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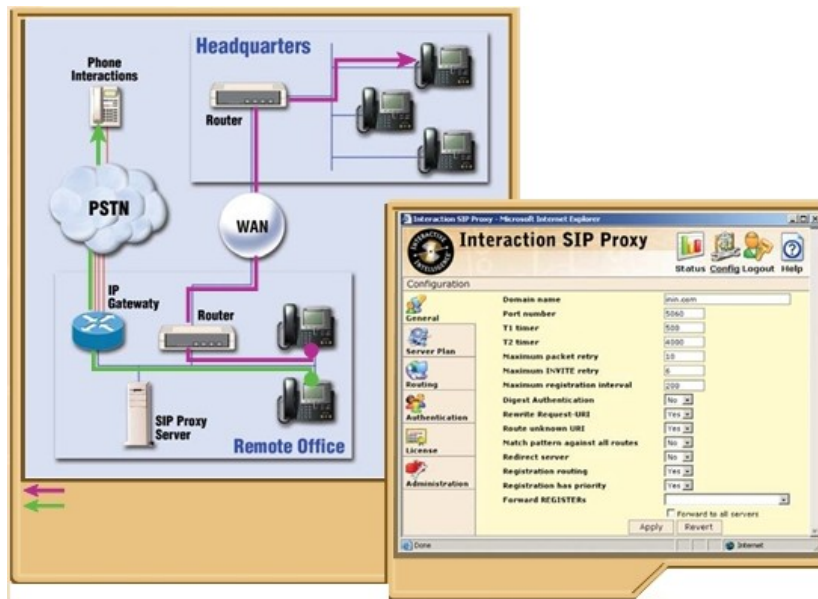
Interaction SIP Proxy™

Only from Interactive Intelligence®

SIP Explained

The international SIP standard—Session Initiation Protocol—is taking the world by storm by allowing organizations to route voice traffic over their data networks using interoperable devices from many different vendors. One critical component of the SIP architecture is the **proxy server**, which acts as the traffic cop by helping calls take the correct turns to get to the proper destination.

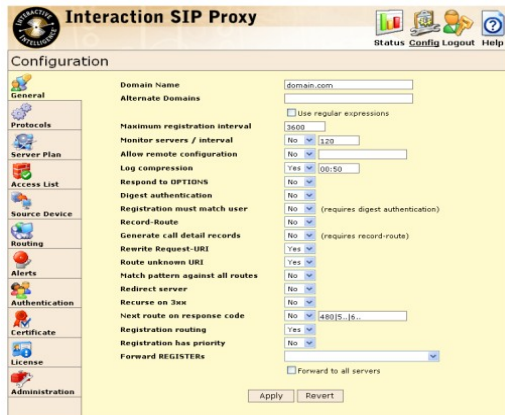
Consider the example shown below involving a headquarters building and a small remote office in another city.



A SIP Proxy allows calls to be routed over the WAN to headquarters or out the gateway to the public telephone network, as necessary

Calls destined for headquarters should travel over the company's WAN, while local calls should use the gateway to the public telephone network (PSTN) at the remote office. The SIP proxy provides this and other types of routing. Phones at the remote office "talk" to the SIP proxy, telling it the number they wish to dial. The proxy then sends their calls down the correct path. The proxy server also is critical for remote survivability. For example, if the WAN to headquarters goes down, the proxy server can send all calls out the gateway.

Interaction SIP Proxy



Interaction SIP Proxy provides easy Web-based administration

Interaction SIP Proxy is an extremely easy to install and easy to use proxy server. Just load the software on an inexpensive Windows®-based machine and you have a sophisticated SIP proxy server with the following capabilities:

- Supports RFC 3261, the IETF standard for SIP

- Built-in, secure web server for easy configuration and remote access from any browser.

- Flexible yet powerful call routing with intuitive regular expression pattern matcher. Routes can even be grouped based on where the call originates from. For example, "all calls from subnet A, which contain a local phone number, shall be sent to the PSTN".

- Unlimited number of destination entries. The Interaction SIP Proxy will try each destination in turn until it finds one available, providing critical fault tolerance (e.g., "First try to send this call over the WAN to New York. If that's not possible, send it out the gateway over the public telephone network.").

- Dial plan simulation. Enter a phone number in the Web interface and see where it would go—great for troubleshooting.

- Strict security using HTTPS, digest authentication, access control lists and administrative protection of all status and configuration pages.

- Real-time information about registered devices (e.g., phones) on the registration status Web page.

- Real-time server status. Visual display of the status of all destination servers on the server status Web page.

- Fault tolerant routing (The status of each server in the server plans is monitored at the application level. If not available the proxy will use the next available route to deliver the message.

- Load balancing (SIP messages can be balanced across multiple routes using a round-robin, random or prioritized distribution model.

- Call detail records and real-time active call status. Visual display of active calls on the session status web page. Call detail records are generated for completed calls in a comma-separated file format.

- Supports SIP over UDP, TCP or TLS (Basic only supports UDP)

- SMTP email alerts and SNMP traps for critical events

Note: Some features are only available with specific licensing. Please refer to the side-by-side feature comparison document for details on feature availability for a specific license type.

Benefits and Advantages

Compared to SIP proxy servers that cost tens of thousands of dollars, the Interaction SIP Proxy from Interactive Intelligence is an incredible bargain. With its ability to handle an unlimited number and variety of call routing plans—along with interoperability with SIP gateways, phones, and other devices—the Interaction SIP Proxy allows you to assemble a powerful but low-cost voice over IP (VoIP) solution. And since it's software, you can load the Interaction SIP Proxy on any Windows-based server and be up and running in minutes. The Interaction SIP Proxy's Web-based administration even allows you to set up and manage your new proxy server from any Web browser.

Features and Specifications

SIP Features

Supports RFC 3261. RFC 3261 is the latest IETF standard for Session Initiation Protocol.
Operates as a stateful proxy or redirect proxy. As a stateful proxy, it will proxy SIP requests on behalf of SIP devices. As a redirect proxy, it will redirect the SIP device to the appropriate destination.
Supports SIP Registrar features. If enabled, this feature will accept REGISTER messages from SIP devices and use those registrations as routing item entries.
Supports forwarding of REGISTER messages to downstream SIP devices.
Provides a visual display of registered SIP devices using the Web interface. All registered devices are shown on the registrations status page along with their addresses and expirations.
Supports SIP digest authentication as described in RFC 2617 to prevent unauthorized access to the proxy's services (**BCM and Load balancing version**).
Supports SIP over UDP, TCP and TLS protocols (Basic only supports UDP)

Dial Plan and Routing Features

Unlimited routing item entries. A routing item uses regular expressions and pattern substitution to match dialed number patterns in SIP messages and route them to the configured destination.
Dial plan simulation. The Web interface lets a user enter a dialed address and presents the route plan after running it through the dial plan. This is useful when setting up dial plans and for diagnosing routing problems.
Unlimited server plan entries. A server plan is a list of destinations, that are attached to route item entries. A request that matches a route item entry will be forwarded to each destination in the route item's server plan until a response is received.
Dynamic registration routing. The SIP proxy will attempt to deliver SIP messages to destinations for which it has received a valid REGISTER. This feature allows SIP devices using the proxy to communicate with each other in the event of failure of the server plan entries. This feature is critical for remote survivability.
Dialed number modification. The SIP proxy can be configured to change the dialed number based on the next destination. For example, an internal dial plan may use 5-digit dialing, but in the case of failure to a PSTN gateway, the SIP proxy can seamlessly change the number to a 7-digit or 10-digit format so that the call can be completed correctly.
Supports distinct dial plans per source device or per subnet of devices (BCM and Load balancing version)
Real-time status of servers in all server plans (**BCM and Load balancing version**). This is useful for diagnosing the availability of servers. Servers that are not available are also disabled during normal proxy routing. Each disabled server is monitored and brought into service when available.
Load-balancing (**Load balancing version**). The SIP proxy supports round-robin, random, and prioritized distribution models for routing SIP messages. The models can be mixed and matched on the same system and are associated with a route item. This means the proxy can chose distribution models based on dialed numbers.

Administration Features

Built-in Web server for local or remote administration. All configuration parameters are available through the Web interface and can be done from any standard Web browser.
HTTPS digest authentication. All configuration and status Web pages are protected with an administrative account.
XML formatted configuration files. All configuration files are stored in XML format, making it easy to archive configuration for disaster recovery and configure the management of redundant proxy servers.

Requirements

Operating system – Microsoft® Windows® 2003, Windows® 2000, or Windows® XP
Hardware – Intel®-based server with at least 512 MB of RAM and 25 GB of free disk space

It's Easy!

The Interaction SIP Proxy is ideal for central or remote sites requiring advanced routing capabilities that SIP phones can't provide. In addition to being powerful, the Interaction SIP Proxy is inexpensive compared to products from major vendors offering similar functionality. The Business Continuity Manager and Load balancing versions add sophisticated fault tolerance, load balancing, call detail records, and security for only a modest fee, and gives you access to telephone support.

Just go to <http://www.inin.com/sipproxy> to download a free trial* license for the Basic Interaction SIP Proxy. BCM and Load Balance trial licenses are available upon request.

FOR MORE INFORMATION AND AVAILABILITY

For more information on the SIP-ready Interaction Center Platform and our suite of IP telephony-based interaction management software products, visit www.ININ.com or e-mail info@ININ.com.

*Free temporary trial license of the Basic version will expire. . A permanent license must be purchased after expiration of trial. Permanent licenses can be easily ordered at <http://www.inin.com/sipproxy>

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