

1 SIP Carriers

1.1 SpeakUP



1.1.1 Warnings

Check the SIP Carrier Matrix of the Interactive Intelligence Testlab website for certification status and supported features:

<http://testlab.inin.com>

1.1.2 Vendor Contact

Vendor Web Site : www.speakup.nl

1.1.3 Versions Verified

Interaction Center 4.0 SU4

1.1.4 PreInstall

SpeakUp will provide users with a set of authentication credentials, a block of DIDs and a reference server (IP, FQDN, or other means to connect to the service). These must be obtained before setup can begin.

1.1.5 Install

SpeakUp requires a fully configured SIP enabled IC server. A SIP line must be created. See Line Configuration for more information.

1.1.6 Required Post Installation Steps

Confirm capacities and capabilities of purchased service.

2 IC Configuration Guide

2.1 Line Configuration

The line page has a vast majority of the configuration options required for SIP Carrier setup.

This is the section that configures the connection to the carrier's servers, any authentication or registration information, and basic configuration needs.

Any reference to a menu, while talking about the line configuration, will refer to the options on the left side of the line configuration page, and tabs will refer to the standard tab interface across the top of the line configuration page.

2.1.1 Line Menu

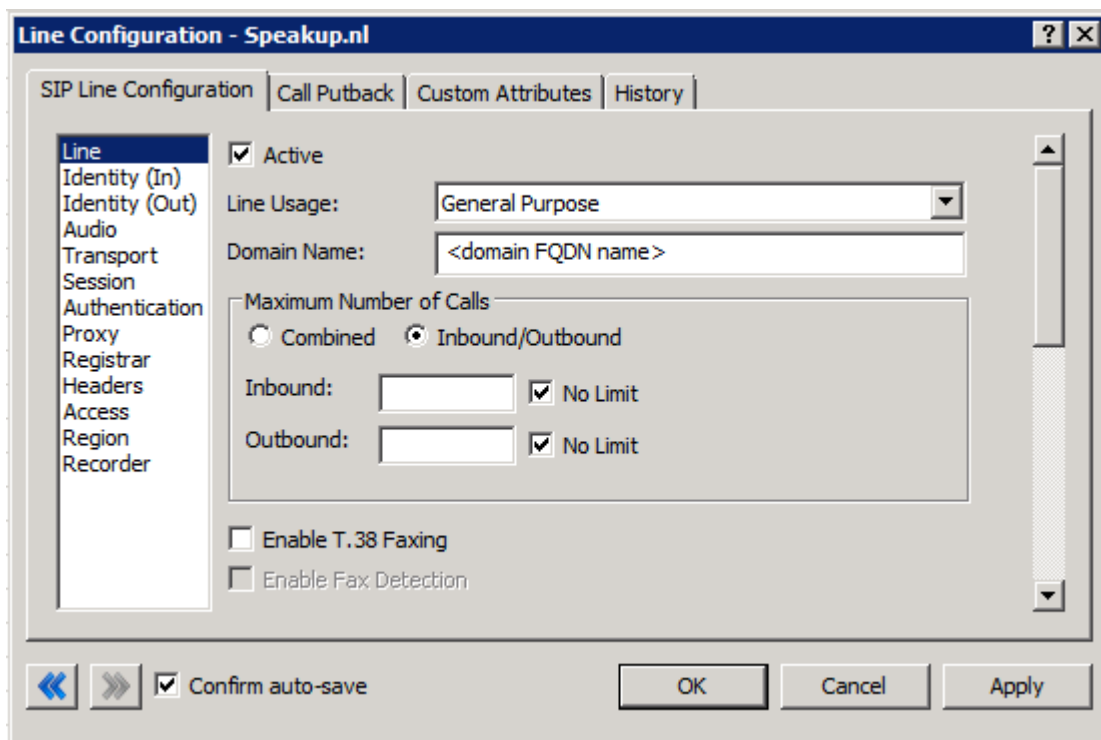


Figure 1: Line Menu Line Configuration Page

- **Active**
The active box should be checked. This activates the line. If this box is not checked, the line will not be available for any function. This can also be affected by right clicking on the line in Interaction Administrator, dropping to the *Set Active* menu option, and selecting Yes.
- **Domain Name**
This box should contain the Fully Qualified Domain Name (FQDN) of the customer provided by *SpeakUp*. It is used for the registration and/or authentication of the line.

- **Enable T.38 Faxing**

SpeakUp's SIP Carrier service does not support the T.38 faxing protocol. Uncheck this box.

- **Remainder of Line Menu Options**

These have no major direct impact on the SIP carrier configuration, and should be addressed according to business needs.

2.1.2 Identity (Out) Menu

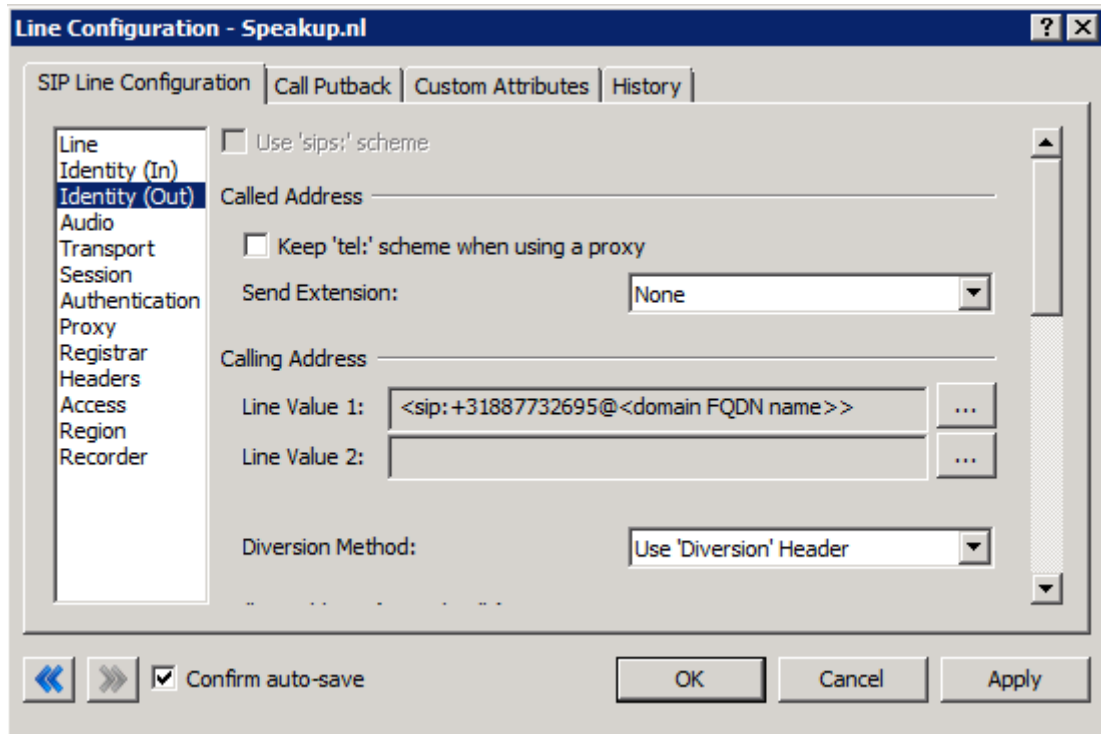


Figure 2: Identity (Out) Menu Line Configuration Page

- **Calling Address**

Clicking the “...” button next to the Line Value 1 brings up the Configure Line Value dialog.

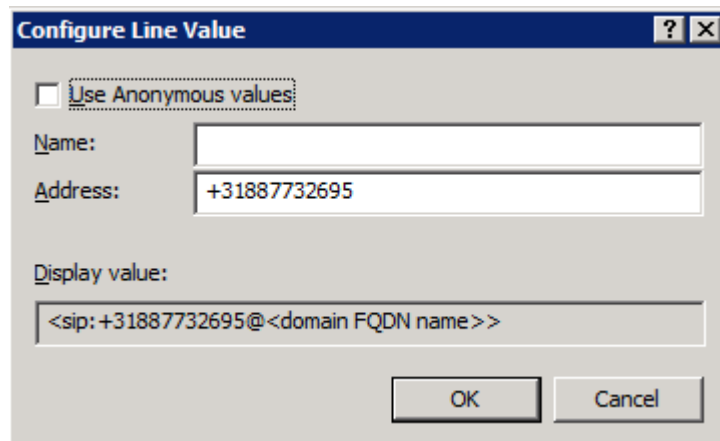


Figure 3: Configure Line Value Configuration Page

Address: enter the main DID to be used as the default User Portion on outgoing calls.

- **Calling Address (Diverted Calls)**

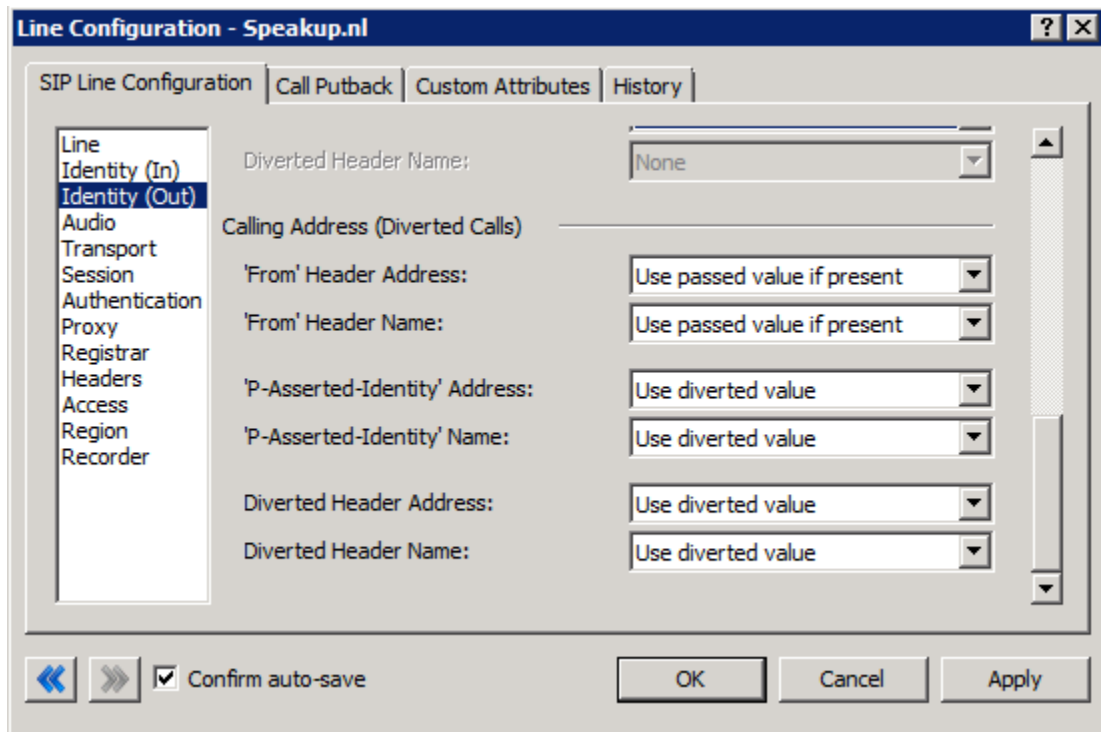


Figure 4: Calling Address (Diverted Calls) Configuration Page

Set values according to Figure 4.

- **Remainder of Audio Menu Options**

These have no major direct impact on the SIP carrier configuration, and should be addressed according to business needs.

2.1.3 Audio Menu

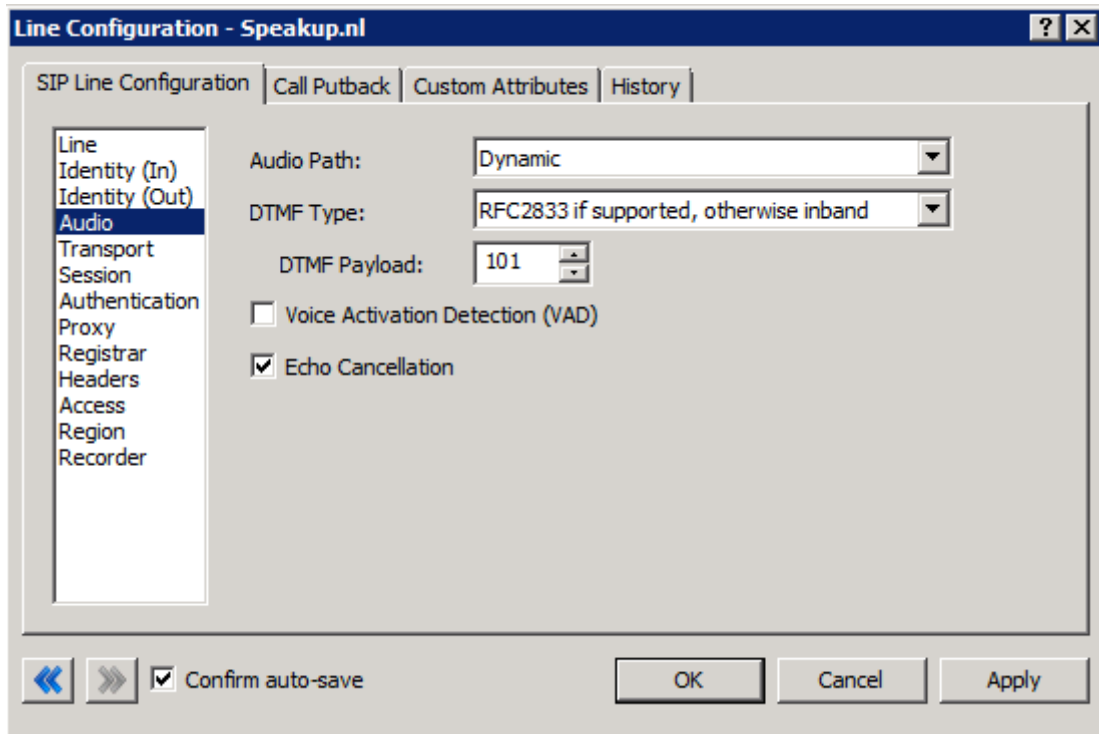


Figure 5: Audio Menu Line Configuration Page

- **Audio Path**
The choice between Dynamic or Always-in is, for the most part, the choice of the client with respect to the business being done on the server.
- **DTMF Type**
Recommended settings: “RFC2833 only” or “RFC2833 if supported...”.
- **Remainder of Audio Menu Options**
These have no major direct impact on the SIP Carrier configuration and should be addressed according to business needs

2.1.4 Transport Menu

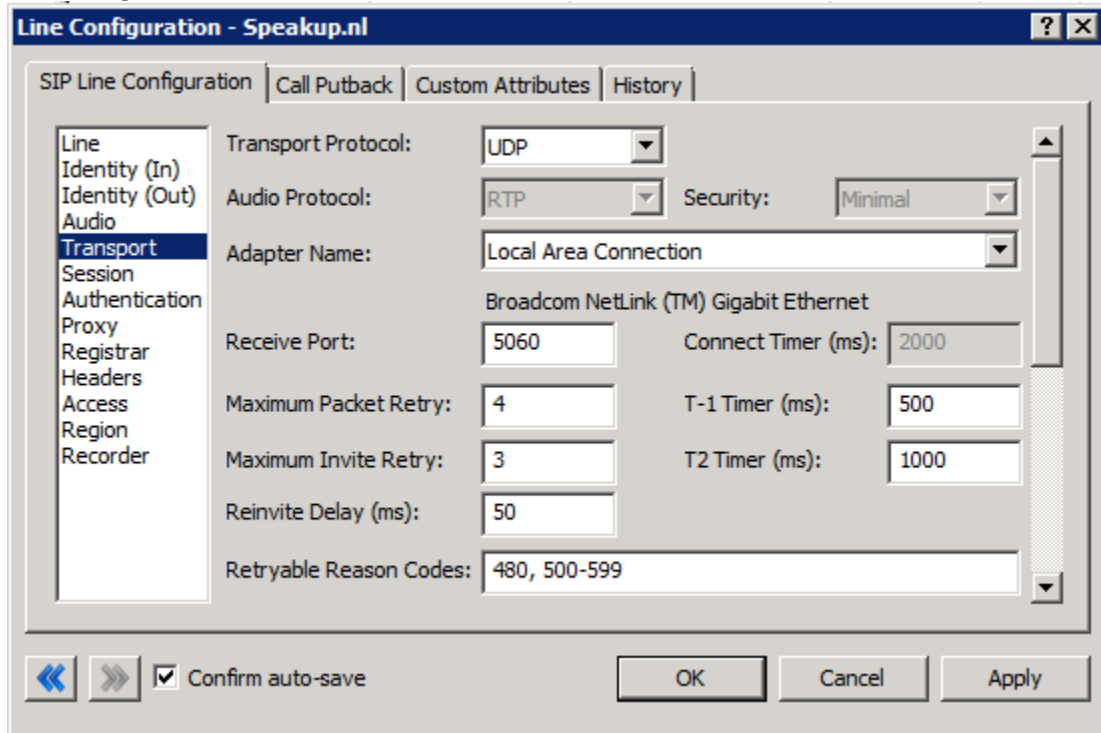


Figure 6: Transport Menu Line Configuration Page

- **Transport Protocol**
This option should be set to UDP
- **Receive Port**
This option should be set to 5060
- **Remainder of Transport Menu Options**
These have no major direct impact on the SIP Carrier configuration and should be addressed according to business needs.

2.1.5 Session Menu

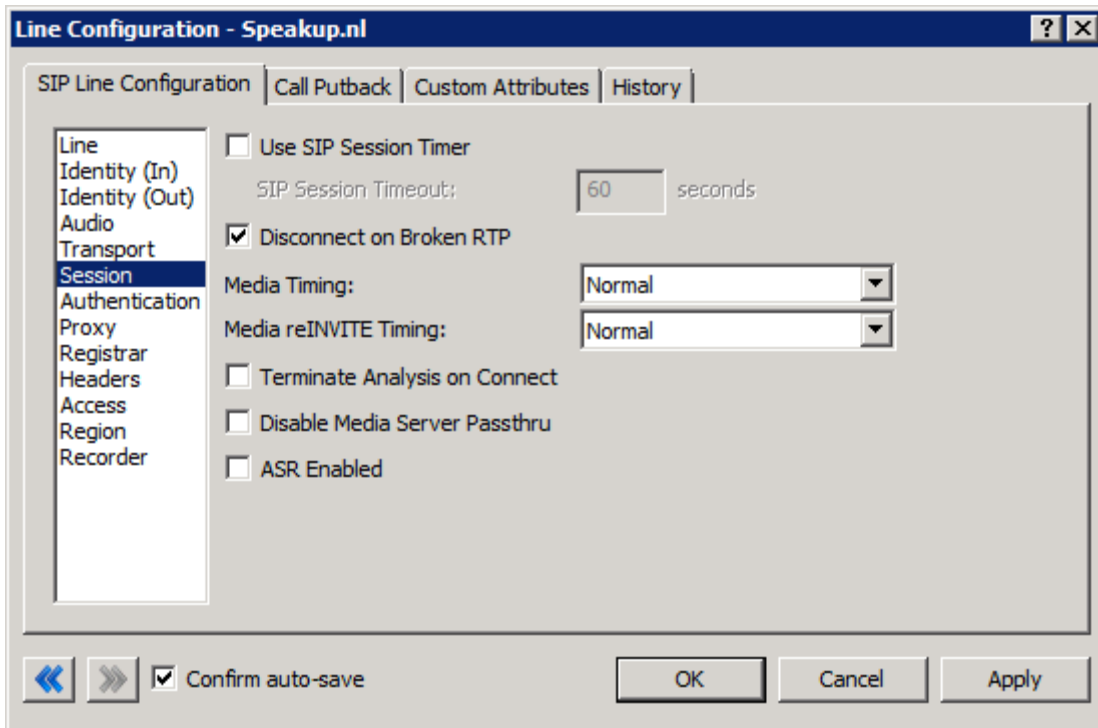


Figure 7: Session Menu Line Configuration Page

- **Media Timing**
Recommended value: “Normal”
- **Remainder of Session Menu Options**
These have no major direct impact on the SIP Carrier configuration and should be addressed according to business needs.

2.1.6 Authentication Menu

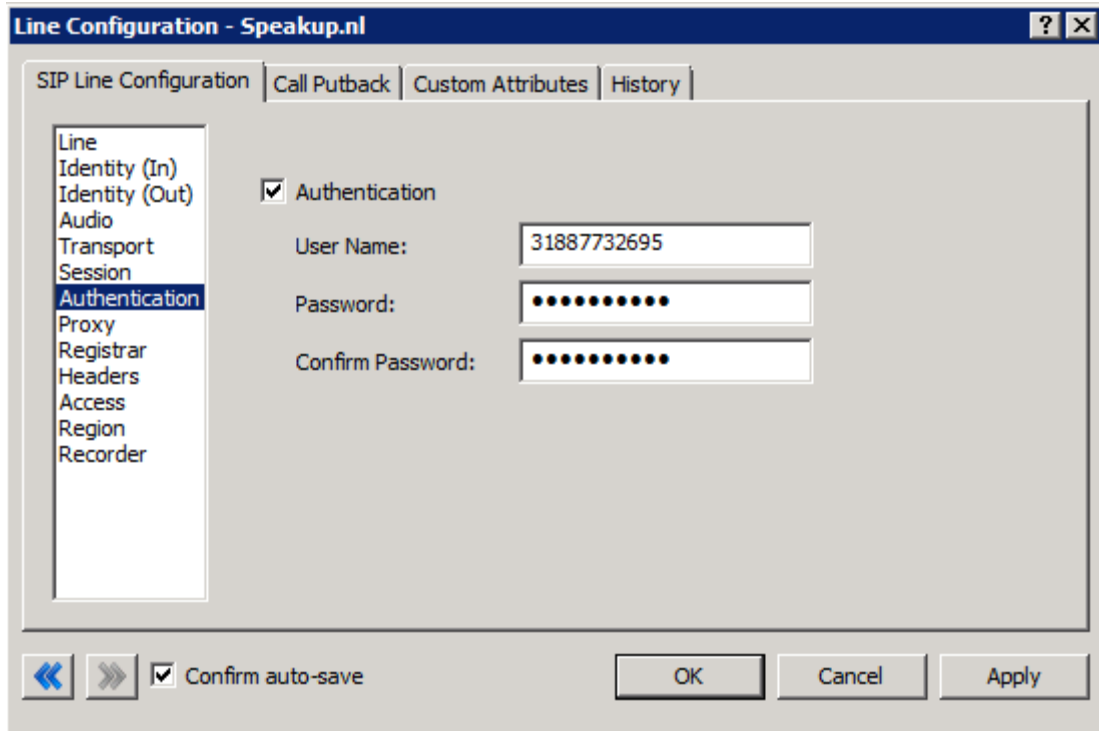


Figure 8: Authentication Menu Line Configuration Page

- **Authentication**

This box must be checked to enable authentication to the SIP Carrier. The *User Name* and *Password* fields should be filled out with the appropriate information provided by the SIP Carrier.

2.1.7 Proxy Menu

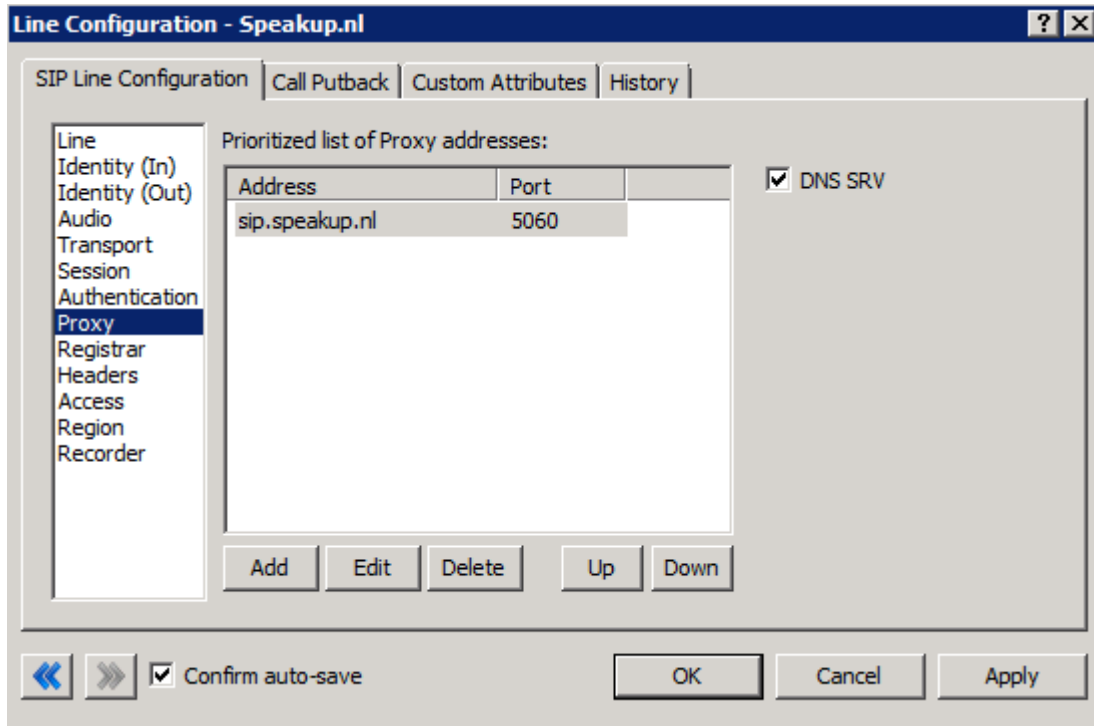


Figure 9: Proxy Menu Line Configuration Page

- **Prioritized list of Proxy IP addresses**

This box is somewhat of a misnomer in the case of some SIP Carriers. In the case of *SpeakUp*, there is not a single IP that is needed. Instead they provide a Fully Qualified Domain Name (FQDN) to a machine or cluster that handles the requests*. When configuring the proxy for *SpeakUp*, this **FQDN** must be entered completely with the port (generally 5060 unless otherwise directed) to enable the service to work properly**.

*DNS SRV is a common method that SIP Carriers use to create a cluster of proxy IP addresses.

**A FQDN must be used because a NAT firewall located between the IC server and the carrier will create an MD5 hash mismatch when authenticating to the SIP carrier. The NAT firewall replaced the IP address of the proxy with an external IP address to create the MD5 hash. This will be a different IP address the carrier used when creating their version of the MD5 hash.

- **DNS SRV**

Enable “DNS SRV” option.

2.1.8 Registrar Menu

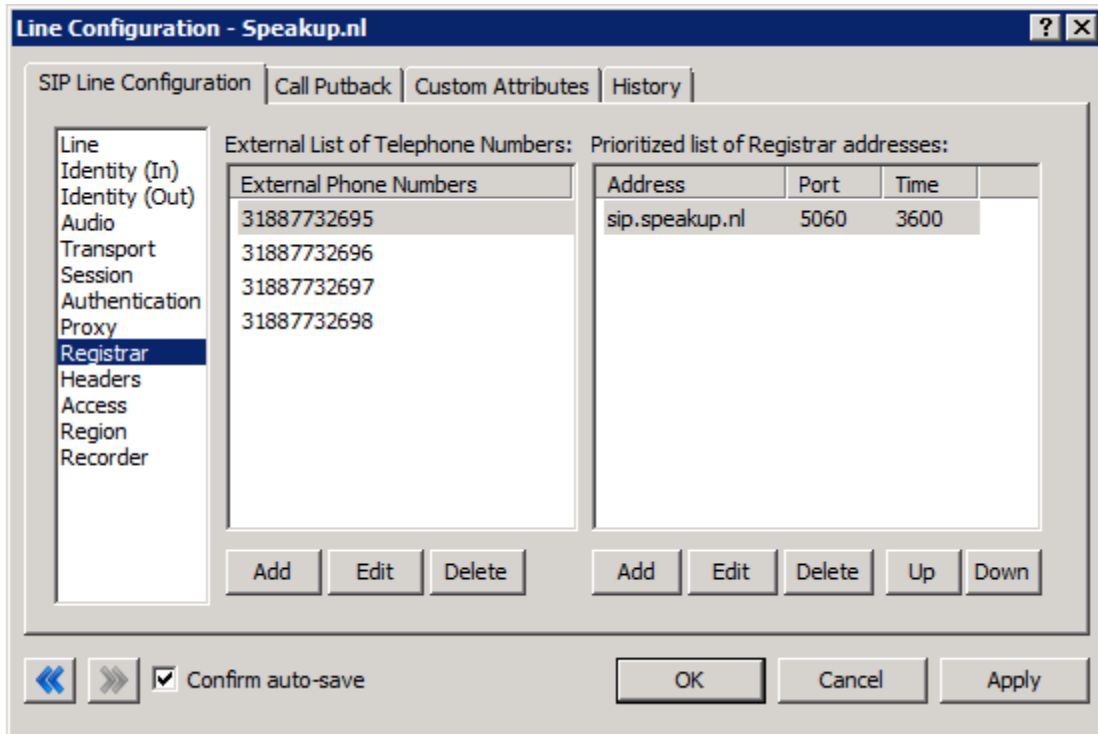


Figure 10: Registrar Menu Line Configuration Page

- **External Phone Numbers**

This box should have the respective group of phone numbers allocated to the customer from the SIP Carrier. If more than one number has been provided, then they should all be placed in this box to allow the IC server to register to all numbers with the SIP Carrier. This in turn will tell the SIP Carrier that it may send calls to all of said numbers to the IC server.

- **Prioritized list of Registrar IP addresses**

This box is used to provide an alternative server or set of servers in which to register. Some SIP Carriers do not handle registration requests on the same server that processes the calls, or have more than one server/cluster that can handle registration requests for redundancy purposes. This information should be provided by the SIP Carrier, and when entered will cause the IC server to send registration requests for all numbers in the *External Phone Numbers* box to all the servers in this registrar server list.

2.1.9 Access Menu

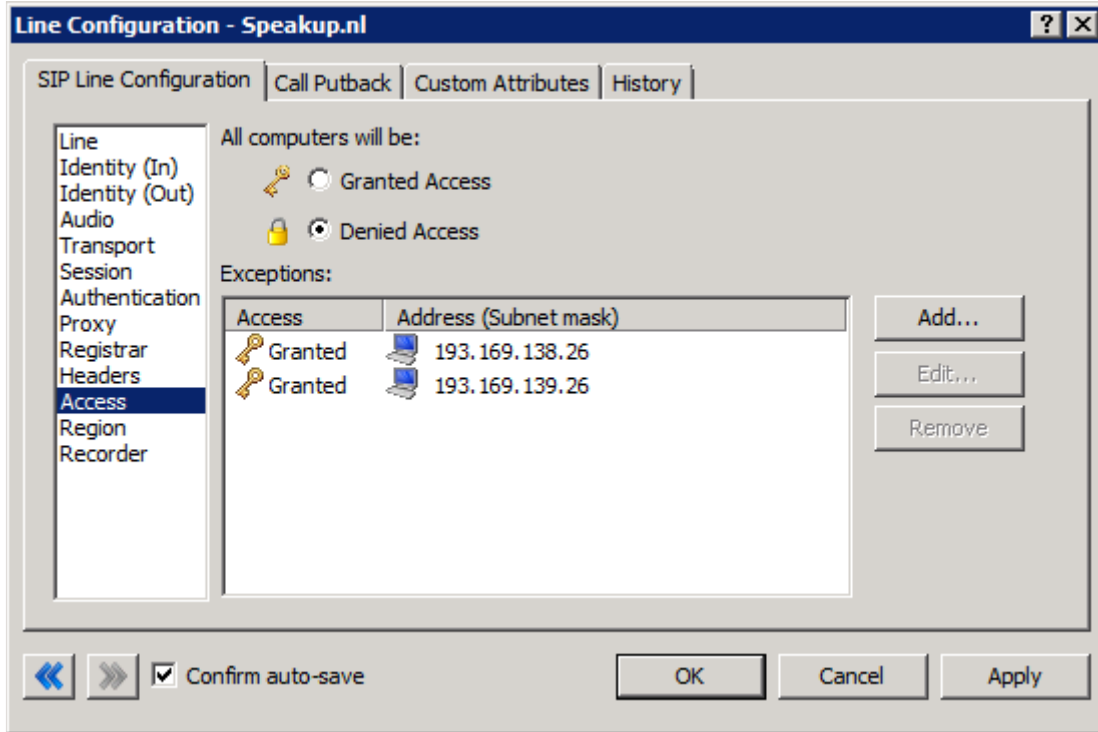


Figure 11: Access Menu Line Configuration Page

- **All computers will be**
This should be set to Denied Access to not only limit the remote end-points that this line will accept calls from, but also so that other lines that use the same Protocol/Port (eg UDP/5060) can be created.
- **Exceptions**
Add each IP Address provided by SpeakUp to the list to grant them access to this Line. This list can be configured to look up the IP Addresses by FQDN.

2.1.10 Region Menu

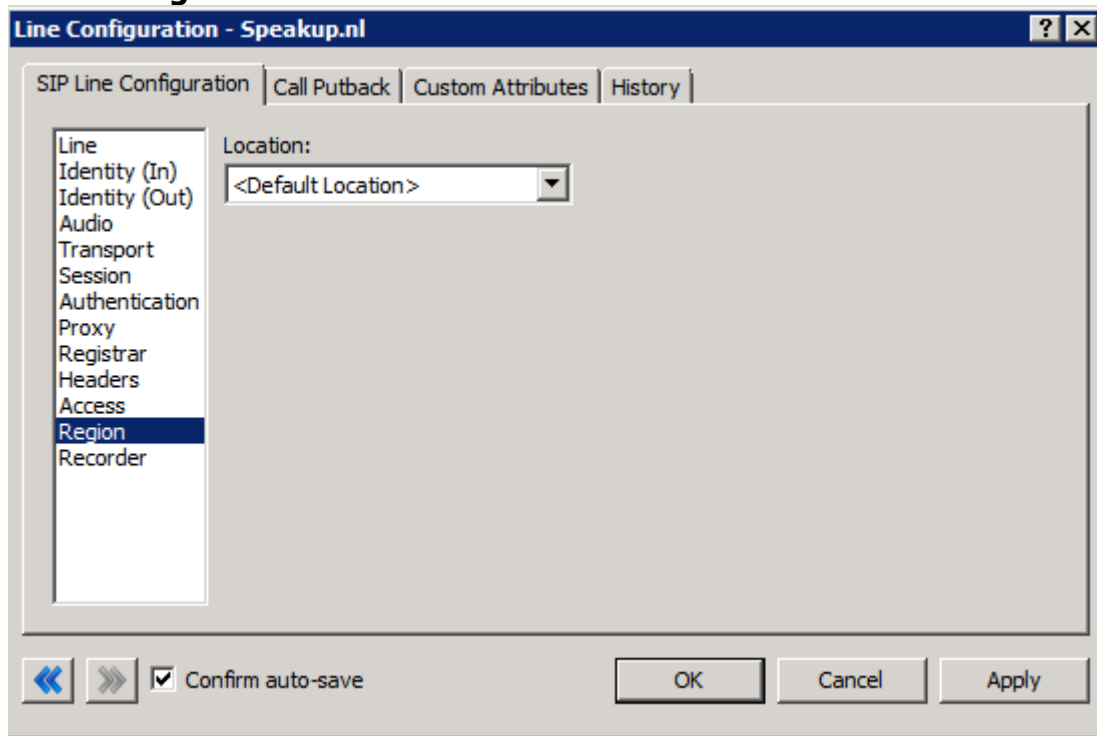


Figure 12: Region Menu Line Configuration Page

- **Location**

This should be set according to business needs, however should take care to assure the assigned Location supports the proper codecs for SpeakUp. SpeakUp supports following codecs:

- G711 alaw (preferred)
- G711 ulaw
- G729

3 SIP Proxy Support

For *SpeakUp*, and all carriers that use the SIP Authentication model, the Interaction SIP proxy is not supported. This information is included for completeness and in the case that it may possibly be supported in the future.

4 Fax Considerations

SpeakUp does not supports useable and functioning T.38 faxing. However if the customer would like to use an analog fax machine, Interactive Intelligence recommends using Audiocodes or Cisco FXS devices.