1 SIP Carriers

1.1 Macquarie Telecom

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 : *http://www.macquarietelecom.com/*

1.1.3 Versions Verified

SIP Carrier status as of *12 Jan 2015 : validated on CIC 4.0 SU6*.

1.1.4 PreInstall

Macquarie Telecom will provide users with a set of authentication credentials, a Number Range(s) 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

Macquarie Telecom SIP Connect requires a fully configured SIP enabled IC server. Two SIP lines must be created. The configuration for these lines will be covered in section 1.1.7.1.1 below.

1.1.6 Required Post Installation Steps

Confirm capacities and capabilities of purchased service.
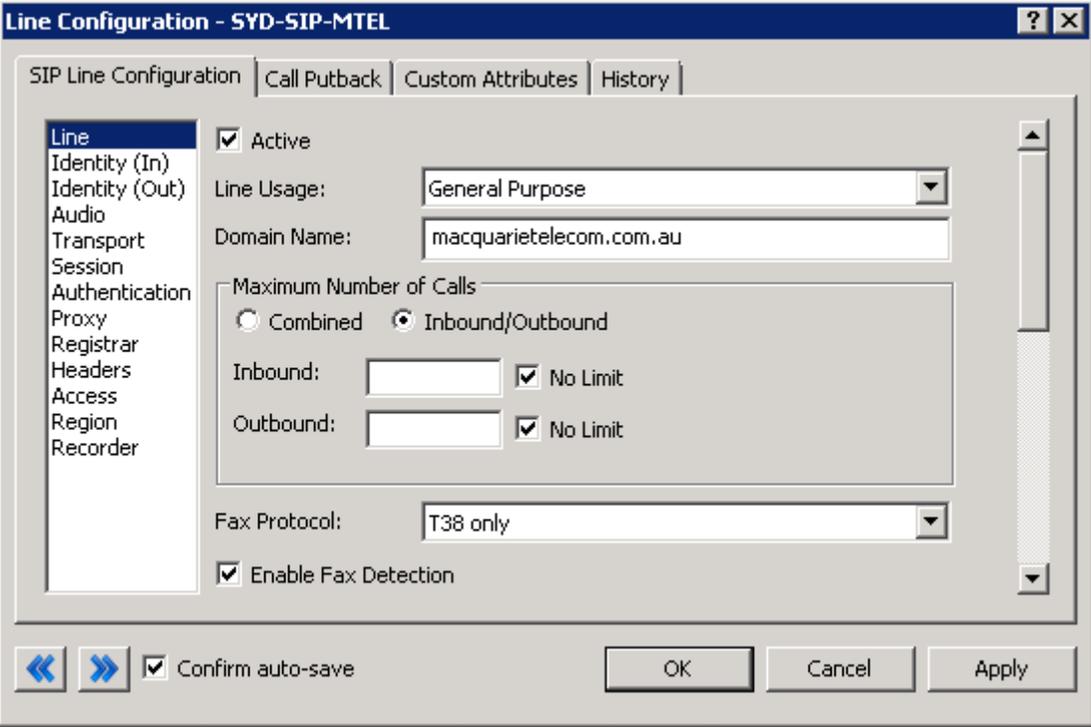
2   IC Configuration Guide

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.
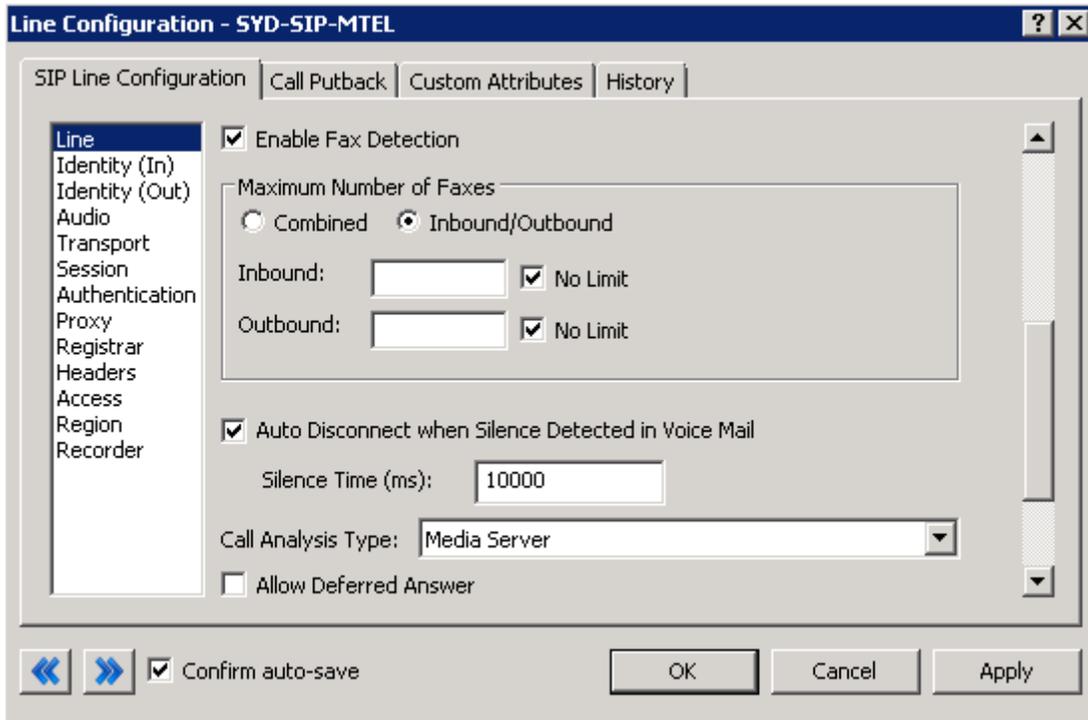
As stated before, two lines must be created. These lines are required, one for the Macquarie Telecom connection, and one for the stations. Each portion of the lines page will be explained as it relates to the Macquarie Telecom Service. For this document, the Macquarie Telecom connection line will be referred to as Macquarie Telecom SIP Line, and the station line will be referred to as *stations*. Also, 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.
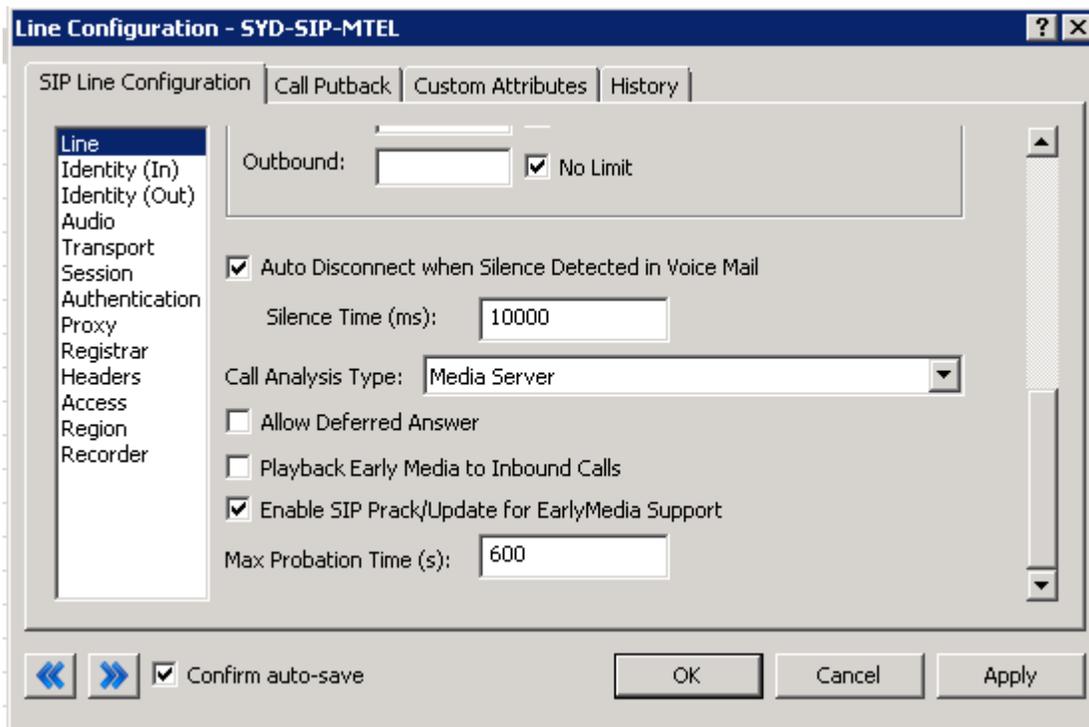
2.1.1 Line Menu

Line Menu Line Configuration Page

Line Configuration - SYD-SIP-MTEL

SIP Line Configuration | Call Putback | Custom Attributes | History

Line
Identity (In)
Identity (Out)
Audio
Transport
Session
Authentication
Proxy
Registrar
Headers
Access
Region
Recorder

☑ Enable Fax Detection

Maximum Number of Faxes
○ Combined   ⦿ Inbound/Outbound

Inbound: [   ]   ☑ No Limit

Outbound: [   ]   ☑ No Limit

☑ Auto Disconnect when Silence Detected in Voice Mail

Silence Time (ms): [10000]

Call Analysis Type: [Media Server   ▼]

☐ Allow Deferred Answer

《 》 ☑ Confirm auto-save   [OK]   [Cancel]   [Apply]

Line Menu Line Configuration Continued

Line Configuration - SYD-SIP-MTEL

SIP Line Configuration | Call Putback | Custom Attributes | History

Line
Identity (In)
Identity (Out)
Audio
Transport
Session
Authentication
Proxy
Registrar
Headers
Access
Region
Recorder

Outbound: [   ]   ☑ No Limit

☑ Auto Disconnect when Silence Detected in Voice Mail

Silence Time (ms): [10000]

Call Analysis Type: [Media Server   ▼]

☐ Allow Deferred Answer

☐ Playback Early Media to Inbound Calls

☑ Enable SIP Prack/Update for EarlyMedia Support

Max Probation Time (s): [600]

《 》 ☑ Confirm auto-save   [OK]   [Cancel]   [Apply]

Line Menu Line Configuration Continued

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   Phone Number

The phone number provided by the SIP Carrier should be entered into this box. The number entered is used in the "From" header in outbound SIP calls. Incorrect numbers can lead to some functionality not working as expected or at all.

2.1.1.3   Domain Name

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

2.1.1.4   Enable T.38 Faxing

MACQUARIE TELECOM's SIP Carrier service supports the T.38 faxing protocol by default.

2.1.1.5   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

All left at default. Enabled usage of numeric portion only. Called address 'use request URI' and Calling address is set to 'Use PAE then FROM'.
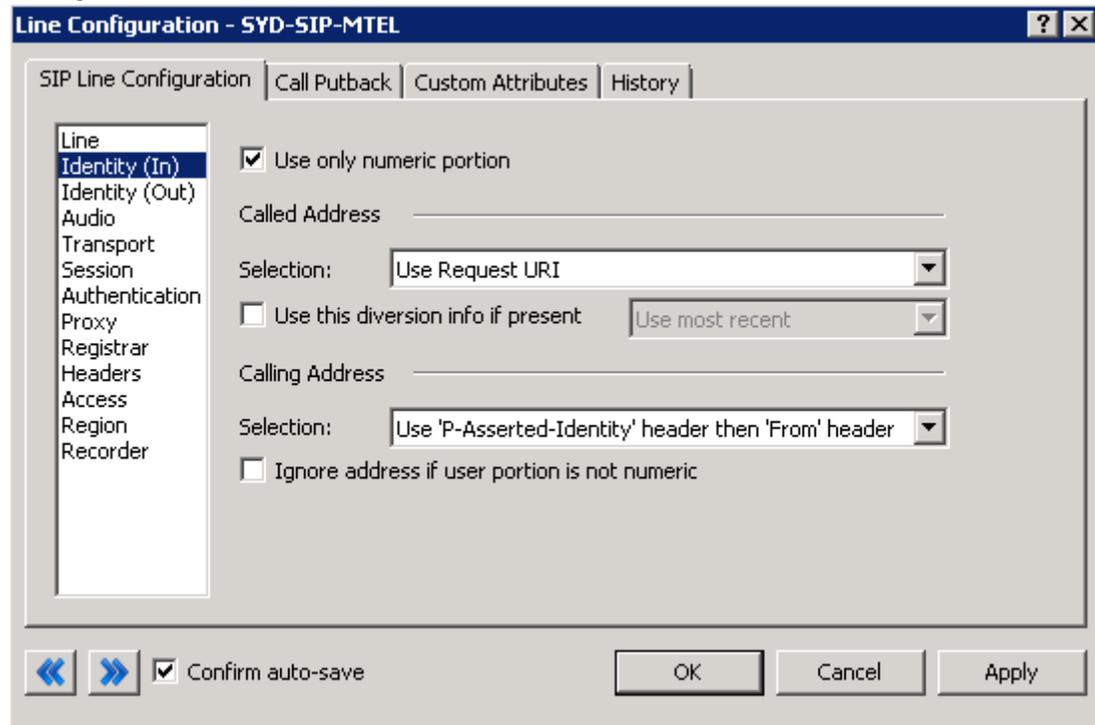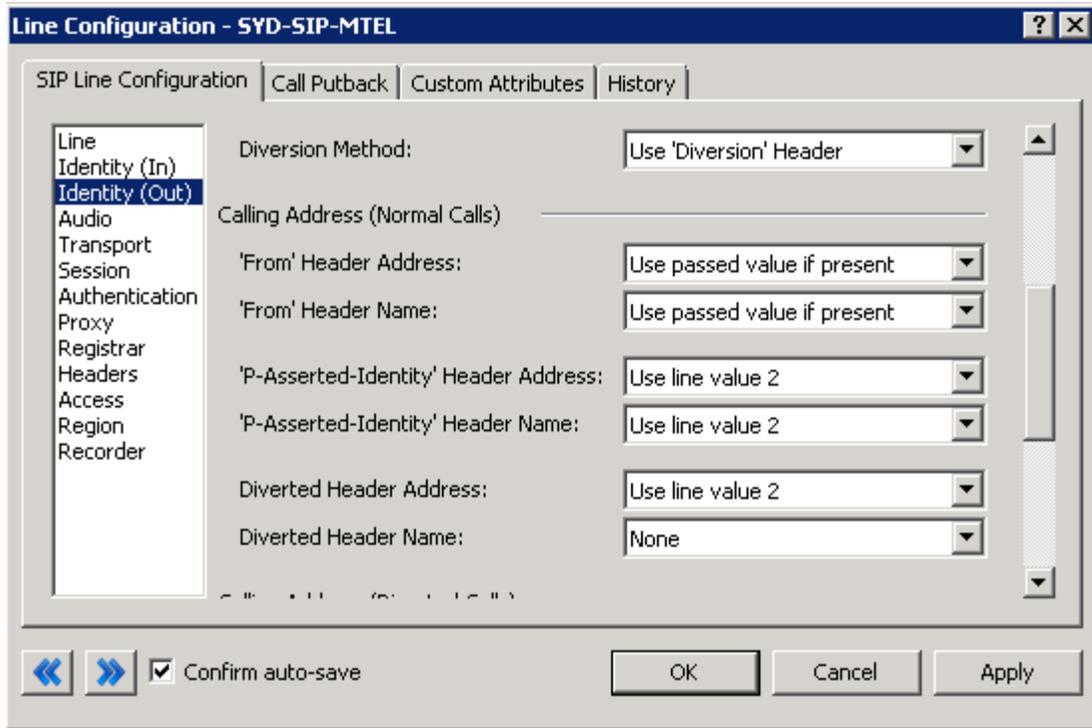

2.1.3 Identity (Out) Menu

Restricting CLI on inbound calls doesn't work as per test cases using 'RESTRICTED' value in 'Outbound ANI' on user. The call fails to proceed. Different settings were used to make it work.
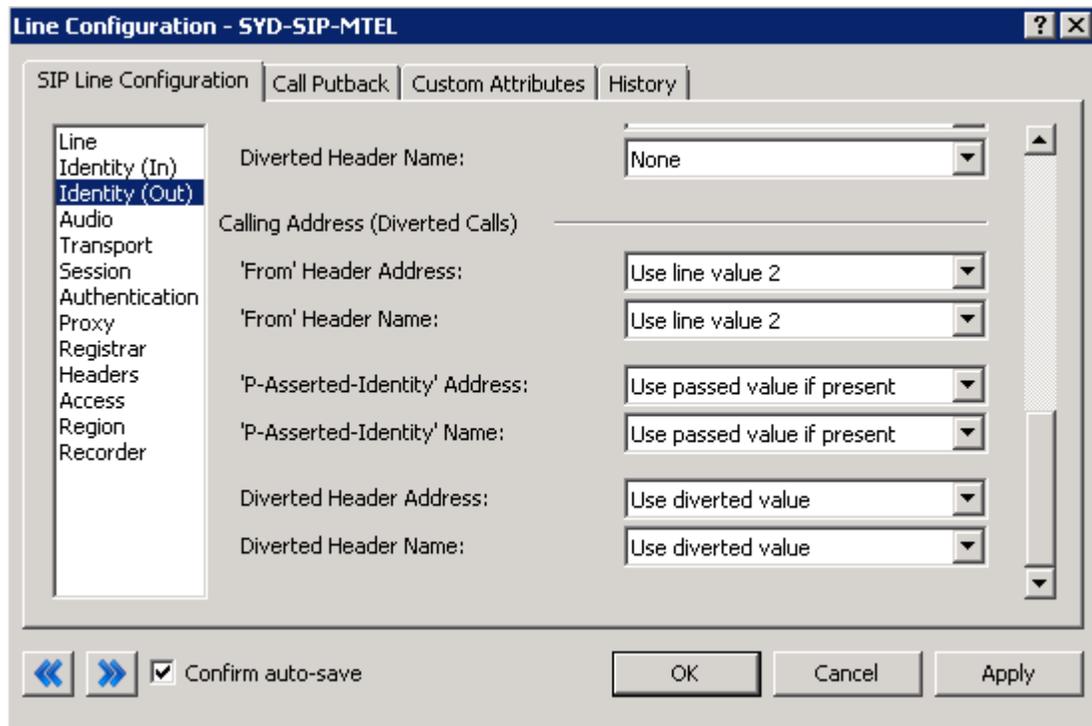
Line Value 1 – set to anonymous, that way, which is further checked at the SBC and message manipulation is done to change the outgoing From number, thus not revealing the true number, but a pilot number instead.
Line Value 2 – Pilot number configured

Line Configuration - SYD-SIP-MTEL

SIP Line Configuration | Call Putback | Custom Attributes | History

Line
Identity (In)
Identity (Out)
Audio
Transport
Session
Authentication
Proxy
Registrar
Headers
Access
Region
Recorder

Diversion Method: `Use 'Diversion' Header`

Calling Address (Normal Calls)

'From' Header Address: `Use passed value if present`

'From' Header Name: `Use passed value if present`

'P-Asserted-Identity' Header Address: `Use line value 2`

'P-Asserted-Identity' Header Name: `Use line value 2`

Diverted Header Address: `Use line value 2`

Diverted Header Name: `None`

☑ Confirm auto-save   OK   Cancel   Apply

From header is used to pass the configured ANI and name of the agent. If no ANI is specified, Line Value 1 is used, which is "anonymous". The SBC checks if the user is anonymous based on the From header, and adds Privacy header as well as changing the From to the Pilot number.
PAI includes the Pilot details, it is used to let the carrier know which account is making the call. If incorrect Pilot number is passed, in most cases, the response message would say "The device is not allowed to make calls on this line…" or similar.

Line Configuration - SYD-SIP-MTEL

SIP Line Configuration | Call Putback | Custom Attributes | History

Line
Identity (In)
Identity (Out)
Audio
Transport
Session
Authentication
Proxy
Registrar
Headers
Access
Region
Recorder

Diverted Header Name: `None`

Calling Address (Diverted Calls)

'From' Header Address: `Use line value 2`

'From' Header Name: `Use line value 2`

'P-Asserted-Identity' Address: `Use passed value if present`

'P-Asserted-Identity' Name: `Use passed value if present`

Diverted Header Address: `Use diverted value`

Diverted Header Name: `Use diverted value`

☑ Confirm auto-save   OK   Cancel   Apply

For Diverted calls, the Pilot number is displayed instead of the real agent numbers, but that can be changed.

In the PAI field, it is set to use passed values. The SBC also checks for the Diversion header, and if it is set to "Anonymous", it adds the Privacy header, indicating to the carrier that the call should be anonymous.

*Note: For Diverted calls, PAI is populated with automatically with Pilot number, because the system makes a new call. Through testing that was found to be the working combination.

However, it may be required to change PAI value to use diverted value instead.

The screenshot of the manipulation rules on the SBC is attached at the end of this document.
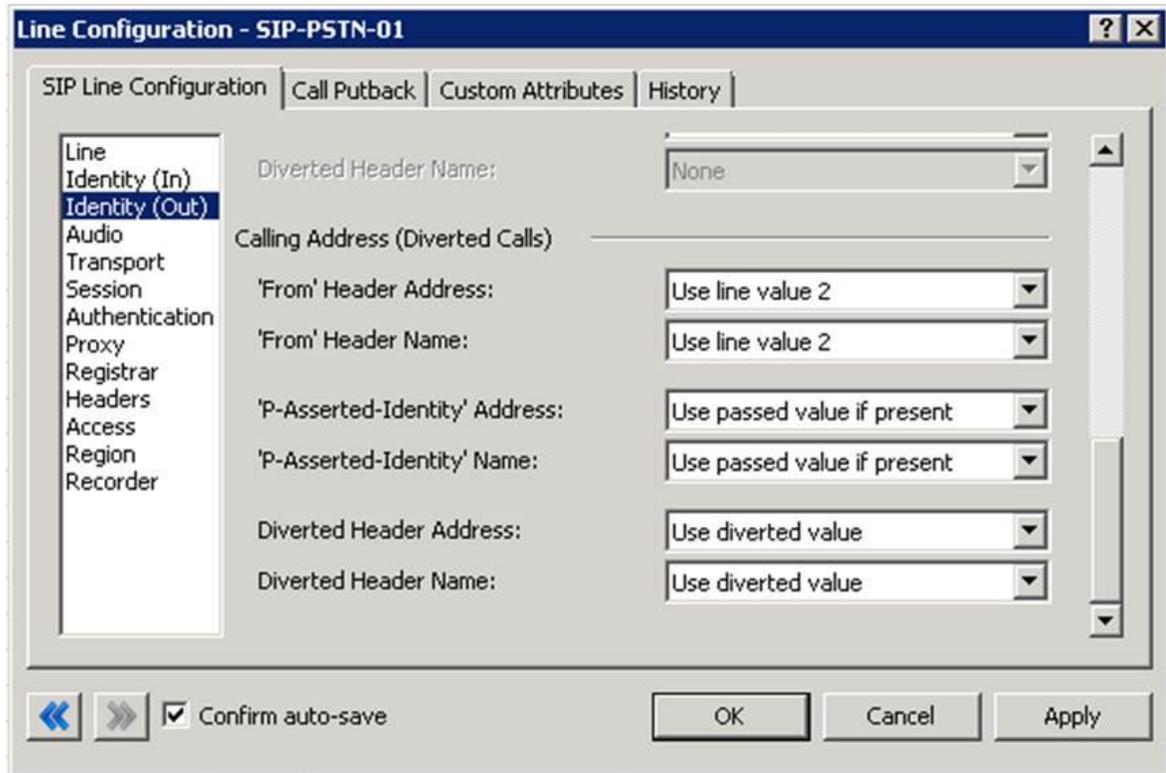
Identity (Out) Menu Line Configuration Page

2.1.3.1   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.
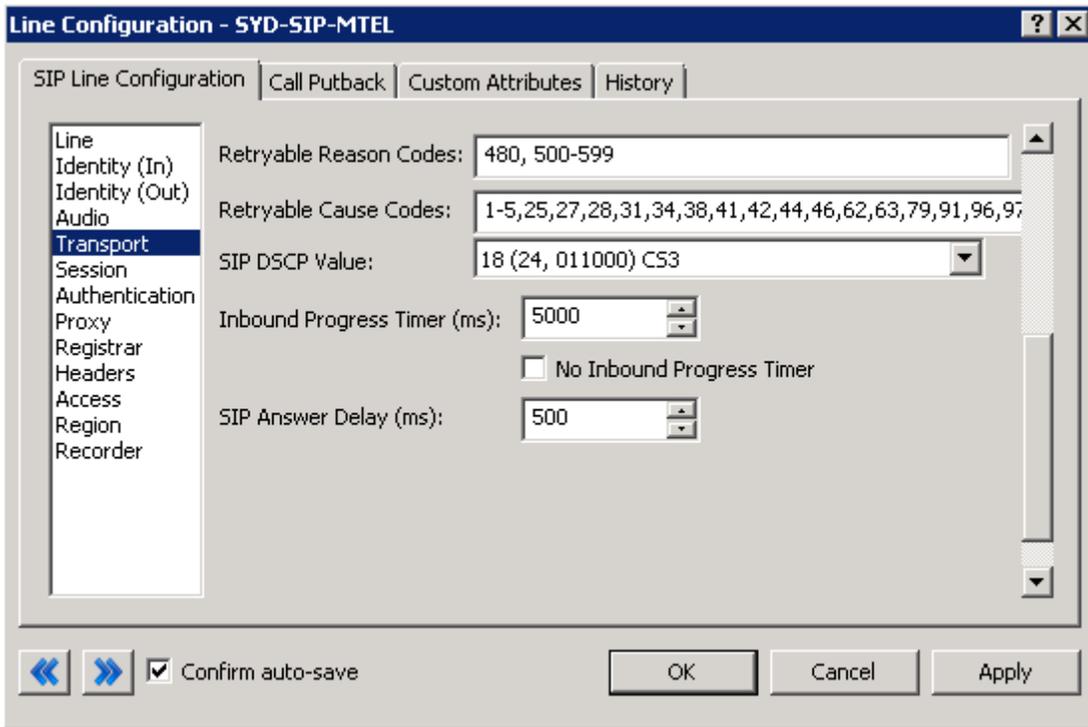
2.1.4 Audio Menu
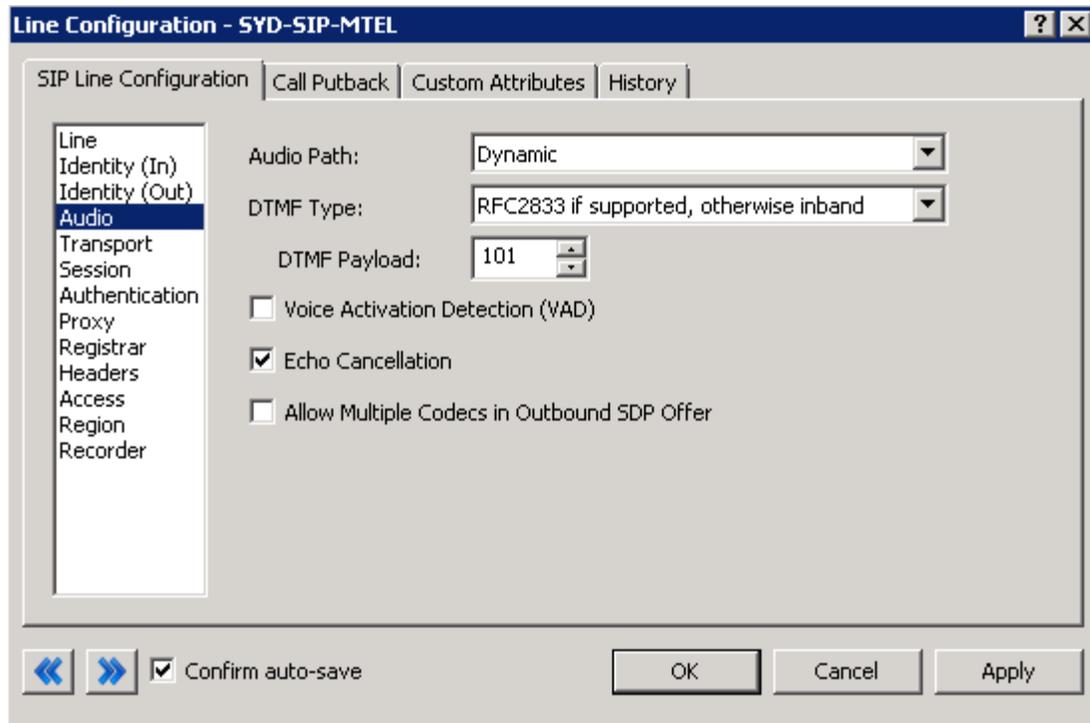
Audio Menu Line Configuration Page

2.1.4.1 Audio Path

This is for the most part, the choice of the client with respect to the business being done on the server. However, there are **several important caveats**.

1. Dynamic audio for SIP carriers has significantly less delay as compared to Always In audio (~100ms).

2. The audio will be brought into the IC server when set to Dynamic Audio for any call that is recorded (just for that call, not permanently). If using a Media Server recorded calls will not travel through the IC server, and very little, if any, latency will be added.

2.1.4.2 DTMF Type

DTMF has three options, In-Band, RFC2833, and RFC2833 Only. These are up to the discretion of the user. Macquarie Telecom supports Out-of-Band (RFC2833) DTMF Type.

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

Transport Menu Line Configuration Page

Transport Menu Line Configuration Page continued

2.1.5.1   Transport Protocol

This option should be set to UDP. As of *12 Jan 2015* Macquarie Telecom has support for UDP by default. TCP and TLS are not currently supported.

2.1.5.2   Receive Port

This option should be set to 5060 (the standard SIP port), unless an agreement for an alternative port has been agreed upon with the SIP Carrier. As of *12 Jan 2015* Macquarie Telecom only has support for port 5060 in the standard offering.

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

Session Menu Line Configuration Page

2.1.6.1   Media Timing/Media reINVITE Timing

This dropdown pair controls Delayed Media support. Setting both to Normal is the recommend method by Interactive Intelligence for all SIP Carriers.

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

Authentication is not configured, since SBC authenticate with SIP trunk in the test lab. It is possible to configure SBC to pass through 401request from SIP trunk for authentication, if required. May be applicable to CaaS setup, if the SBC's are shared.
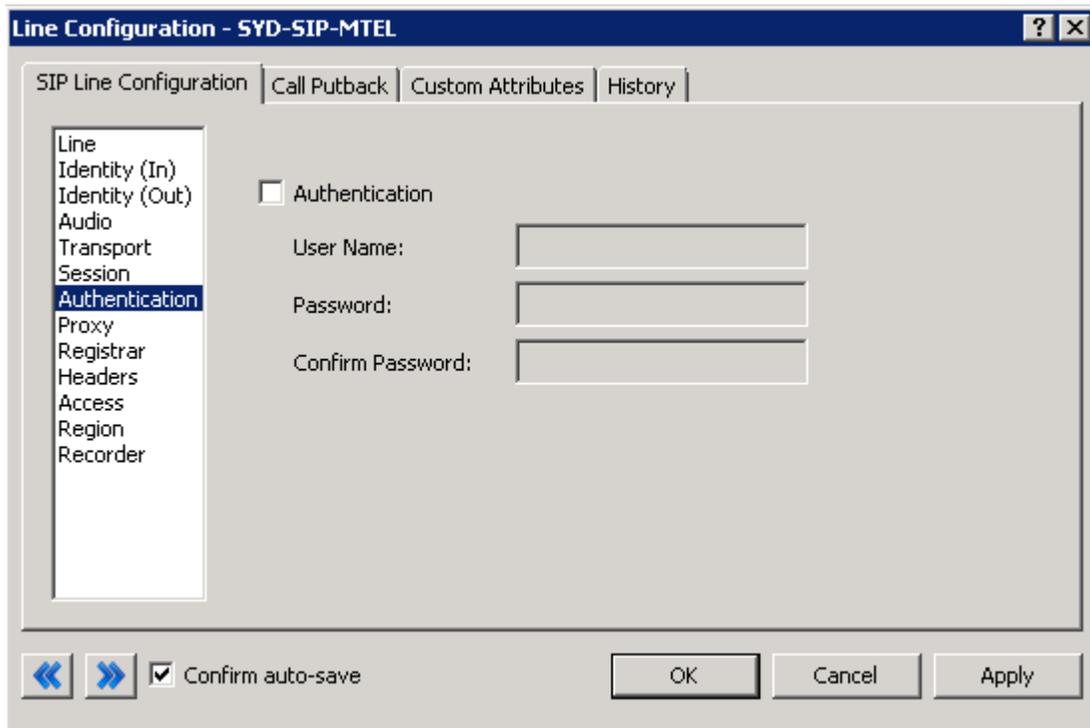
Authentication Menu Line Configuration Page

2.1.8 Proxy Menu

Proxy Menu Line Configuration Page

2.1.8.1   Prioritized list of Proxy IP addresses

This the SBC's address.

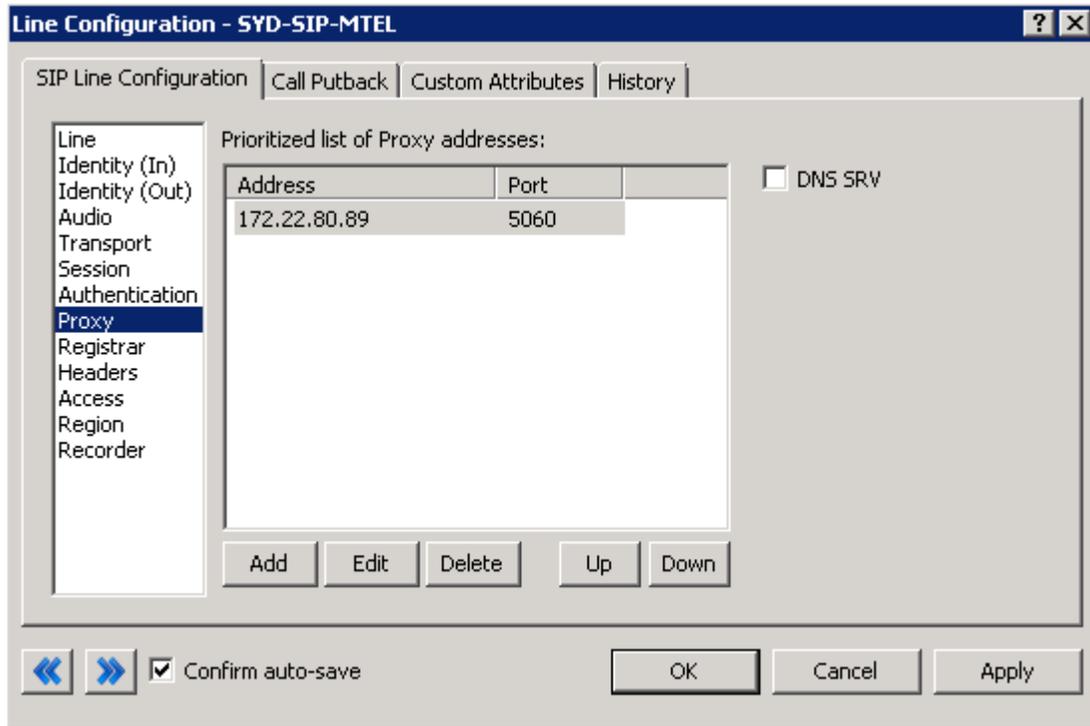This box is somewhat of a misnomer in the case of some SIP Carriers. In the case of Macquarie Telecom, there is not a single IP that is needed. Instead they provide a Fully Qualified Domain Name (FQDN) to a machine or cluster that handles the requests. When configuring the proxy for Macquarie Telecom, this **FQDN** must be entered completely with the port (generally 5060 unless otherwise directed) to enable the service to work properly. If a resolved IP address is entered the service may not work as advertised, if at all due to the random port selection of the carrier.

2.1.8.2   Remainder of Proxy Menu Options

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

2.1.9 Registrar Menu

No additional/custom headers have been configured/tested.
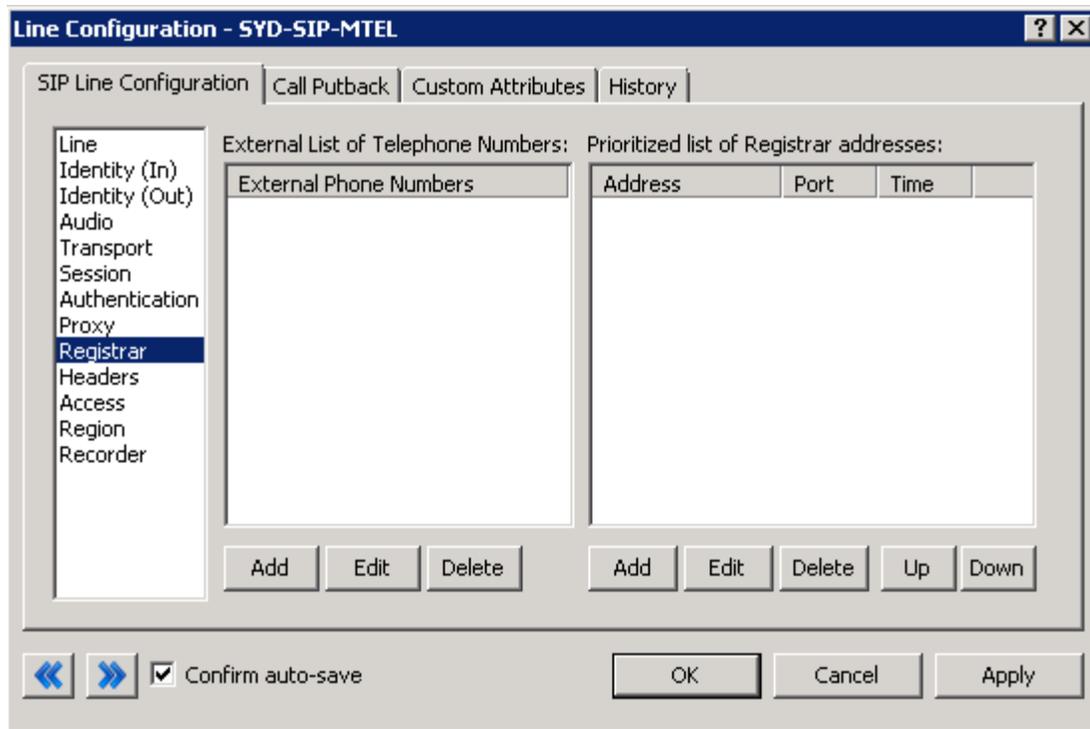
Registrar Menu Line Configuration Page

2.1.9.1 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.

2.1.9.2 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.

2.1.10 Access Menu (Access Control lists)

If business needs require endpoints (i.e. phones) use port 5060, Access Control lists are recommended. The 3.0 and higher versions of the IC server come with default station lines that are set to 8060. If using these default station lines for your endpoints, and not requiring multiple lines that are using the same protocol, and port, this section can be skipped. These lists are recommended if not using

the default station lines because separate lines allow better tracking of resource utilization.
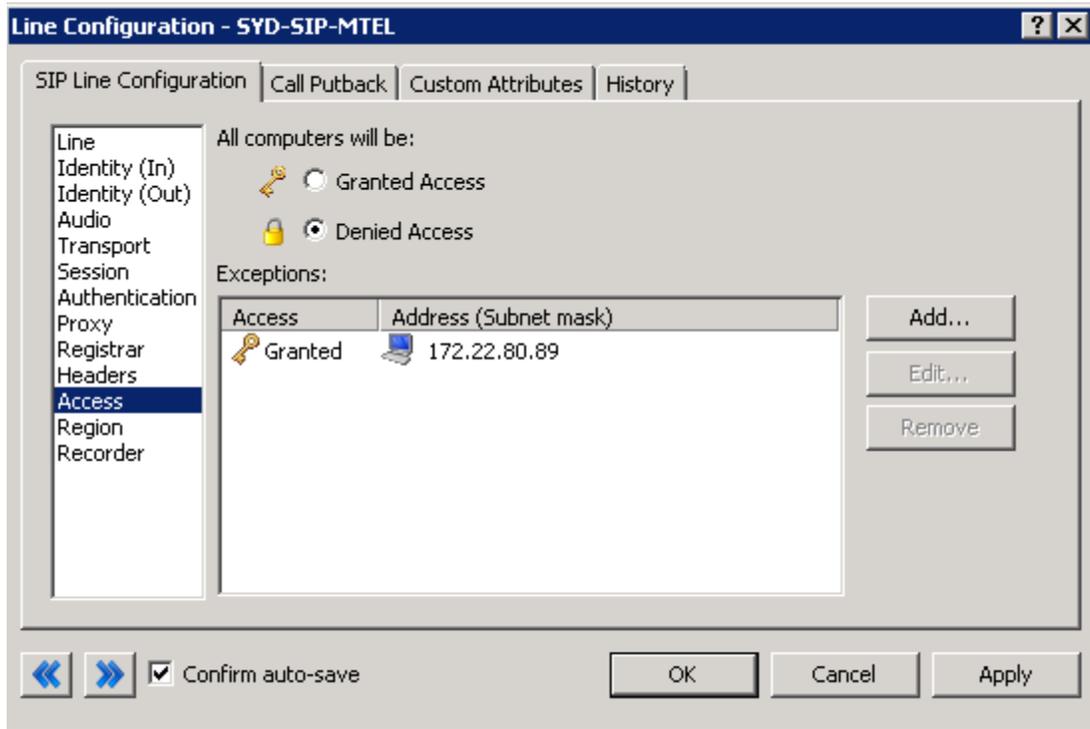
Access Menu Line Configuration

2.1.10.1 Macquarie Telecom 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. The "add menu" has a DNS lookup option if the only information provided by the carrier were FQDNs. This allows the IC server to talk to all the required elements of the SIP carrier.

2.1.10.2 Remainder of Access Menu Options

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

2.1.10.3 Region Menu
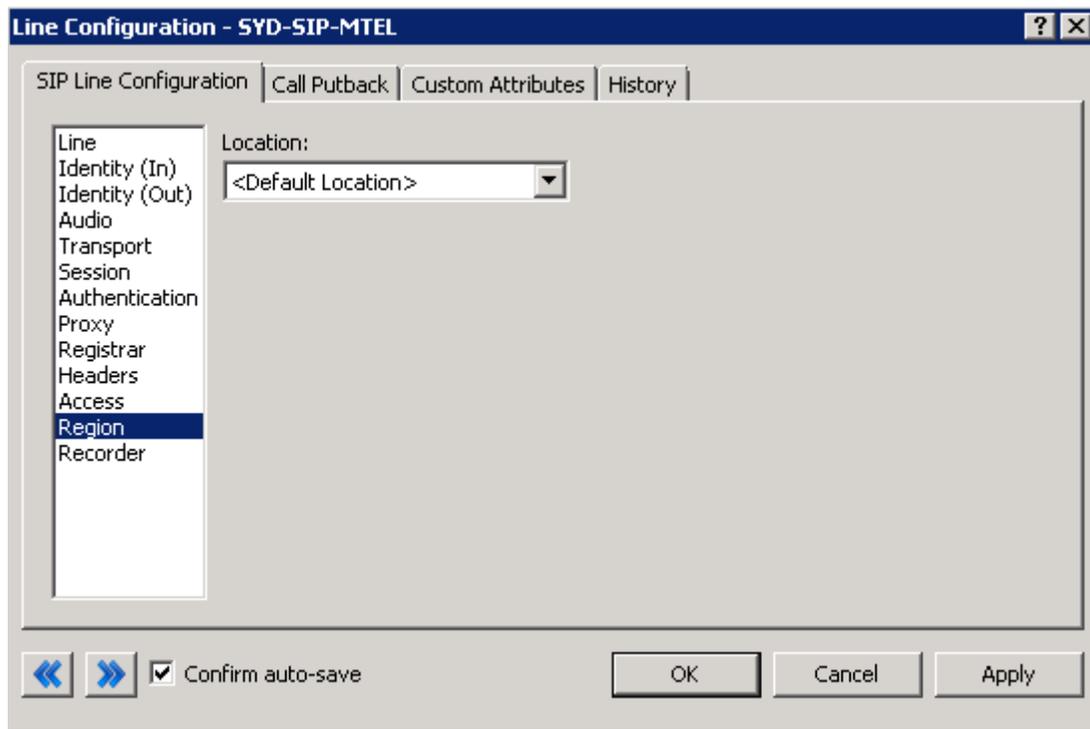
Region Menu Line Configuration

This should be set at the user discretion, however the user should take care to assure the location supports the proper codecs supported by the SIP Carrier.

3 SIP Proxy Support

Macquarie Telecom is a service provider, however the SBC's are not managed by Macquarie Telecom, therefore use of SIP proxy is possible. Depending on the SBC used, different options are possible.

For 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.

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 MACQUARIE TELECOM 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 Macquarie Telecom. 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

Macquarie Telecom supports 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)

Macquarie Telecom requires Privacy header. Eg. Privacy:user
The message manipulation rules were configured so that the header is added when the From header is "anonymous".
MT doesn't seem to accept RESTRICTED or other non-numbers in FROM header, hence another rule overwrites the user's ANI from "anonymous" to the pilot number, but since it doesn't get displayed, the call remains anonymous.
For Diverted calls, the Diversion header is checked if it's "Anonymous"

▼ Message Manipulations

| Add + | Insert + | Edit | Delete | Up ↑ | Down ↓ | | | |

Index	Manipulation Name	Manipulat Set ID	Message Type	Condition ⬍	Action Subject	Action Type	
0		3	Invite	header.from.url.user=='anonymous'	header.Privacy	Add	'user'
1		3	Invite	header.from.url.user=='anonymous'	header.from.url.user	Modify	'0862200301'
2		3	Invite	header.Diversion.name=='Anonymous'	header.Privacy	Add	'user'

**Note: the condition strings in the SBC manipulation rules are case-sensitive. "Anonymous" != "anonymous"