

1 SIP Carriers

1.1 Bandwidth.com



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 sites below:

- <http://testlab.inin.com>
- <http://testlab.vonexus.com>

This document is created using IC 2.4, which is reflected in some of the configuration steps. If not using IC 2.4, then the same options should still be available, but may be located in different configuration menus.

1.1.2 Vendor Contact

Vendor Web Site : <http://www.bandwidth.com/sip-trunking>

1.1.3 Versions Verified

SIP Carrier status as of August, 30 2006.

1.1.4 PreInstall

Bandwidth.com will provide users with an IP address to make contact with their systems, and require a static IP from the customer. These must be exchanged before setup can begin.

1.1.5 Install

Bandwidth.com 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.

1.1.7 Configuration

1.1.7.1 IC Server

1.1.7.1.1 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 SIP carrier connection, and one for the stations. 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 *Bandwidth.com 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.

1.1.7.1.1.1 Line Menu

1.1.7.1.1.1.1 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*.

1.1.7.1.1.1.2 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.

1.1.7.1.1.1.3 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.

1.1.7.1.1.1.4 Remainder of Line Menu Options

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

Note: The disable T.38 fax option does nothing in this case, as T.38 is not currently supported by Bandwidth.com.

1.1.7.1.1.2 Audio Menu

1.1.7.1.1.2.1 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.

1.1.7.1.1.2.2 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:

Bandwidth.com 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.

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

1.1.7.1.1.2.3 Remainder of Audio Menu Options

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

1.1.7.1.1.3 Transport Menu

1.1.7.1.1.3.1 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 Sept. 1st 2006, Bandwidth.com has support for UDP by default. TCP is supported by specific request only, and TLS it not currently supported.

1.1.7.1.1.3.2 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.

1.1.7.1.1.3.3 Remainder of Transport Menu Options

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

1.1.7.1.1.4 Session Menu

1.1.7.1.1.4.1 Disable Delayed Media

This checkbox controls Delayed Media support. It must be checked to allow RFC2833 DTMF tones to work, as stated above (1.1.7.1.1.2.2 DTMF Type). *Checking this box is the recommend method by Interactive Intelligence for all SIP Carriers.*

1.1.7.1.1.4.2 Remainder of Session Menu Options

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

1.1.7.1.1.5 Authentication Menu

This box must be checked to enable authentication to the SIP Carrier. At the moment, Bandwidth.com 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.

1.1.7.1.1.6 Proxy Menu

1.1.7.1.1.6.1 Prioritized list of Proxy IP addresses

This box is somewhat of a misnomer in the case of some SIP Carriers. In the case of Bandwidth.com, there may not be a single IP that is needed. Instead they may provide a Fully Qualified Domain Name (FQDN) to a cluster that handles the requests*. When configuring the proxy for Bandwidth.com, this **FQDN** should 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.

*This is a completely plausible scenario, however during testing only an IP address was provided. In that case, the IP address was placed in the proxy field, and everything worked as expected.

1.1.7.1.1.6.2 Remainder of Proxy Menu Options

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

1.1.7.1.1.7 Registrar Menu

1.1.7.1.1.7.1 External Phone Numbers

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

1.1.7.1.1.7.2 Prioritized list of Registrar IP addresses

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

1.1.7.1.1.8 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.

1.1.7.1.1.8.1 Bandwidth.com 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.

1.1.7.1.1.8.2 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.

1.1.7.1.1.9 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 Bandwidth.com, only G.711 and G.729 is supported, so selecting a location that does not have either of these as an option would cause the line not to function properly.

1.1.7.1.2 SIP Proxy Support

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 FQDN of the **SIP Proxy** must be entered in the following places in the Bandwidth.com 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 must have the IP address provided to Bandwidth.com. 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.

1.1.7.1.3 Fax Caveats

Most SIP Carriers do not support T.38 faxing while HMP only supports T.38 fax. 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 passthrough (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.

1.1.7.1.3.1 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

1.1.7.1.3.2 Cisco Gateway FXS Card Configuration

To configure the Cisco Gateway FXS Card to use G.711 passthrough 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
  
```

1.1.7.1.3.3 E911 Support

Bandwidth.com currently supports E911 support via giving registered numbers of customers to an intermediary that then updates them with the emergency services. 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.

