

AT&T IP Flex Reach/ IP Toll Free Configuration Guide

IC 3.0 with Interaction SIP Proxy

Version 1.7

9/2/2009

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SIP Carriers

1 AT&T



1.1 Introduction

This document is intended for the setup and configuration of options for the Interactive Intelligence Interaction Center (IC) server and Interaction SIP Proxy for use with the AT&T IP Flex Reach and IP Toll Free offerings. This document covers setup and configuration relative only to the IC server and AT&T. Other configurations (stations, permissions, QoS, etc...) are outside the scope of this document. The interoperability test consisted of the following products:

AT&T Service Tested	Status
AT&T IP Flex Reach	100%
AT&T IP Toll Free	100%

Product Tested	Version Tested
Interaction Center (xIC)	3.0 SU-4
Dialogic HMP	3.0 SU-221
Interaction SIP Proxy	4.0
Interaction Media Server	3.0 SU-4
IP Phones	
Polycom IP 430, 550, 560, 650	3.0.3.0401
Cisco 7960	8.8

1.2 Product Descriptions

Interaction Center (xIC) – Interaction Center delivers an innovative pre-integrated application suite to manage all business communications on one platform. xIC's powerful contact center applications and PBX / IP PBX call processing, voicemail, fax server and unified messaging capabilities extend its reach throughout the enterprise - connecting and empowering agents, supervisors and business users, to elevate productivity, performance and customer service.

Interaction SIP Proxy – The SIP Proxy is a standalone proxy server that enables load balancing, redundancy, and protocol conversion in a simple, easy to configure application. The Interaction SIP Proxy will maintain the current status of all outbound destinations to ensure that real-time failover to a backup call path will occur. It will also be the destination IP address given to AT&T.

Interaction Media Server – A standalone server used with a xIC integration that provides a dramatic increase in system scalability and reliability by processing the majority of all media RTP audio flows. The audio processing is offloaded from the xIC server to the dedicated media server or servers registered to the xIC servers.

2 Special Notes

Transport Protocol Support

UDP is the only supported transport protocol available with the AT&T IP Flex Reach and IP Toll Free services. The default protocol for IC 3.0 is TCP. Calls will not connect if this is not changed to UDP.

T.38 Faxing – Requires that the call originate as a G.729 call. An inbound G.711 fax cannot be re-invited to T.38.

Interactive Intelligence does not support T.38 SG3 faxing at the time this document was created. It does however, support G3 faxing.

Multiple SIP Trunk Connections

If remote sites have independent AT&T IP Flex Reach/IP Toll Free service, refer to [section 6](#) for specific requirements.

xIC Version

AT&T IP Flex Reach and IP Toll Free were certified using Interactive Intelligence IC server 3.0. This integration requires a minimum of SU4; which includes all resolutions uncovered during testing.

Remote Priority Codecs

The IC server does not accept remote priority codecs. The codec that will be selected for use is the highest one in the IC server list that matches one offered by the carrier. Please put the carrier desired codec at the top of the list to avoid any issues.

G.729 Annex B Support

The IC server can use G.729 with the Annex B option, however it is not dynamically configurable per call. It is enabled on a given line (as demonstrated in the appropriate configuration section below), and is only an always on, or always off option.

G.726 Support

The IC server does not support G.726 in a standalone fashion. An Interactive Intelligence Media Server is required for G.726 support.

Failover Support

The Interactive Intelligence Interaction SIP Proxy is required for redundant IC server support. The SIP Proxy maintains the route availability to both the IC server pair as well as any number of IP Border Elements provided by AT&T.

IP Flex Reach/IP Toll Free Phone Numbers

A customer may receive one of 2 types of DID's from AT&T: Virtual TNs and non-virtual TNs.

A Virtual TN is one that has an NPA that is different from the NPA at the customer site to which it is being routed. For a virtual TN, AT&T will pass 10 digits to the PBX. For example, if a PBX telephone is associated with a VTN, the number received from AT&T would be 10 digits (i.e. 732-216-2700).

A non-virtual TN has an NPA that is the same NPA as the customer site. For a non virtual TN, AT&T will pass the length of the phone extension plus some prefix if needed (typically a 4 digit extension without a prefix). If a PBX telephone is associated with a non virtual TN, the number received from AT&T would be 4 digits (i.e. 2701 for a 908-216-2701 TN).

However, when originating calls to AT&T, the calling party number must be a 10 digit number regardless of the type of TN associated with the phone that is originating the call.

On the PBX, a specific 10 digit AT&T TN will always be used as the calling number for PBX to AT&T calls.

Emergency 911/E911 Services Limitations

While AT&T IP Flex Reach Services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide in detail to understand these limitations and restrictions.

Overview

2.1 Network Diagram

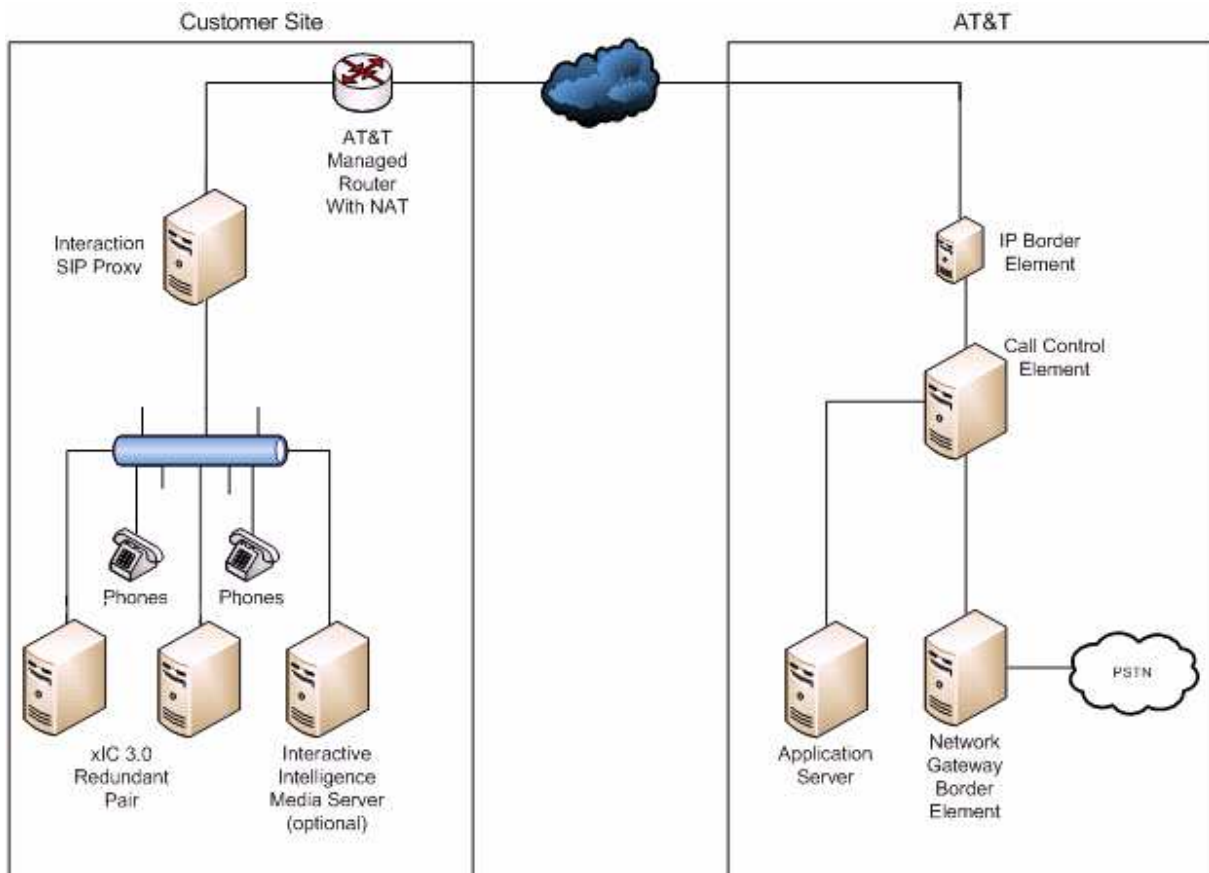


Figure 1: Diagram of General Network Setup

2.2 Proxy Description Overview

2.2.1 Customer Site

IC SIP Proxy: Two Server Plans will be created on the SIP Proxy:
(1) AT&T Server Plan will contain the IP addresses(es) of the IP Border Elements provided by AT&T.
(2) IC Sever Plan will contain the two IP addresses of the IC servers.

Refer to [section 5.1](#) for the dialog menu.

IC Server(s): The outbound proxy of the AT&T SIP line will be the Interaction SIP Proxy.

Refer to [section 4.1.4.4](#) for the dialog menu.

IP Border Element: The destination Proxy address of the IP Border Element will be directed to the Interaction SIP Proxy IP address.

2.2.2 AT&T Network

AT&T will request the IP address of the customer's border element interface connected to the AT&T Managed Router. This is the Interaction SIP Proxy. Shown in **Figure 1** above.

3 IC Configuration Guide

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

3.1.1 Line Menu

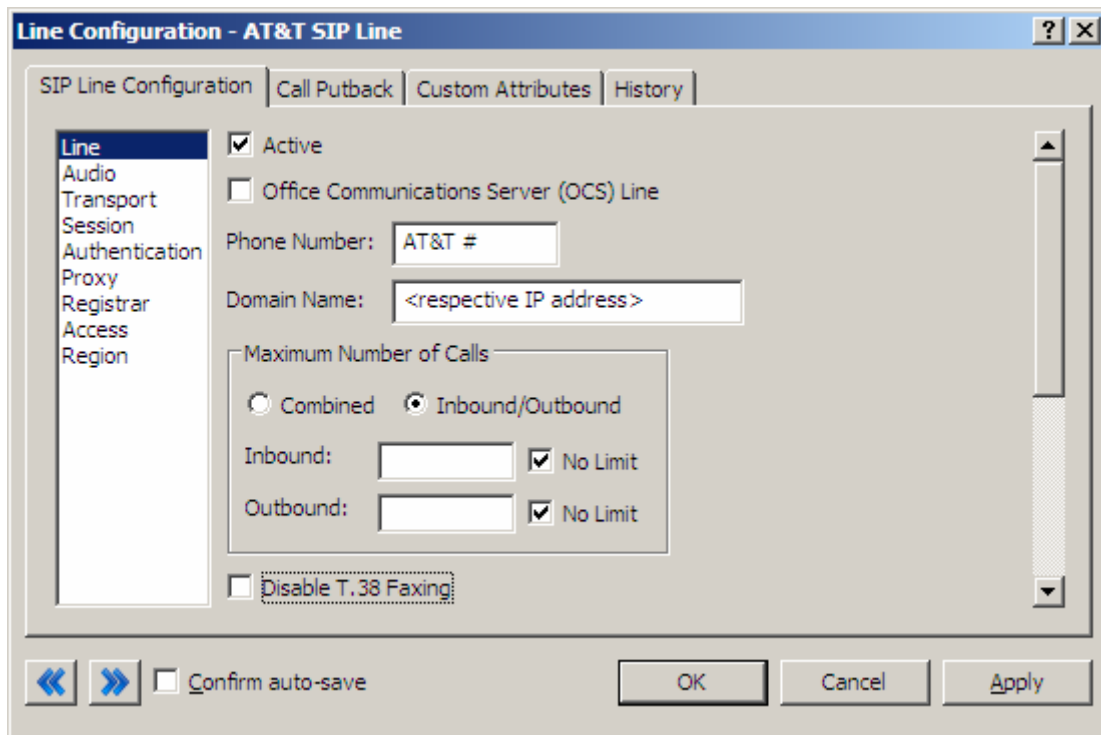


Figure 2: Line Menu Line Configuration Page

3.1.1.1 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*.

3.1.1.2 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. This is critical to be accurate as AT&T validation is done based on this number. Incorrect numbers can lead to some functionality (e.g. international calling, etc...) not working as expected or not working at all.

3.1.1.3 Domain Name

This box should contain the Interaction SIP Proxy IP address.

3.1.1.4 Disable T.38 Faxing

AT&T's SIP carrier service supports 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.

3.1.1.5 Remainder of Line Menu Options

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

3.1.2 Audio Menu

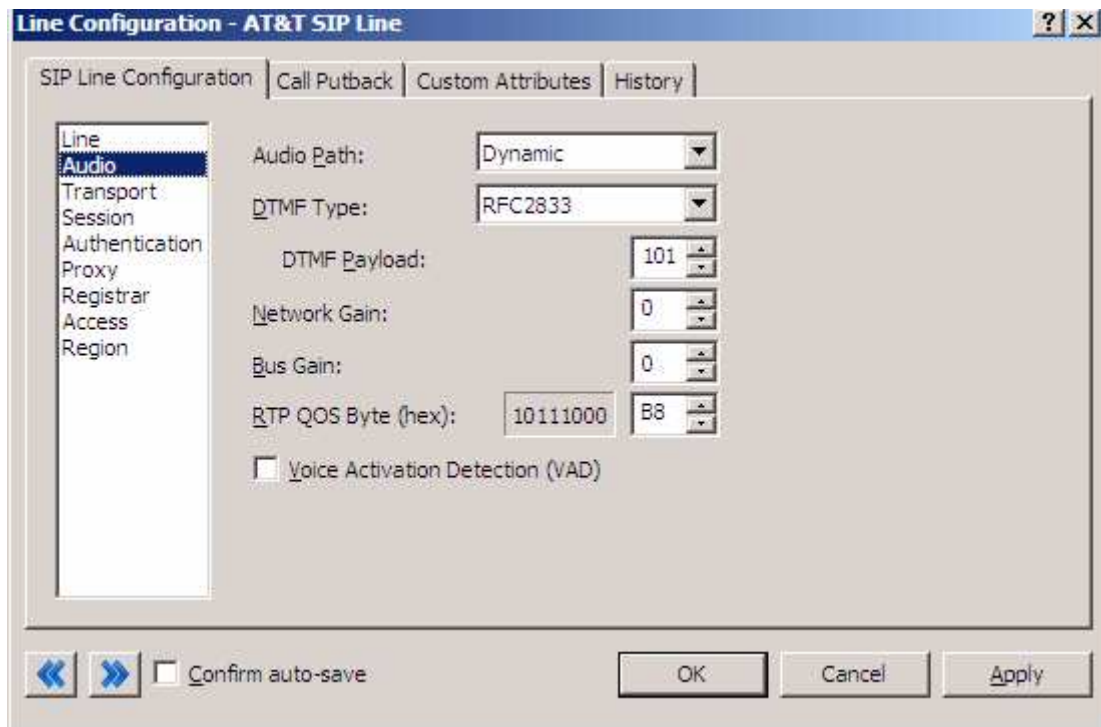


Figure 3: Audio Menu Line Configuration Page for IP Toll Free and IP Flex Reach

3.1.2.1 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.

3.1.2.2 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:

AT&T 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 on the *AT&T SIP Line* 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

3.1.2.3 DTMF Payload

To use DTMF options (especially the control codes in the AT&T IP Toll Free offering,) AT&T and the IC server must negotiate the same payload. IP Flex Reach uses a Payload type of 100 by default and IP Toll Free uses a Payload type of 96 by default.

Normal Media should be configured on the *station* connection Line Configuration to avoid any compatibility issues with DTMF negotiation.

DTMF negotiation with Delayed Media support will be added in SU6.

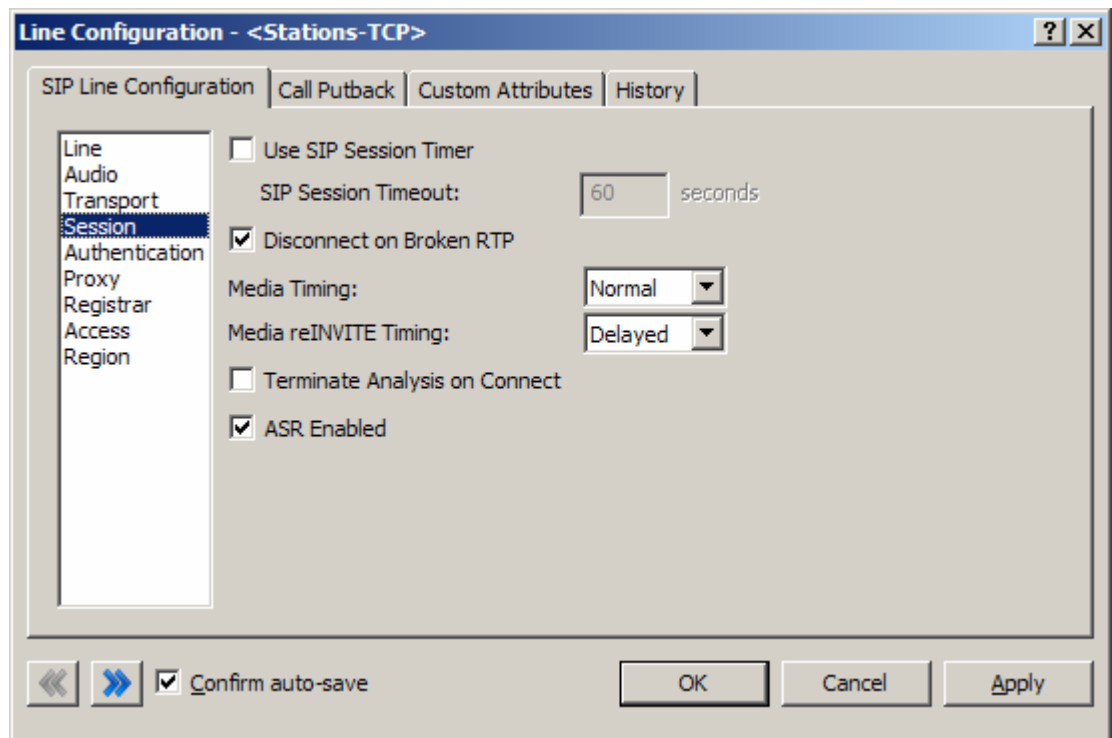


Figure 4: Station connection Line Configuration

3.1.2.4 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.

3.1.2.5 Remainder of Audio Menu Options

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

Transport Menu

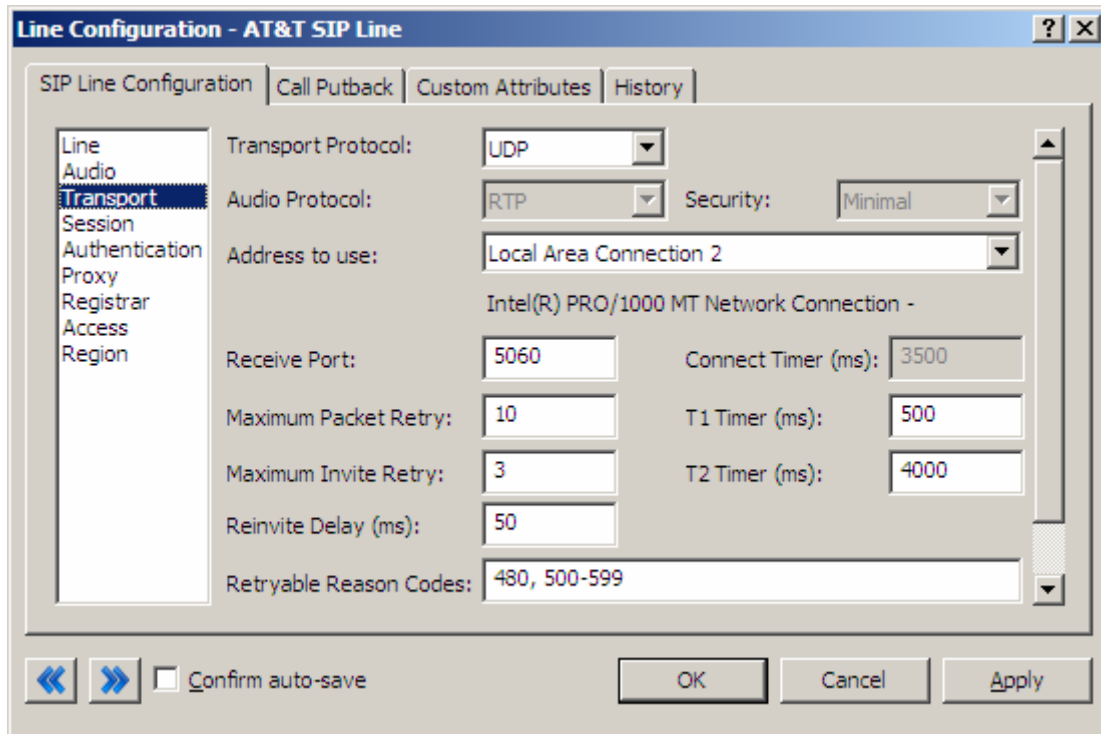


Figure 5: Transport Menu Line Configuration Page

3.1.2.6 Transport Protocol

This option should be set to UDP. As of March 25, 2009 UDP is the only available protocol. TCP and TLS are not currently supported.

3.1.2.7 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 AT&T.

3.1.2.8 Remainder of Transport Menu Options

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

3.1.3 Session Menu

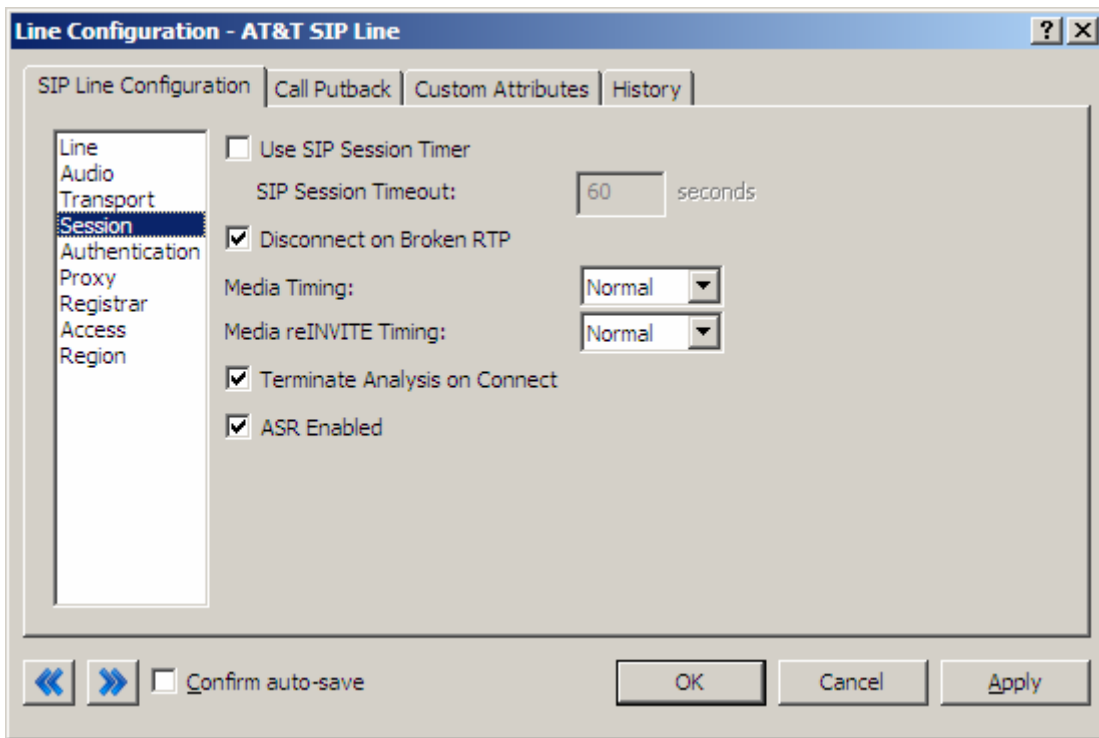


Figure 6: Session Menu Line Configuration Page

3.1.3.1 Disable Delayed Media

This checkbox controls Delayed Media support (in a post-GA IC 3.0 server, it is a dropdown that has Delayed, and Normal options). It must be checked to allow RFC2833 DTMF tones to work, as stated above (3.1.2.2 DTMF Type). Checking this box (or setting to normal) is the recommend method by Interactive Intelligence for all SIP Carriers, and is required for the AT&T service to function properly.

3.1.3.2 Remainder of Session Menu Options

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

3.1.3.3 Authentication Menu

This box must be checked to enable authentication to the SIP Carrier. At the moment, AT&T uses a static IP model with no authentication, so nothing should be done with this page. However, were they to require authentication, the *User*

Name and Password fields should be filled out with the appropriate information provided by the SIP Carrier.

3.1.3.4 Proxy Menu

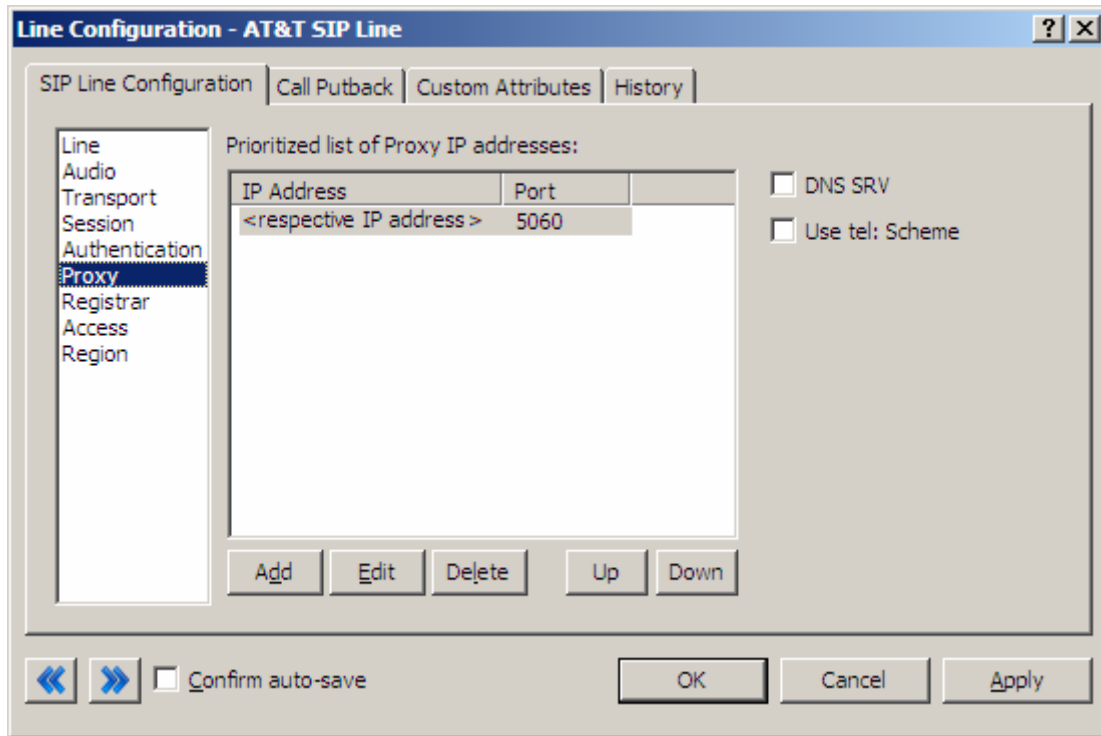


Figure 7: Proxy Menu Line Configuration Page

3.1.3.5 Prioritized list of Proxy IP addresses

In the case of AT&T, the proxy entry should be configured to send information directly to the Interaction SIP Proxy.

3.1.3.6 Remainder of Proxy Menu Options

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

3.1.4 Registrar Menu

3.1.4.1 External Phone Numbers

This box is not currently used by AT&T's configuration. In most cases it would be used to register multiple numbers to the same IC server. However as AT&T 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.

3.1.4.2 Prioritized list of Registrar IP addresses

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

3.1.5 Access Menu (Access Control lists)

If your business needs require that your 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.

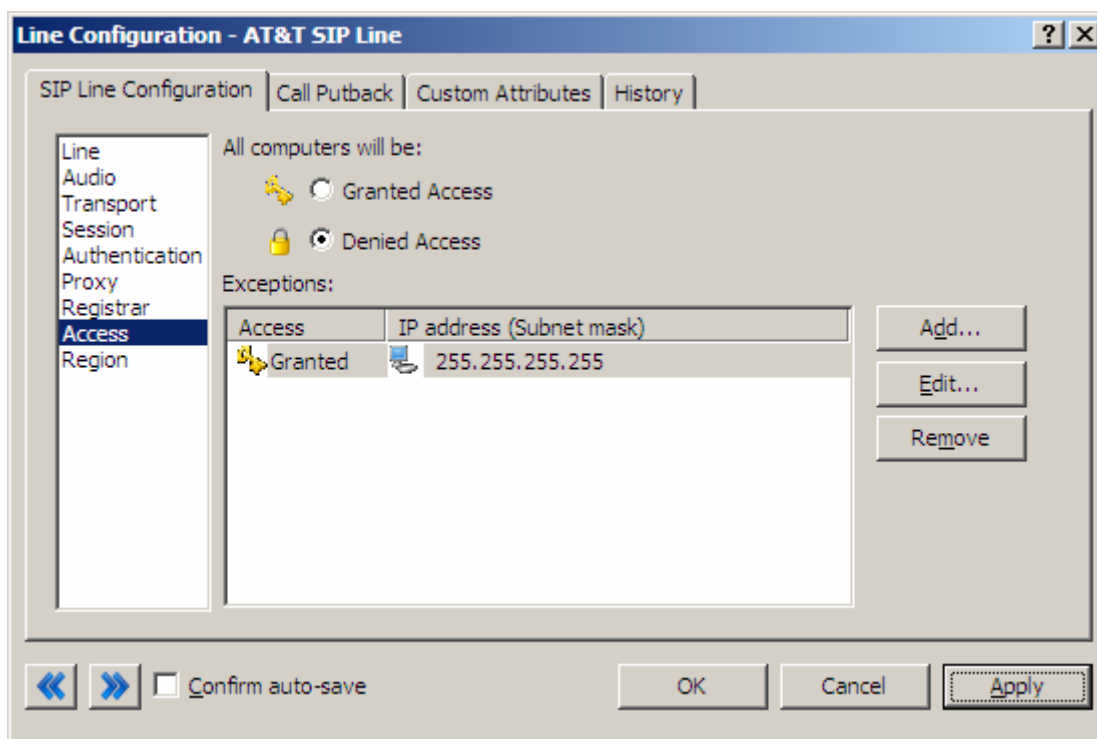


Figure 8: Access Menu Line Configuration Page (note the 255.255.255.255 address is a sample, and the actual number should be respective to the customer needs).

3.1.5.1 AT&T 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.

3.1.5.2 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 IPs, 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.

3.1.5.3 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 AT&T, only G.711 (mu-law and A-law), G.726 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. Given bandwidth and situational information, AT&T typically recommends using G.729 (this can be the default by moving it to the top of the list, if it is not supported by the other end device, then it should fall back to the second in the list and so on).

Important Note: The IC server does not by default support G.726. It should not even be included in the supported list in a 3.0 GA IC server, as it could cause issues. SU4 or later will have the ability to have G.726 in the list without an issue. It will not be able to be used unless an Interactive Intelligence Media Server is used in conjunction with the IC server.

4 SIP Proxy

4.1 Configuration Page

The Sip Proxy requires two Server Plans to be created –

- (1) For inbound calls to AT&T to the IC server switchover pair
- (2) For outbound calling to the AT&T Border Elements

Plan Name	Servers	Protocol	Distribution
ATT	207.242.225.200 207.242.225.201	UDP	Prioritize
IC Server	209.43.1.139 209.43.1.145	UDP	Prioritize

Also, the SIP Proxy must be the IP address provided to AT&T. Otherwise it will reject messages coming to it from an unknown IP. IP addresses of the CIC server(s) and the SIP Proxy must be configured in the AT&T Managed router's static NAT table.

Additional configuration needed is to ensure all SIP signaling is being routed through the SIP Proxy. This requires that Record Route is selected on the Main configuration page under the General tab.

5 Multiple SIP Trunk Connections

5.1 Remote Site Architecture

Some integrations require that each remote site have a IP Flex Reach/IP Toll Free termination. For this to be supported, a local media server is required at each of these locations. This is to ensure an available route for RTP traffic back to the origination site. Each remote site in this architecture would include a SIP Proxy, Media Server, and phone endpoints.

6 Fax Caveats

AT&T 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 AT&T, it may be possible to use G.729 to do the pass-through faxing, however 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.

6.1 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

6.2 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

7 Troubleshooting

Depending on how the service was contracted, support is available for the Interactive Intelligence solution set. If any Interactive Intelligence related materials were obtained via direct sale from Interactive Intelligence then a support contact procedure should have been established at that time. If Interactive Intelligence related materials were obtained through a reseller or partner, then that party should be contacted for support options.

Testlab.inin.com

E-Mail: support@ININ.com

Phone: 1-800-267-1364