



Micromedia International

Technical Study
SIP configuration
for Alert

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SIP driver configuration in ALERT

Réf. : ETT_20081218_000001.docx

This document specifies how to configure Alert to use with the SIP driver for VOIP.

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Attached Documents:

Of all the media available to make calls with ALERT, the VoIP driver offers the possibility to interface with IP telephony equipment in accordance with the SIP (Session Initiation Protocol) protocol.

This document describes the configuration of the SIP Alert driver and on call operators using this medium.

Prerequisites

VoIP (Voice Over IP), as its name suggests, is based on the IP protocol. An IP network interface is required to use this driver.

The VoIP Alert driver is subject to license. The user must have acquired a license based on the number of simultaneous calls (using SIP) desired.

Devices that Alert has to communicate with need to be compatible with the SIP protocol, they may be telephones, PBXs, SoftPhone...

These devices must meet the following specifications:

Technical specifications:

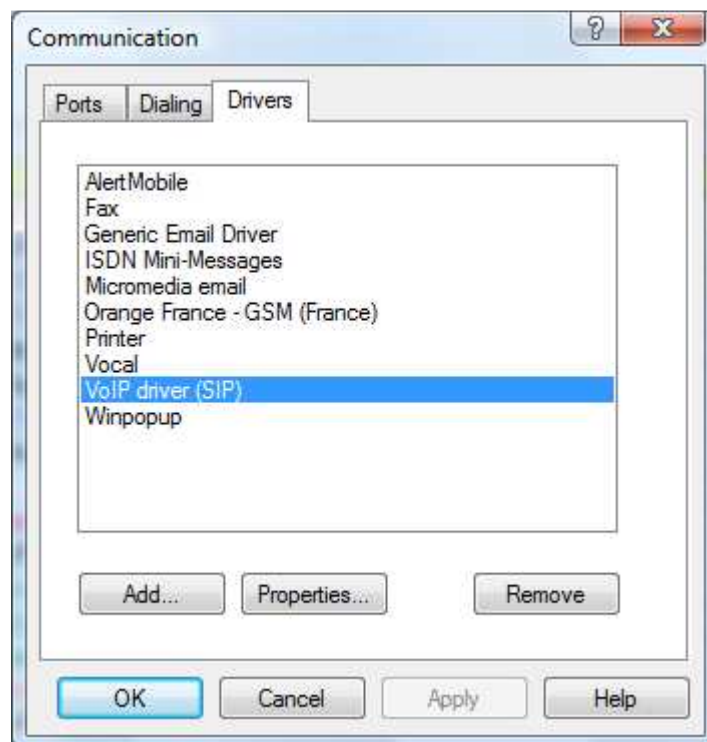
SIP protocol supported: RFC 3261

Audio formats supported: G711 Mu Law and G711 A Law

DTMF management: RFC 2833

ALERT configuration

The driver configuration is done through the properties box. Menu "Configuration / Communication Then select the "Drivers" tab.



Select the driver "VoIP Driver (SIP)" and press "Properties ...".
The properties box of the driver will appear.

The image shows a Windows-style dialog box titled "VoIP driver settings". At the top, it says "VoIP driver (SIP) -- v1. 0. 1. 7 (Unicode)". There are "OK" and "Cancel" buttons. Below this, there is a field for "Maximum simultaneous calls number:" with a value of "0" and a note "(0 : no limit)".

The "SIP Transport" section has two radio buttons: "UDP" (selected) and "TCP". A "Port:" field contains "5080". Below this, "IP address:" has two radio buttons: "Default" (with a note "(192.168.38.123)") and "192.168.38.113" (selected).

The "SIP Settings" section has a "User uri:" field with "sip:alert@localhost". There is a checkbox for "Use outbound proxy" which is unchecked. Below it is a "Proxy uri:" field with "sip:host".

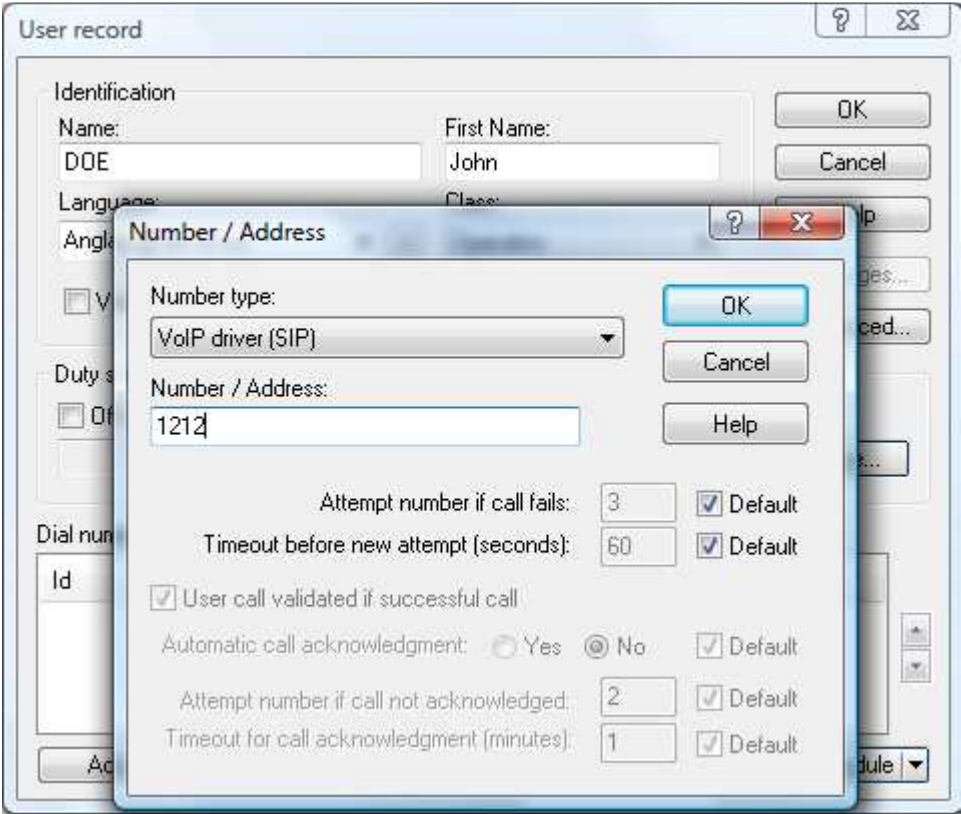
The "SIP Registration" section has a checkbox for "Register user uri" which is unchecked. Below it is a "Registration failure retry period:" field with "0" and "(seconds)". There is another unchecked checkbox for "Use authentication". Below that are "Username:" and "Password:" fields.

The "Miscellaneous" section has a checked checkbox for "Trace in alert log". Below it is a "Max call time (0=infinite):" field with "0" and "seconds". At the bottom is a "DTMF payload type:" field with "101".

The values of the parameters to specify will depend on the telephone architecture. Depending on the telephone system, three types of configuration stand out:

- Setup without proxy
- Setup with proxy without registration (Registrar)
- Setup with registration

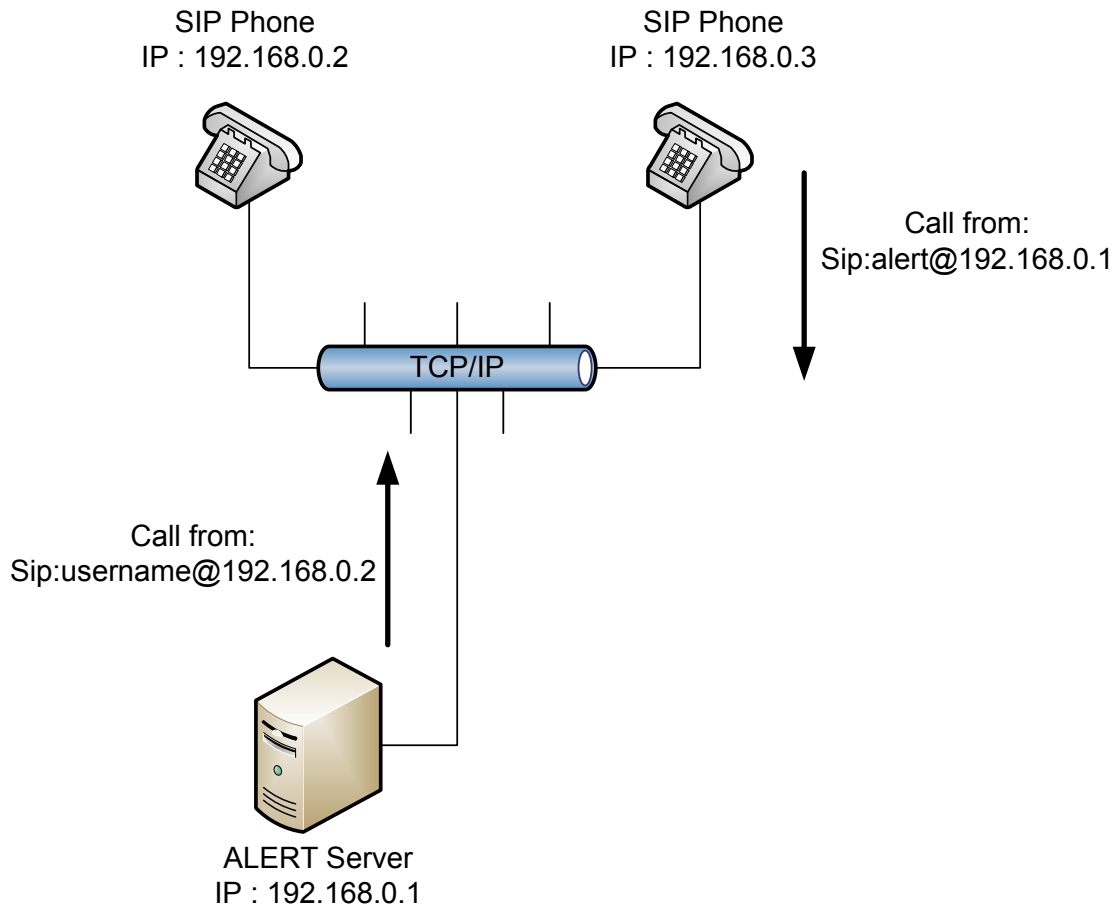
In the user record properties, the telephone numbers supplied must be associated with the VOIP driver. Here is a sample configuration with a SIP number.



These numbers will also be dependent on the telephone architecture.

Setup without proxy

In this configuration, ALERT calls SIP phones or softphones directly. Since there is no intermediary (proxy or PBX) for routing calls, ALERT must know the IP addresses of each device.



SIP driver

To configure the driver, here is an example of an installation without a proxy.

For SIP transporting, the values to indicate are the default values of the SIP protocol. The IP address to choose is the one of the machine on the network where the devices to contact are located. The same address will be used by the devices to call ALERT

The uri of the user matches the call number of ALERT which is dialed from the SIP phones.

VoIP driver (SIP) -- v1.0.1.7 (Unicode) [OK] [Cancel]

Maximum simultaneous calls number: 0 (0 : no limit)

SIP Transport

Protocol: UDP TCP Port: 5060

IP address: Default (192.168.38.123) 192.168.0.1

SIP Settings

User uri: sip:alert@192.168.0.1

Use outbound proxy

Proxy uri: sip:host

SIP Registration

Register user uri

Registration failure retry period: 0 (seconds)

Use authentication

Username: []

Password: []

Miscellaneous

Trace in alert log

Max call time (0=infinite): 0 seconds

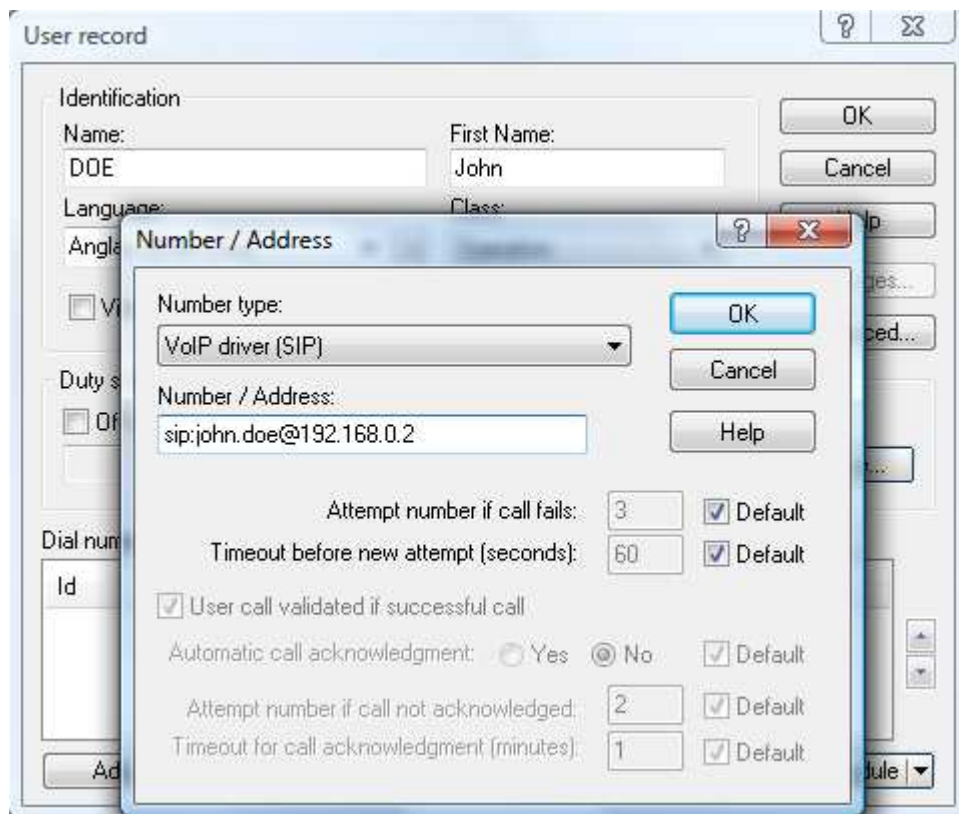
DTMF payload type: 101

User record

Regarding the setting of on call operators: For the number to call, you must indicate the complete sip address (uri) of the correspondent.

[Sip:username@ipaddress:port](#)

The port is optional if you use the default sip port: 5060.



Usage examples

This type of configuration is used for small installations or for testing. The SIP equipment is not managed centrally.

This configuration also allows you to have a softphone on the same machine as ALERT. In this case, you need to set the port number to, for example 5080, to transport sip for ALERT and 5060 for the softphone. Both programs will then be able to coexist.

Alert will call address <sip:username@192.168.0.1>

The softphone will call ALERT using <sip:alert@192.168.0.1:5080>

Setup with proxy without registration

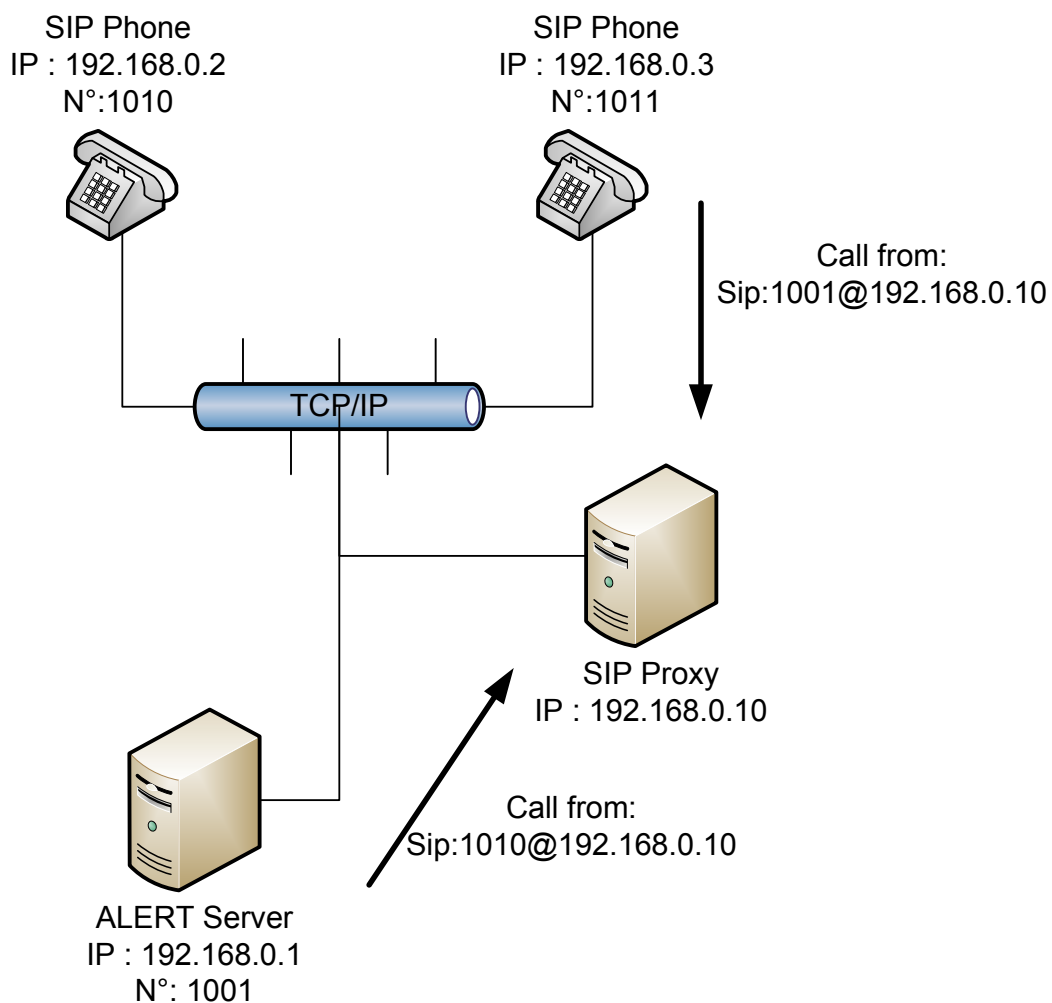
Proxy configuration with no registration

With this architecture, ALERT no longer needs to know the IP addresses of the phones. The proxy, if it is an IPBX, will generally manage telephone number vs. IP address correspondence.

User record configuration is thus simplified; only their phone number needs to be specified.

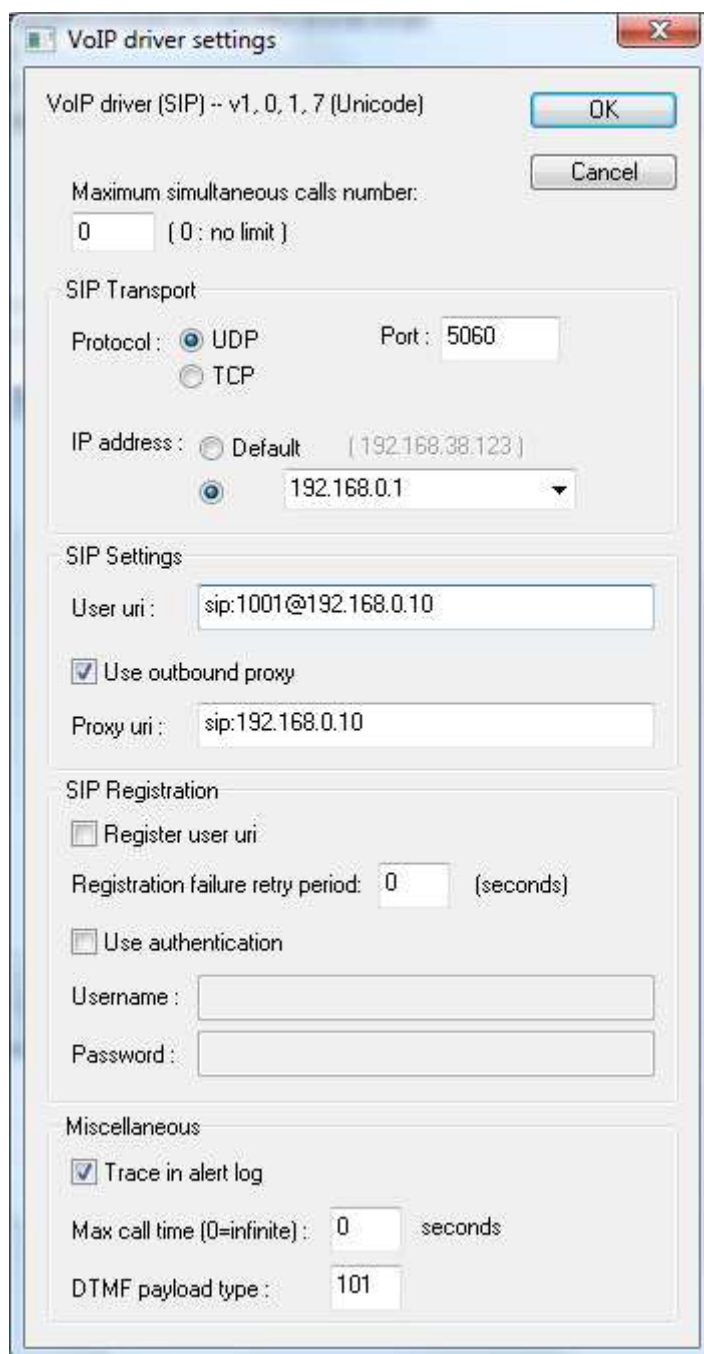
With this type of installation, the uri's are all of type: sip:username@realm, where the realm field is the domain name (usually the address of the proxy or the name of the machine) and the username field is a phone number.

In this case, all calls go through the proxy.



SIP driver

In this properties box, the proxy field needs to be filled in.
The uri of the user can, as in the following example, be a phone number.



The image shows a screenshot of the 'VoIP driver settings' dialog box. The window title is 'VoIP driver settings' and it has a close button (X) in the top right corner. The dialog is divided into several sections:

- VoIP driver (SIP) -- v1.0, 1.7 (Unicode)**: Includes 'OK' and 'Cancel' buttons.
- Maximum simultaneous calls number:** A text box containing '0' with '(0: no limit)' next to it.
- SIP Transport**:
 - Protocol:** Radio buttons for 'UDP' (selected) and 'TCP'.
 - Port:** A text box containing '5060'.
 - IP address:** Radio buttons for 'Default (192.168.38.123)' and '192.168.0.1' (selected).
- SIP Settings**:
 - User uri:** A text box containing 'sip:1001@192.168.0.10'.
 - Use outbound proxy**
 - Proxy uri:** A text box containing 'sip:192.168.0.10'.
- SIP Registration**:
 - Register user uri**
 - Registration failure retry period:** A text box containing '0' with '(seconds)' next to it.
 - Use authentication**
 - Username:** An empty text box.
 - Password:** An empty text box.
- Miscellaneous**:
 - Trace in alert log**
 - Max call time (0=infinite):** A text box containing '0' with 'seconds' next to it.
 - DTMF payload type:** A text box containing '101'.

User record

The configuration of the user record is made much simpler. All the addresses having the same proxy, ALERT offers the possibility to specify the full uri, just like the setup without proxy or only the phone number.

In this case: sip:1010@192.168.0.10 or simply 1010.



Usage example

This is the type of setup to use with the Cisco CallManager. The CallManager is the proxy. The table of correspondence between telephone numbers and IP addresses is managed by the proxy.

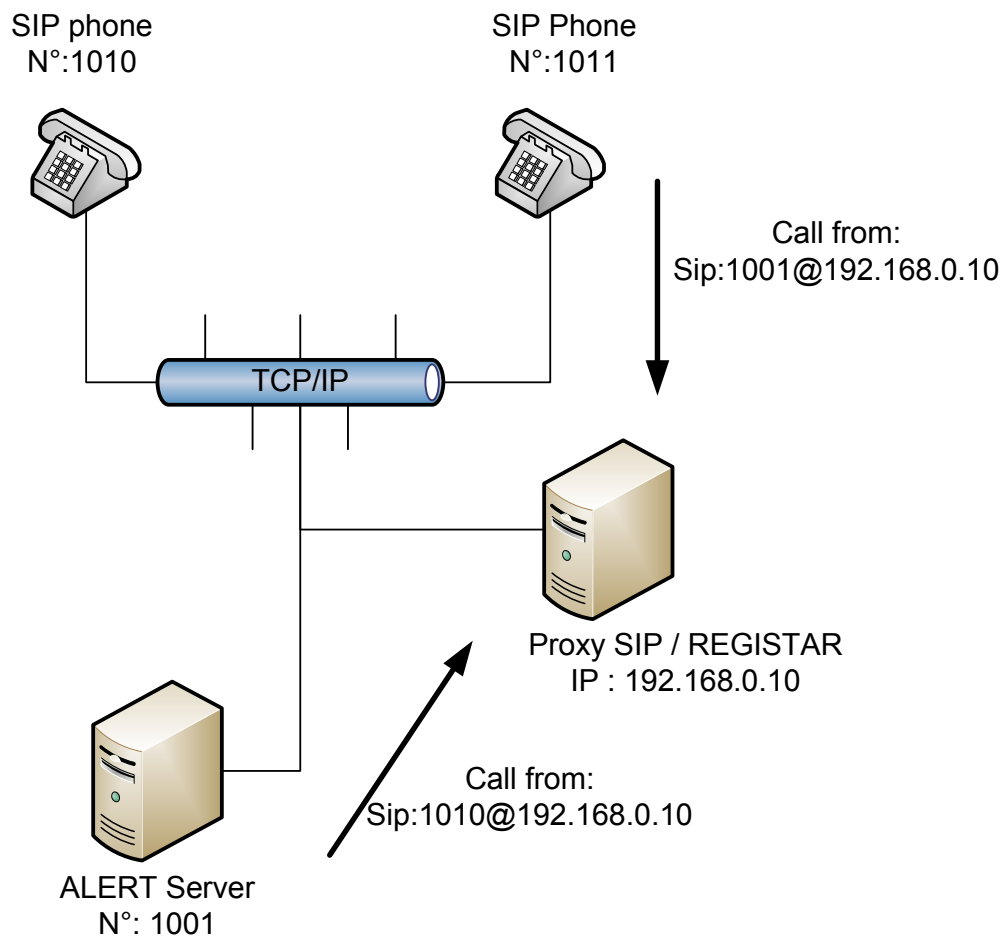
ALERT must also have a phone number if operators want to contact the voice server application.

This configuration also corresponds to the use of the MV-310 VoIP gateway / GSM. In this case, the proxy is the gateway and ALERT can then only call via this gateway.

Setup with registration

This type of installation allows dynamic IP addresses. Each piece of equipment (including ALERT) registers at startup with the current registrar sip. The latter is often the proxy.

The operation is the same as with the proxy alone... The difference is that the IP / phone numbers correspondence is done dynamically.



SIP driver

In this particular case, you need to specify the data identification supplied by the telephony administrator.

In our case, ALERT registers itself with its login and password for the call number 1001

VoIP driver (SIP) -- v1. 0. 1. 7 (Unicode) [OK] [Cancel]

Maximum simultaneous calls number: 0 (0 : no limit)

SIP Transport

Protocol : UDP Port : 5060
 TCP

IP address : Default (192.168.38.123)
 192.168.0.1

SIP Settings

User uri : sip:1001@192.168.0.10

Use outbound proxy

Proxy uri : sip:192.168.0.10

SIP Registration

Register user uri

Registration failure retry period: 0 (seconds)

Use authentication

Username : alert

Password :

Miscellaneous

Trace in alert log

Max call time (0=infinite) : 0 seconds

DTMF payload type : 101

User record

The user record setup is the same as the setup with proxy.

The call addresses are of type: sip:1010@192.168.0.10 or simply 1010.



Usage example

With a subscription to a SIP provider.

The configuration of the driver then depends on data supplied by the provider.

With Asterisk or another IPBX. This time it is the administrator of the telephone system that provides the data to specify in ALERT.