VisNetic MailServer

SIP Server Reference

Version 9.1

Contents

CHAPTER 1

SIP Service - Introduction

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

The SIP Service in VisNetic MailServer is actually a SIP domain which should be defined within VisNetic MailServer 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 PC's, PDA's, Cell phones etc. or they can be SIP-enabled network devices such as SIP-phones, or even, via SIP Gateways, or 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. It then forwards the session invitation directly to the User Agent if it is in the same Domain or to another Proxy Server if the User Agent is in another Domain.

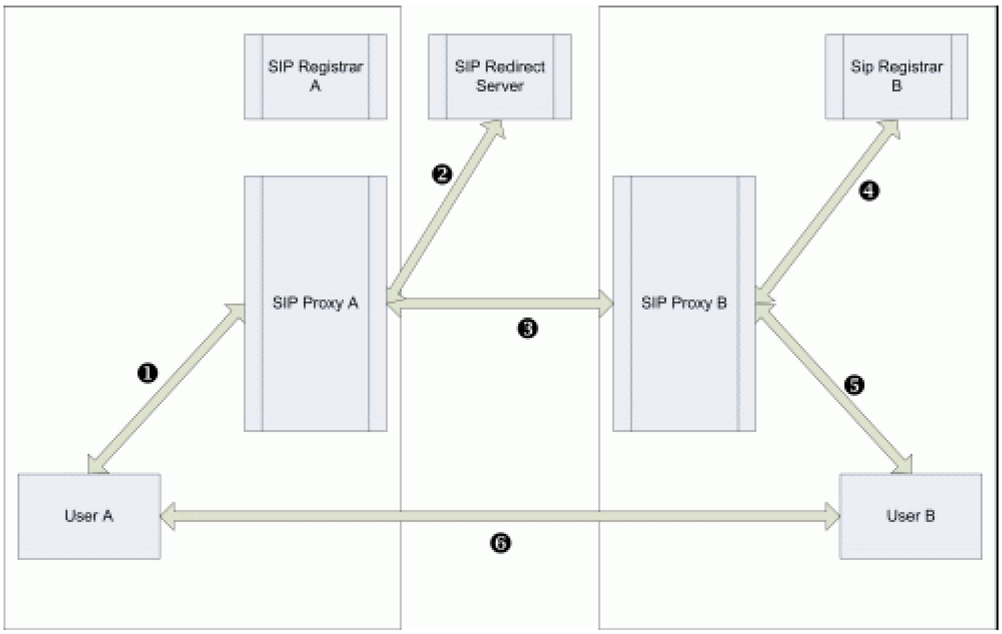
SIP Redirect Servers

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

NOTE that in VisNetic MailServer the Registrar, Proxy and Redirect Servers are integral to the software, no further software is required.

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

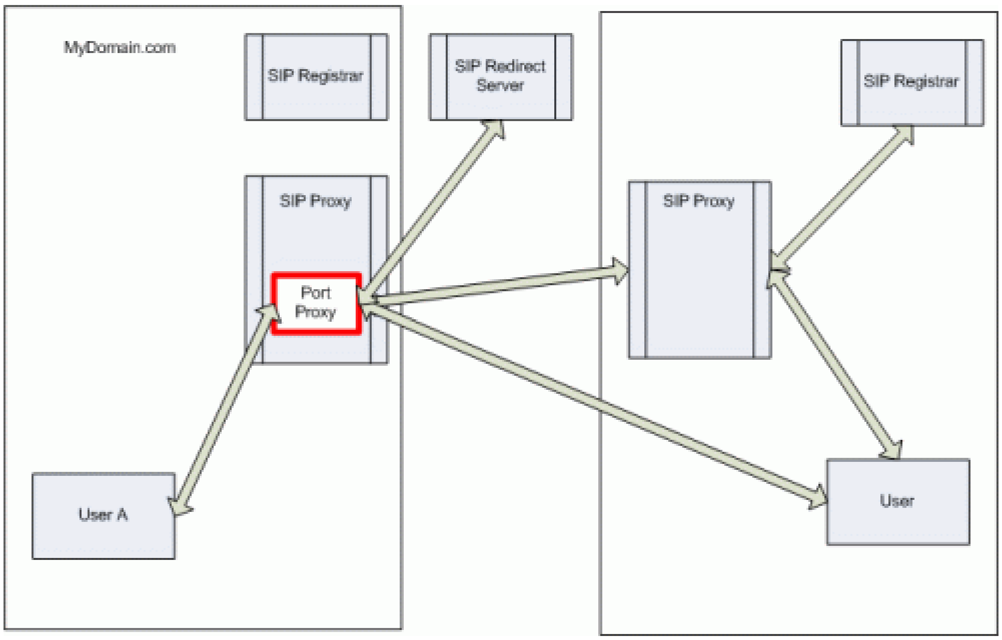
In This Chapter



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 ***SIP - Advanced*** (see "Advanced" on page 12) tab).

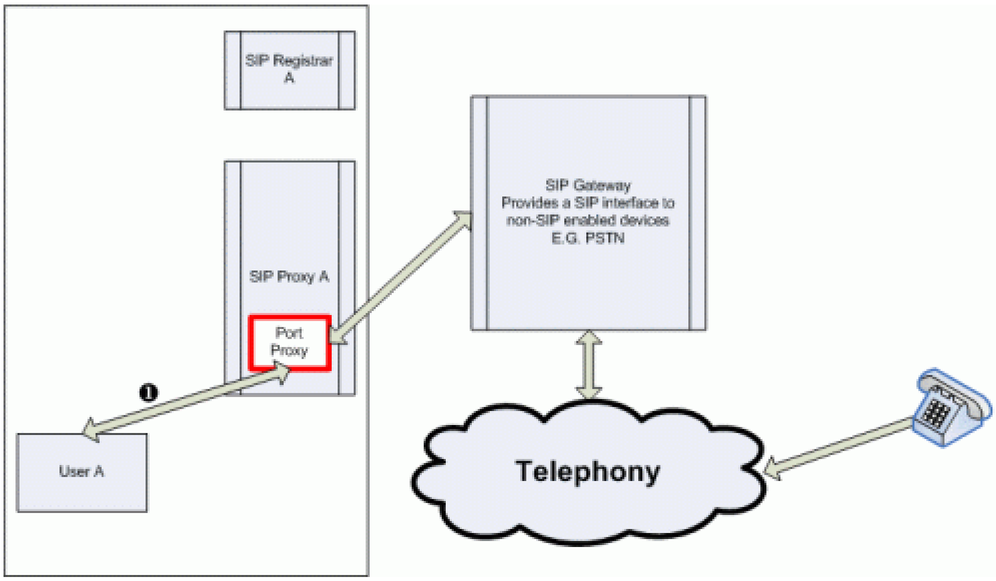
The following diagram shows a call using NAT traversal and Proxy ports.



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.

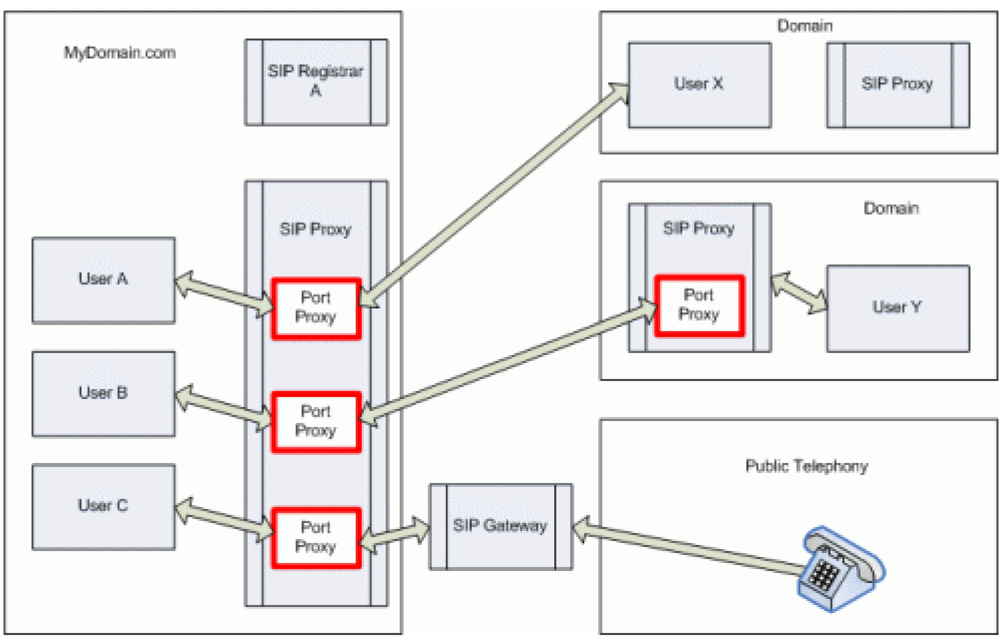
A SIP gateway is a service provided that allows you to connect to non-SIP devices, such as the public telephone network. These services usually have to be paid for.



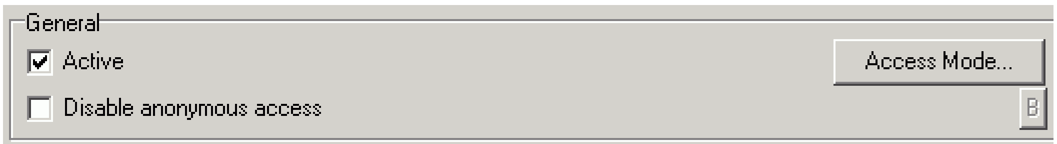
The initiation of the call is the same up to the point where the SIP Gateway is reached.

The Big Picture

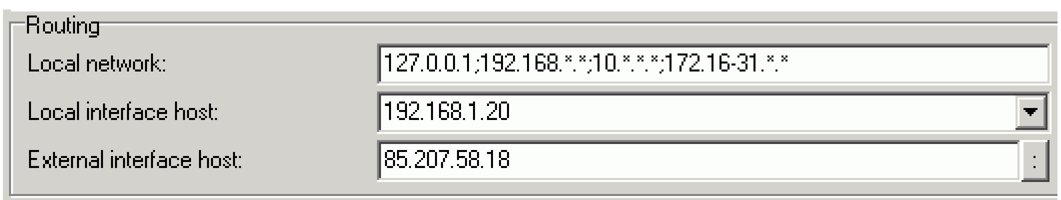
The SIP server allows you to offer a complete voice communications solution to your Users.



General

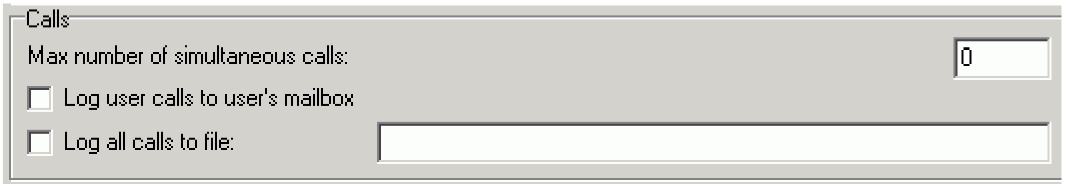


Field	Description
Active	Check this box to enable the SIP server
Disable anonymous access	Check this box if you do not wish to allow anonymous users access to the SIP service. If you select this option you can use the B button to edit a bypass file, allowing IP ranges, Users and Domains anonymous access.
SIP access mode:	Use the **Access Mode** button to specify which Accounts, Domains etc. will have access to the SIP Service. This opens the standard Access Mode dialog.



Field	Description
Local Network	Here you need to specify all the local IP addresses that this SIP server should be available for.
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.



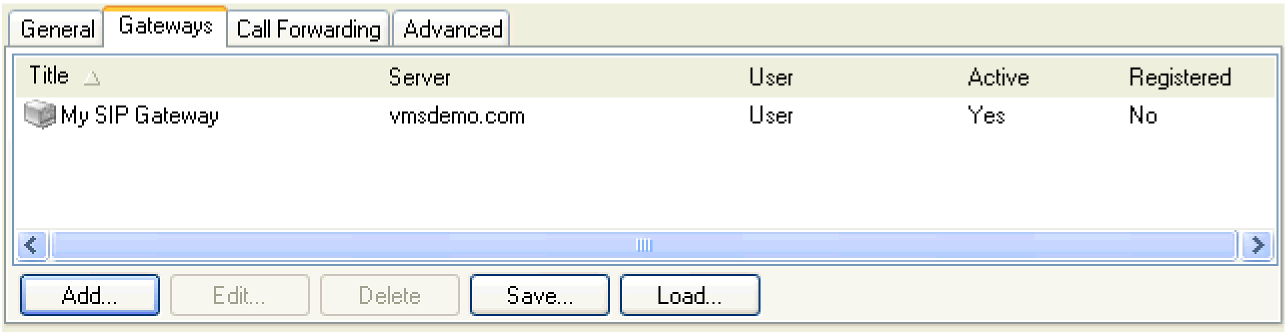
Field	Description
Max number of simultaneous calls	Once the maximum number of simultaneous calls is reached any further attempted calls will be rejected by the Service. This can be useful if you want to limit the bandwidth that is used by the SIP Server. A typical voice call is around 8kB/s.
Log user 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 should be stored. Note that yyyy, mm and dd can be used in the directory name, For example: <InstallDirectory>\SIPyyyymmdd\sip.log

Gateways

Here you should specify any Gateways you wish to route calls to.

Gateways are usually an interface to a non-SIP communications system, such as Public Telephony, and you would normally have to pay subscription or usage charges to the Gateway provider.

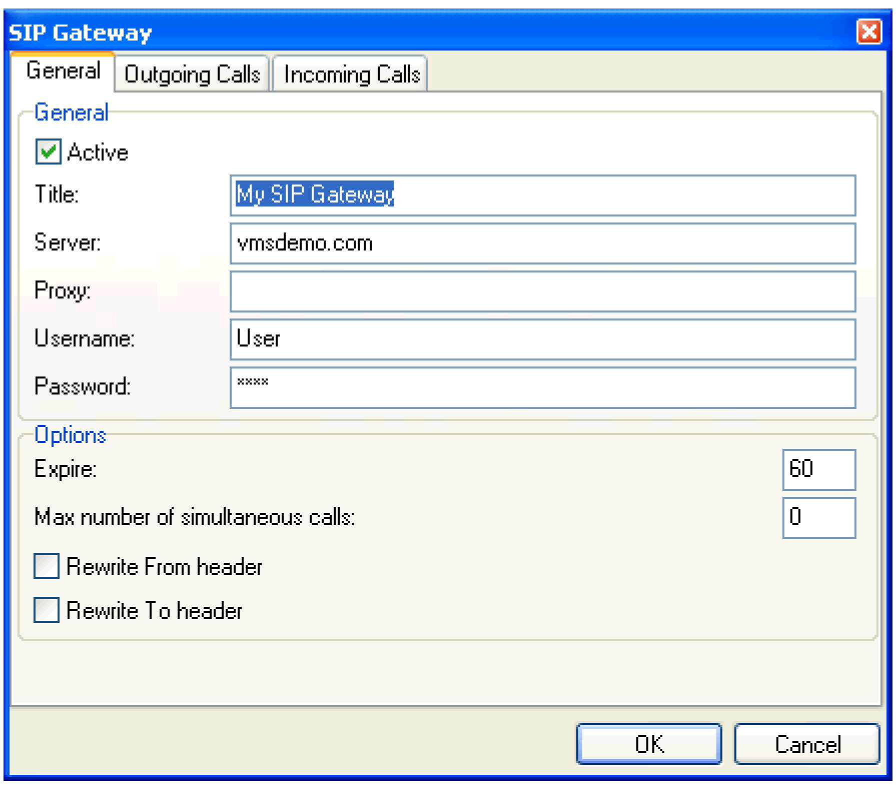
Selecting the Gateways tab presents you with a list of defined Gateways.



Use the **Delete** button to delete a selected Gateway.

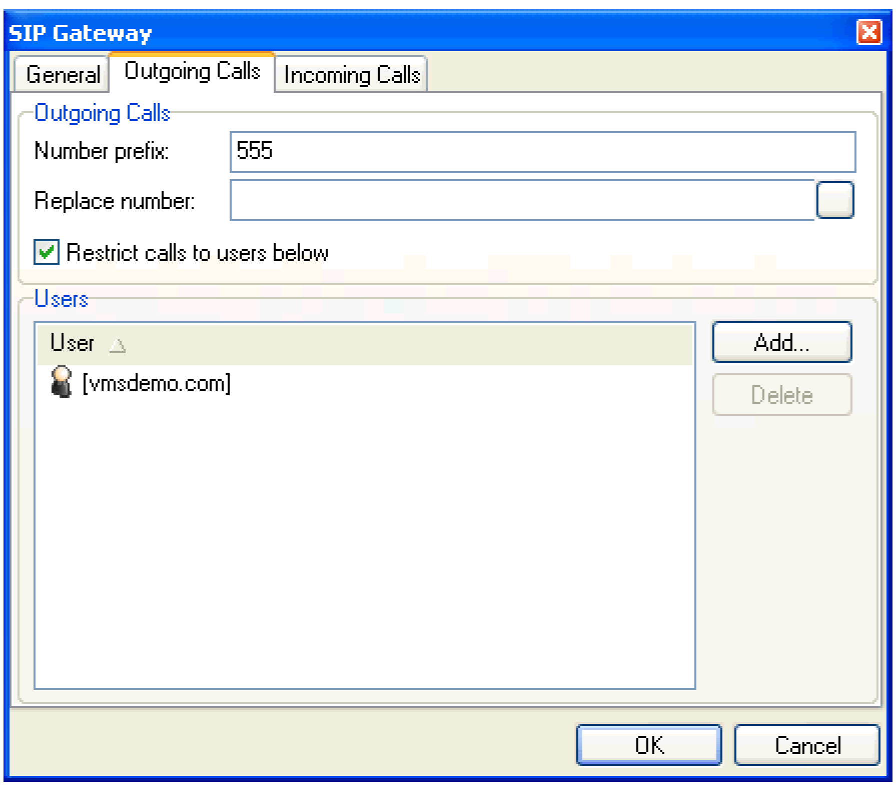
Use the **Save** and **Load** buttons to respectively save and load a list of Gateways. A standard file browser dialog will be presented.

Pressing the **Add** or **Edit** button will open the **SIP Gateway** dialog:



Field	**Description**
Active	Check this box to make this Gateway active.
Title	A descriptive name for the Gateway.

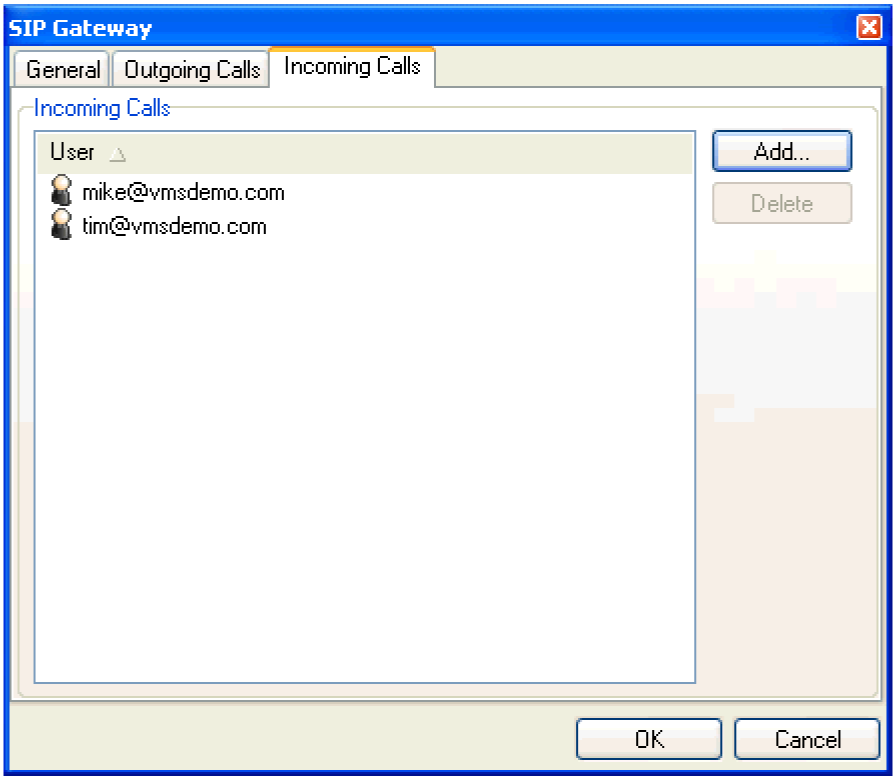
Server	Specify the IP address or hostname of the Gateway.
Proxy	Specify the IP address or hostname of any proxy server VisNetic MailServer should use to get to this Gateway.
Username	Specify the username supplied by your Gateway provider.
Password	Specify the password for the above username.
Expire	Specify here how often, in seconds, the SIP Gateway should re-register with the Gateway. Basically, this tells the Gateway that the server is still here and available.
Max. number of simultaneous calls	Specify here the maximum number of simultaneous calls allowed via this Gateway. Can be useful in limiting bandwidth usage.
Rewrite From header	Normally this field should be left un-checked. This is here in case your gateway provider requires it.
Rewrite To header	Normally this field should be left un-checked. This is here in case your Gateway provider requires it.



The **Outgoing Calls** tab allows you to select particular calls to use this Gateway, using the **number prefix**.

If you have multiple gateways then this is effectively a way to select which gateway the call will be routed through. You may have different SIP to phone providers for different countries and want to route calls appropriately. You can select you own prefix for each gateway and publish the list within your organization.

Field	**Description**
Number prefix	Specify the number prefix that will cause a call to be routed through this Gateway. The above example shows that any recipient address that starts with 555 will be processed via this Gateway. Multiple prefixes can be entered, separated by semicolons.
Replace number	This field allows you to modify the prefix if necessary. You can enter numbers here and there are three special stings that are substituted by part of the original dialed number: **%&** - is substituted with the original number, minus the prefix. **%^** - is substituted with just the prefix **%*** - is substituted with the whole number These stings can be selected via the button to the right of the text field. Example This gateway is a SIP to Phone gateway offering cheap calls to the UK, but the gateway requires that the full UK number is specified for the call (without the country code but with the leading zero). Specify Number prefix as **0044;44;+44** Specify Replace number as **0%&** and if a user dials **+4479795551234**, the call would be routed to **079795551234** via this gateway.
Restrict calls to users below	Check this box to restrict the Users who can use this Gateway to place an outgoing call. This option is recommended as otherwise you could be leaving your Gateway open to abuse by anyone that 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.



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

Use the **Add** button to specify which User or Users the call should be routed to.

If no Users are specified VisNetic MailServer will do nothing with the call request, not even reject it.

If one User is specified then VisNetic MailServer will attempt to route the call to that User.

If multiple Users are specified VisNetic MailServer will attempt to contact all of those Users simultaneously, and will wait until either:

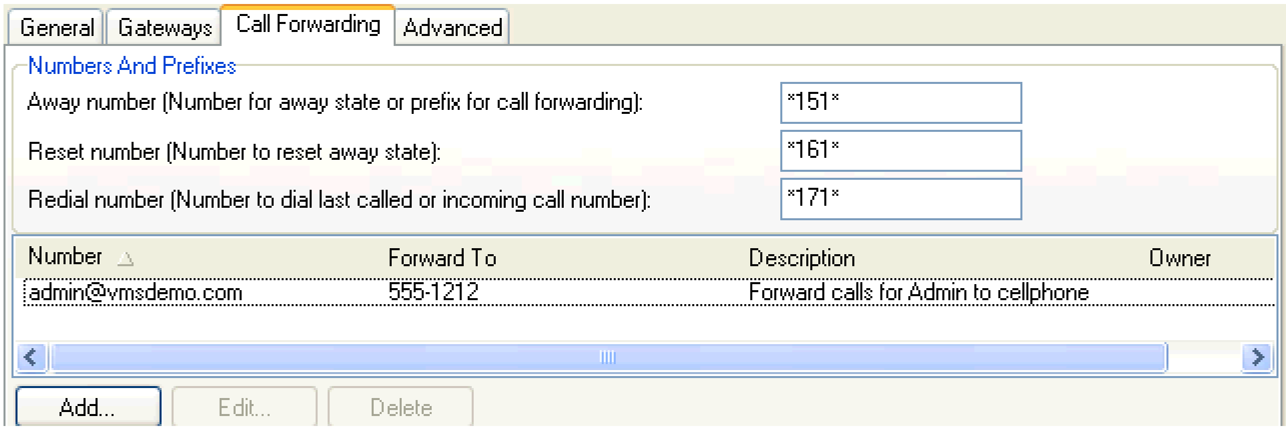
- A User accepts the call, in which case it is routed to that User
- All Users reject the call or the request times out, in which case VisNetic MailServer will reject the incoming call

Use the **Delete** button to delete a selected Gateway

Call Forwarding

The **Call Forwarding** tab allows you to specify rules on how to handle SIP requests, **Accept**, **Reject** or **Drop** based on where the request is coming from, going to, etc.

You are presented with a list of defined rules:



Away number

The Away number is a number which any user can dial to have his phone "switched off" or forwarded to another number, by the server.

The user initiates this on demand from his own SIP device, and VisNetic MailServer instantly writes a Call Forwarding rule for the number.

Specify here the number a User should dial.

Reset number

Specify here the number a User should dial to "switch on" his phone and cancel any Call Forwarding.

In the above example:

- if a User dials *151* then the server will reject any calls to that User.
- if a User dials *151*3105551234 then the server will forward calls for that user to 3105551234
- if a User dials *151*john@vmsdemo.com then the server will forward any calls to john@vmsdemo.com

NOTE that the above Call Forwarding is dynamic, initiated and cancelled by individual Users via their SIP device. The Call Forwarding described later in this section is controlled by the system administrator, and is not dynamic.

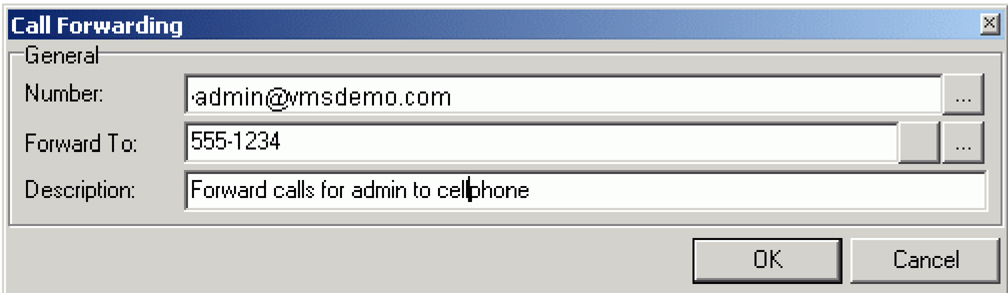
Redial number

Specify here a number that a User can dial to redial the last number for his address.

NOTE - that the number dialed will be the number associated with the last event associated with this User, whether it is an outgoing or incoming call.

Use the **Delete** button to delete a selected rule.

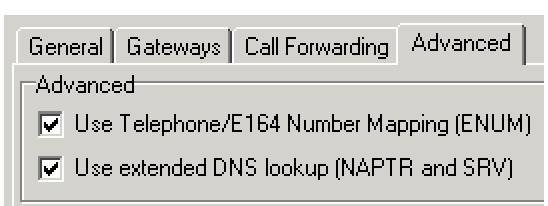
Pressing the **Add** or **Edit** button opens up the **Call Forwarding** dialog:



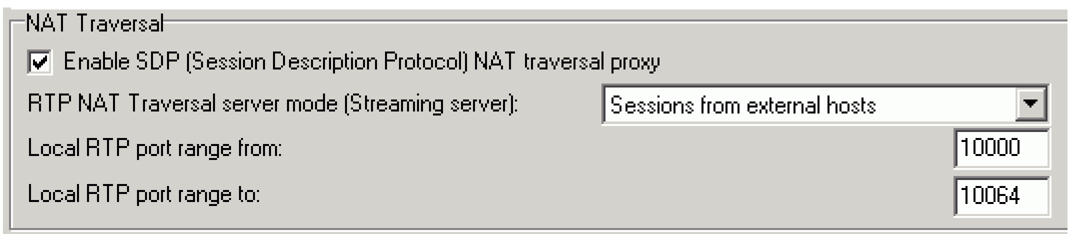
Field	Description
Number	Specify here the original intended recipient of the call. Multiple recipients can be specified, separated by semicolons. Masks can be used, using %, instead of *, to specify any string. (This is because * is valid in telephony terms) Examples: +11% - means any string starting with +11. %@vmsdemo.com - means any string ending with @vmsdemo.com. %domain% - means any string containing domain.
Forward To	Specify here a number or address for the new recipient. Multiple addresses/numbers can be specified, separated by semicolons.
Description	Short descriptive text for identification purposes.

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.

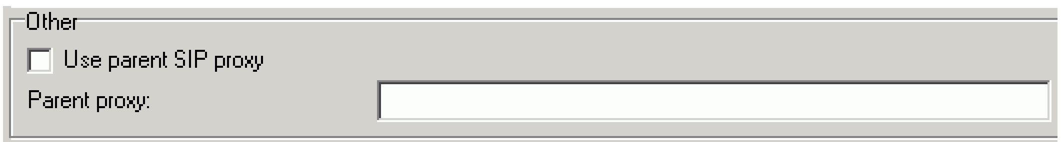


Field	Description
Use Telephone/E164 Number Mapping (ENUM)	Check this box to enable your users to place calls to standard telephone numbers. The ENUM system allows a standard telephone number to be dialled from a SIP client. The SIP server will check for a NAPTR DNS record based on the number dialled and, if found, the record will tell the SIP Server where to connect to to place the call.
Use extended DNS lookup (NAPTR and SRV)	Select this option to have VisNetic MailServer check for SRV and NAPTR DNS records to determine the hostname for a SIP Server. VisNetic MailServer first checks for a NAPTR DNS record and , if none is found, it will check for an SRV DNS record.



Field	Description
Enable SDP NAT traversal proxy	Check this box to enable the SDP (Session Description Protocol) NAT (Network Address Translation) Proxy. NAT is used to correctly route incoming data to the correct local network recipient.
RTP NAT Traversal server mode	RTP (Real-time Transport Protocol) is the protocol used by the SIP server for streaming data. The SIP server can create Proxy Ports for each SIP call. This is useful if you have no control over the ports being used by your User's SIP clients. Choose from the following options: **Disabled** Select this option and no Port Proxies will be created **Sessions from external hosts** Select this option and Port Proxies will be created for communications with external hosts. **All sessions** Select this option and Port Proxies will be created for all SIP server communications. This is especially useful if you have many Windows XP Users.

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 that the port range specified here must be open in your router/firewall setup.



Field	**Description**
Use parent SIP proxy	You can have multiple SIP servers with only one server having access to the outside world. In this case you would specify this external server as the "parent" server. Check this option if you wish to use a parent server.
Parent proxy	Specify the IP address or hostname of the parent SIP server.

Using the Dial via SIP functionality

Both the VisNetic Outlook Connector and WebMail have the ability to dial out via SIP clients.

In the VisNetic Outlook Connector

- 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 VisNetic MailServer" from the VisNetic Outlook Connector 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 WebMail

- 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 that the call is in no way routed by VisNetic Outlook Connector or WebMail, they are just used to initiate the call. The SIP server dials your registered client and once connected route the call to the destination you specified. This method will work with any SIP client.

CHAPTER 2

Setting up a SIP Client - X-Lite

There are numerous SIP clients available, both software and hardware.

This section describes how to set up X-Lite to access your SIP server.

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

In This Chapter

First run of X-Lite

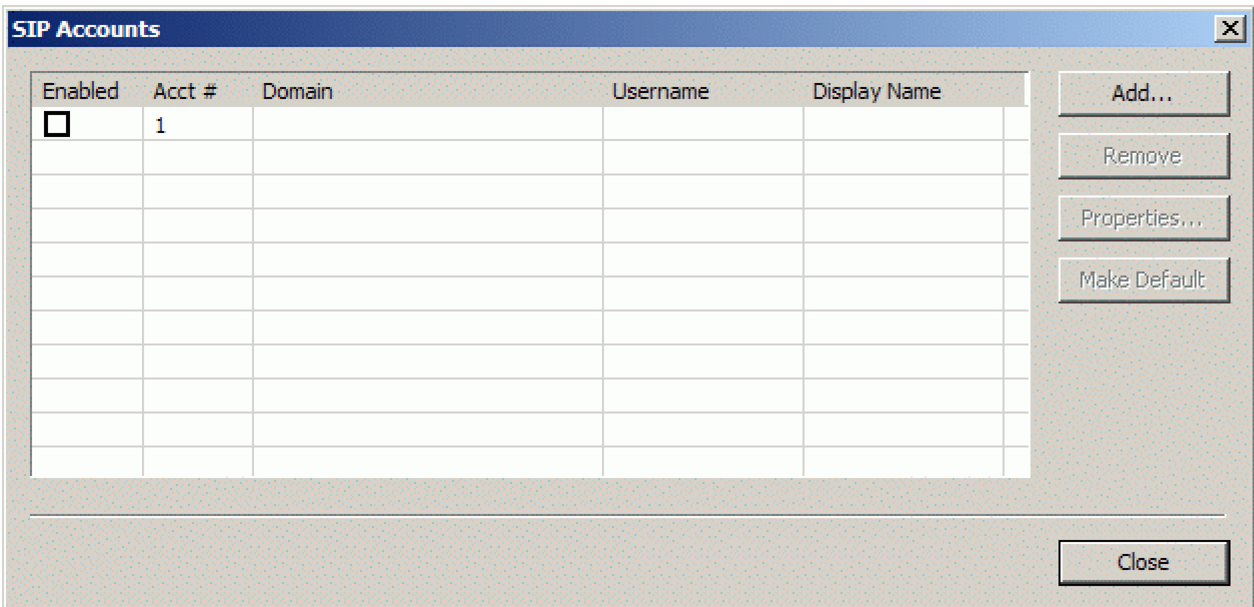
When you first run X-Lite it will discover that you have no SIP account defined and will show the message "No SIP accounts are enabled""

And the "SIP accounts" dialog will we be displayed automatically.

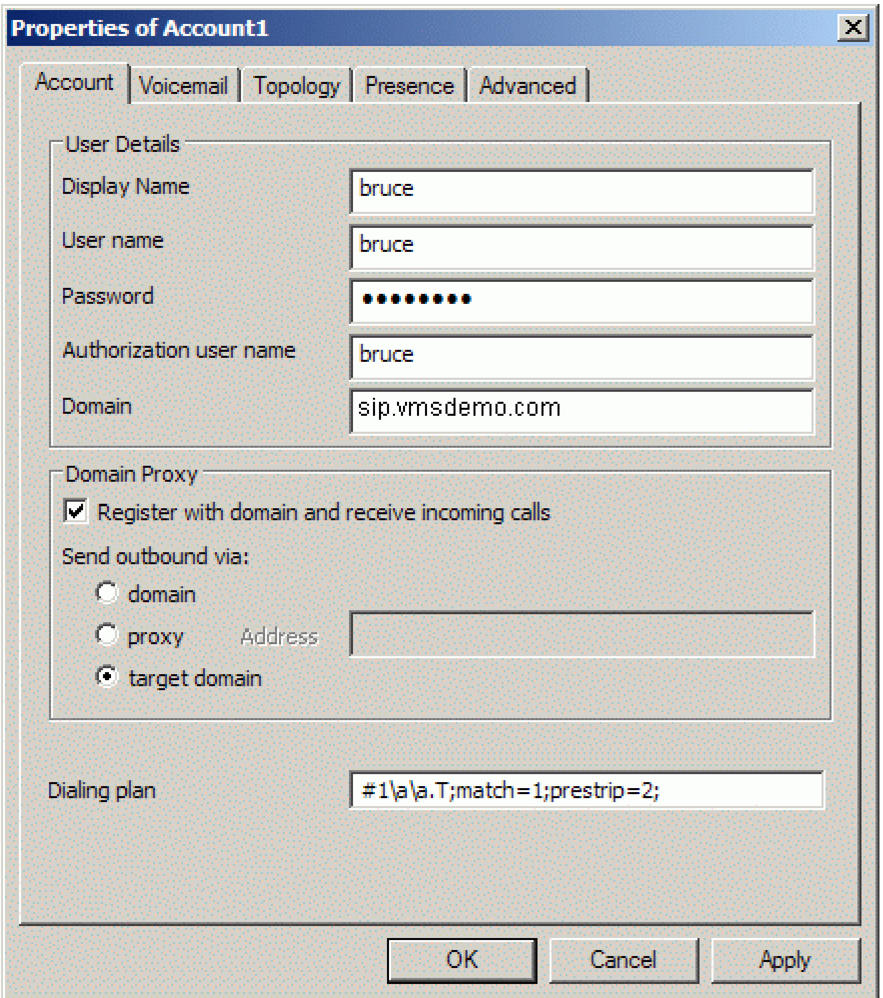
If you wish to add your VisNetic MailServer account to X-Lite you should open the "SIP accounts" manually:

SIP Accounts dialog

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

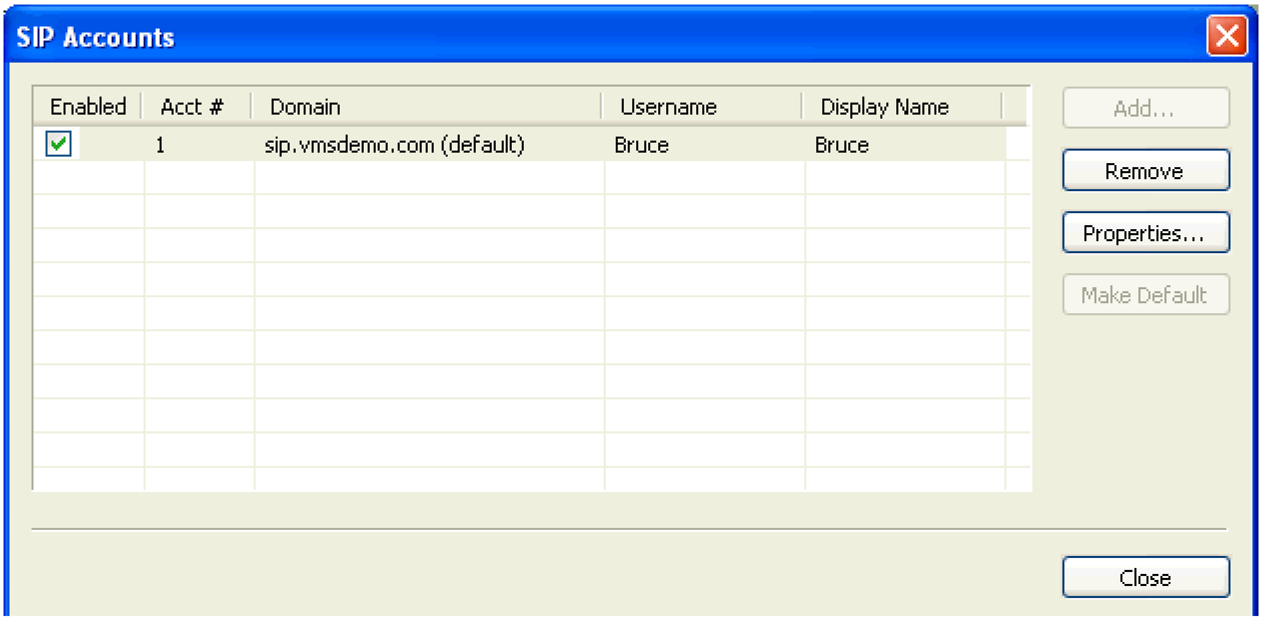


Press the Add button to define your VisNetic MailServer account. The properties dialog will be displayed:



Field	Description
Display name	Enter the display name you would like people to see when you are in a call with them.
User name	The User name supplied by your SIP service provider.
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.

Press OK to return to the SIP Accounts dialog:

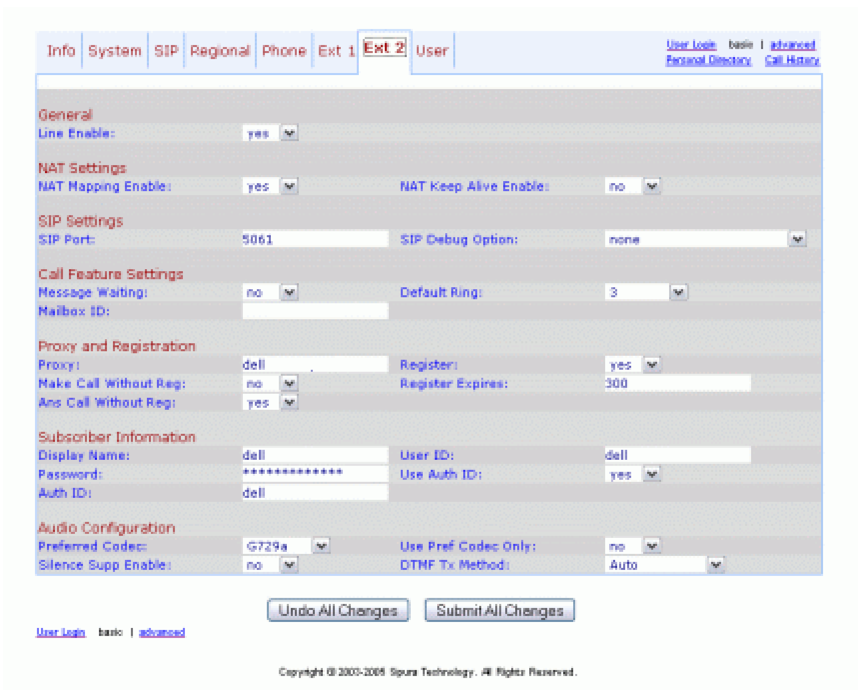


Press Close to return to X-Lite.

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

Settings for the Sipura Hardware SIP phone

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



The settings you need to change for VisNetic MailServer are:

Proxy - The SIP server domain name

Display name - the name you wish others to see when you are in a call

Password - the password for your SIP server

Auth ID - Your SIP server username

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