

SIP Carrier

Allstream



1.1 Introduction

This document is a guide to install and configure the Allstream SIP carrier with the Interactive Intelligence Interaction Center Server. The document refers to IC version 3.0 and HMP SU176, configuration with previous versions of IC will remain similar.

1.2 Vendor contact

<http://www.allstream.com/contact/support.html>

1.3 Pre installation

The Interactive Intelligence Interaction Center must be up and running with HMP installation without any errors.

Carrier information: Allstream will require the static IP address of the IC server and they will provide a static IP address which will be the proxy to be configured in the lines. (Configuration explained below in section 2)

Two SIP lines must exist on the IC server; one for station connection calls and the other for interoperability to Allstream. The configuration for these lines will be covered in section 2 below.

Installation

IC Server Configuration

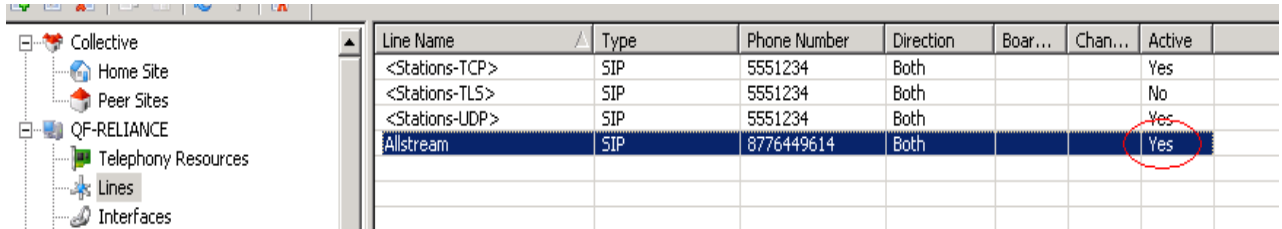
2.1 Line Configuration

The line page has a vast majority of the configuration options required for SIP Carrier setup. As stated before, two lines must exist on the IC server. In IC version 3.0 a Station connection line is created during installation and must be added to a registration group if using managed phones. The second line must be created for the SIP carrier connection. Each portion of the lines page will be explained as it relates to the SIP carriers. For this document, the SIP carrier connection line will be referred to as Allstream SIP Line, and the station line will be referred to as stations-TCP. 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.

As a note, Allstream only supports UDP at this time (3/10/08). This does not prevent TCP or TLS connections to the stations; IC will make the signaling conversion dynamically. The configuration below is for Allstream SIP line.

2.1.2 Active

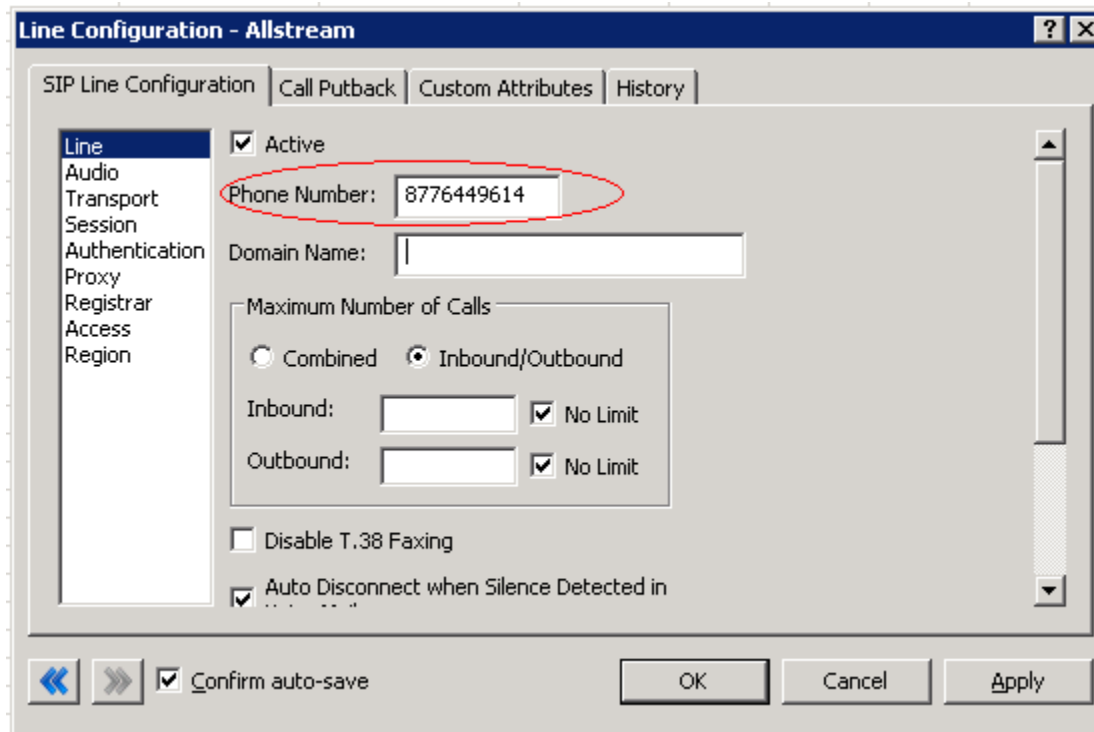
The active box should be checked. This activates the line. If this box is not checked, the line will not be available for any function. This can also be affected by right clicking on the line in Interaction Administrator, dropping to the Set Active menu option, and selecting yes.



Line Name	Type	Phone Number	Direction	Boar...	Chan...	Active
<Stations-TCP>	SIP	5551234	Both			Yes
<Stations-TLS>	SIP	5551234	Both			No
<Stations-UDP>	SIP	5551234	Both			Yes
Allstream	SIP	8776449614	Both			Yes

2.1.3 Phone Number

The phone number provided by Allstream should be entered into this box. The number entered is used in the "From" header in outbound SIP calls



Line Configuration - Allstream

SIP Line Configuration | Call Putback | Custom Attributes | History

Line Active

Audio

Transport Phone Number: 8776449614

Session

Authentication Domain Name: |

Proxy

Registrar

Access

Region

Maximum Number of Calls

Combined Inbound/Outbound

Inbound: | No Limit

Outbound: | No Limit

Disable T.38 Faxing

Auto Disconnect when Silence Detected in

Confirm auto-save OK Cancel Apply

2.1.4 Domain Name

This box should contain the domain information of the SIP Carrier, not the IC server. This is appended to the URL in registration requests.

The screenshot shows a window titled "Line Configuration - Allstream". It has several tabs: "SIP Line Configuration", "Call Putback", "Custom Attributes", and "History". The "SIP Line Configuration" tab is active. On the left, there is a tree view with "Line" selected. The main area contains the following settings:

- Active
- Phone Number: 8776449614
- Domain Name: AllstreamDomain.com (circled in red)
- Maximum Number of Calls:
 - Combined
 - Inbound/Outbound
 - Inbound: [] No Limit
 - Outbound: [] No Limit
- Disable T.38 Faxing
- Auto Disconnect when Silence Detected in []

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

2.1.5 Remainder of Line Menu Options

These have no major direct impact on the SIP carrier configuration, and should be addressed according to business needs. The disable T.38 fax option will prevent both T.38 inbound and outbound faxing from working properly. Allstream supports only G.711 codec as of now (3/10/08)

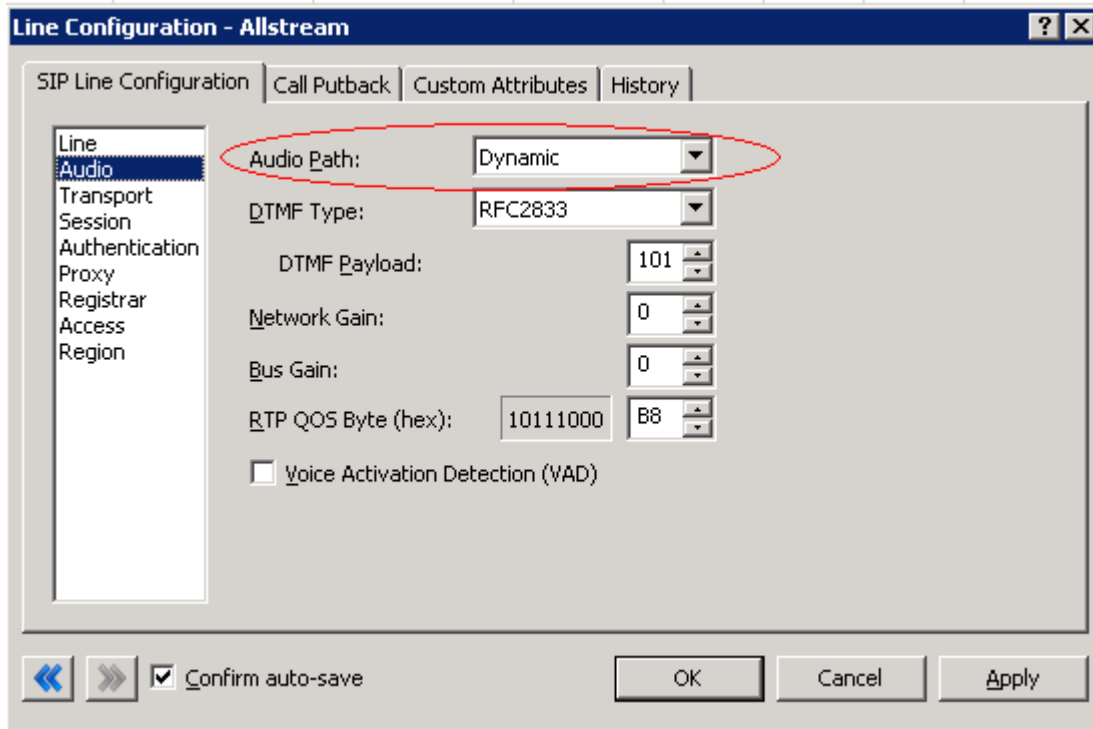
They do not support G.729 and T.38 for now (3/10/08). These codecs are in their testing phase and will be support them when they release. Interactive Intelligence has not tested G.729 and T.38 for now (3/10/08).

Audio Menu

2.1.6 Audio Path

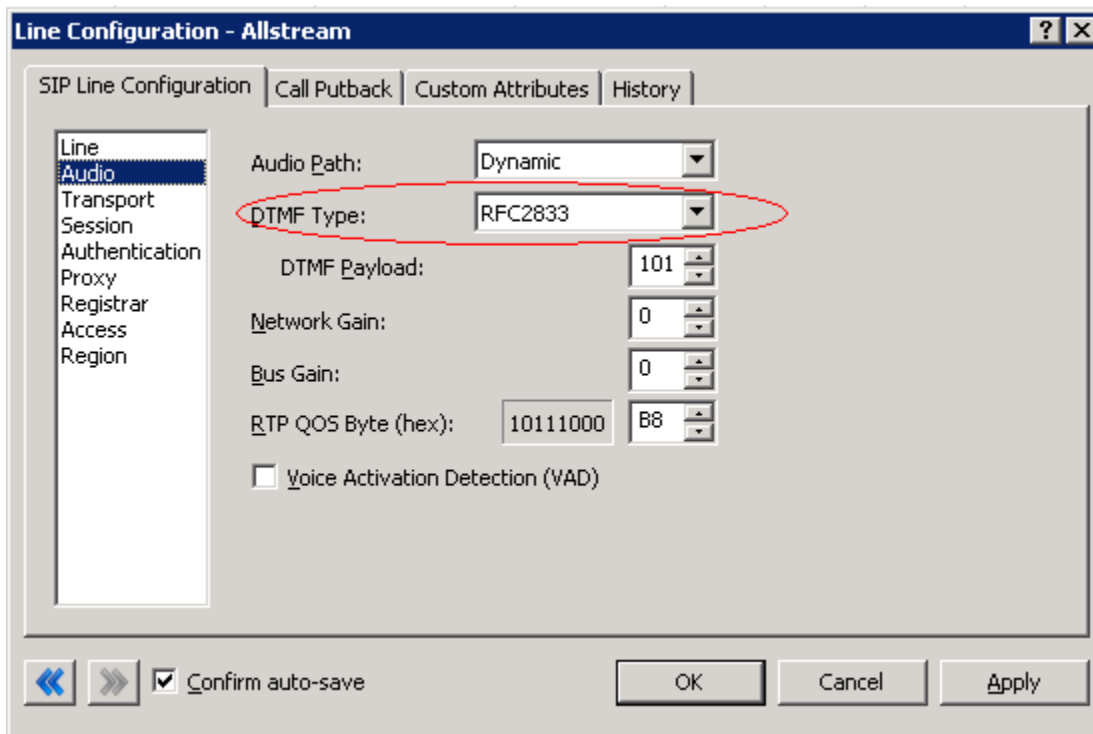
This is for the most part, the choice of the client with respect to the business being done on the server. However, there are several important caveats.

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.



2.1.7 DTMF Type

DTMF has three options, Inband, RFC2833, and RFC2833 Only. These are up to the discretion of the user.



2.1.8 Remainder of Audio Menu Options

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

2.1.9 Transport Menu

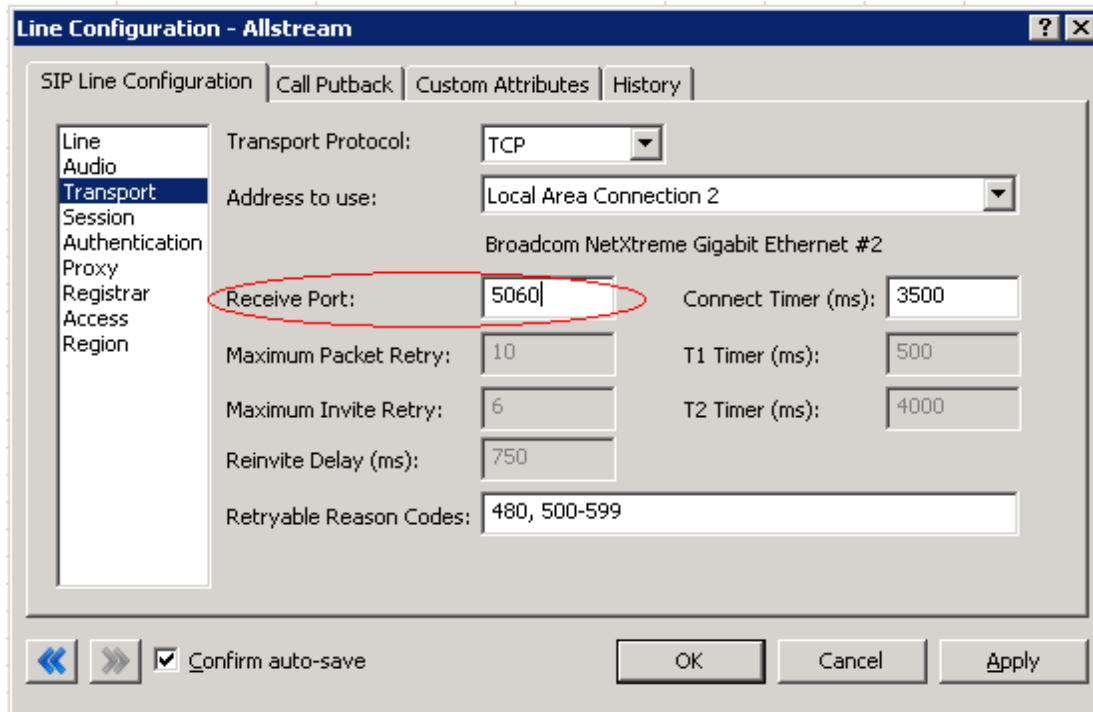
2.1.9.1 Transport Protocol

This option should be set to UDP, unless an agreement for TCP or TLS support has been agreed upon with Allstream. As of March 10, 2008, Allstream has support for UDP only.

The screenshot shows the 'Line Configuration - Allstream' dialog box with the 'Transport' tab selected. The 'Transport Protocol' dropdown menu is highlighted with a red circle and set to 'TCP'. Other settings include 'Address to use: Local Area Connection 2', 'Receive Port: 5060', 'Maximum Packet Retry: 10', 'Maximum Invite Retry: 6', 'Reinvite Delay (ms): 750', 'Retryable Reason Codes: 480, 500-599', 'Connect Timer (ms): 3500', 'T1 Timer (ms): 500', and 'T2 Timer (ms): 4000'.

2.1.9.2 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 Allstream.



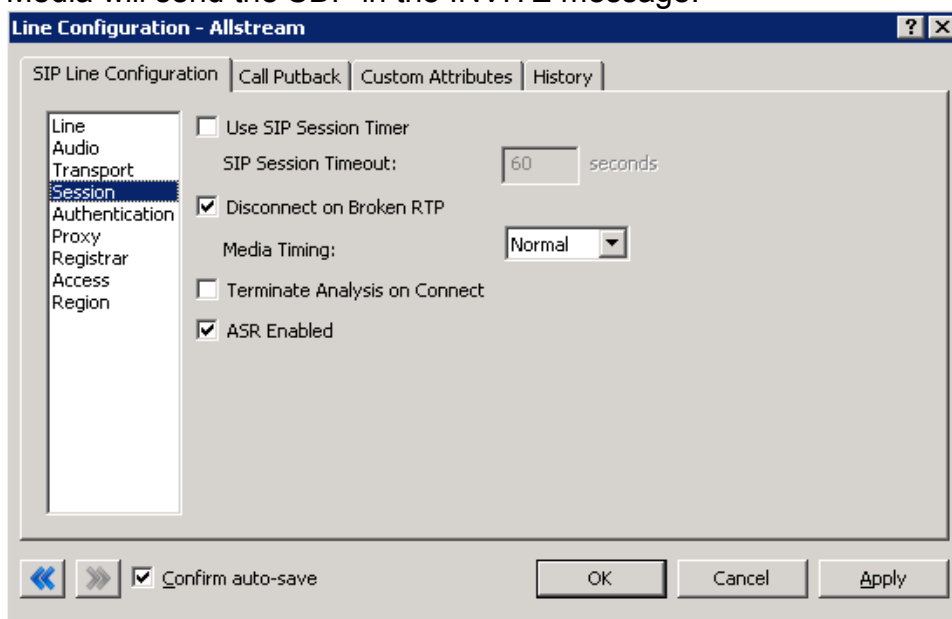
2.1.9.3 Remainder of Transport Menu Options

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

2.1.10 Session Menu

2.1.10.1 Media Type

This drop down (IC version 3.0) controls who offers the SDP information first on outbound calls; Delayed Media will let Allstream determine the codec and Normal Media will send the SDP in the INVITE message.



2.1.10.2 Remainder of Session Menu Options

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

2.1.11 Authentication Menu

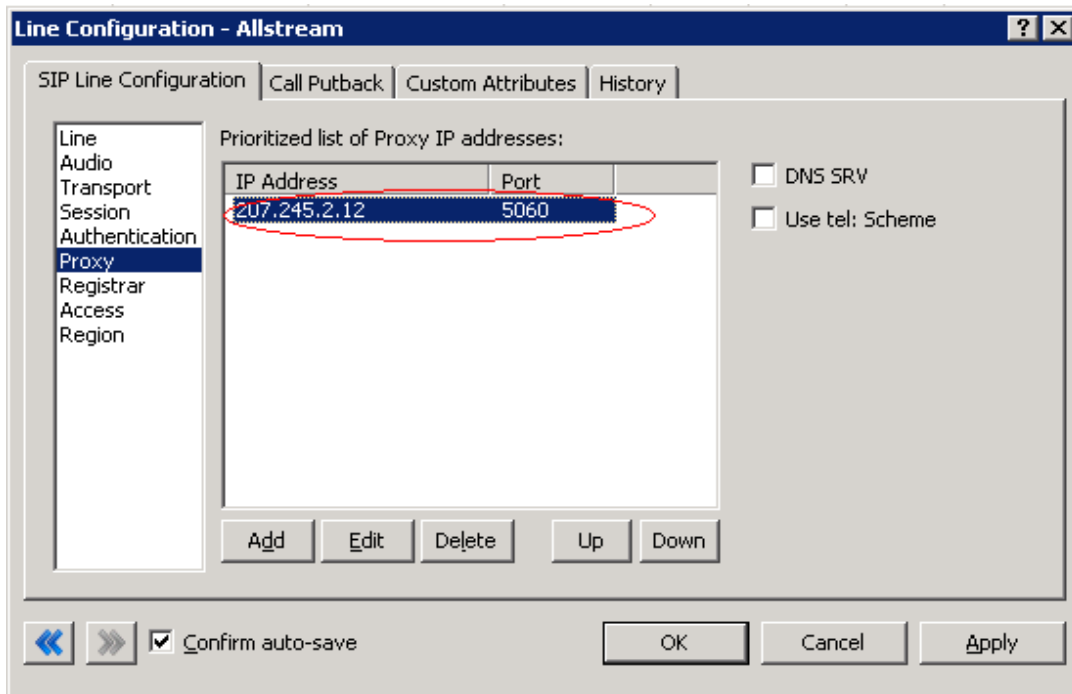
This box must be checked to enable authentication to the SIP Carrier. At the moment, Allstream uses a static IP model with no authentication, so nothing should be done with this page. Allstream does support digest authentication but currently in their environment they don't use it.

The screenshot shows a window titled "Line Configuration - Allstream". It has four tabs: "SIP Line Configuration", "Call Putback", "Custom Attributes", and "History". The "SIP Line Configuration" tab is active. On the left, there is a vertical list of menu items: "Line", "Audio", "Transport", "Session", "Authentication" (highlighted in blue), "Proxy", "Registrar", "Access", and "Region". In the main area, there is a checkbox labeled "Authentication" which is currently unchecked. Below this are three text input fields: "User Name:", "Password:", and "Confirm Password:". At the bottom of the window, there are navigation arrows (left and right), a checked checkbox labeled "Confirm auto-save", and three buttons: "OK", "Cancel", and "Apply".

2.1.12 Proxy Menu

2.1.12.1 Prioritized list of Proxy IP addresses

Allstream, by default will supply the customer with a single IP address for which all signaling must be directed too. A second backup destination proxy address can be requested to support redundancy. When configuring the proxy for Allstream, this IP Address should be entered completely with the port (generally 5060 unless otherwise directed) to enable the service to work properly.



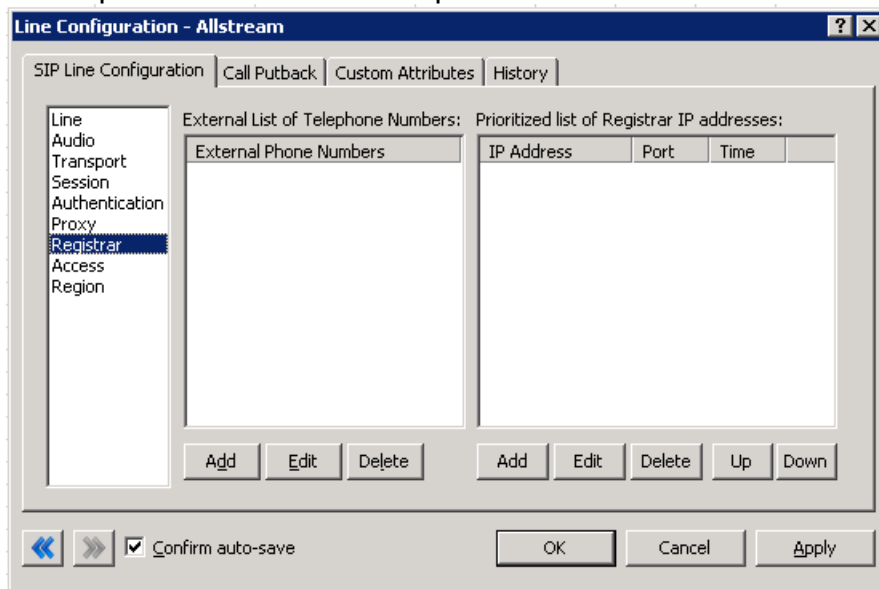
2.1.12.2 Remainder of Proxy Menu Options

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

2.1.13 Registrar Menu

2.1.13.1 External Phone Numbers

This box is not currently used by Allstream's configuration. In most cases it would be used to register multiple numbers to the same IC server. However as Allstream uses a static IP method, they do the registration/routing setup on their end and do not require the IC server to request the various numbers itself.



2.1.13.2 Prioritized list of Registrar IP addresses

This box is not used in Allstream's current configuration. The current system of providing static IP or FQDN makes registration messages unnecessary.

2.1.14 Access Menu (Access Control lists)

If your business needs require that your endpoints (i.e. phones) use port 5060, Access Control lists are recommended. These lists are recommended because separate lines allow better tracking of resource utilization. The default ports of the station connection lines (stations-TCP) are as follows:

UDP = 8060

TCP = 8060

TLS = 8061

If these defaults are used access lists are unnecessary and the following two sections can be skipped (2.1.14.1 and 2.1.14.2).

2.1.14.1 Allstream 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 FQDN's. This allows the IC server to talk to all the required elements of the SIP carrier.

2.1.14.2 Stations Line

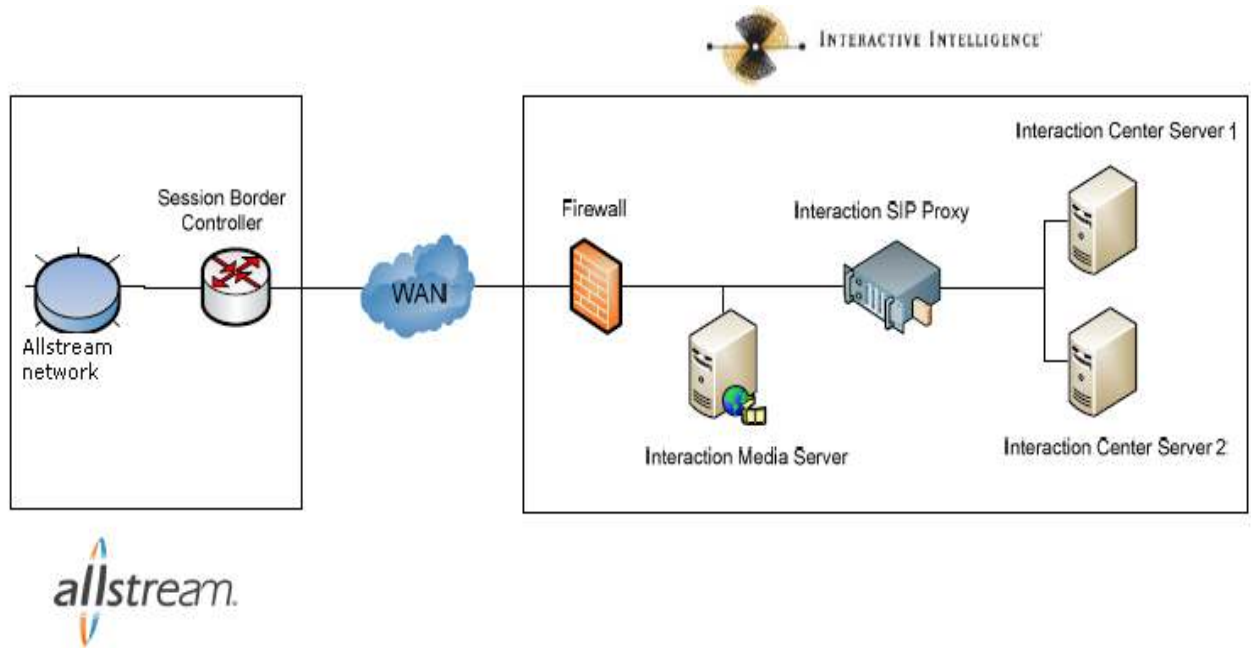
In the case of the stations line, this is up to the discretion of the user. It is possible to enter in single IP's, IP groups (using subnet masks), or allow everything. The user has several options based on business needs and security requirements. However note that only one line can be selected to "Granted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

2.2 Regionalization Container

This should be set at the user discretion; however the user should take care to assure the location supports the proper codec's supported by the Allstream. At this time only G.711 can be offered by Allstream.

3 SIP Proxy Support

Note: If using a NAT/PAT type solution with Allstream, a SIP Proxy can be used to provide redundancy and/or load balancing. The tested configuration consisted of:



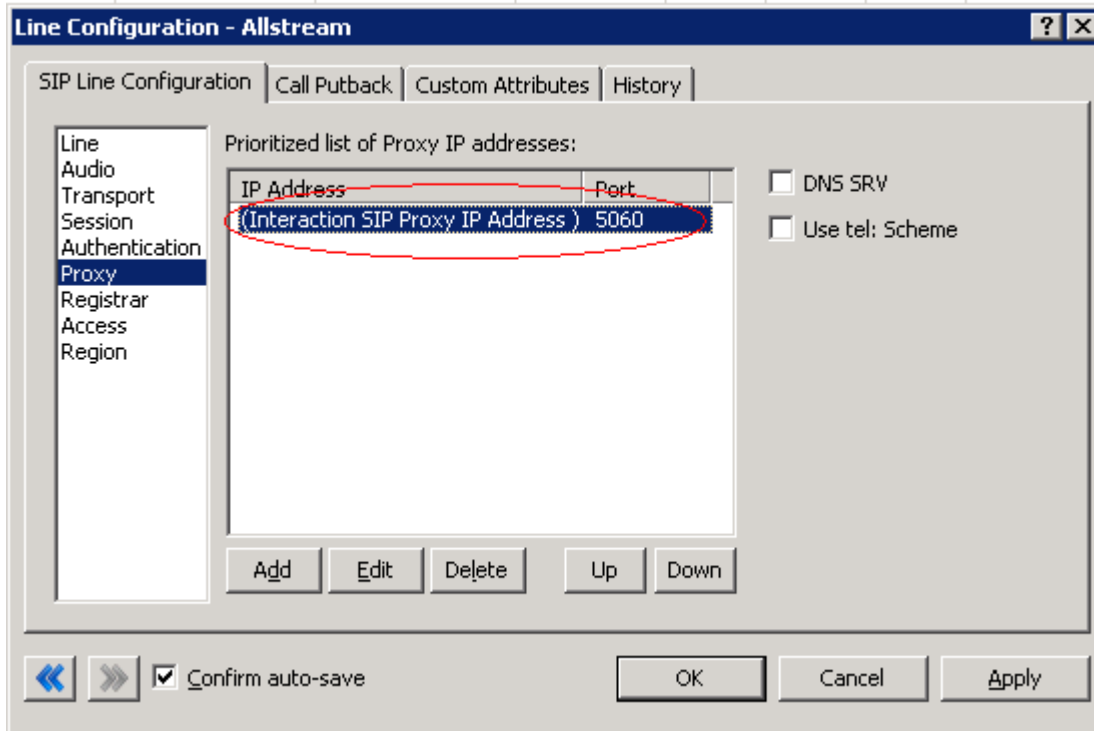
The external address of the firewall is the IP address given to Allstream to verify the origination of the call, the customer, and services being supplied. As a reminder, Allstream uses a static IP address to identify an end user and will deny the call if this IP address does not match.

Proxy for Inbound calls only:

If using an Interaction SIP Proxy as outlined in the diagram above for inbound calls, the primary and backup IC servers will be added to a Server Plan and a route will need to be created to link the incoming DID(s) from Allstream to the Server Plan. The proxy defined in IC for the Allstream SIP Line should remain pointed at Allstream. If using a PIX or other firewall device, the IP address given to Allstream will be the external side of the firewall. Otherwise the Interaction SIP Proxy's IP address should be given.

Proxy for Outbound calls:

If using an Interaction SIP Proxy as outlined in the diagram above for outbound calls, then the IP Address of the Interaction Center SIP Proxy must be entered in the Allstream SIP Line configuration on the IC server instead of the Allstream's IP address. A server plan, with Allstream's IP address, and a route will need to be created on the proxy. The route will consist of all outbound dialed numbers needing to be directed to Allstream's network. If using a PIX or other firewall device, the IP address given to Allstream will be the external side of the firewall. Otherwise the Interaction Center's IP address should be given. Proxy for both



Inbound and Outbound:

If a SIP Proxy is to be used for both inbound and outbound calls both of the two previous sections will need to be followed. Two Server Plan's will be created, one for inbound and one for outbound. The Inbound server plan consists of the IC server IP addresses and the outbound server plan consists of Allstream's addresses. The Allstream SIP Line on the IC server will be directed at the Interaction SIP Proxy. If using a PIX or other firewall device, the IP address given to Allstream will be the external side of the firewall. Otherwise the Interaction SIP Proxy's IP address should be given and Record Route selected on the Interaction SIP Proxy "Settings" page.

Special Notes:

- 1) Allstream will support only G.711 codec at this time (3/10/08)
- 2) The support for G.729 and T.38 is not done at this moment (3/10/08)
- 3) Allstream supports primary and backup IC Server configuration by registering the primary and backup IP addresses of the respective servers in their system. Calls are routed through the back up IC server when the primary is down.
- 4) Blind transfer with IC client cannot be done, you must use consult transfer in the client.
- 5) We have tested for forty simultaneous calls on our IC server using Allstream SIP Carrier.