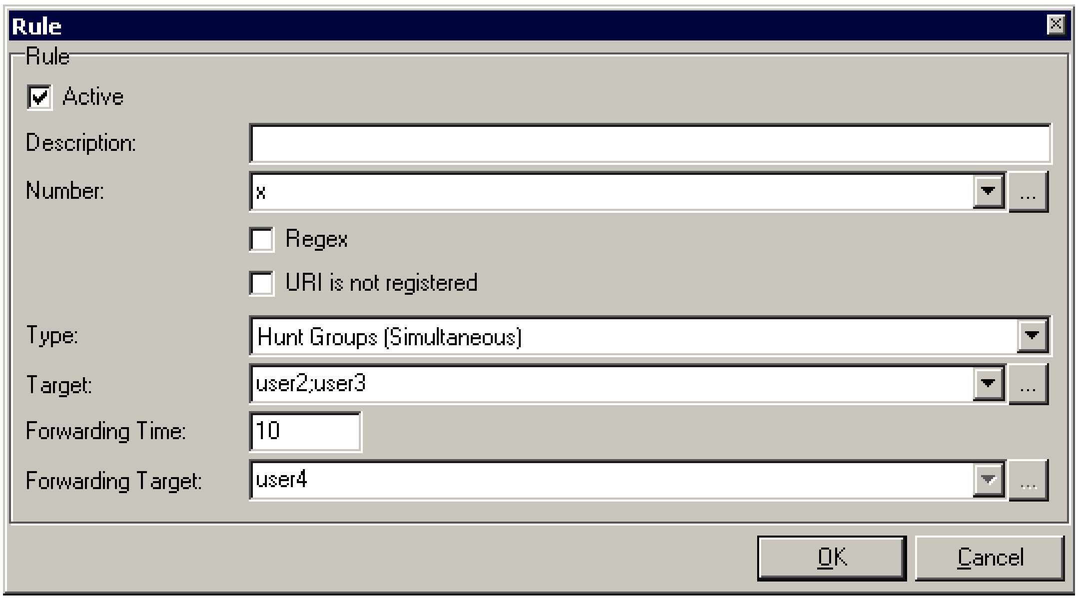
	▪ Time condition is met. ▪ None of the defined recipients is online/registered. In this case, the call is forwarded immediately. ▪ All of the defined recipients rejected the call or are busy.
Forwarding To	If none of possible recipients defined in the ***Target*** field accepts a call (because they are busy, not on-line, etc.), this call is forwarded to users, groups, etc. defined here. You can use a drop down to select from a list of extensions and groups defined on the **Extensions/Groups** tab.



Field	Description
Trunk	Select one of the options: ▪ *None* – does not distinguish trunk/non-trunk calls ▪ *Call form trunk* – a call is realized via a trunk (gateway) defined within the **Trunks** tab ▪ *Call not from trunk* – a call is not realized via a trunk (gateway) defined within the **Trunks** tab
Time Filter	You can set a time interval when the rule is to be active. Click the "**...**" button to open the **Schedule Task** dialog. For more information about this dialog, refer to the **manual.chm – Shared Topics – Schedule chapter – Schedule** section.
Apply if URI not registered in the location service	If checked, the rule is used only if a SIP user is not online.

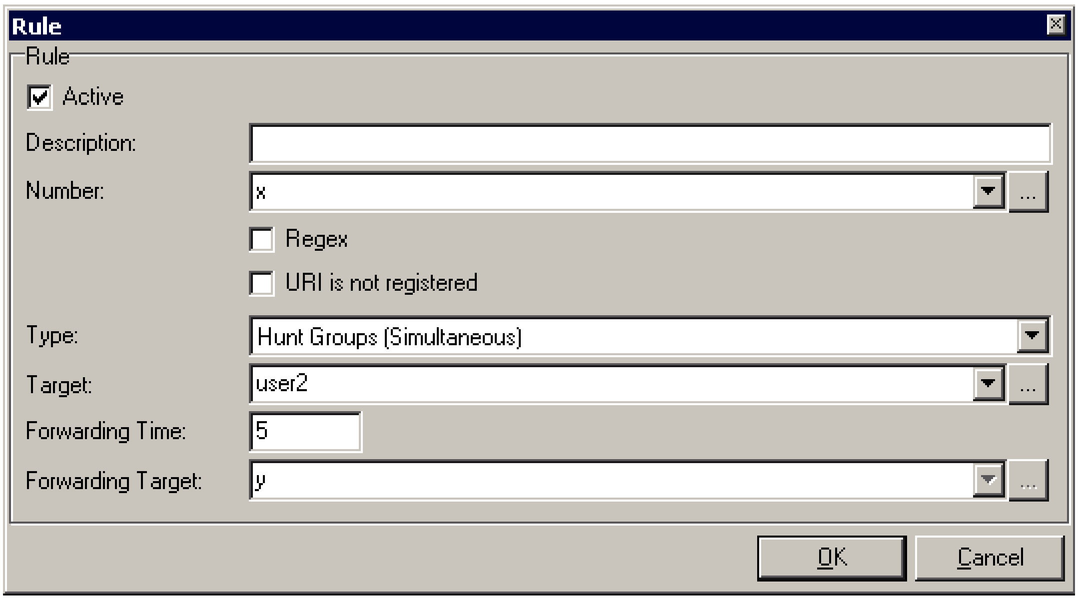
Dial Plan Examples

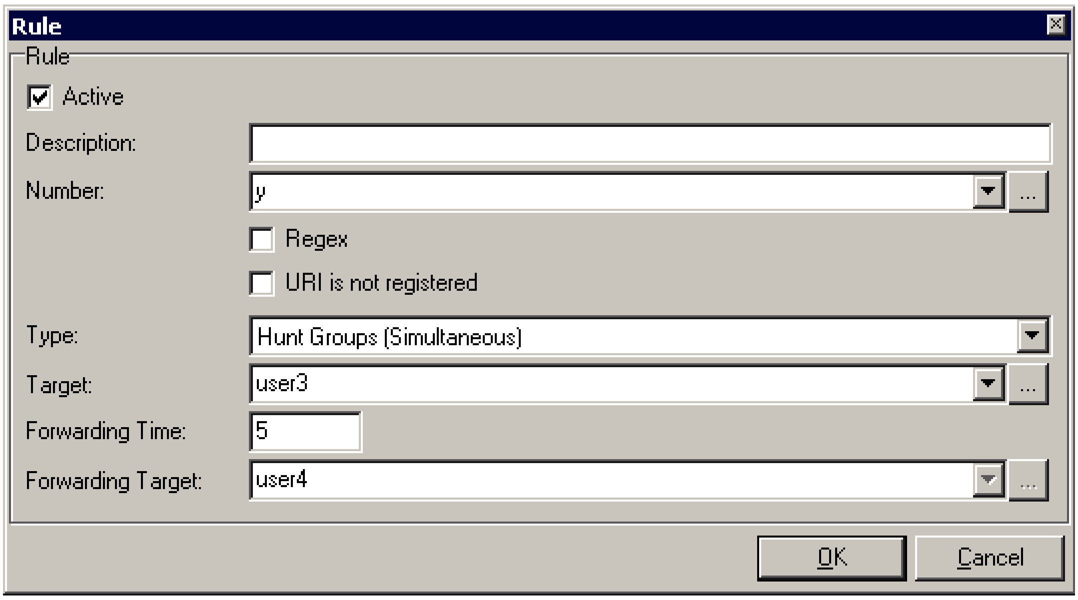
Hunt Groups



When ***x*** is called, ***user2*** and ***user3*** are dialed simultaneously and after 10 seconds (unless one of them answers the call) ***user4*** is dialed.

Another example:

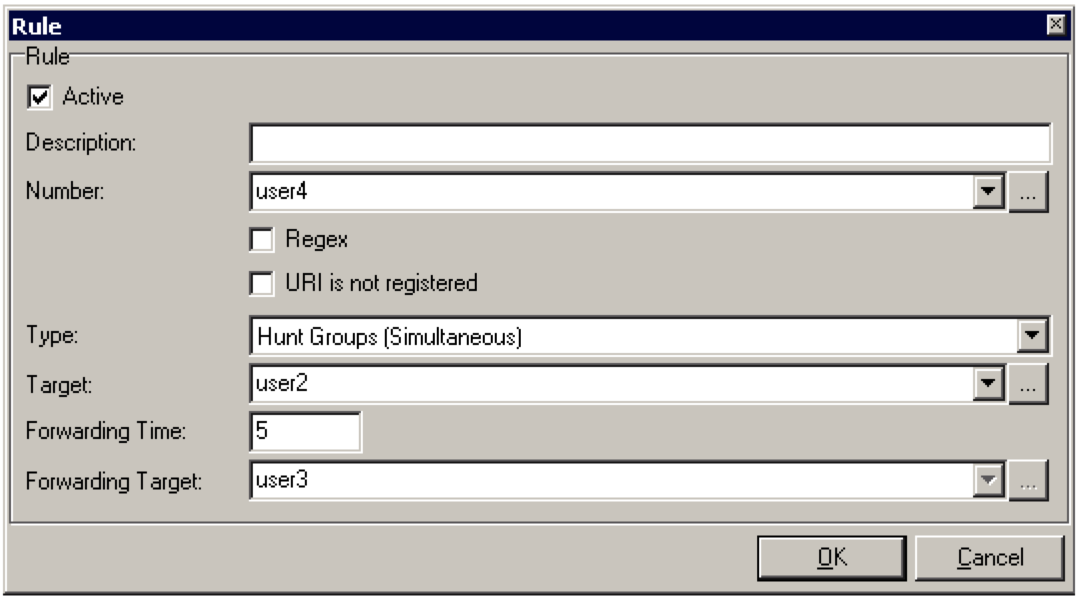




When ***x*** is dialed, ***user2*** is called and after 5 seconds (unless he/she answers the call) ***user3*** is called and after 5 seconds (unless he/she answers the call) ***user4*** is called.

*NOTE: The order of dial plans in the list (within the **Dial Plan** tab) is important! If the dial plan with **y** is placed above the **x** one in the dial plan list, users 3 and 4 will not be called.*

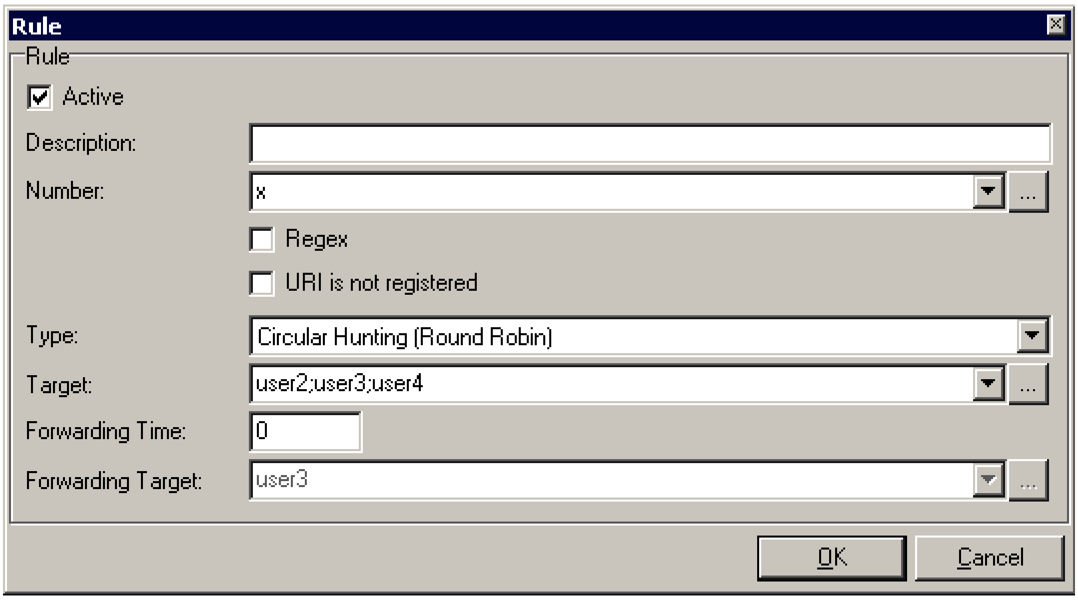
URI Is Not Registered



In this case, ***user4*** is both the name of a dial plan and also an existing user account on the server. When ***user4*** is dialed then ***user2*** is called.

If you want to prefer the user account before the name of the dial plan, the ***URI is not registered*** box must be ticked. In this case, calls are redirected when ***user4*** is not logged on.

Circular Hunting

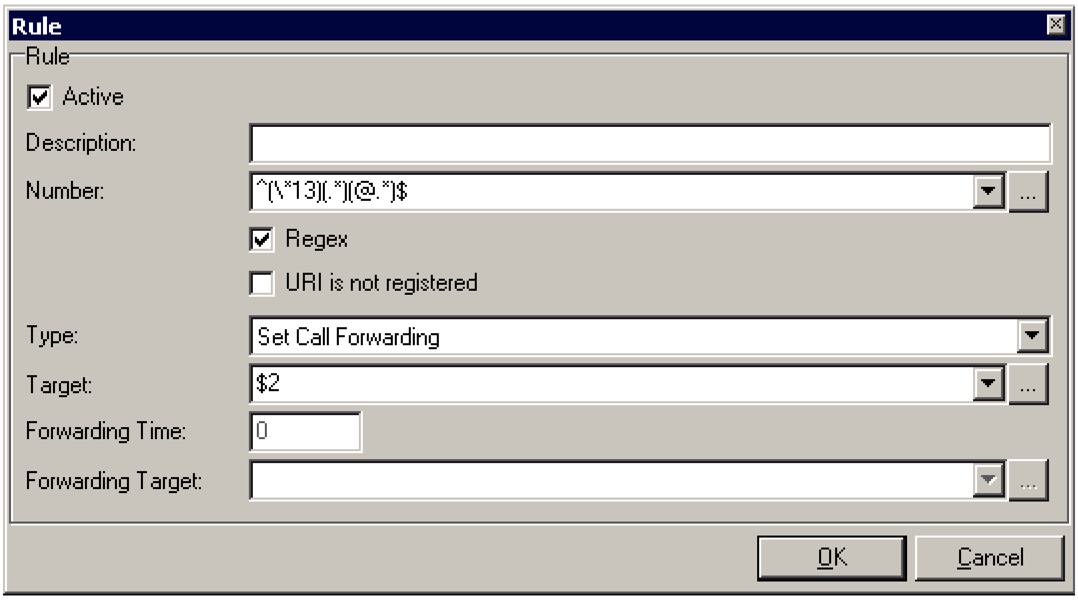


If ***Forwarding Time*** is set to 0 and ***x*** is dialed, all target users (***2***, ***3*** and ***4***) are ringed simultaneously.

If ***Forwarding Time*** is set e. g. to 5, ***user2*** is dialed, after 5 seconds ***user2*** and ***user3*** are called simultaneously and after another 5 seconds ***user2***, ***user3*** and ***user4*** are called simultaneously.

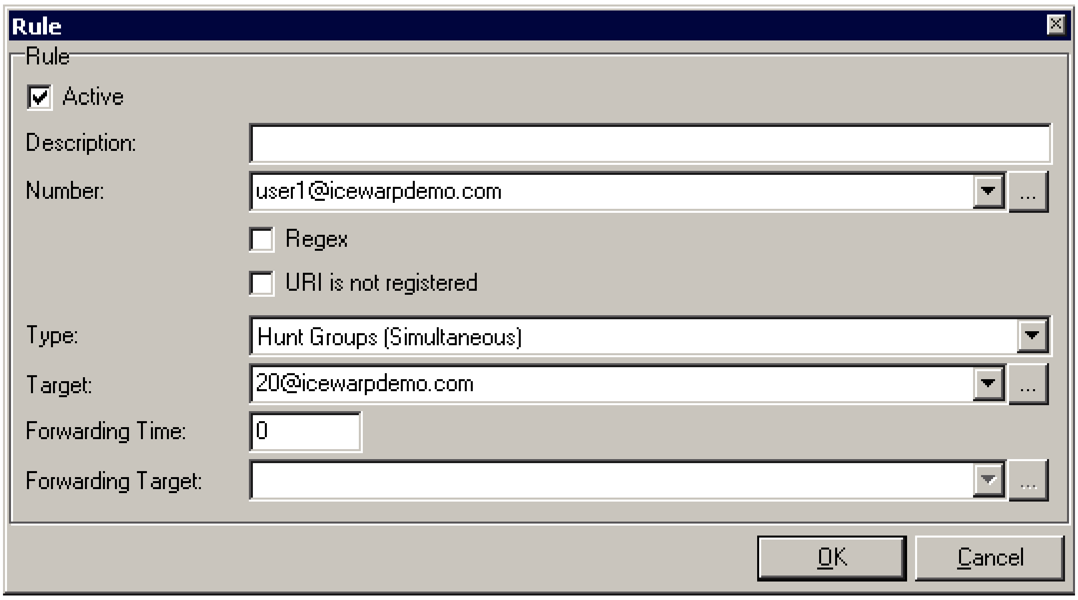
*NOTE: In this case if **user3** is inaccessible, the next user in the list (**user4** in this case) is called instead.*

Setting of Call Forwarding



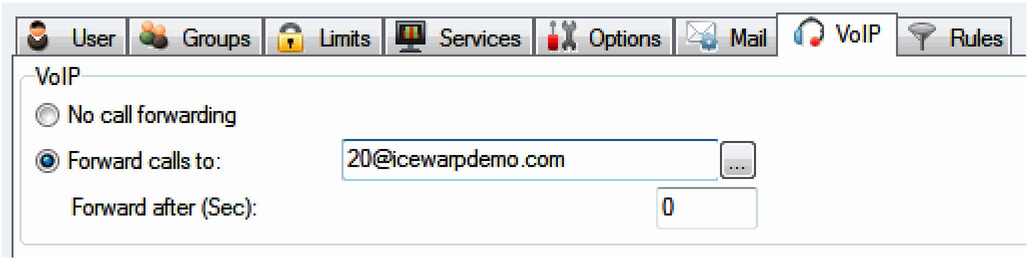
When an administrator creates this rule, the regex says that all calls to strings starting with ****13*** will be forwarded to the given string which follows after ****13*** (up to the ***@***).

So when ***user1@icewarpdemo.com*** dials e. g. ****1320***, IceWarp Server creates the following rule:

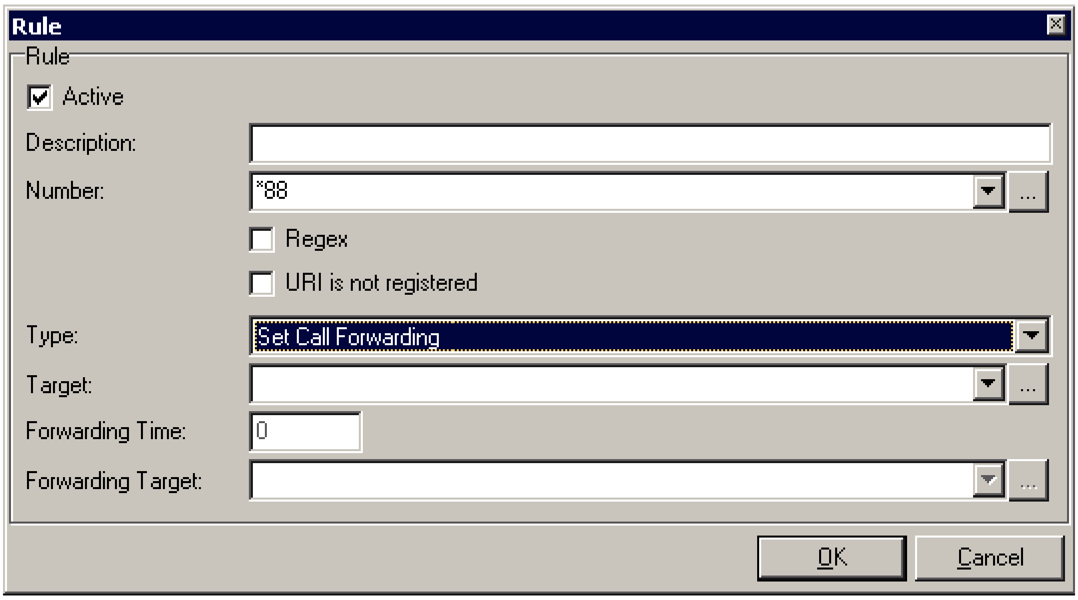


It means that all calls to ***user1*** will be forwarded to ***20@icewarpdemo.com***.

This can also be seen within user's account settings:



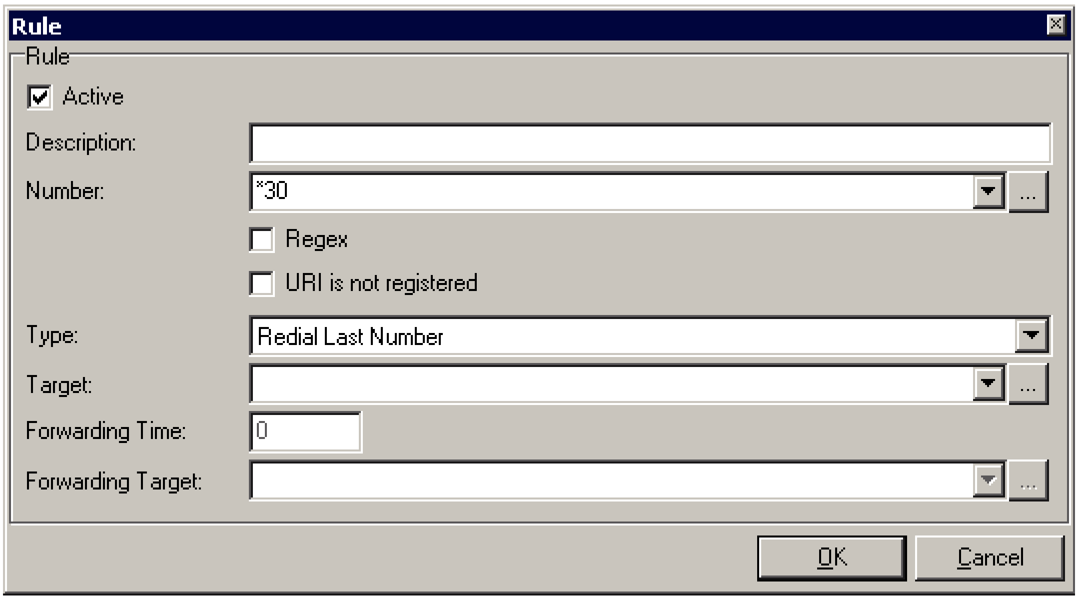
To allow users to disable call forwarding, create a rule similar to this one:



Leave the ***Target*** field blank. Users can disable set call forwarding by dialing ****88***.

Re-dial Last Number

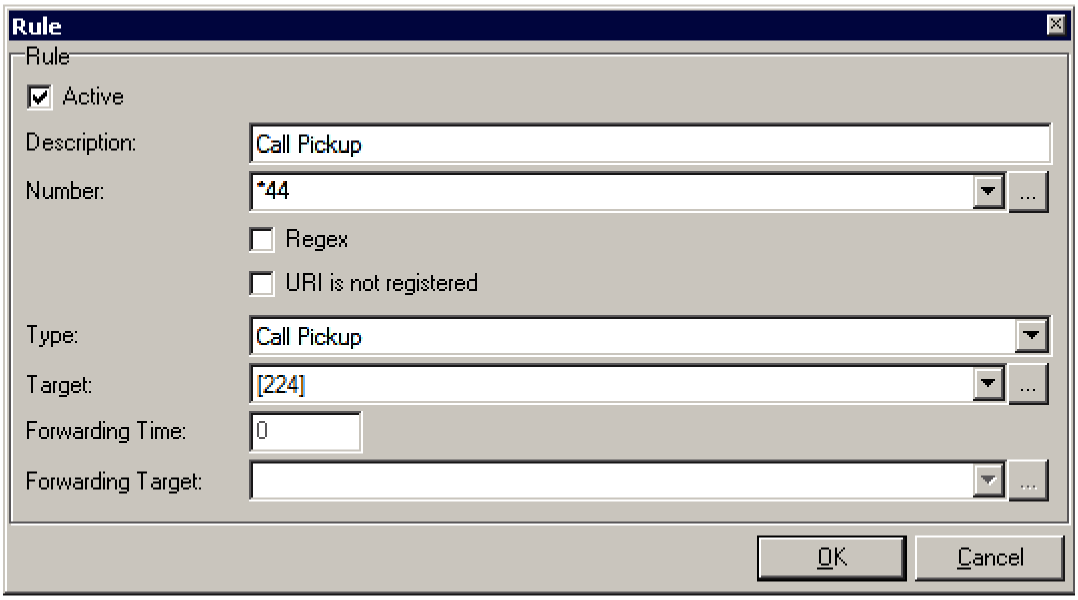
This rule allows users to dial the number of the last connection regardless of whether the user initiated, received, rejected or missed this call.



Call Pickup

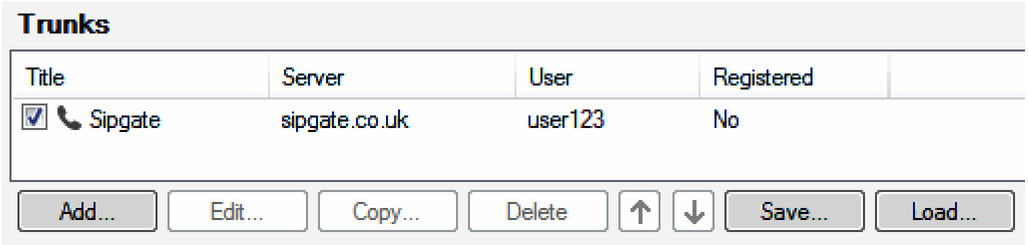
Imagine the situation there is just one employee left in the office and his colleague's phone begins to ring on the other side of the office. The employee does not need to physically go to his colleague's desk and pick up the phone. He/she can simply dial a given number to answer the phone.

This group of colleagues has to be entered into the ***Target*** field (and before defined within the **VoIP – Extensions/Queues** tab).

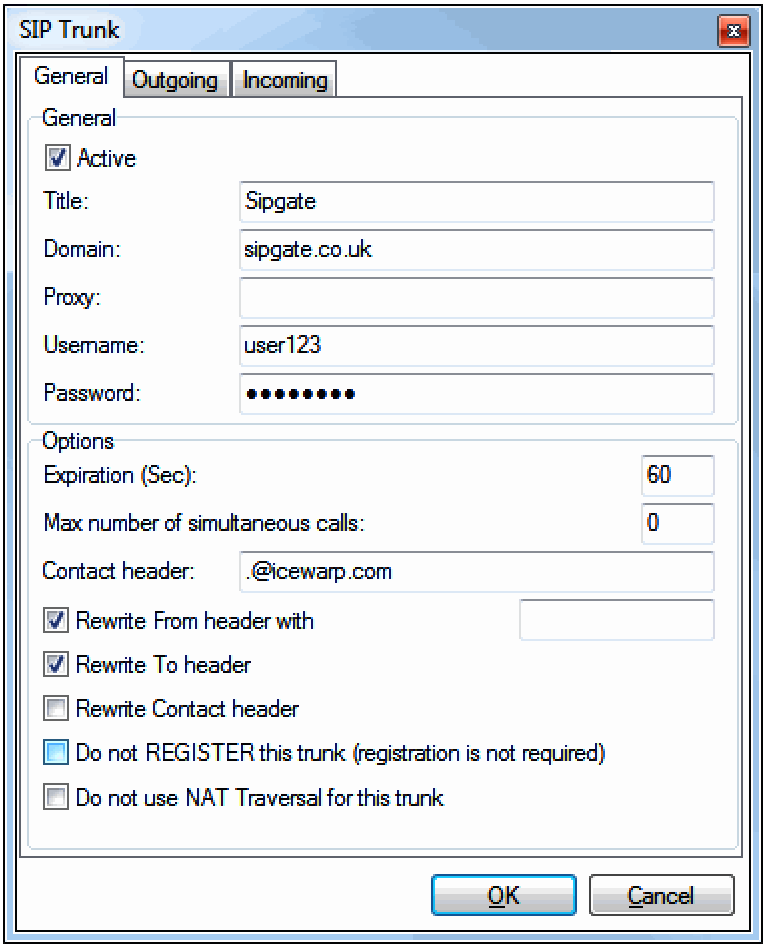


Trunks

Here you can specify trunks (gateways) you wish to route calls to. Trunks are usually an interface to a non-SIP communication system, such as public telephony, and you would normally have to pay subscription or usage charges to the gateway provider.

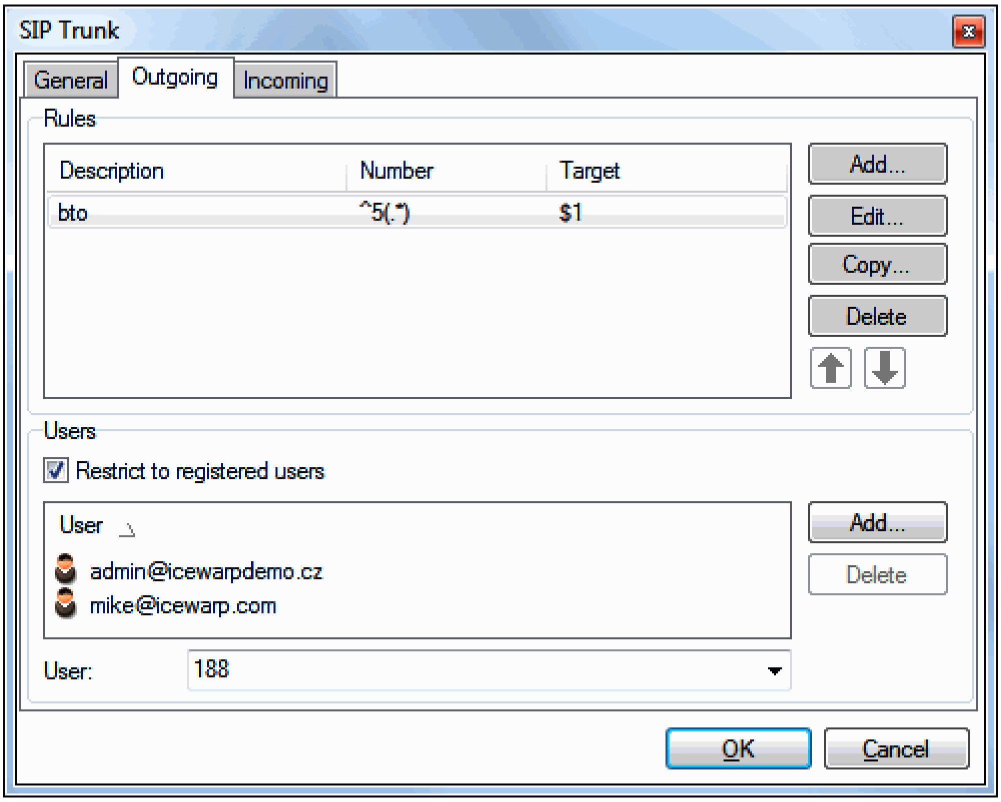


Button	Description
Add	Use the button to add a new trunk. The **SIP Trunk** dialog opens – see further.
Edit	Use the button to edit properties of the selected trunk. The **SIP Trunk** dialog opens – see further.
Delete	Click the button to delete the selected trunk.
Save	Click the button to save a list of trunks. A standard file browser dialog opens.
Load	Click the button to load a list of trunks. A standard file browser dialog opens.



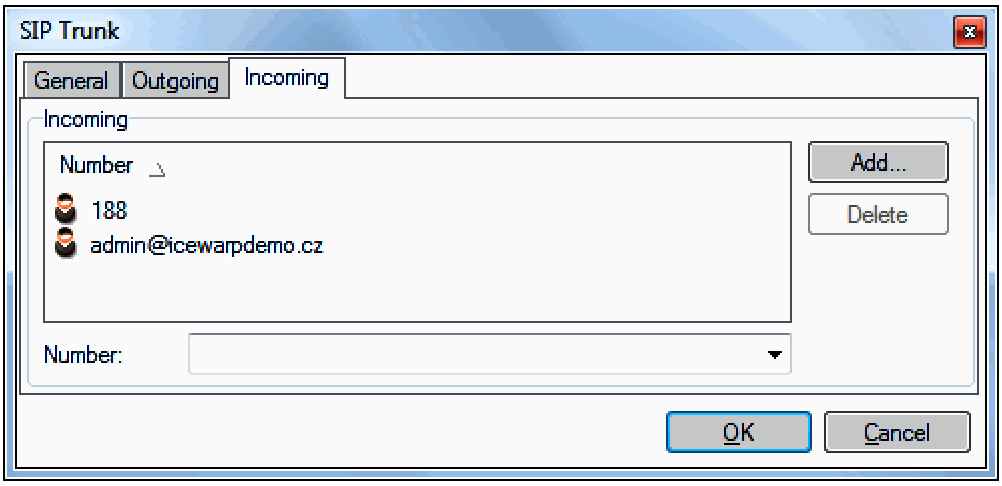
Field	Description
Active	Check this box to make this trunk active.
Title	A descriptive name for the trunk.
Domain	Specify the domain of the trunk provider.
Proxy	Specify the IP address or hostname of any proxy server IceWarp Server should use to get to this trunk.
Username	Specify the username supplied by your trunk provider.
Password	Specify the password for the above username.
Expiration (Sec)	Specify here how often, in seconds, the IceWarp Server should re-register with the gateway. Basically, this tells the trunk that the server is still here and available.
Max. number of simultaneous calls	Specify here the maximum number of simultaneous calls allowed via this trunk. Can be useful in limiting bandwidth usage.
Contact header	Fill in this field incoming calls from the Skype SipToSis to be properly detected. ***Contact header*** is used to register to a SIP gateway (remote SIP server). The remote server contacts us using this ***Contact header***. Some servers do not accept the original format and strictly require the format as follows: ***user@127.0.0.1*** It is possible to set this value here. *NOTE: It is also possible to define e. g. this:* ***272048628@sip.mujtelefon.cz;272048628*** *In this case, IceWarp Server will also treat all incoming calls with URIs beginning with* ***272048628****.* For additional information, refer to the **Skype Gateway** section.
Rewrite From header with	If enabled, the *From* header of the packets going into the gateway is changed to correspond with the gateway user (given *Username* is used). Optionally, exact value of the *From* header can be entered into the text box. The entered value can be in one of the following forms: *user* or *user@domain*.
Rewrite To header	Normally this field should be left un-checked. This is here in case your gateway provider requires it.
Rewrite Contact header	Tick this box if one of the following applies: ▪ The remote gateway can deal only with digits – the domain from the ***Contact heade***r field is rewritten to the appropriate IP address. ▪ Your trunk provider requires it. (Deeper understanding of SIP advantageous.)
Do not REGISTER this trunk	Tick this box if you want to prevent the trunk from registration. E. g. SipToSis (see the **Skype Gateway** section) behaves as a SIP client. I. e. it is not possible to register but it is used as a trunk. Use this box to deal wit this situation.
Do not use NAT Traversal for this trunk	Tick this box if you do not want to use NAT Traversal. If you know, that your provider has NAT, you can switch your one off. All will work properly and a stream will not be slower. Another situation when to use this feature is the case you use a local SIP client. There is no need to use NAT.

The **Outgoing** tab allows you to select particular calls to use this trunk, using regex conditions.



Field	Description
Add, Edit	Click the button to add/edit a routing rule and rewrite expressions. The **Rule** dialog opens. For details about this dialog, refer to the **Dial Plan** section.
Copy	Select a rule from the list and click the button to copy this rule. The **Rule** dialog opens – here you can change some settings. Useful if you want to create a similar rule.
Delete	Click the button to delete the selected condition.
Arrows	Use the buttons to move the selected condition up or down in the list.
Restrict to registered users	Check this box to restrict the users who can use this gateway to place an outgoing call. This option is recommended as otherwise you would leave your gateway open to anyone who knows of its existence. Use the ***Add*** button to open the **Select Item** dialog, allowing you to add accounts and/or domains to the list.
User	Use this field to select and add users pre-defined within the **Extensions/Queues** tab. (Click the ***Add*** button to add.)

The **Incoming** tab allows you to specify where an incoming call to your number on this trunk is routed.



Button	Description
Add	Use the button to specify which user or users the call should be routed to. 1. If no users are specified, IceWarp Server will do nothing with the call request, not even reject it. 2. If one user is specified, IceWarp Server will attempt to route the call to that user. 3. If multiple users are specified, IceWarp Server will attempt to contact all of those users simultaneously, and will wait until either: ▪ A user accepts the call, in which case it is routed to that user. ▪ Or all users reject the call or the request times out, in which case IceWarp Server will reject the incoming call.
Delete	Use the button to delete the selected user.
Number	Use this field to select and add numbers pre-defined within the **Extensions/Queues** tab. (Click the ***Add*** button to add.)

Skype Gateway

This gateway can be used for phone calls from WebClient WebPhone to Skype users.

Configuration of this gateway is based on the SipToSis project: **http://www.mhspot.com/sts/siptosis.html**

It is a Java Skype SIP gateway and is for free. It supports running multiple instances and multiple Skype instances too. A Skype client is required to be installed on the machine.
It can be easily integrated via VoIP gateways and users can just use both their soft-phones or hard-phone as usually.

How to Set Gateway

There is an old free version of SipToSis from December 2009. (The new version is paid – $2.50.)

Closely follow the guidelines how to setup SipToSis in the **ATA Setup** section. Basically, you just install Skype, extract SipToSis and run it via a batch file on Windows. Then you need to make a few configuration changes:

1. In the **siptosis.cfg** file:

   Set the port where SipToSis should be bound.

   E. g. to port 5080:
   ...
   host_port=5080
   ...
   Do not change anything else.

2. In the **SkypeOutDialingRules.props** file:

   This file specifies the rules to dial out. You do not need to touch it. It already contains something and there are simple rules you can skip and use our VoIP gateway rules instead.
   The SipToSis acts as a SIP client which when called will direct a call to the alias of the SIP URI (e. g. ***echo123@127.0.0.1:5080***).

3. In the **SkypeToSipAuth.props** file:
   This is a file for specification of incoming Skype calls and how they should be handled.

   It can be set e. g. to:
   ****,sip:siptosis@127.0.0.1***

   Which means forward calls to ***sip:siptosis@127.0.0.1***

   The IP address of 127.0.0.1 (without a port) will contact the local SIP server and the address will be matched to the ***VoIP Gateway Contact*** header and will be linked with the Skype gateway setting on your VoIP server.

4. Setting up an IceWarp VoIP Skype gateway:

   Create a new gateway:

   Server: ***127.0.0.1:5080*** (required)

   Proxy:

   User: ***siptosis***

   Pass: ***siptosis***

   Outgoing rules:

   Condition: ***^(1)(.*)@(.*)$***

   Rewrite: ***$2@127.0.0.1:5080***
   This means that if a number starts with "1", take everything after than and call via the SipToSis gateway.

   E. g.: ***1echo123*** would call ***echo123*** on Skype.

5. Restart SipToSis, start Skype and you can make and receive your first calls.

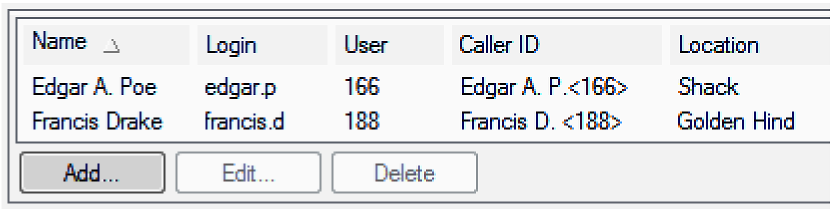
Extensions

This section allows you to define:

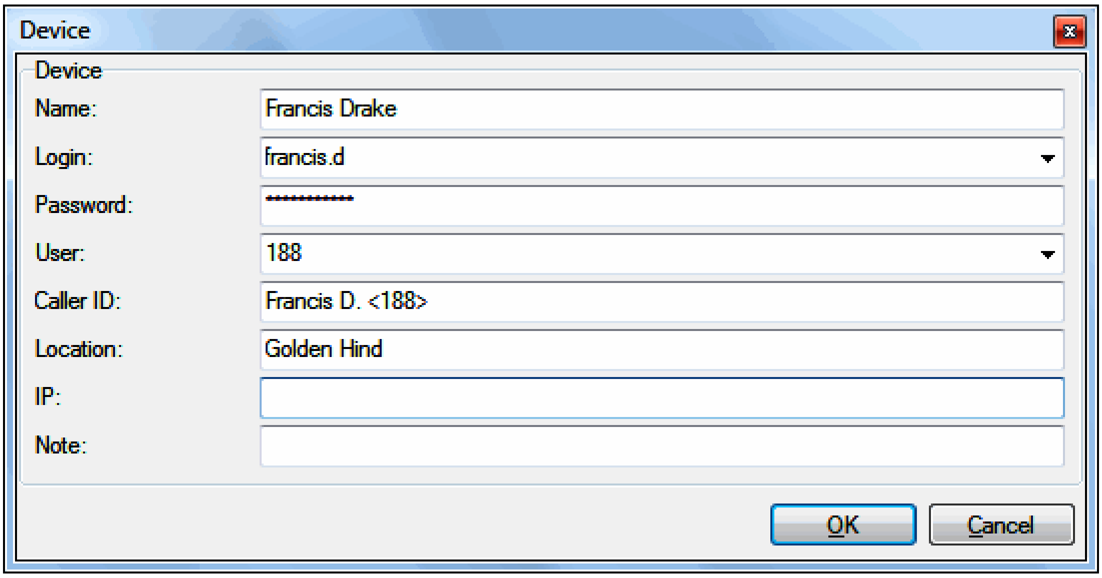
- Devices
- Extensions
- Queues

Devices

This tab allows you to define extensions for devices (phones) that will authenticate to SIP as standalone SIP accounts.



Button	Description
Add	Click the button to add a new extension. The **Device** dialog is displayed.
Edit	Select an extension and click the button to edit this extension's entries.
Delete	Select an extension and click the button to remove this extension.

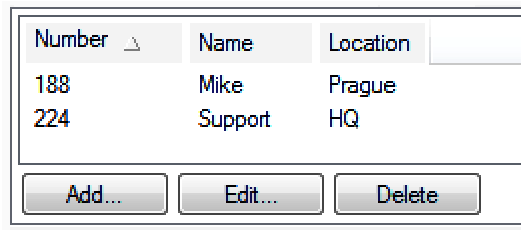


Field	Description
Name	Enter a descriptive name (user or device).
Login	Enter a login name.
Password	Enter a login password.

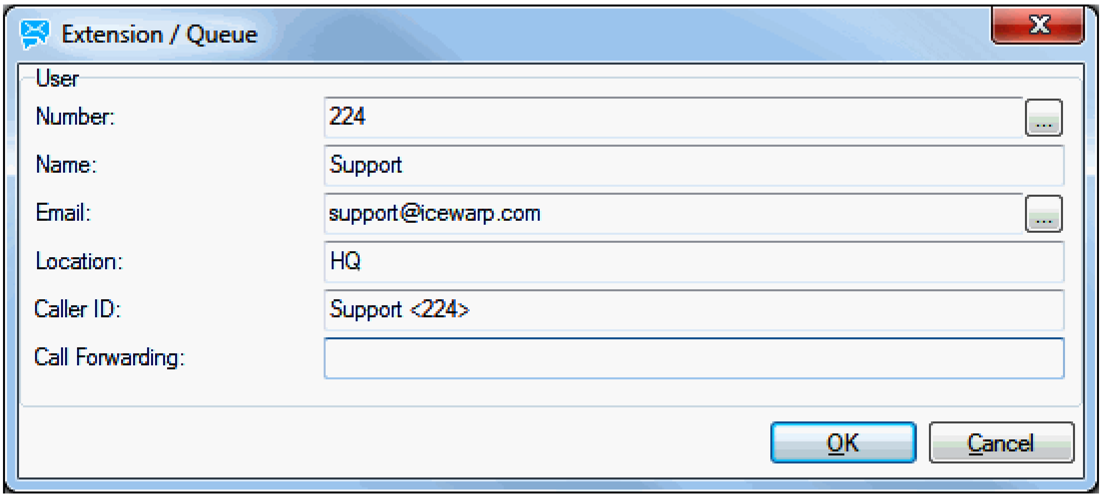
User	Link a device with a user defined upon the **Extensions/Queues** tab.
Caller ID	Specify what ***From:*** header will be used for calls originating from this device (It does not matter what information you set in the phone itself, it will be overwritten by this value.)
Location	Enter a device location (optional).
IP	The IP can be used as a security feature to limit an authentication of the device just from certain IP addresses (LAN etc.).
Note	Enter a note about the device, if desired.

Extensions

This tab allows you to define users that can be linked with SIP devices or other SIP feature (trunks, dial plans, etc.).



Button	Description
Add	Click the button to add a new extension. The **Extension/Queue** dialog opens.
Edit	Select an extension and click the button to edit this extension's entries.
Delete	Select an extension and click the button to remove it.

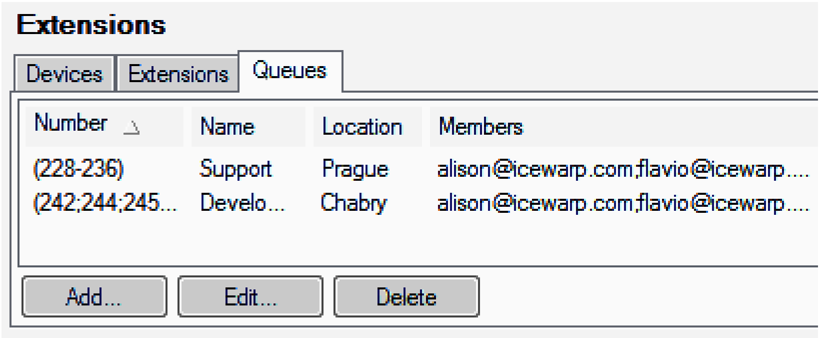


Field	Description
Number	Fill in a single number, interval (300–315) or a group of numbers (300;301;302). These numbers are to be unique in the system.
Name	Enter a user name.
Email	Enter or select a user's email.
Location	Enter a user's location (optional).

Caller ID	***Caller ID*** overrides the device ***Caller ID*** and has the same functionality. If any device is linked with the user, the user's ***Caller ID*** will be used (if not blank).
Call Forwarding	This field lets you define call forwarding directly on this user level. This can be any number controlled via the ***Dial Plan*** or other number or SIP account.

Queues

This tab allows you to define groups that can be linked with SIP devices or other SIP feature (trunks, dial plans, etc.)



Button	**Description**
Add	Click the button to add a new queue. The **User** dialog opens.
Edit	Select a queue and click the button to edit this queue's entries.
Delete	Select a queue and click the button to remove it.

Field	**Description**
Number	Fill in a single number, interval (300–315) or a group of numbers (300;301;302). These numbers are to be unique in the system.
Name	Enter a group name.
Email	Enter or select a group email.
Location	Enter a group location (optional).
Caller ID	***Caller ID*** overrides the device ***Caller ID*** and has the same functionality. If any device is linked with the group, the group ***Caller ID*** will be used (if not blank).

Call Forwarding	This field lets you define call forwarding directly on this group level. This can be any number controlled via the ***Dial Plan*** or other number or SIP account.
Members	Enter group members; use the "**...**" button to select them.

Services

To obtain needed credentials, you have two possibilities:

- Use the **Device** tab (VoIP) and define the appropriate devices there (e. g. ***voicemail*** and ***echo***). This way they will be able to authenticate and work as SIP clients. Recommended.
- Create a system account for each, but it is not to handy (smart) to have them in the account list.

*NOTE: You may face a warning message announcing that JAVA is not installed on your computer. It is needed for this service. In this case, download and install the latest **JAVA 7 (or 8) Update** from **http://www.java.com**.*

*NOTE: With 32-bit IceWarp Server, use 32-bit Java, with 64-bit one, use 64-bit Java. This is needed because Java registers its path only for the application with same bitness. Still you can register path to 32-bit Java app for 64-bit applications and it will work, only verify that the path **c:\windows\syswow64** is indeed in the system **%path%**. When you go download Java 64-bit, click the **See all Java downloads** at **java.com** and choose it manually. Or else, if your browser is 32-bit one, it will download the 32-bit version instead.*

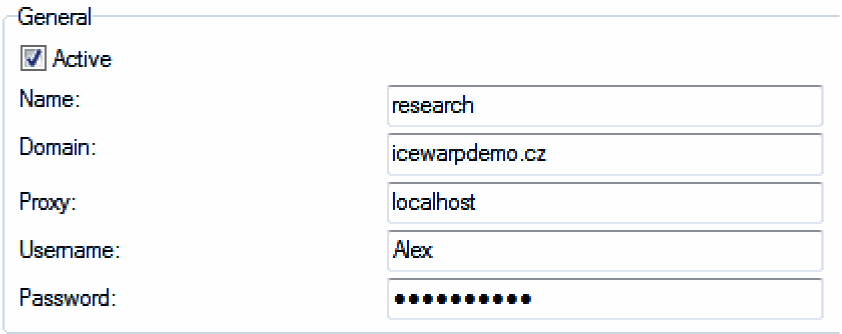
IVR

Interactive Voice Response (IVR) is a technology that allows a computer to interact with humans through the use of voice and DTMF (Dual-Tone Multi-Frequency signaling) keypad inputs.

This feature allows you to define automatic directing of callers. They select from offered voice choices and push the appropriate keys.

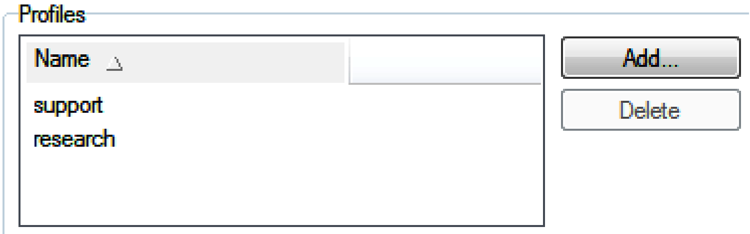
The text to speech section within the **VoIP – Advanced** tab allows IVR to generate audio files from textual data on-the-fly.

General



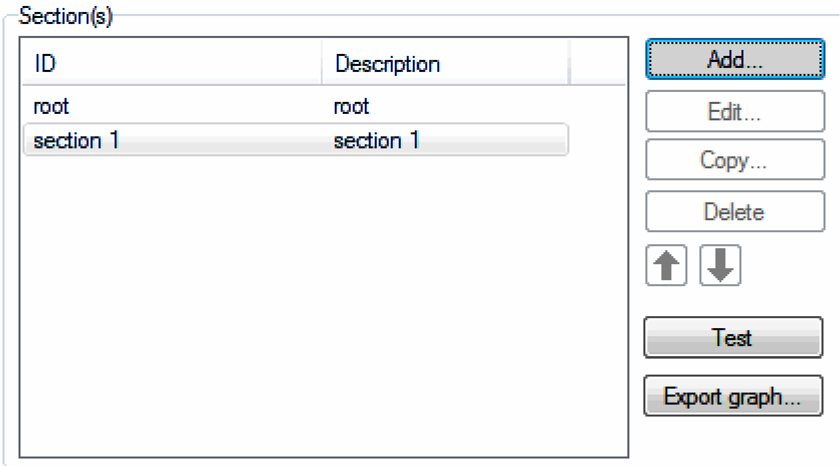
Field	Description
Active	Tick the box if you want to have this service active.
Name	Enter a service name. BE AWARE: This name has to be unique.
Domain	Fill in the name of domain where the account created for this purpose exists. *NOTE: It is also possible to use a user defined under the **Devices** tab. In this case, it is not necessary (and even possible) to specify this field.*
Proxy	Fill in the IP address of the server where SIP (VoIP) service is running. Default value and mostly used one is ***127.0.0.1***. BE AWARE: The ***<proxy,username>*** pair has to be unique.
Username/Password	Fill in the credentials of the auto attendant account. This account can be defined within one of the server domains (only for this reason) or under the **Devices** tab. *NOTE: IVR is another Java service. It needs to connect to the SIP server (just like **Echo** and **Voicemail**). For this reason there must be an account for IVR and IVR needs to know what account to connect to.* *This user (his/her number or email) is called. The root rule is executed first, then other rules/actions as defined.*

Profiles



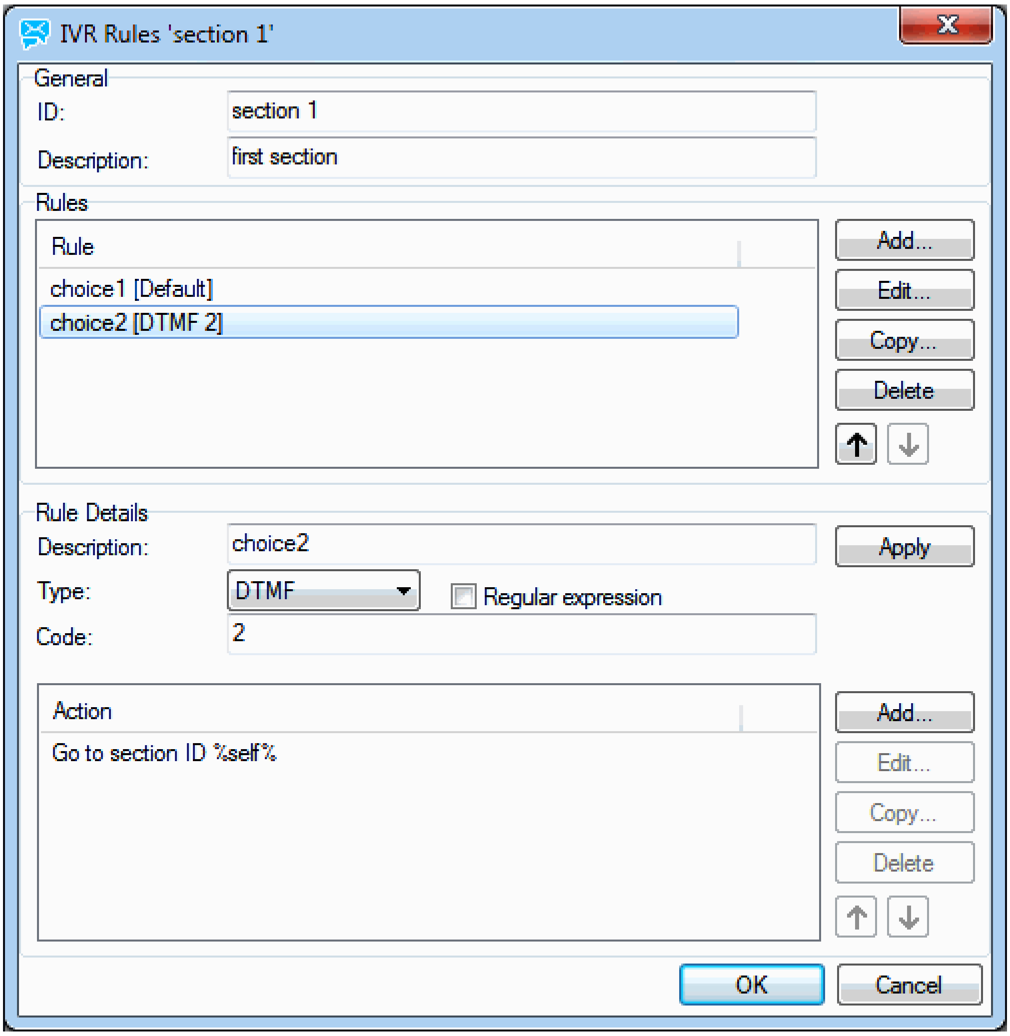
Field	Description
Add	Click the button to define a new profile. The **Profile** dialog opens. Fill in the profile ***Name*** here. After entering all needed profile information, click the ***Apply*** button in the lower right corner.
Delete	Select a profile and click the button to remove this profile.

Section(s)



Field	Description
Add	Click the button to add a new routing. The **Auto Attendant Routing** dialog is shown.
Edit	Select a routing and click the button to edit properties of this routing. The **Auto Attendant Routing** dialog is shown. See the **Type – Text to speech** field bellow.
Copy	Select a routing and click the button to open the **Auto Attendant Routing** dialog. You can add a similar routing using this button. See the **Type – Text to speech** field bellow.
Delete	Select a routing and click the button to delete it.
Arrows	Select a routing and move it up or down in the list using these buttons.
Test	Click the button to have a phone dial pad shown. This is the way how to test defined rules. *TIP: Test starts on the selected row. It is not necessary to go always through all defined rules.*
Export Graph	Click the button to export a graph of your IVR rules. The ***.gv*** format is used (supported by **Graphviz**).

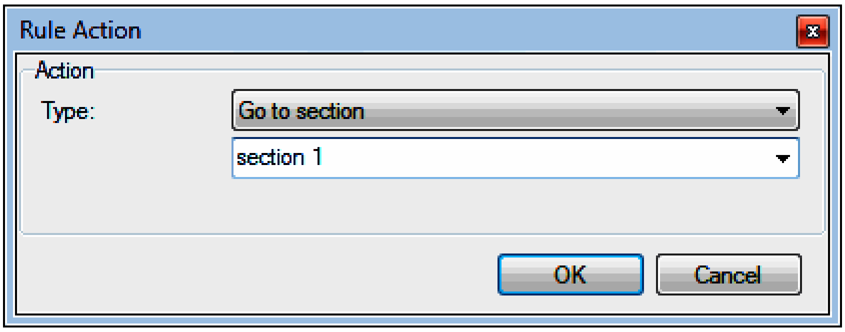
IVR Rules Dialog

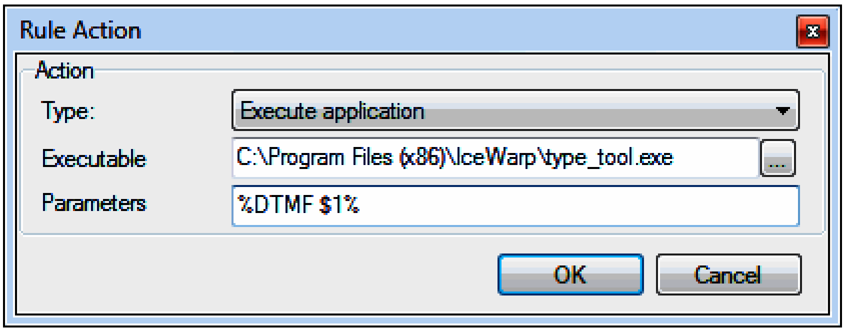

IVR Rules 'section 1'
General
ID:
section 1
Description:
first section
Rules
Rule
choice1 [Default]
choice2 [DTMF 2]
Add...
Edit...
Copy...
Delete
Rule Details
Description:
choice2
Apply
Type:
DTMF
Regular expression
Code:
2
Action
Go to section ID %self%
Add...
Edit...
Copy...
Delete
OK
Cancel

Field	Description
General	
ID	Fill in a routing name.
Description	Fill in a short descriptive text.
Rules	
Add	Click the button to add a new rule.
Edit	Select a rule record and click the button to enable editing.
Copy	Select a rule and click the button to copy it. This allows you to add a similar rule.
Delete	Select a rule and click the button to remove this rule.
Arrows	Select a rule and use arrows to move this rule up/down in the list.
Rule Details	
Description	Fill in a short descriptive text.
Type	Select from the list: ▪ **Default** – the default/initial set of actions that will be called when new action entered. ▪ **DTFM** – actions are based on a DTMF condition. ▪ **Timeout** – actions are based on a specified timeout. *Regex expression*s can be used (tick the box) – see the **example** below.
Code	Fill in a value: ▪ **DTFM** – a choice identifier. E. g. a phone key number. ▪ **Timeout** – define after how long time is the defined action (***Go to section*** or ***Transfer call to***) performed.
Add	Click the button to add a new action. The **Rule Action** dialog opens.
Edit	Select an action record and click the button to enable editing.
Copy	Select an action and click the button to copy it. This allows you to add a similar action.
Delete	Select an action and click the button to remove this action.
Arrows	Select an action and use arrows to move this action up/down in the list.
OK	Click the button to add a rule or save its changes.

Rule Action Dialog

This dialog changes according to the ***Type*** option selected:





Field	**Description**
Type	Select the appropriate action type: ▪ **Play audio file** – fill in an audio file address and name to the value field. This file will be played. ▪ **Go to section ID** – select one of defined sections from a list (or write its name). This section will be performed subsequently. ▪ **Forward call to** – fill in or select what number defined within the system is to be called (individual or group one). ▪ **Record audio file** – fill in a path where recorded audio files will be saved. Use when you require caller's answers. For a file name you may want to use the ***%tempfile%*** variable. This allows you to have unique file names. E. g. ***20110821_1024.mp3***. There are two additional options: DTFM used for finishing recording and timeout used for finishing recording. ▪ **Execute application** – use the "..." button to select a path to the application that is to be executed. See also the ***Parameters*** field. If the application has no output, it is considered as error and operation is retried. If the application output consist of formated XML tag ***<action>*** (same syntax as in **voicemail.xml**), this action is performed, otherwise nothing special happens and the following actions are evaluated. ▪ **Copy file** – use this feature to copy a recorded file to another directory (e. g. for backup reasons). Use the following syntax: ***[source];[destination]*** Example: ***data/%tempname%.wav;data/users/%caller_email%.wav*** ▪ **Delete file** – use to delete a recorded (and backed up) file. Example: ***data/%tempname%.wav*** ▪ **Silence** – specify how long the silence should be (in seconds). It is useful e.g. when you want to wait between two replays of some audio files (or between another two actions). ▪ **Text to speech** – enter some text into the value field. It will be converted into

	speech. *NOTE: To get better idea, see the complex example further.*
Parameters	*NOTE: This field is only visible when the **Execute application** type is selected.* Fill in parameters that are to be passed to your application. E. g. the ***%DTMF $1%*** variable will pass the actual DTFM key choice.
OK	Click the button to save this rule action.

Text to Speech Engine Change

To change a text to speech engine (in Windows), do the following:

1. Run the **cmd.exe** tool under a local **system** account.

   There are several ways how to do it. One of them is to schedule it:

   - Within **cmd.exe**, use the ***time*** command.

     You will obtain something like this: *The current time is: 16:29:06.96 Enter the new time*.

     Select ***Ctrl + C***.
   - Use the *at 16:30/interactive cmd.exe* command.

     You will obtain the *Added a new job with job ID = 1* message.
2. Within **cmd.exe** (running under your local system account) run the following command:

   "C:\WINDOWS\system32\rundll32.exe" C:\WINDOWS\system32\shell32.dll,Control_RunDLL "C:\Program Files\Common Files\Microsoft Shared\Speech\sapi.cpl",Speech
3. The **Speech Properties** dialog – **Text to Speech** tab opens. Select your desired text to speech engine from the ***Voice selection*** list. (Click ***OK***.)

Examples

This is a xml code (**<install_dir>/config/voicemail.xml**) with the following procedure defined:

1. Play audio
2. Record audio
3. Play recorded audio
4. Copy file
5. Delete file

```
<item id="voicemail" description="Voicemail">
          <action>
            <playaudiofile>audio/beep.mp3</playaudiofile>
          </action>
          <action>
            <dtmf>0</dtmf>
            <goto>%self%</goto>
          </action>
        <action>
            <dtmf>1</dtmf>
            <playaudiofile>audio/beep.mp3</playaudiofile>
            <recordaudiofile>
              <path>%tempname%.wav</path>
              <dtmf>#</dtmf>
            </recordaudiofile>
            <goto>%self%</goto>
          </action>
          <action>
            <dtmf>2</dtmf>
            <playaudiofile>audio/beep.mp3;data/%tempname%.wav</playaudiofile>
            <goto>%self%</goto>
          </action>
          <action>
            <dtmf>3</dtmf>
            <copyfile>
              <source>data/%tempname%.wav</source>
              <destination>data/users/%caller_email%.wav</destination>
            </copyfile>
            <deletefile>data/%tempname%.wav</deletefile>
            <playaudiofile>audio/beep.mp3</playaudiofile>
            <goto>%self%</goto>
          </action>
          <action>
            <timeout>20</timeout>
            <goto>%self%</goto>
          </action>
        </item>
```

The following text gives an example of the ***Execute application*** type use:

DTMF: ***^(.*)#$***
Path: ***/bin/ivr.sh***
Param: ***%DTMF $1%***

The rule will take any DTMF sequence ending with #, call the **ivr.sh** script and pass the DTMF number to it.
The script can do anything required – validate against a database or whatever else.

*NOTE: The **vocemail.xml** file contains information about **Voicemail**, **Echo** and **Attendant**. The order is of no importance here. Just account types are to be distinguished.*

voicemail.xml File

This file includes data as set on both **IVR** and **Services** tabs. It is possible to set these data using IceWarp Server GUI or edit this file directly. To be able to do it, you need to know variables, formats , etc.:

profile type values:

VOICEMAIL – voicemail service that sends recorded messages via email with embedded MP3

ECHO – echo service ala Skype

AA – auto attendent service

%email% – email of the callee taken from To:

%alias% – alias of the callee

%domain% – domain of the callee

%time <timeformat>% – time of voicemail received using any syntax

%callee_email% – email of the callee taken from To:

%callee_alias% – alias of the callee

%callee_domain% – domain of the callee

%caller_email% – email of the caller taken from From:

%caller_alias% – alias of the caller

%caller_domain% – domain of the caller

%length% – call length of recorded message

time format:

Letter	Date or Time Component	Presentation	Examples
G	Era designator	Text	AD
y	Year	Year	1996; 96
M	Month in year	Month	July; Jul; 07
w	Week in year	Number	27
W	Week in month	Number	2
D	Day in year	Number	189
d	Day in month	Number	10
F	Day of week in month	Number	2
E	Day in week	Text	Tuesday; Tue
a	am/pm marker	Text	PM
H	Hour in day (0-23)	Number	0
k	Hour in day (1-24)	Number	24
K	Hour in am/pm (0-11)	Number	0
h	Hour in am/pm (1-12)	Number	12
m	Minute in hour	Number	30
s	Second in minute	Number	55
S	Millisecond	Number	978
z	Time zone	General time zone	Pacific Standard Time; PST; GMT-08:00
Z	Time zone	RFC 822 time zone	-0800

Example:

"EEE, d MMM yyyy HH:mm:ss Z" = Wed, 4 Jul 2001 12:08:56 -0700

maxcalltime tag values:

-1 – unlimited

0 – call is immediately stopped after playing audio file

<seconds> – otherwise specifies number of seconds after which call will be stopped

playaudiofile and playbyeaudiofile can be specified the settings aread or in each profile

<proxy> Tag

In the case you do not want to use the default VoIP service port (5060), you can specify different ports for the **Voicemail**, **Echo** and **Conference** services either in GUI (**VoIP – Services**) or within the **voicemail.xml** file (**<install_dir>/config/**) – *<proxy>* tags for individual accounts/services.

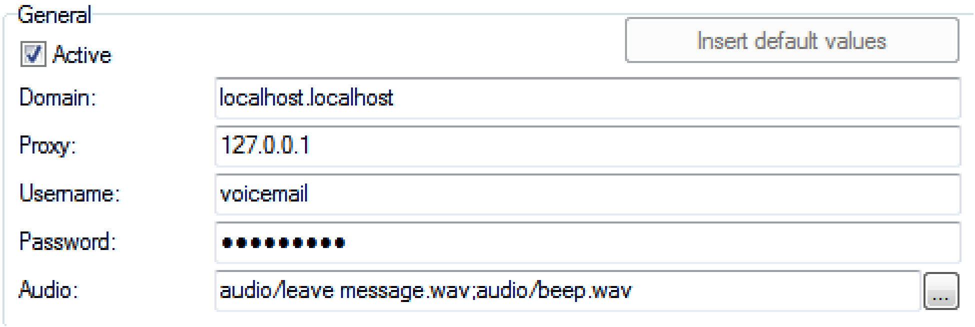
The syntax is as usually – ***IP:port***.

Voicemail

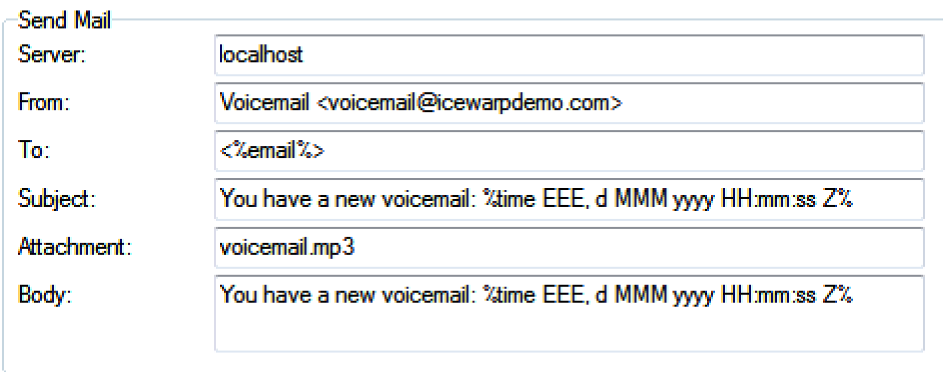
The Voicemail service is used as an answering machine. It is not used for direct calls but instead you make call forwarding either via the **Dial Plan** or the **<user> – VoIP** tab – ***Forward calls to*** option.

For example, if a user does not answer within 30 seconds, transfer this call to ***Voicemail***. It will pick up the call, play an audio message and record a caller's message. The message will be converted to MP3 and sent as an email attachment to the original call receiver.

It is also recommended to record your own audios and put them in the **voip/audio** directory. This ensures that they will not be overwritten in contrast the default ones.



Field	**Description**
Active	Tick the box if you want to have this feature active.
Insert default values	Click the button to use default values within this section. *NOTE: The button is active only if one (or more) of the fields is empty (except for the Password one).*
Domain	Use data for either devices or accounts you have created for – see this chapter introductory text.
Proxy	
Username	
Password	
Audio	Specify path(s) to default audio file(s) that are to be played when Voicemail is reached. ("Leave a message after the tone.") Use semicolons to separate paths.

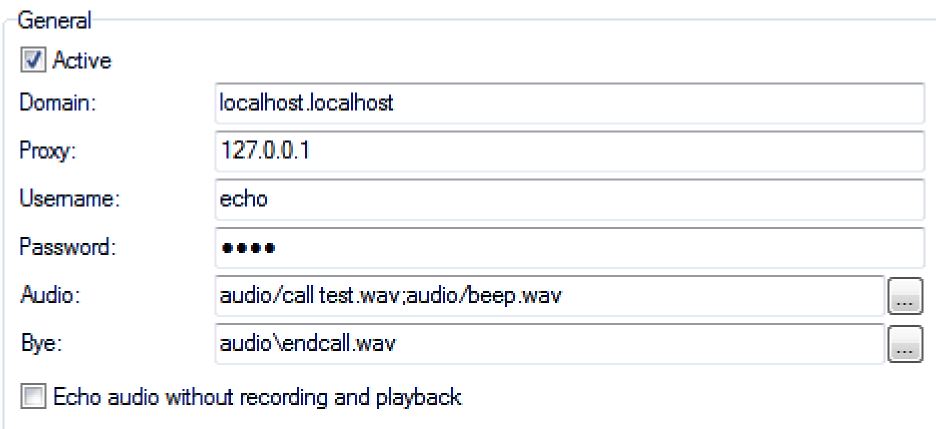


Field	**Description**

Server	Use data for either devices or accounts you have created for.
From	Specify headers and a body of the email that will be sent to the original receiver. The ***%email%*** variable sets the receiver's email address into the ***To:*** header. Unicode and any customization is possible. Even HTML mails are, if required. (In the **settings.xml** file.)
To	
Subject	
Attachment	
Body	

Echo

This service allows users to test the quality of their audio connections. They can call this service directly. Their calls are answered, they can record test messages that are played back afterwards.



Field	**Description**
Active	Tick the box if you want to have this feature active.
Domain	Use data for either devices or accounts you have created for – see this chapter introductory text.
Proxy	
Username	
Password	
Audio	Specify path(s) to audio file(s) asking to leave a test message after the tone etc. Use semicolons to separate paths.
Bye	Specify the path to an audio file saing "Good bye" at the end of an echo call.
Echo audio without recording and playback	Tick the box if you want to use the echo service without recording and playback of a test message.

WebMeetings

This powerful tool allows both system and external users to organize and join online conference meetings.

Service Pre-requisites

The server where you run IceWarp Server has to have Java installed.

Before using this service, it is necessary to configure the VoIP service (the **Advanced** tab), to switch on NAT Traversal and the **WebMeetings** service (**System – Services**) has to be running.

Users have to have installed some SIP clients on their computers (or Java, to be able to use WebPhone).

About

WebMeeting (conference) can be:

- instant
- attached to a calendar event (both recurrent and non/recurrent).

WebMeeting participants can use:

- WebClient
- Desktop Client
- Outlook Sync
- a generic client (i.e. usual Outlook, Thunderbird, EAS device).

Each participant can use more than one client – i.e. accepts an invitation in one client and then joins the meeting from another one.

WebMeeting participants can have their accounts:

- on the same IceWarp Server as organizer
- on another IceWarp Server
- on different kind of server (e.g. Exchange).

The meeting can be:

- audio and desktop sharing
- audio only.

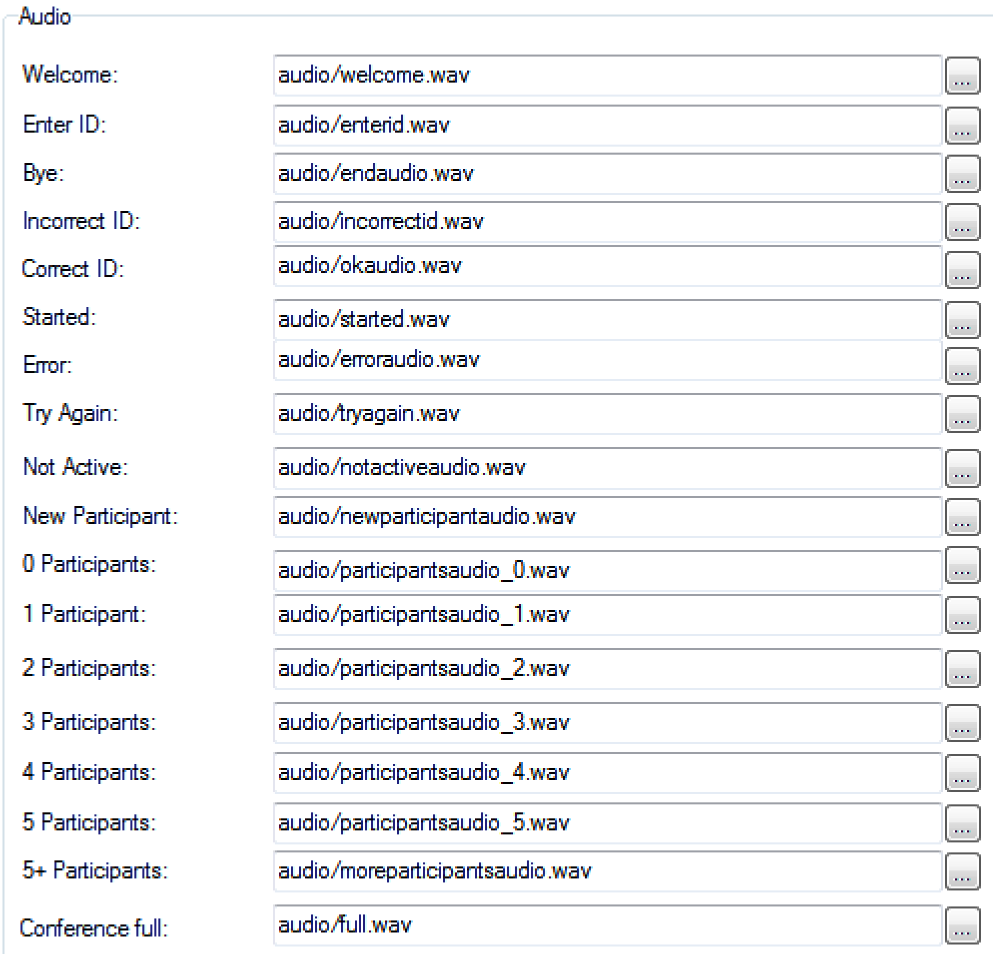
Participants can join the meeting via:

- WebClient
- anonymous page
- normal phone calling a special number
- software SIP phone calling a SIP number
- hardware SIP phone.

It is possible to launch more meetings concurrently – every meeting has its own ID.

Field	**Description**
Active	Tick this box to activate the service.
SIP Number	Email address of the "conference" account. By default, it is *conference@[your_primary_domain]*. This account is implemented and behaves like a "virtual device" defined under the **Devices** tab.
PSTN	Participants that want to join to a meeting have to call a specific phone number. You have to create a SIP gateway

Number	(under **Trunks**) with this number users to be able to call. It is just an information here – participants obtain an information message generated by the server. This message includes all information needed to join the meeting. Meeting organizers do not need to find out this number. They just add participants (via WebClient).



This pane lets you define audio files that are to be used to greet meeting participants, to inform them about inserted ID, number of other participants joined in the meeting, etc.

By default, audio files in English are provided. You may want to localize or change them. This is possible but the *.wav* format is obligatory.

It is also possible to place these files into a different directory. Use the "..." buttons to select this directory. Both an absolute path and relative (to the **audio/** folder) one can be used.

It is not necessary to use all these audio files.

Field	**Message**
Welcome	Greeting and request to enter ID
Enter ID	When calling in to the meeting with a normal phone, one is required to dial the meeting ID.
Bye	End of the meeting
Incorrect ID	Wrong ID entered
Correct ID	Correct ID entered
Started	The message played for an organiser every time when starting a meeting
Error	Meeting error
Try Again	After something went wrong – "Please try to join again later."

Not Active	Meeting established but not started
New Participant	Another participant has joined the meeting
Zero to 5+ Participants	Number of other participants
Conference full	Played when no more participants are allowed

Using the Dial via SIP Functionality

Both the IceWarp Outlook Sync and IceWarp WebClient have the ability to dial out via SIP clients.

In the IceWarp Outlook Sync:

- Locate and select the contact you wish to call (if the person you wish to call is not in your contacts, skip this step).
- Select ***Dial via IceWarp Server*** from the IceWarp Outlook Sync dropdown menu.
- Check the correct contact is displayed and click ***Dial***.
- Your SIP client will start to ring, answer it.
- After a couple of seconds the other person's SIP client will be contacted and your conversation can start.

In IceWarp WebClient:

- Click ***Dial*** on the menu bar.
- Select the contact you wish to call (or type in the email address) and click ***Dial***.
- Your SIP client will start to ring, answer it.
- After a couple of seconds the other person's SIP client will be contacted and your conversation can start.

NOTE: The call is in no way routed by IceWarp Outlook Sync or IceWarp WebClient, they are just used to initiate the call. The SIP server dials your registered client and once connected routes the call to the destination you specified. This method will work with any SIP client.

Dialling over Third Party SIP Server

For this feature description, refer to the **IceWarp WebClient Administration Guide – Administration Options – Dialling over Third Party SIP Server** chapter.

Setting up a SIP Client – X-Lite

There are numerous SIP clients available, both software and hardware ones. This section describes how to set up X-Lite to access your SIP server.

X-Lite is available from ***http://www.counterpath.com/***

First run of X-Lite

When you first run X-Lite, it will discover that you have no SIP account defined and will show the following message: "No SIP accounts are enabled".

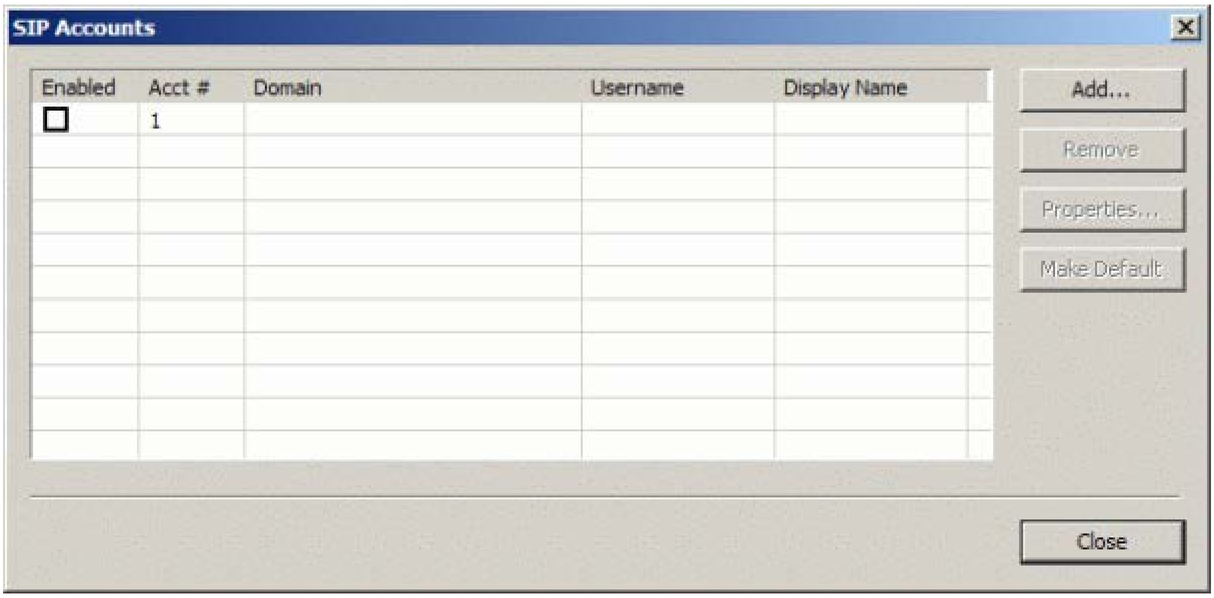
The **SIP Accounts** dialog will we be displayed automatically.

If you wish to add your IceWarp Server account to X-Lite, open the **SIP Accounts** dialog manually:

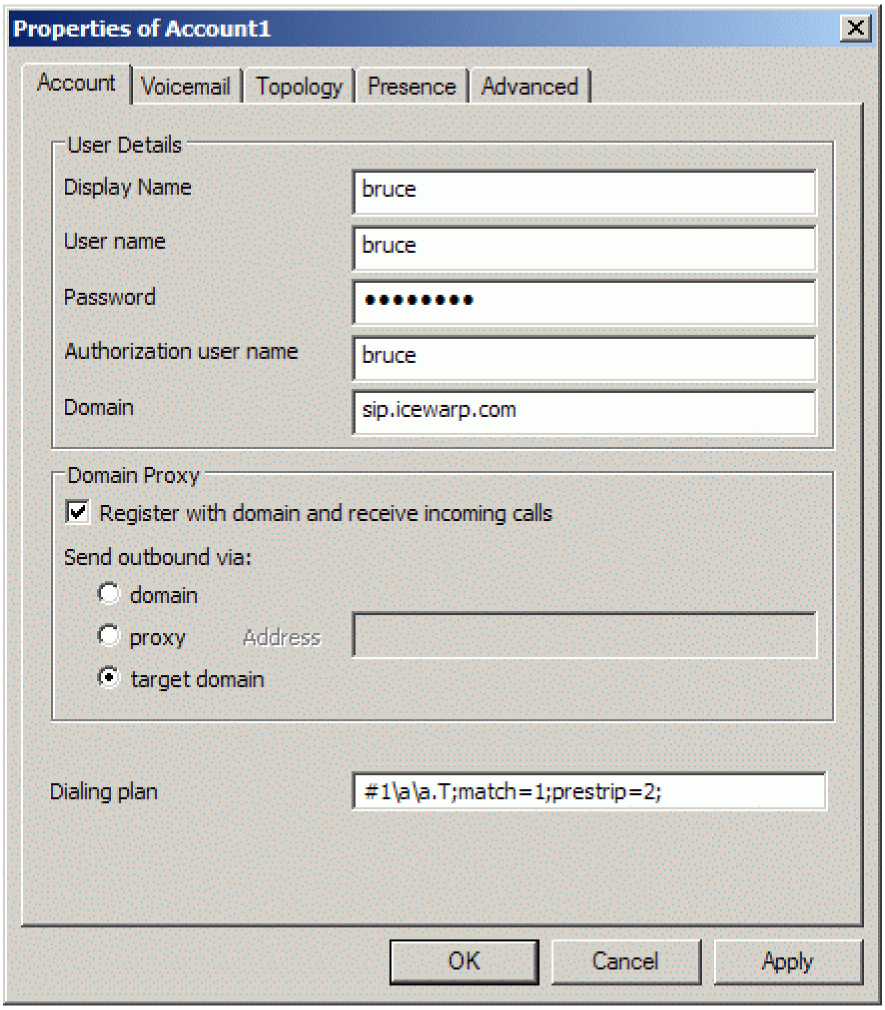

No SIP accoun
Options...
SIP Account Settings...
Privacy Rules...
Open Diagnostic Log
Open Diagnostic Folder
Help
About...
Exit
XFER
HOLD
FLASH
REDIAL
1
2 ABC
3 DEF
4 GHI
5 JKL
6 MNO
7 PQRS
8 TUV
9 WXYZ
*
0
#
CLEAR
MUTE
SPEAKER PHONE
Powered by
COUNTERPATH

SIP Accounts Dialog

This dialog shows a list of all SIP accounts you have defined.

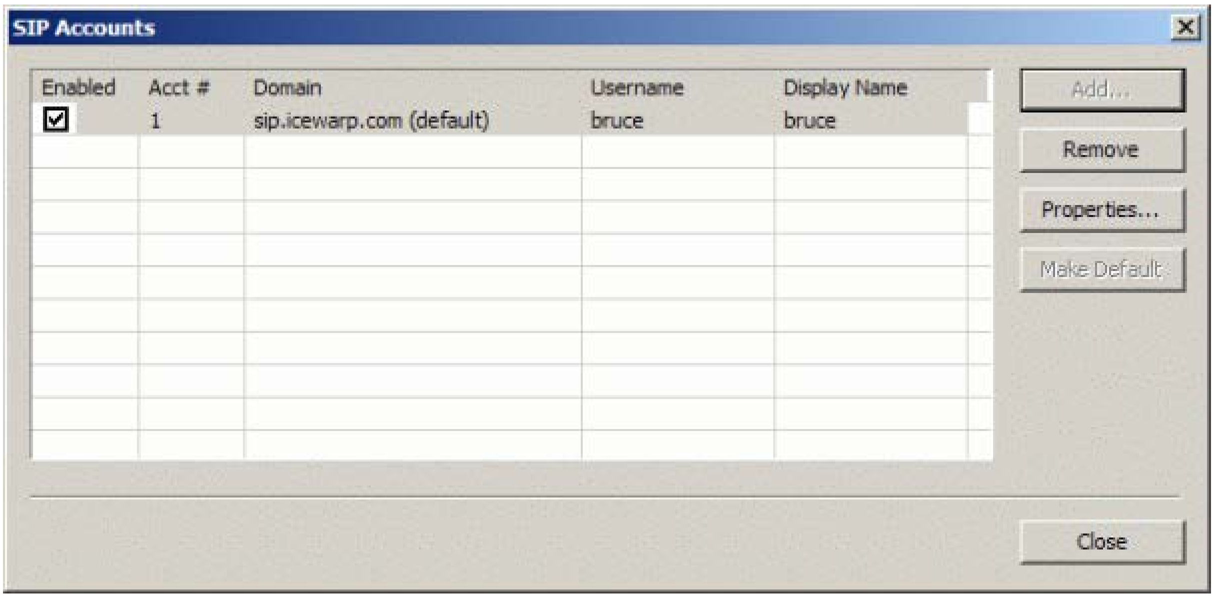


Click the ***Add*** button to define your IceWarp Server account. The **Properties ...** dialog is displayed:



Field	Description
Display name	Enter the display name you would like people to see when you are calling them.
User name	The user name supplied by your SIP service provider. *NOTE: It is necessary to log in using this user name only (not the user's e-mail address). In the case of IceWarp Server, use the alias of the appropriate account.* *NOTE: This applies even if the **Users login with their email addresses** option (**Domains & Accounts – Policies – Login Policy – Login Settings**) is selected.*
Password	The password supplied by your SIP service provider.
Authorization user name	Same as your ***User name***.
Domain	Enter the domain name of your SIP service supplied by your SIP service provider.
Domain Proxy	Leave as is.
Dialing plan	Leave as is.

Click ***OK*** to return to the **SIP Accounts** dialog:



Click ***Close*** to return to X-Lite.

X-Lite will now attempt to connect to your SIP service provider and will show the following if successful:

3CX PBX Integration

3CX Phone

This new SIP phone supports Android, iPhone and Windows. The SIP phone itself is free and supports all the SIP extensions such as TLS, TCP, UDP, and secure RTP, REFER, Replaces, Video, etc.

For more information, follow this link: ***http://www.3cx.com/VOIP/voip-phone.html***.

Integration

Integration requirements are:

- WebPhone compatibility
- Support for local calls to email addresses of IceWarp Server accounts
- Support for incoming calls from 3rd party PBX
- Support for outgoing calls to 3rd party PBX
- All external calls to the 3rd party PBX have to be routed to HW phones already linked to that PBX

All of the above is possible, you just need to know how. The configuration is really simple.

Just create a new SIP gateway to the 3rd party PBX and preferably set it up so authentication is not required (does not really matter though). Now, all you have to do is to specify the outgoing rule:

Number: .*

Regex: True

URI is not registered: True

This will make sure that all calls made through IceWarp PBX (probably WebPhone or somebody connected to IceWarp Server making a call) will be routed to the 3rd party PBX if the destination is not a local registered number. If it is a local registered number, the call will be established locally. The rest is up to the 3rd party PBX to support all the calls made by our VoIP.

The whole scenario covers all requirements above. You can use WebPhone with the default settings (no special settings for the 3rd party PBX required – although possible). Local calls will still work and external calls will work easily from both local and external VoIP.

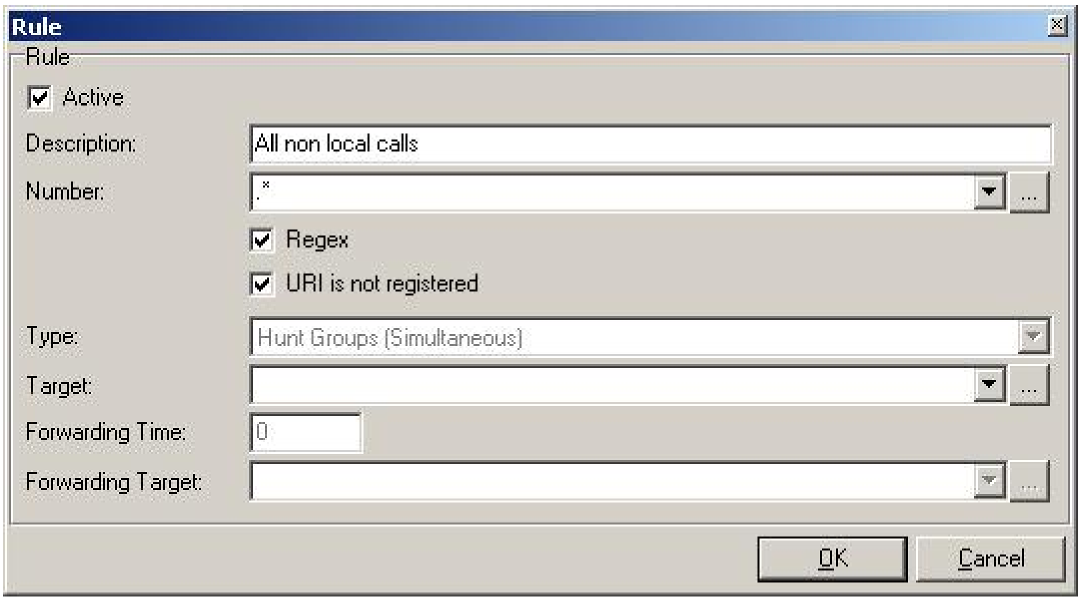
IP Detection Functions

SOCKS and VoIP IPs are automatically set and you do not need to set them manually while configuring the server for the first time. This handles 99% of problems in configuring SOCKS and VoIP.

In WebPhone (and in **refer.dat** of the mailbox), you can specify an external SIP server to use for ***Call Now*** functionality (internal ***API SIPReferCall()***). Originally, it had to be a SIP address. Now, it is possible to use a URL too.

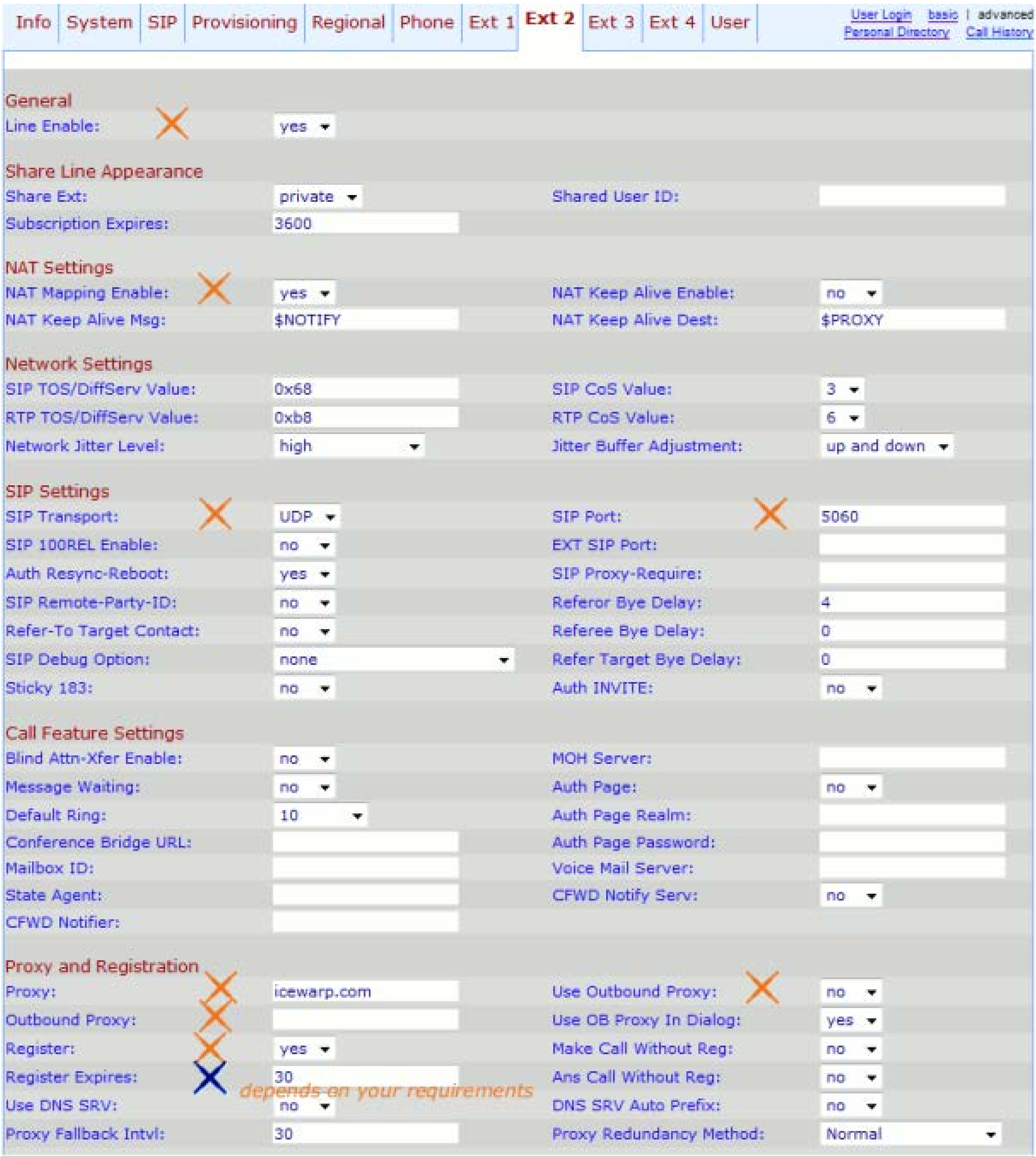
Such as: ***http://mypbx/call.php?number=%number%***

This way you can use any dialing capabilities not only via SIP but via external http scripts.



Settings for the Grandstream Hardware SIP Phone

The following screenshot shows the settings for the Grandstream hardware SIP phone:



The settings you need to change for IceWarp Server are:

- Proxy – the SIP server domain name
- Display name – the name you wish others to see when you are calling them
- Password – the password for your SIP server
- Auth ID – your SIP server user name

DNS SRV Records Configuration

For information about this topic, refer to the **DNS Records Configuration** chapter (**manual.chm – Shared Topics**).

Access Mode – Policies

Access mode for individual services is set on both domain and user levels:

- Upon the **[domain] – Policies** tab (**Domains and Accounts – Management**) for domains.
- Upon the **[user] – Policies** tab (**Domains and Accounts – Management – [domain]**) for users.