1 SIP Carriers

1.1 PAETEC

1.1.1 Warnings

>>> At this time, PAETEC has successfully completed interoperability testing with Interactive Intelligence on one (Broadsoft) of three geographically dispersed VoIP platforms. Depending on demand, there may be future testing on the other two. Please consult with a PAETEC representative for service availability.

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

http://www.paetec.com

1.1.3 Versions Verified

SIP Carrier status as of *October 23, 2009*

1.1.4 PreInstall

PAETEC 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

PAETEC 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.

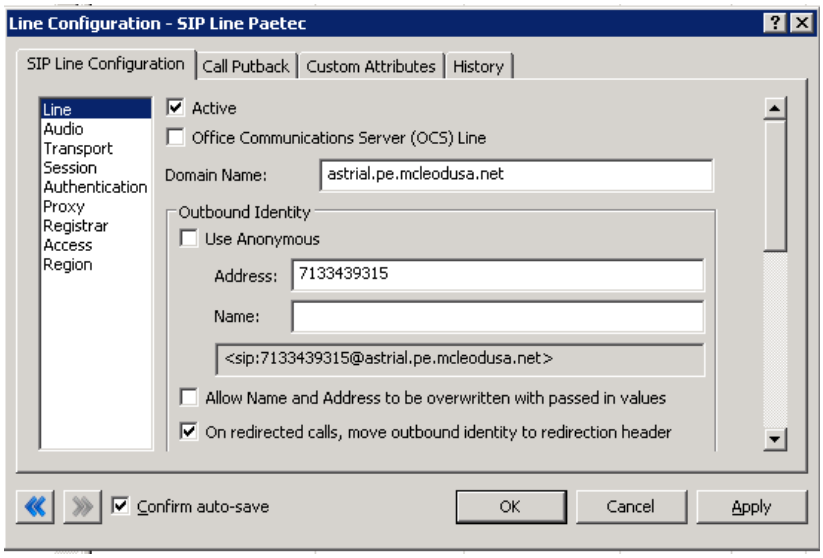
2 IC Configuration Guide

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

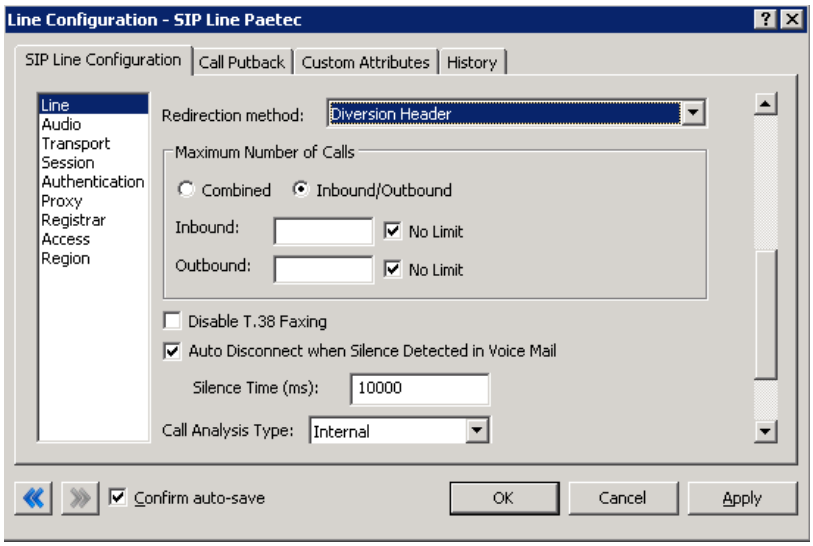


Figure 1: Line Menu Line Configuration Page

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 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 the Paetec SIP Carrier Service. It is used for the registration and/or authentication of the line.

2.1.1.4 Disable T.38 Faxing

At the time of this certification, PAETEC's SIP Carrier service does not support the T.38 faxing protocol. Check this box 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.

2.1.1.5 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 Audio Menu

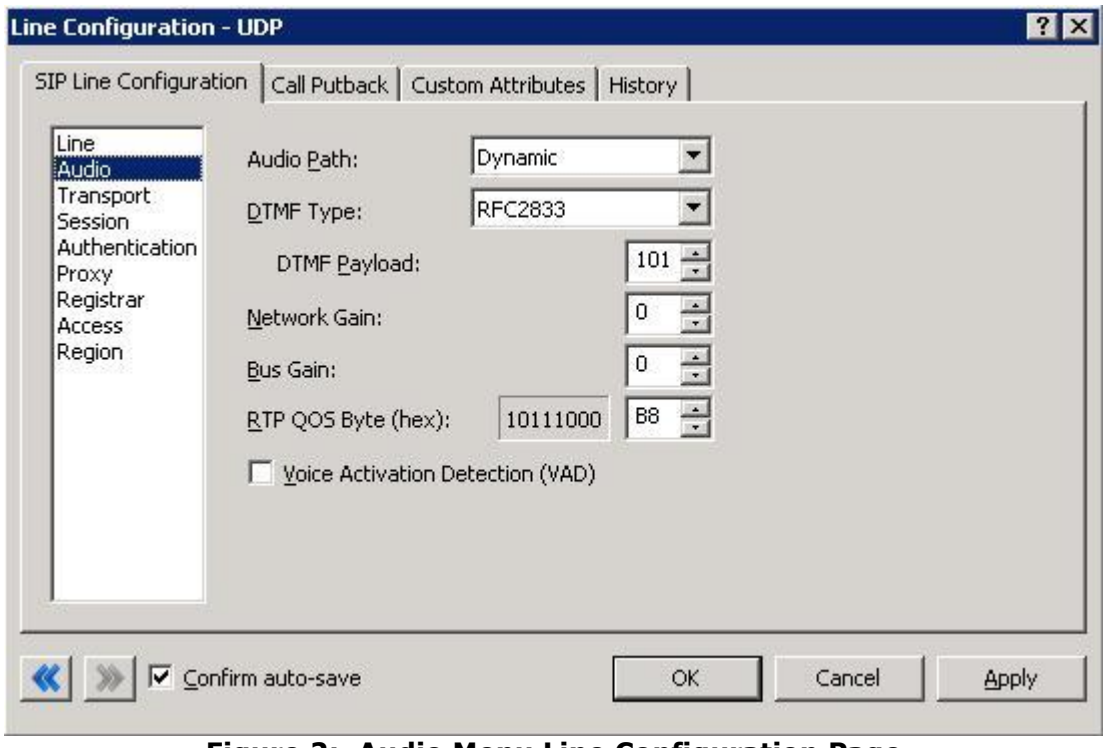


Figure 2: Audio Menu Line Configuration Page

2.1.2.1 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.

2.1.2.2 DTMF Type

DTMF has three options, *Inband*, *RFC2833*, and *RFC2833 Only*. These are up to the discretion of the user. All three are supported with the following caveats:

PAETEC requires the RFC2833 to be identified in the Invite message which requires Normal Media. To use Normal Media, the Disable Delayed Media checkbox needs to be selected (or Normal Media selected in the same location from the dropdown in a post-GA IC 3.0 server) from the session menu described later. Disabling Delayed media is the recommended method by Interactive Intelligence for all SIP Carriers.

RFC2833 – If using Delayed Media, the DTMF type will fall back to Inband.

RFC2833 Only – If using Delayed Media, the call will fail.

Inband - Delayed Media will have no effect on Inband DTMF

2.1.2.3 Voice Activation Detection (VAD)

This checkbox controls the Annex B option when using G.729. The IC server will *not* dynamically negotiate G.729 with annexb=yes. If Annex B is desired, this box must be checked, otherwise it will always use the annexb=no option. If it is required to have both another line can be set up with some differentiating factor one with Annex B enabled, and one without, then use the difference to select between the two. The reseller or an Interactive Intelligence support option can give more information on how this can be configured for the desired result.

2.1.2.4 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 Transport Menu

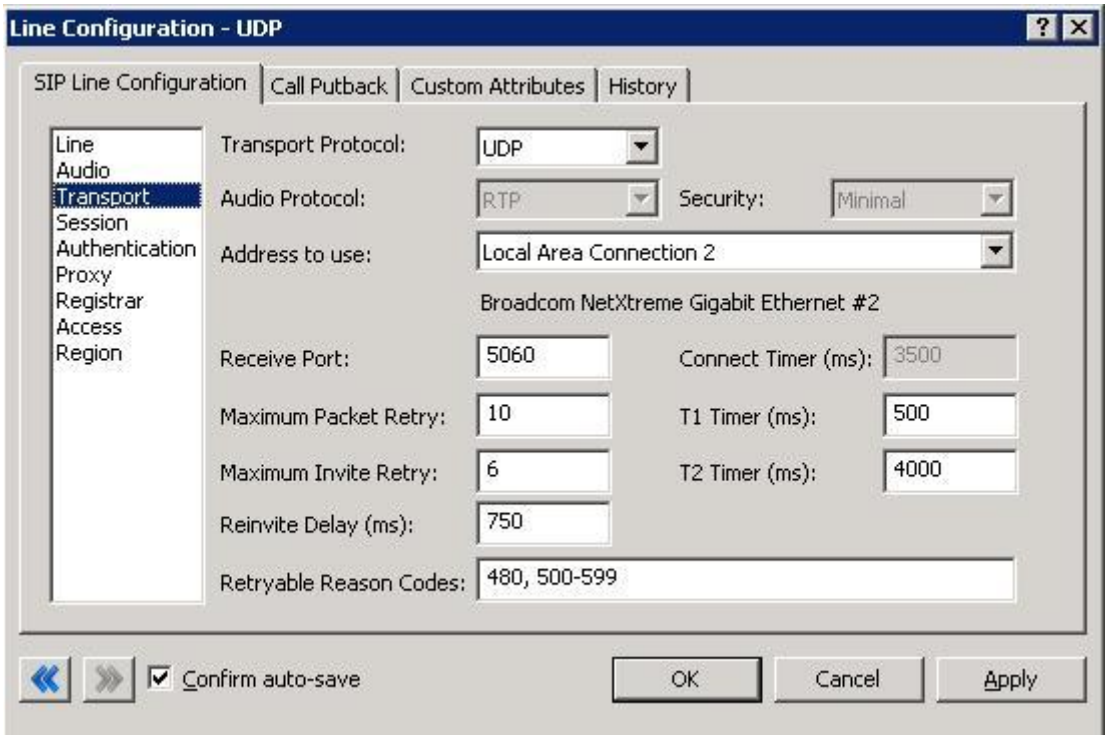


Figure 3: Transport Menu Line Configuration Page

2.1.3.1 Transport Protocol

This option should be set to UDP, unless an agreement for TCP or TLS support has been agreed upon with the SIP Carrier. As of October 23, 2009 PAETEC has support for UDP by default. TCP and TLS are not currently supported.

2.1.3.2 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 October 23, 2009 PAETEC only has support for port 5060 in the standard offering.

2.1.3.3 Remainder of Transport Menu Options

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

2.1.4 Session Menu

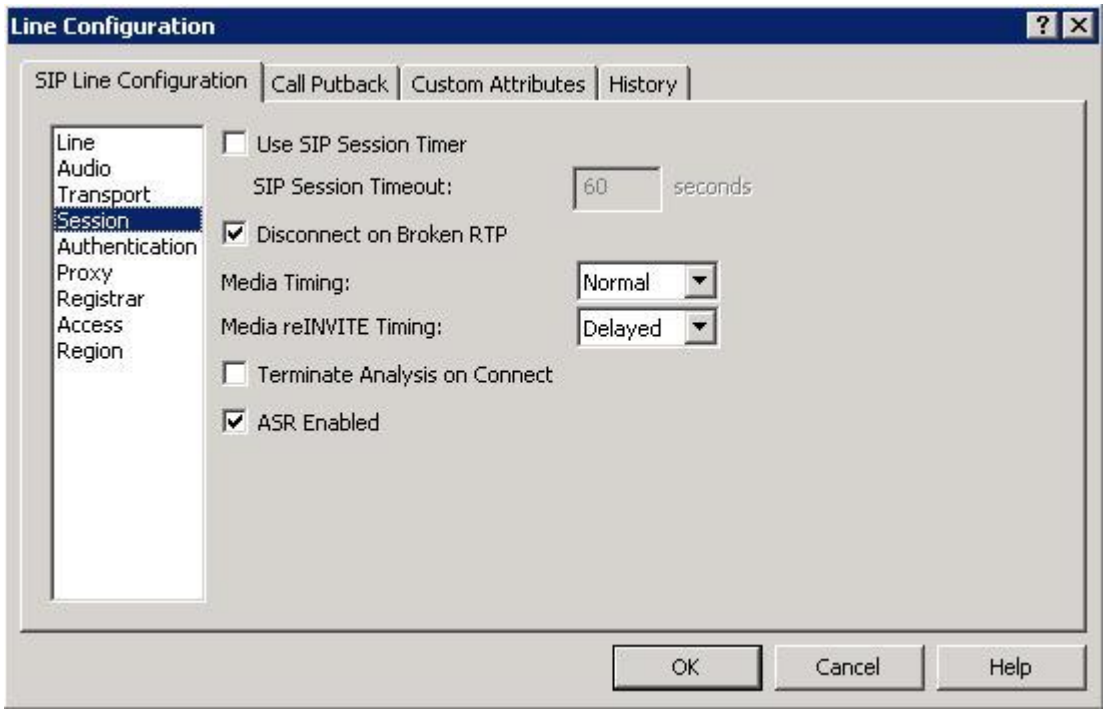


Figure 4: Session Menu Line Configuration Page

2.1.4.1 Media Timing/Media reINVITE Timing

This dropdown pair controls Delayed Media support. If delayed media is not supported by SIP Carrier they will both have to be set to *Normal* for RFC2833 DTMF tones to work, as stated above (2.1.2.2 DTMF Type). Setting both to *Normal* is the recommend method by Interactive Intelligence for all SIP Carriers, and is *required* for the PAETEC service to function properly if delayed media is not supported.

2.1.4.2 Remainder of Session Menu Options

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

2.1.5 Authentication Menu

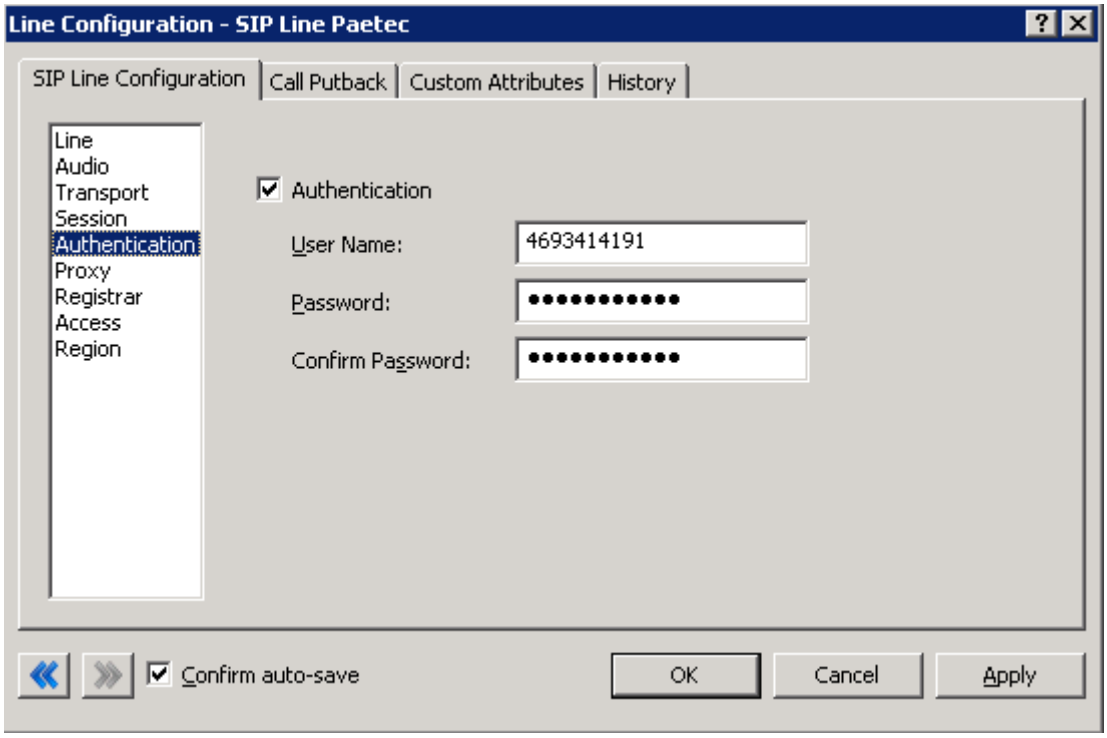


Figure 5: Authentication Menu Line Configuration Page

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.

2.1.6 Proxy Menu

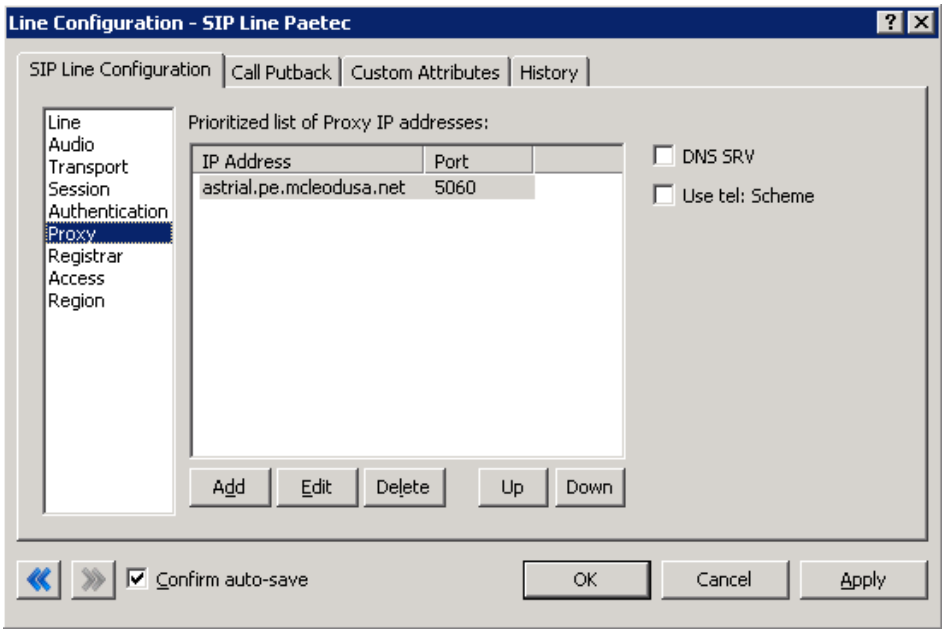


Figure 6: Proxy Menu Line Configuration Page

2.1.6.1 Prioritized list of Proxy IP addresses

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

*DNS SRV is a common method that SIP Carriers use to create a cluster of proxy IP addresses. This does not require checking the DNS SRV checkbox however, due to providing the FQDN as the proxy address and the SIP Carrier handling the resolution.

**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.

2.1.6.2 Remainder of Proxy Menu Options

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

2.1.7 Registrar Menu

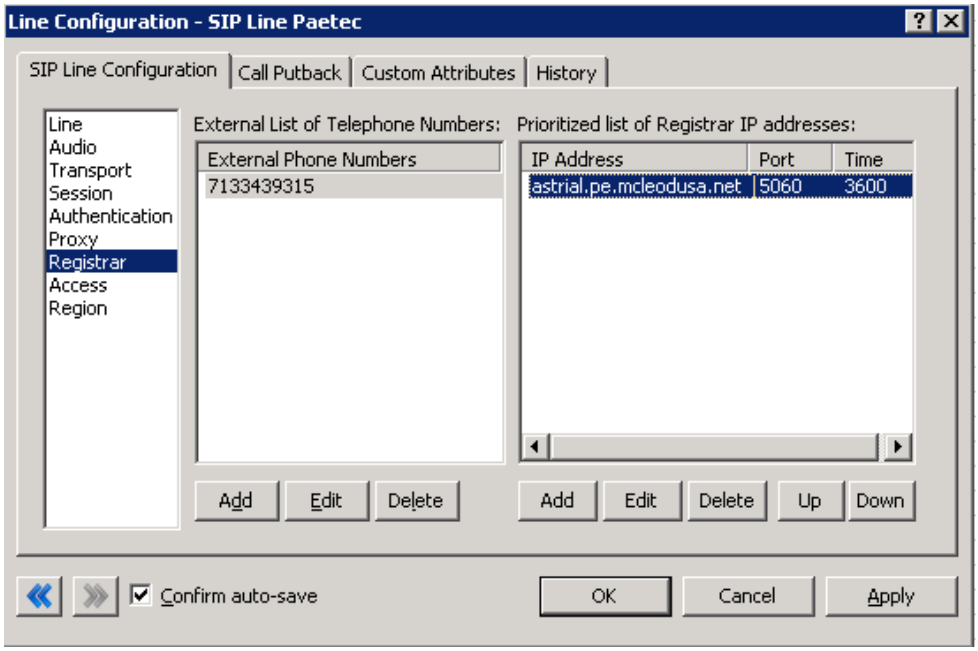


Figure 7: Registrar Menu Line Configuration Page

2.1.7.1 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.

2.1.7.2 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.

2.1.8 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.

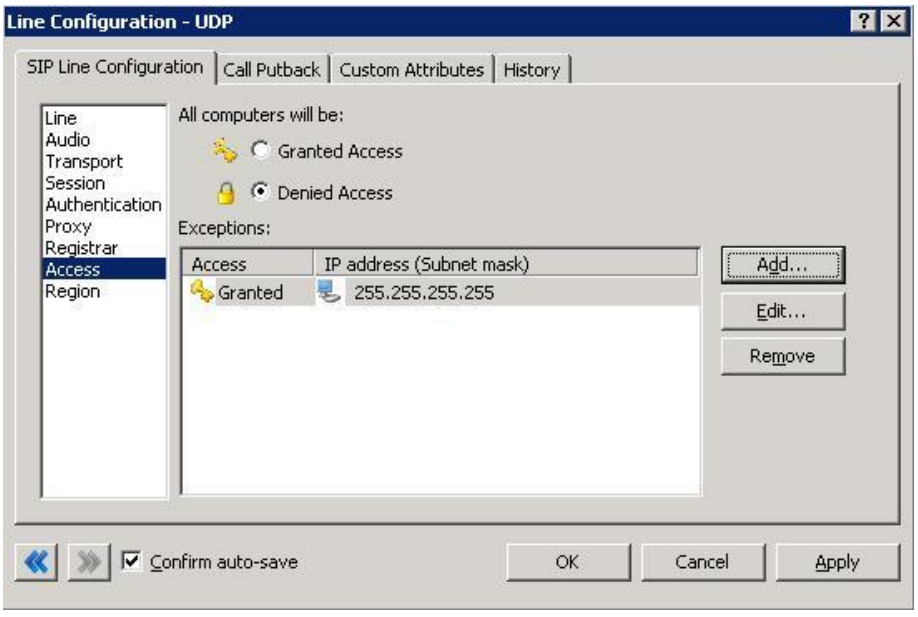


Figure 8: Access Menu Line Configuration Page (note the 255.255.255.255 address is a sample, and the actual number should be respective to customer needs)

2.1.8.1 PAETEC 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.

2.1.8.2 Stations Line

In the case of the stations line, this is up to the discretion of the user. It is possible to enter in single IP's, IP groups (using subnet masks), or allow everything. The user has several options based on business needs and security requirements. However note that only one line can be selected to "*Granted Access"* per port per IC server.

The reason why the SIP Carrier Line was selected to be ***Denied Access*** was because it has far fewer and less complicated entries than the line that will be supporting all the local endpoints.

2.1.8.3 Region Menu

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 PAETEC only G.711 (mu-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. PAETEC 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 PAETEC 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 <*service name*> 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 PAETEC 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

PAETEC does not 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.

Note: In the case of PAETEC it is not possible to use G.729 to do the pass-through faxing. Due to the compression used by the codec, and the sensitivity of fax communications, it is not recommended and not tested by Interactive Intelligence.

Note: Interactive Intelligence does not support T.38 SG3 faxing at the time this document was created. It does however, support G3 faxing, and a vast majority of fax SG3 machines will revert to G3 in the negotiation process.

4.1 AudioCodes Media Pack Configuration

Aside from the standard configuration options that must be entered for general SIP to analog usage (e.g. proxy name, IP address, etc…) two additional features must be set to enable the Media Pack to properly pass the fax.

One is the *Fax Signaling Method*. This must be set to *G.711 Transport*, and can be found by selecting the following links from the main page of the Media Pack configuration.

- Protocol Management
    - Protocol Definition
        - General

The other required configuration setting is Fax*/Modem Bypass Coder Type*, which must be set to *G711Mulaw*. This configuration option can be found by selecting the following links from the main page of the Media Pack configuration.

- Advanced Configuration
    - Media Settings
        - Fax/Modem/CID Settings

4.2 Cisco Gateway FXS Card Configuration

To configure the Cisco Gateway FXS Card to use G.711 pass-through for an analog fax machine, the following information must be entered. The information in parenthesis at the end of the lines is not to be typed in, but provides additional information regarding the line to aid in configuration for various environments.

Also, this information must be entered under the configuration level of IOS (i.e. enable access, then configure access).

For Outbound Faxing:

dial-peer voice X voip (the X is to be respective to the given gateway)

Under the above created dial-peer, the following options must be entered.

service session

destination-pattern .T

session protocol sipv2

session target ipv4:x.x.x.x (use the IP of the IC server in place of x's)

incoming called-number pattern .T

dtmf-relay rtp-nte (This is for RFC2833 support)

codec g711ulaw

fax rate 14400

For Inbound Faxing:

dial-peer voice X pots (POTS Dial peer)

service session

destination-pattern 7777 (IC station extension)

port 0/1/1 (FXS port number)

forward-digits 0

5 E911 Support

PAETEC currently supports E911 support via giving registered numbers of customers directly to the local E911 authority. This does not allow for dynamic updates. This is fairly standard, however those using a large number of remote clients should be aware and take the proper measures to ensure proper coverage. If a purely remote number is requested, PAETEC will make this known and may ask for an alternate solution or a waiver option.