

## 1 SIP Carriers

### 1.1 AMCOM



#### 1.1.1 Warnings

>>> At this time, AMCOM has successfully completed interoperability testing with Interactive Intelligence Communications-as-a-Service (CaaS) on BroadSoft in Australia. Please consult with an AMCOM representative for service availability.

Check the *SIP 3<sup>rd</sup> Party SIP Carrier Matrix* for certification status, and supported features. More info about the *SIP 3<sup>rd</sup> 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 : [amcom.com.au](http://amcom.com.au)

Vendor Support Contact: <http://www.amcom.com.au/support-centre>

#### 1.1.3 Versions Verified

SIP Carrier status as of November 19, 2013: validated on CIC 4.0 SU3 (4.03.1598).

#### 1.1.4 PreInstall

AMCOM 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

AMCOM 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 AMCOM connection, and one for the stations. Each portion of the lines page will be explained as it relates to the AMCOM Service. For this document, the *AMCOM* connection line will be referred to as *AMCOM 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.

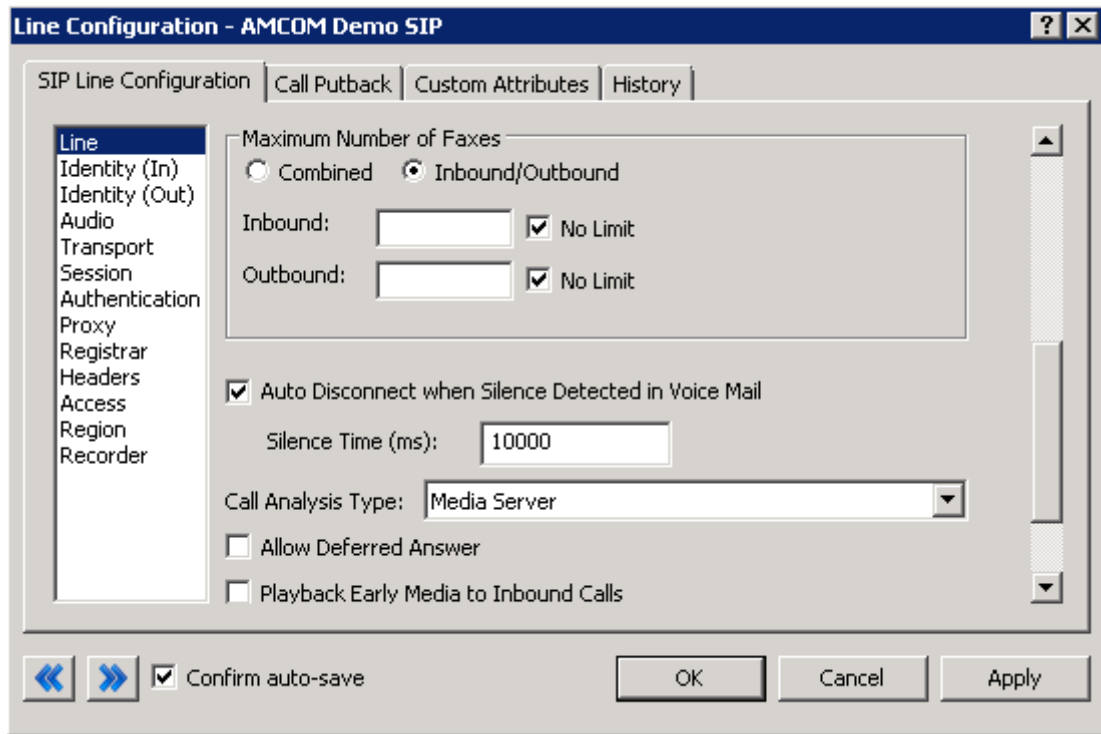
#### 2.1.1 Line Menu

The screenshot shows the 'Line Configuration - AMCOM Demo SIP' dialog box. The 'SIP Line Configuration' tab is selected. The left-hand menu is expanded to show the 'Line' option. The main configuration area includes the following settings:

- Active
- Line Usage: General Purpose
- Domain Name: amcomvoice.ipsystems.com.au
- Maximum Number of Calls:
  - Combined
  - Inbound/Outbound
  - Inbound: [ ]  No Limit
  - Outbound: [ ]  No Limit
- Enable T.38 Faxing
- Enable Fax Detection

At the bottom, there are navigation arrows, a checked 'Confirm auto-save' checkbox, and 'OK', 'Cancel', and 'Apply' buttons.

**Line Menu Line Configuration Page**



**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 AMCOM. It is used for the registration and/or authentication of the line.

### **2.1.1.4 Disable T.38 Faxing**

AMCOM's SIP Carrier service supports (but has not been tested in this process) the T.38 faxing protocol by default. Leave this box unchecked if you do not have

(or wish to use) an analog to SIP capable FXS type device to connect an analog fax machine to the system.

### **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 (Out) Menu

Line Configuration - AMCOM Demo SIP

SIP Line Configuration | Call Putback | Custom Attributes | History

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

Use 'sips!' scheme

Called Address: \_\_\_\_\_

Keep 'tel:' scheme when using a proxy

Send Extension:

Calling Address: \_\_\_\_\_

Line Value 1:  ...

Line Value 2:  ...

Diversion Method:

.. . . . .

Confirm auto-save

OK Cancel Apply

**Identity (Out) Menu Line Configuration Page**

### 2.1.2.1 Calling Address

Configure Line Value

Use Anonymous values

Name:

Address:

Display value:

OK Cancel

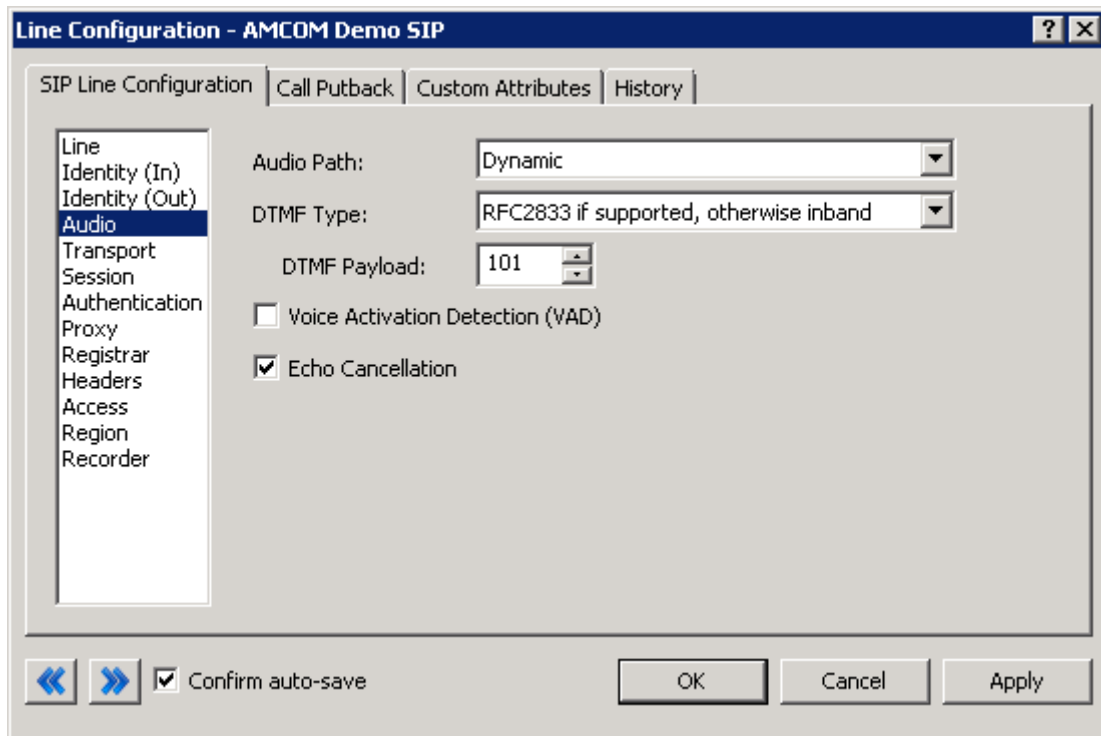
**Configure Line Value Dialog**

Click the "..." button next to the Line Value 1 box to bring up the Configure Line Value dialog. In the Address box, enter the Main DID to be used as the Default User portion on outgoing calls.

### 2.1.2.2 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.

## 2.1.3 Audio Menu



**Audio Menu Line Configuration Page**

### 2.1.3.1 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.

### 2.1.3.2 DTMF Type

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

### 2.1.3.3 Remainder of Audio Menu Options

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

## 2.1.4 Transport Menu

The screenshot shows a window titled "Line Configuration - AMCOM Demo SIP" with a tabbed interface. The "SIP Line Configuration" tab is active. On the left is a tree view with "Transport" selected. The main area contains the following fields:

Transport Protocol:	UDP		
Audio Protocol:	RTP	Security:	Minimal
Adapter Name:	VLAN152		
	Microsoft Virtual Machine Bus Network Adapter #5		
Receive Port:	5060	Connect Timer (ms):	2000
Maximum Packet Retry:	4	T-1 Timer (ms):	500
Maximum Invite Retry:	3	T2 Timer (ms):	1000
Reinvite Delay (ms):	50		
Retryable Reason Codes:	480, 500-599		

At the bottom, there are navigation arrows, a checked "Confirm auto-save" checkbox, and "OK", "Cancel", and "Apply" buttons.

**Transport Menu Line Configuration Page**

### 2.1.4.1 Transport Protocol

This option should be set to UDP. As of **<date certified>** AMCOM has support for UDP by default. TCP and TLS are not currently supported.

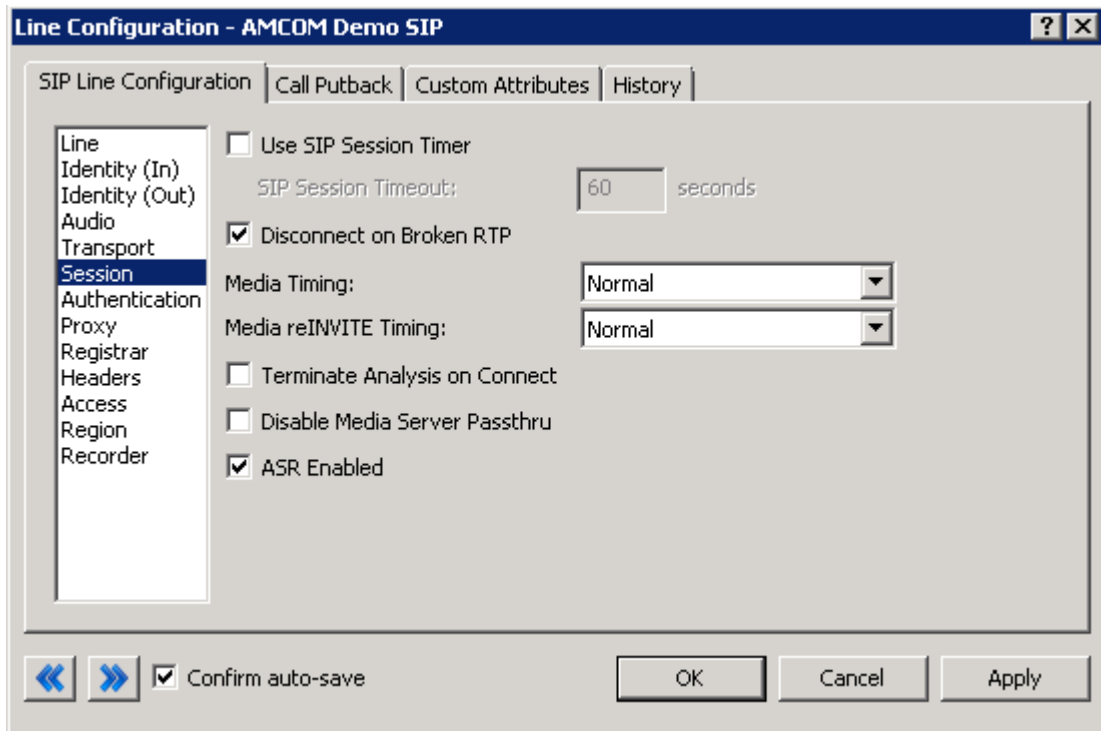
### 2.1.4.2 Receive Port

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 **<date certified>** AMCOM only has support for port 5060 in the standard offering.

### 2.1.4.3 Remainder of Transport Menu Options

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

## 2.1.5 Session Menu



**Session Menu Line Configuration Page**

### 2.1.5.1 Media Timing/Media reINVITE Timing

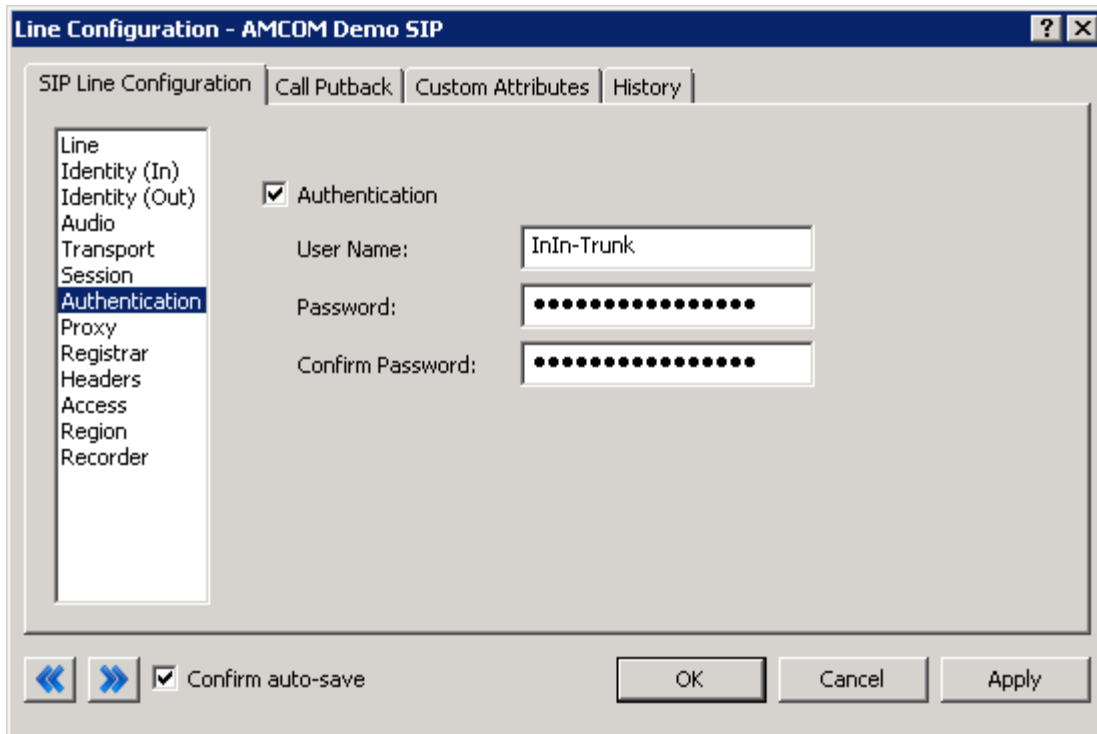
This dropdown pair controls Delayed Media support. Setting both to Normal is the recommend method by Interactive Intelligence for all SIP Carriers.

### 2.1.5.2 Remainder of Session Menu Options

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

## 2.1.6 Authentication Menu

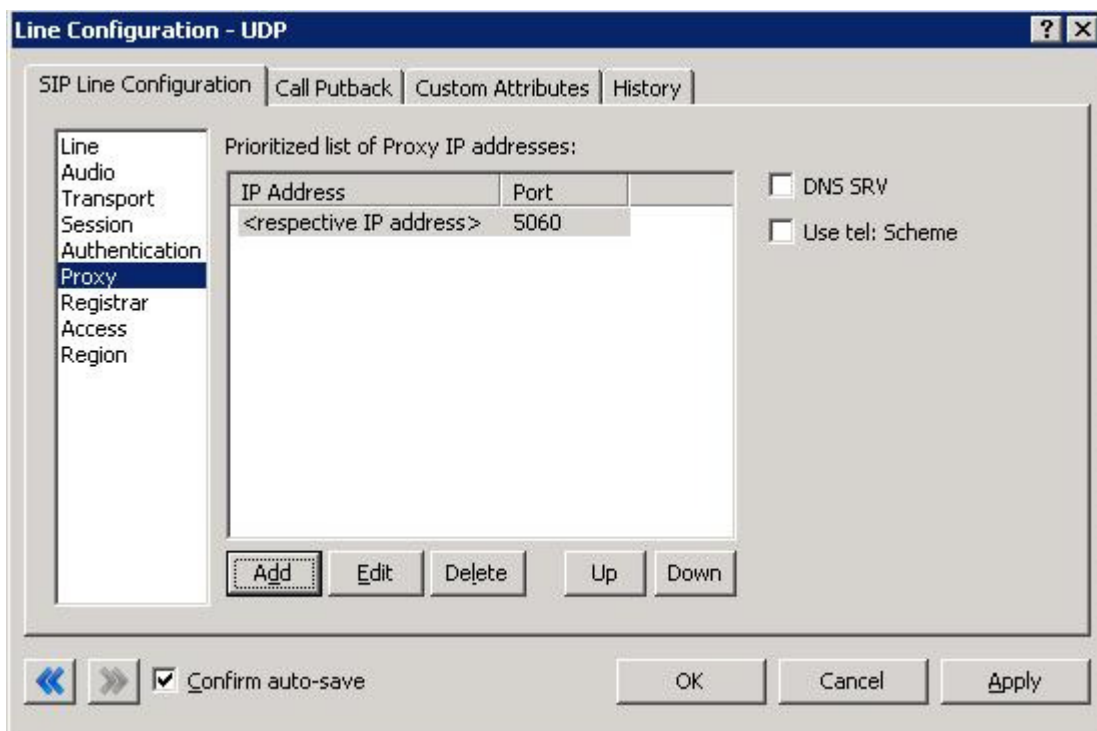




**Authentication Menu Line Configuration Page**

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



**Proxy Menu Line Configuration Page**

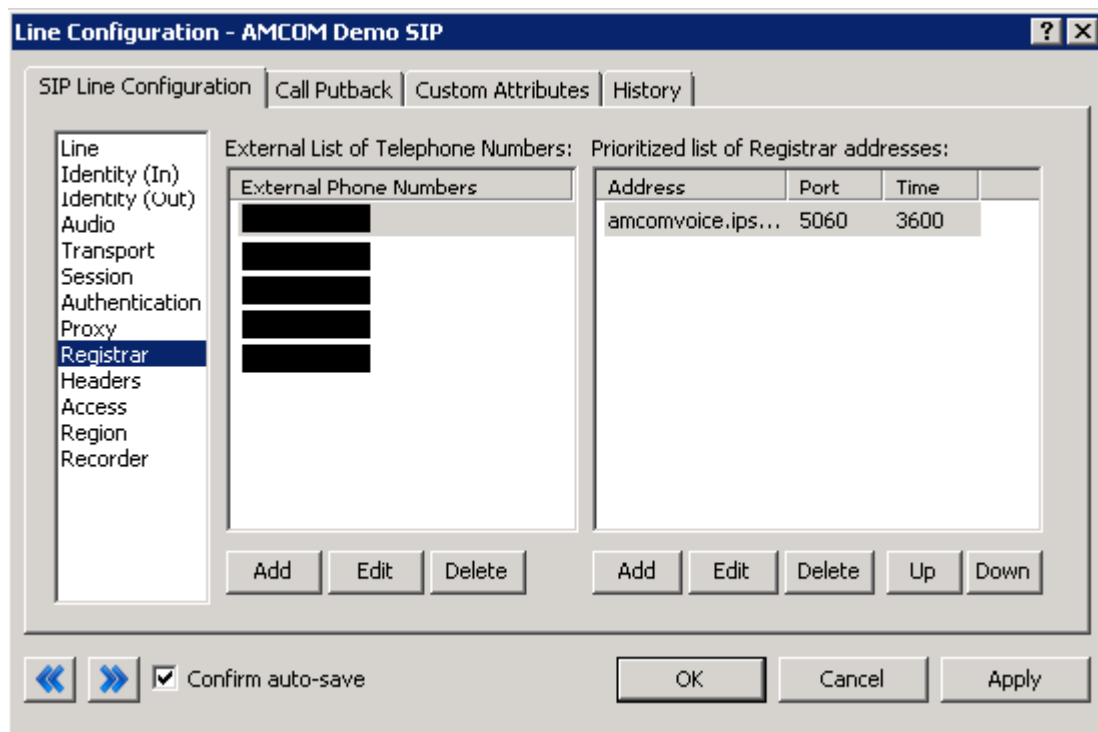
### 2.1.7.1 Prioritized list of Proxy IP addresses

This box is somewhat of a misnomer in the case of some SIP Carriers. In the case of AMCOM, 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 AMCOM, 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.

### 2.1.7.2 Remainder of Proxy Menu Options

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

### 2.1.8 Registrar Menu



**Registrar Menu Line Configuration Page**

#### 2.1.8.1 External Phone Numbers

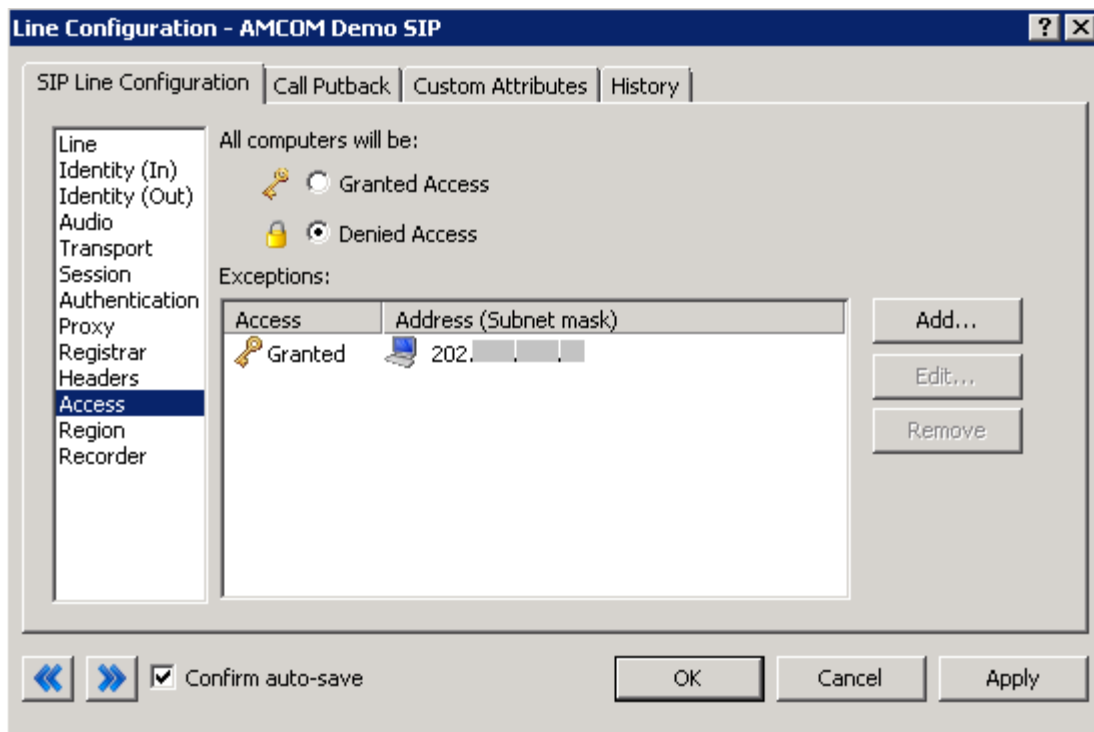
This box should have the respective group of phone numbers allocated to the customer from the SIP Carrier. If more than one number has been provided, then they should all be placed in this box to allow the IC server to register to all numbers with the SIP Carrier. This in turn will tell the SIP Carrier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

## 2.1.8.2 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.

## 2.1.9 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**

### 2.1.9.1 AMCOM 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.

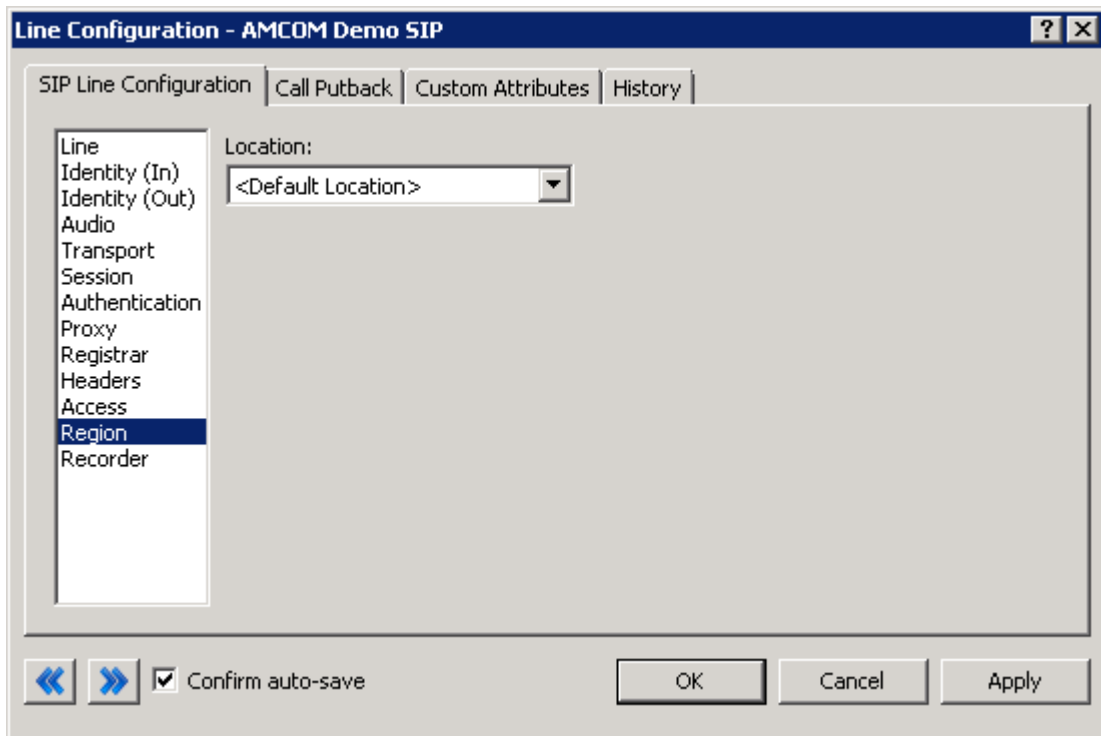
### 2.1.9.2 Exceptions

Each IP Address provided by AMCOM should be added to the list to grant them access to this Line. As of November 19<sup>th</sup> 2013, this list cannot be configured with FQDN or DNS SRV records.

### 2.1.9.3 Remainder of Access Menu Options

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

### 2.1.9.4 Region Menu



#### Access 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.

In the case of AMCOM, only G.711 (a-law), and G.729 are supported, so selecting a location that does not have any of these as an option would cause the line not to function properly. AMCOM does not have a particular business model preference for either codec, so this is up to the discretion and needs of the user.

### 3 SIP Proxy Support

For AMCOM, 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.

**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 AMCOM 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 AMCOM. 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**

AMCOM 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.