1 SIP Carriers

1.1 Tele2

1.1.1 Warnings

Check the *SIP 3rd Party SIP Carrier Matrix* for certification status, and supported features. More info about the *SIP 3rd Party SIP Carrier Matrix* can be found in the SIP Carrier section of the web site(s) below:

http://testlab.inin.com

1.1.2 Vendor Contact

Vendor Web Site : *www.tele2.nl*

1.1.3 Versions Verified

Interaction Center 2015 R2 Patch1

1.1.4 PreInstall

Tele2 VoIP Connect will provide users with a set of authentication credentials, 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

Tele2 VoIP Connect requires a fully configured SIP enabled IC server. A SIP line must be created. See Line Configuration section 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

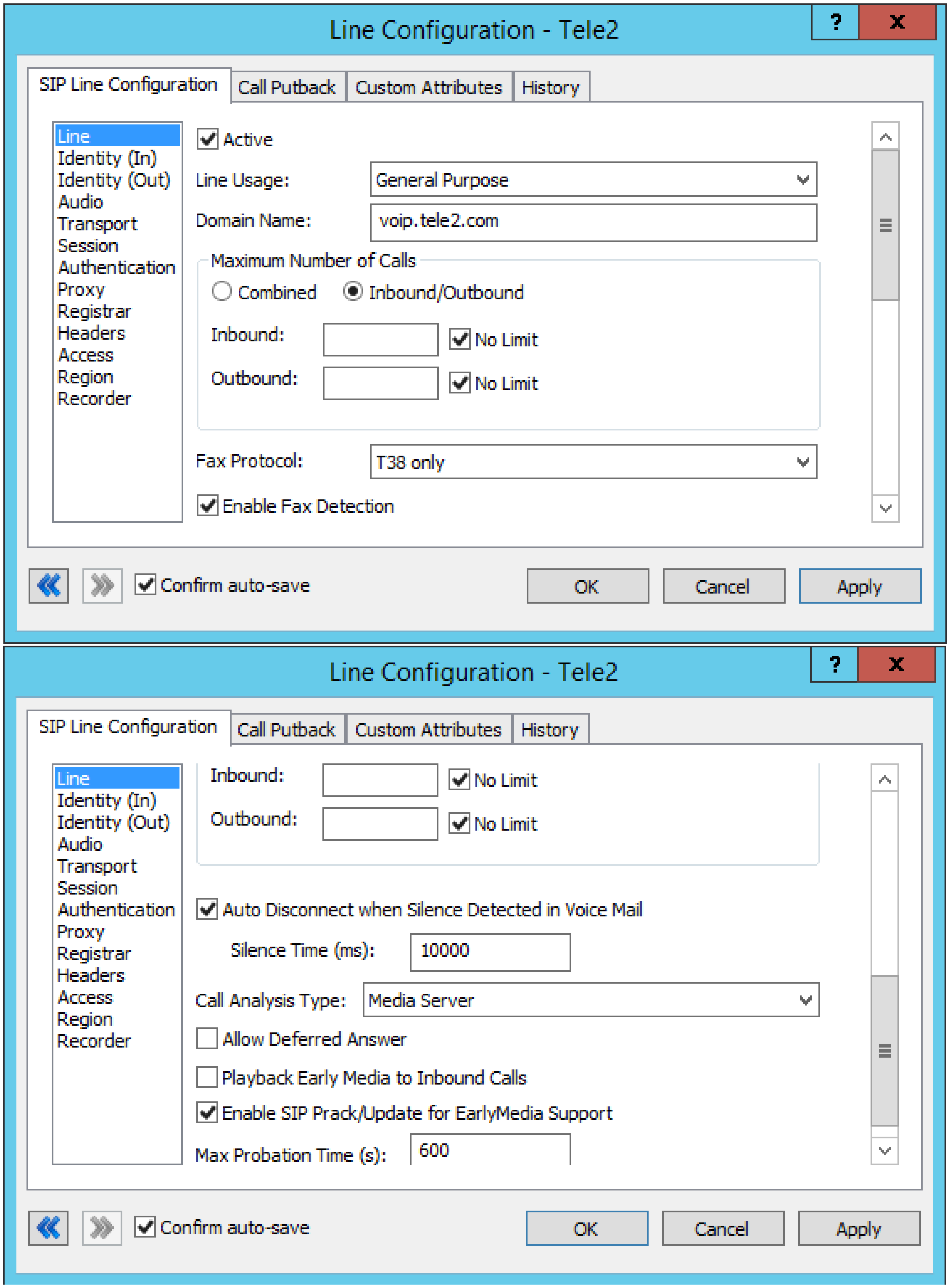


Figure 1: Line Menu Line Configuration Page

2.1.1.1 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*.

2.1.1.2 Domain Name

This box should contain the Fully Qualified Domain Name (FQDN) of the authentication/registration server provided by *Tele2 VoIP Connect*. It is used for the registration and/or authentication of the line.

2.1.1.3 Enable Fax Detection

Tele2 VoIP Connect's SIP Carrier service supports the T.38 faxing protocol by default. Leave this box unchecked if you do not have (or whish to use) an analog to SIP capable FXS type device to connect an analog fax machine to the system.

2.1.1.4 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 (In) Menu

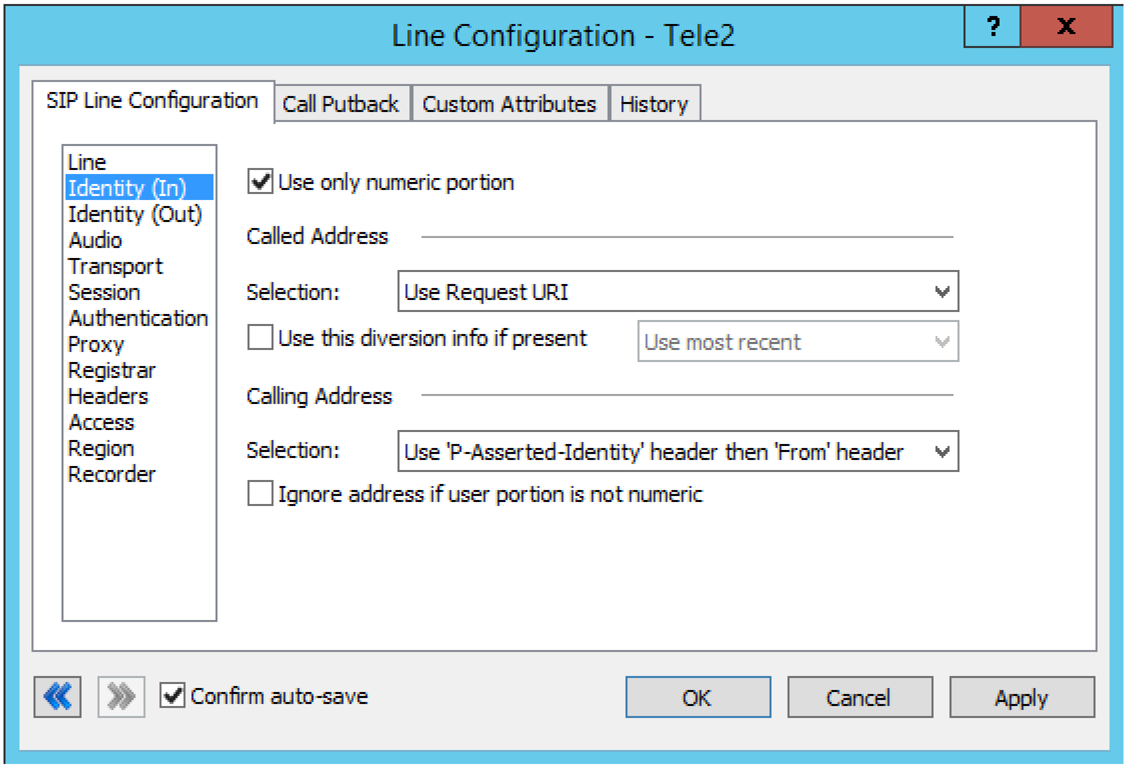
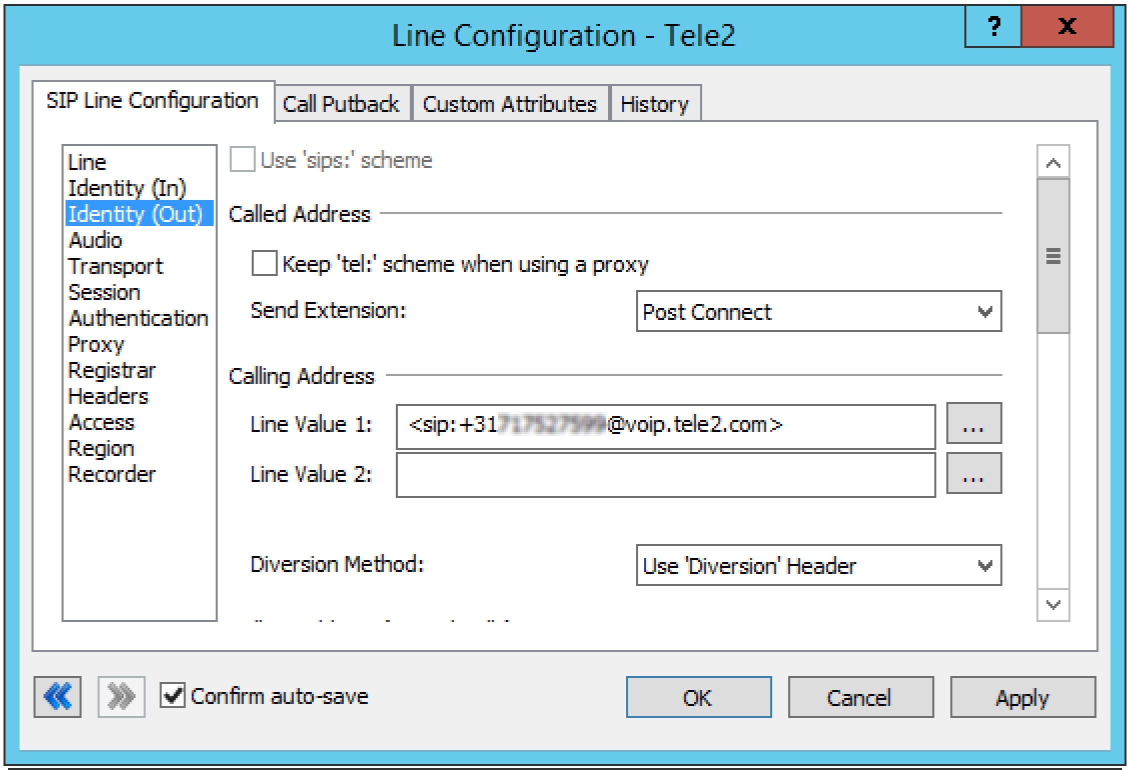


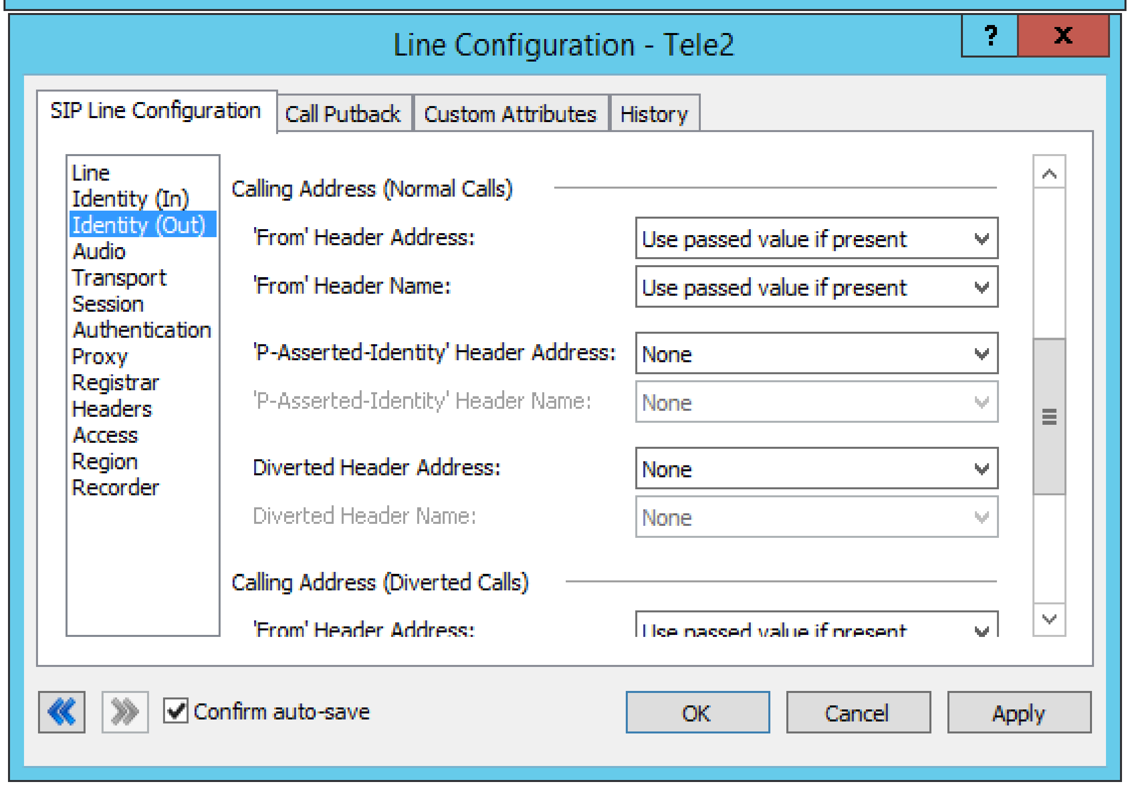
Figure 2: Identity (In) Menu Line Configuration Page

2.1.2.1 Identity (In) Menu Options

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

2.1.3 Identity (Out) Menu





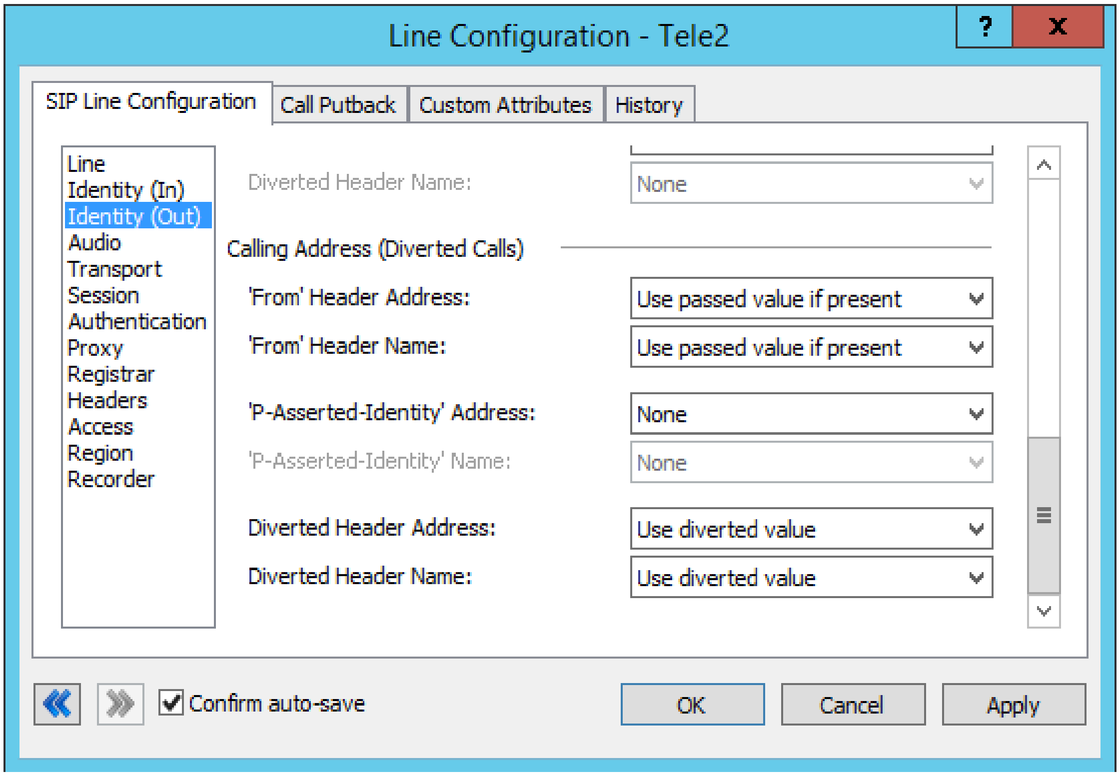


Figure 3: Identity (Out) Menu Line Configuration Page

2.1.3.1 Calling Address

Clicking the "…" button next to the Line Value 1 brings up the Configure Line Value dialog.

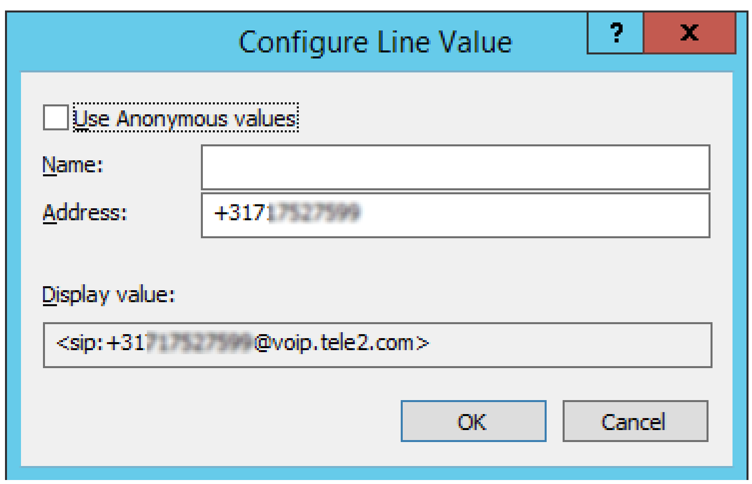


Figure 4: Configure Line Value Configuration Page

2.1.3.2 Calling Address (Diverted Calls)

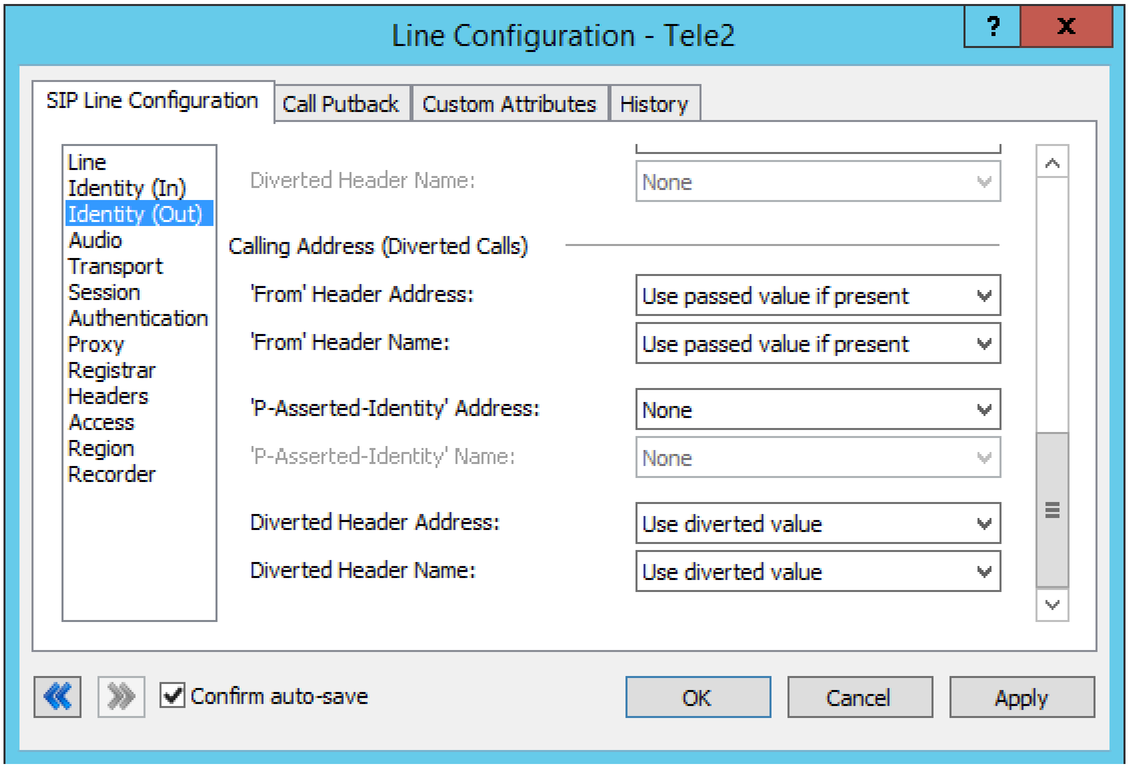


Figure 5: Calling Address (Diverted Calls) Configuration Page

Set values according to Figure 5.

2.1.3.3 Remainder of Identity (Out) Menu Options

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

2.1.4 Audio Menu

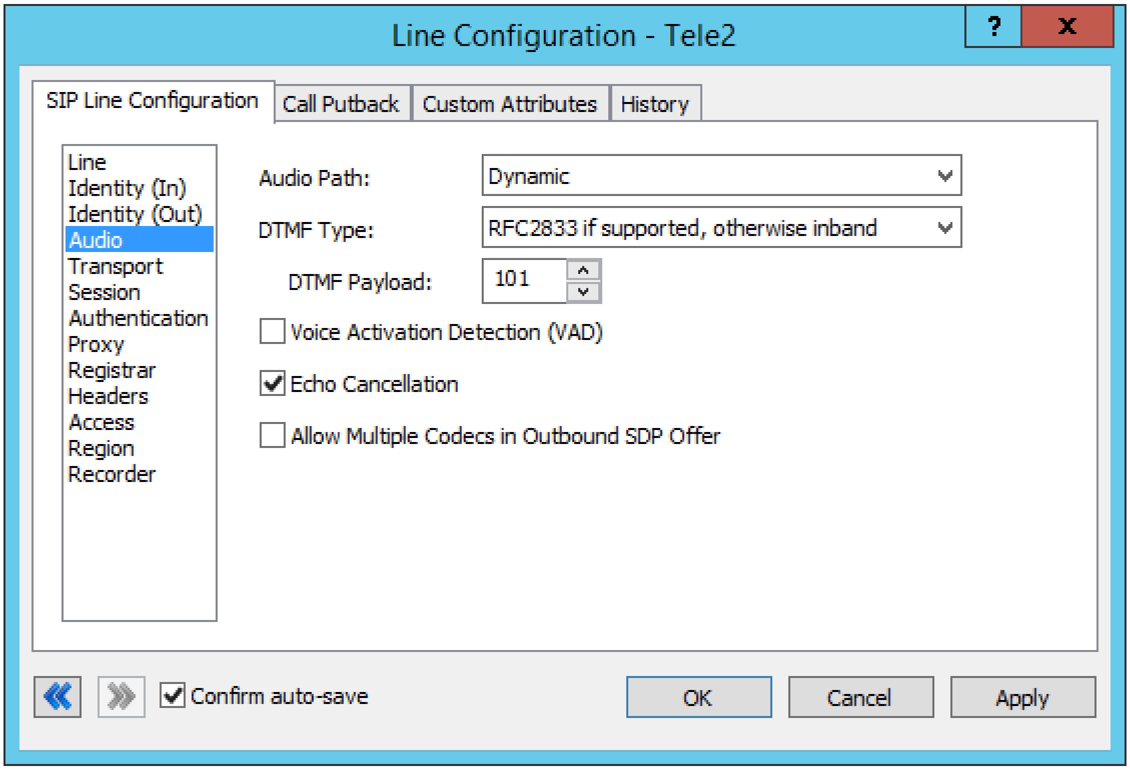


Figure 6: Audio Menu Line Configuration Page

2.1.4.1 Audio Path

The choice between Dynamic or Always-in is the choice of the client with respect to the business being done on the server.

2.1.4.2 DTMF Type

Recommended settings: "RFC2833 only" or "RFC2833 if supported…".

2.1.4.3 Remainder of Audio Menu Options

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

2.1.5 Transport Menu

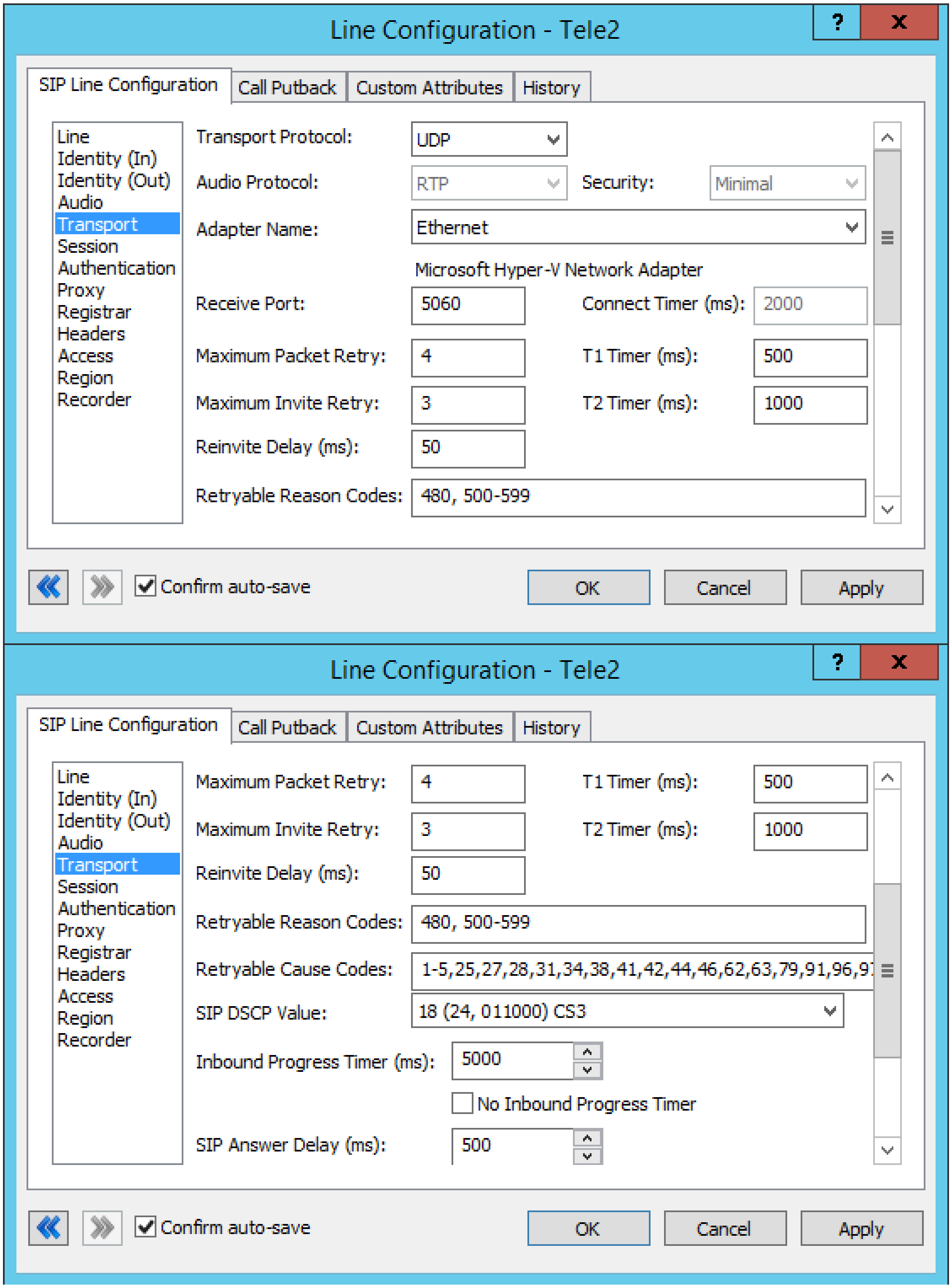


Figure 7: Transport Menu Line Configuration Page

2.1.5.1 Transport Protocol

This should be set to UDP

2.1.5.2 Receive Port

This option should be set to 5060

2.1.5.3 Remainder of Transport Menu Options

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

2.1.6 Session Menu

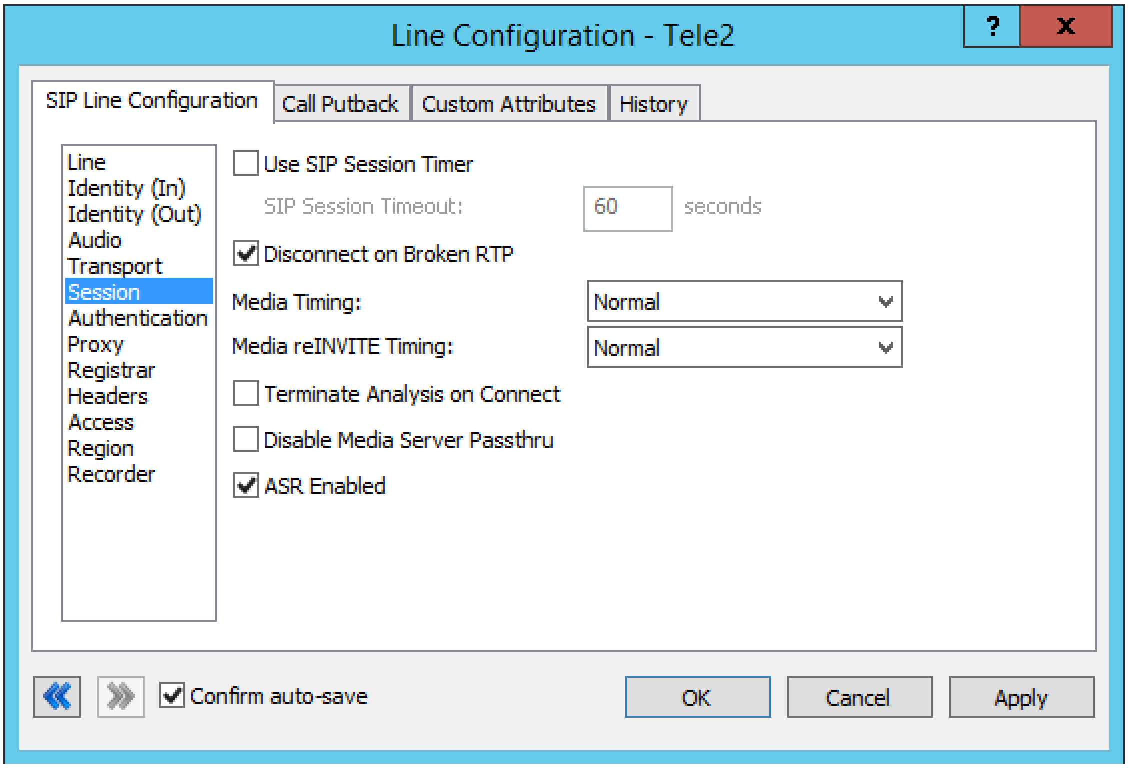


Figure 8: Session Menu Line Configuration Page

2.1.6.1 Media Timing/Media reINVITE Timing

Recommended value: "Normal"

2.1.6.2 Remainder of Session Menu Options

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

2.1.7 Authentication Menu

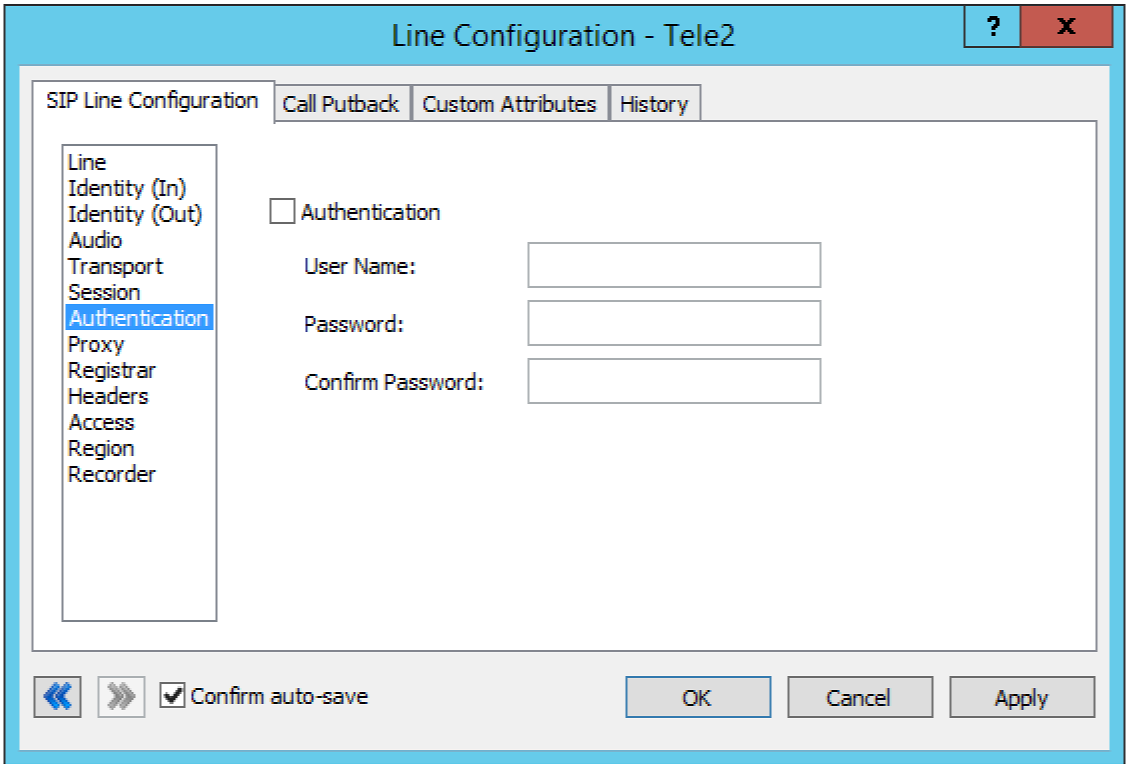


Figure 9: Authentication Menu Line Configuration Page

2.1.7.1 Authentication Menu Options

Leave it unchecked.

2.1.8 Proxy Menu

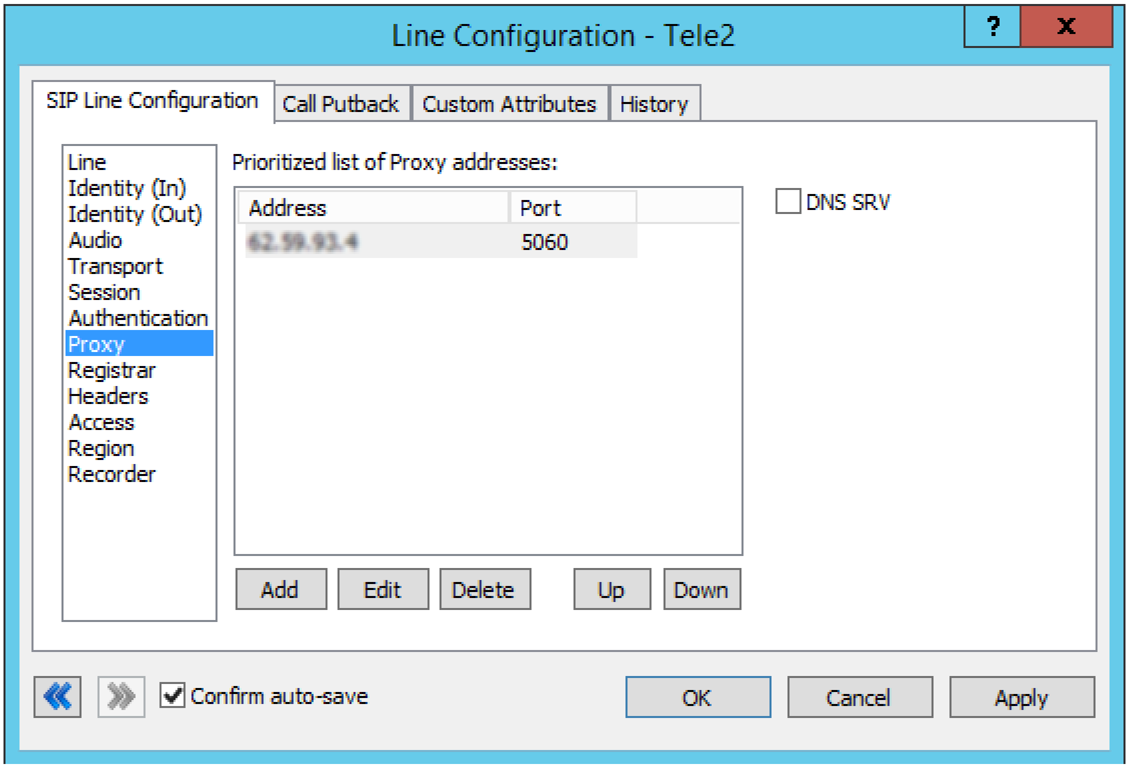


Figure 10: Proxy Menu Line Configuration Page

2.1.8.1 Prioritized list of Proxy IP addresses

Tele2 VoIP Connect provides a single IP address that has to be entered.

2.1.9 Registrar Menu

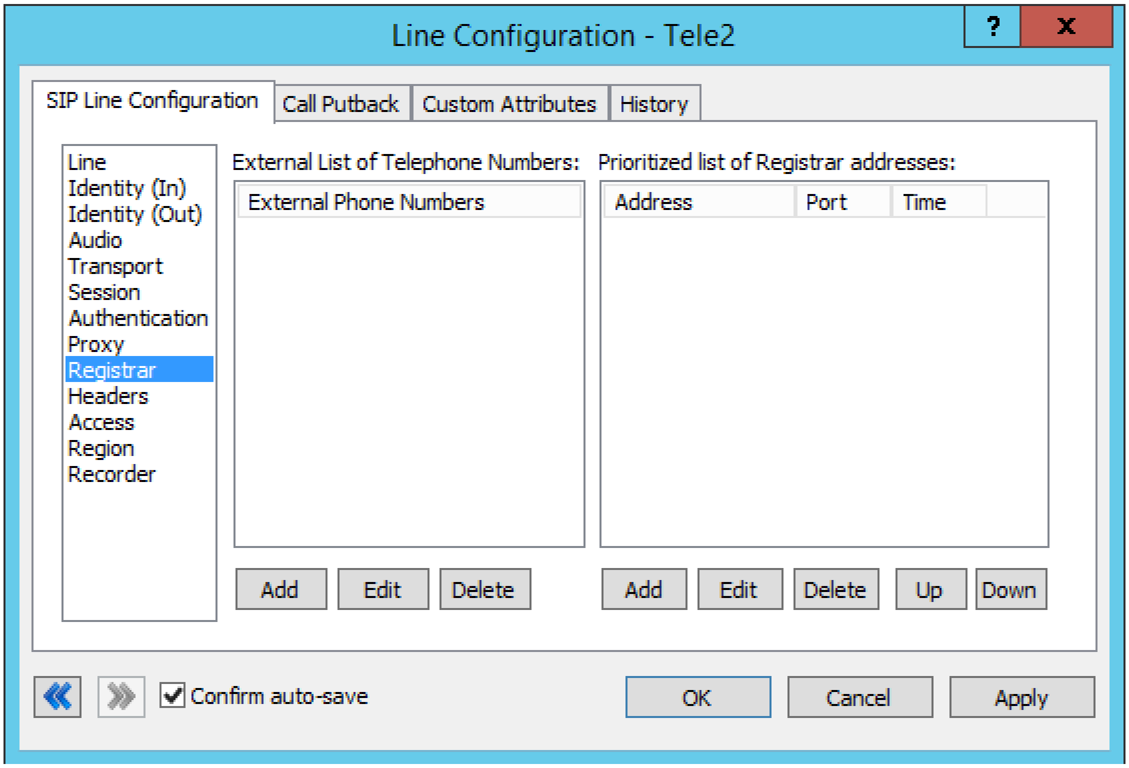


Figure 11: Registrar Menu Line Configuration Page

2.1.9.1 External Phone Numbers

Leave it empty.

2.1.9.2 Prioritized list of Registrar IP addresses

Leave it empty.

2.1.10 Access Menu (Access Control lists)

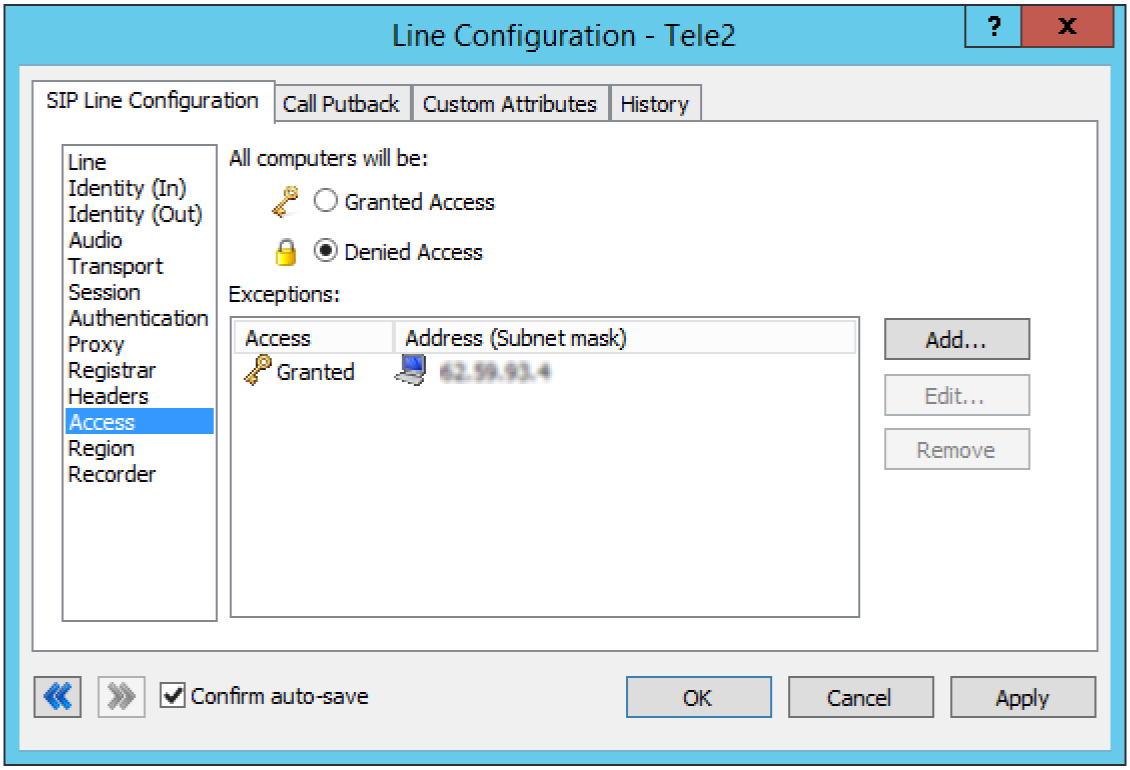


Figure 12: Access Menu Line Configuration Page

2.1.10.1 Access to *Tele2 VoIP Connect* SIP Line

For the access menu, the radio button should be shifted to the value:

By default, all computers will be: ***Denied Access***.

In the access list below the radio button, the resolved IP address for each proxy server **MUST** be added.

3 SIP Proxy Support

Tele2 VoIP Connect supports Interaction SIP Proxy.

Note: If using a NAT/PAT type solution, a SIP Proxy can only be used in conjunction with a SIP Carrier that supports a static IP proxy (on their side, the same thing entered into the proxy menu on the lines page, not the SIP proxy). If this is not supported, the SIP Proxy can not properly pass its return address through to the carrier.

If a SIP Proxy is to be used in a NAT/PAT environment, then the externally facing IP of the **SIP Proxy** must be entered in the following places in the *Tele2 VoIP Connect* SIP Line configuration.

- On the proxy menu, in place of those provided by the Carrier
- On the registrar menu, in places of those provided by the Carrier

Also, the SIP Proxy (in a non NAT/PAT environment, or the NAT/PAT externally facing IP) must have the IP address provided to *Tele2 VoIP Connect*. Otherwise it will reject messages coming to it from an unknown IP.

The information regarding the SIP Carrier is then transferred to the appropriate places in the SIP Proxy. The SIP Proxy then feeds the required info back to the SIP Carrier. It is required to put the SIP Proxy information in the IC server. This is due to the fact that it is no longer directly talking to the SIP Carrier, and all information coming and going must be relative to the SIP Proxy.

4 Fax Caveats

Tele2 VoIP Connect supports useable and functioning T.38 faxing. However if the customer would like to use an analog fax machine connected to the network, or if T.38 faxing is not an option, the way to circumvent this problem is with an analog to SIP FXS device connecting an analog fax machine to the IP network. The FXS device will pass the SIP information on allowing for G.711 pass-through (which is the carrying of the fax signal through the voice packets on the network). This has been tested using an AudioCodes Media Pack, and a Cisco FXS card on its SIP Gateway.

5 Restricted ANI (additional)

Tele2 VoIP Connect requires Privacy header.

The message manipulation rules were configured so that the header is added when the From header is "Anonymous".

Tele2 VoIP Connect doesn't accept RESTRICTED or other non-numbers in FROM header, hence another rule overwrites the user's ANI from "Anonymous" to the main line number, but since it doesn't get displayed, the remains anonymous.