Interaction SIP Proxy™

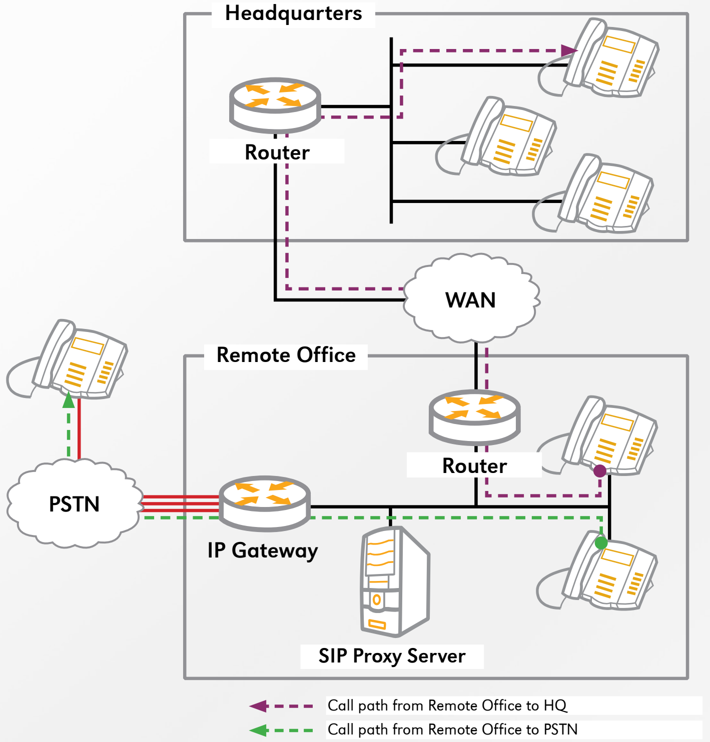
Low-cost solution for VoIP

Interaction SIP Proxy is an extremely easy to install and easy to use proxy server. Compared to SIP proxy servers from major vendors that cost tens of thousands of dollars, it's also an incredible bargain.

With the ability to handle an unlimited number and variety of call routing plans — plus interoperability with SIP gateways, phones and other devices — the Interaction SIP Proxy allows organizations to assemble a powerful yet low-cost proxy solution for voice over IP (VoIP). As software, simply load the Interaction SIP Proxy on any Microsoft® Windows®-based server and be up and running in minutes. Web-based administration also allows you to set up and manage the Interaction SIP Proxy server from any web browser, which further lowers costs.

Routing versatility for central and remote sites

Interaction SIP Proxy is designed for central as well as remote sites requiring advanced routing capabilities that SIP phones can't provide. Available Business Continuity Manager (BCM) and Load balancing versions add sophisticated fault tolerance, load balancing, call detail records, and security for a modest fee, and include telephone access to Interactive Intelligence's tech support teams to handle SIP Proxy issues.



Structure advanced routing routines for central site and multi-site configurations.

Key features

- **Support for RFC 3261**, the IETF standard for SIP
- **A 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 be grouped based on where the call originates from; e.g., "all calls from subnet A, which contain a local phone number, shall be sent to the PSTN"
- **An unlimited number of destination entries**
  - The Interaction SIP Proxy tries 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 to 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
- **Support for SIP over UDP, TCP or TLS** (Basic version supports UDP only)
- **SMTP email alerts and SNMP traps** for critical events

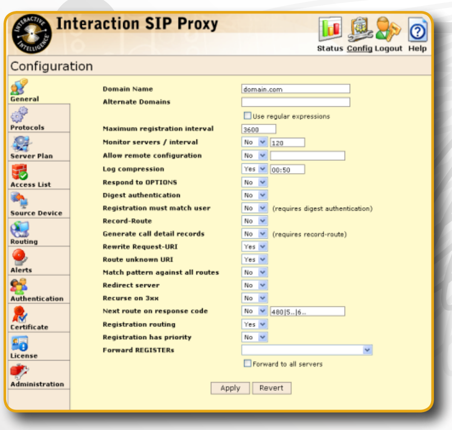
SIP

- Supports RFC 3261, the latest IETF standard for the Session Initiation Protocol
- Operates as a stateful proxy or redirect proxy
  - As a stateful proxy, will proxy SIP requests on behalf of SIP devices
  - As a redirect proxy, 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 the 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 version supports UDP only)

BCM = Business Continuity Manager

Administration

- **Built-in web server for local or remote administration.** Access and manage all configuration parameters via the Interaction SIP Proxy web interface and any standard web browser.
- **HTTPS digest authentication.** Protect all Interaction SIP Proxy configuration and status web pages with an administrative account.
- **XML formatted configuration files.** Interaction SIP Proxy stores all configuration files in XML format, making it easy to archive configuration for disaster recovery and configure the management of redundant proxy servers.



The Interaction SIP Proxy web interface

Dial plan and routing

- **Unlimited routing item entries.** With the Interaction SIP Proxy, 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 Interaction SIP Proxy web interface lets a user enter a dialed address and presents the routing plan after running it through the dial plan. 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 routing item entries. A request that matches a routing item entry will be forwarded to each destination in the routing item's server plan until a response is received.
- **Dynamic registration routing.** Interaction 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 Interaction SIP Proxy to communicate with one another in the event of failure of the server plan entries, and is critical for remote survivability.
- **Dialed number modification.** Configure the Interaction SIP Proxy 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 Interaction SIP Proxy can seamlessly change the number to a 7-digit or 10-digit format so 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).* 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 Interaction SIP Proxy supports round-robin, random, and prioritized distribution models for routing SIP messages. Models can be mixed and matched on the same system and are associated with a routing item. This means the Interaction SIP Proxy can choose distribution models based on dialed numbers.

Requirements

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

Free trial license*

Go to http://www.inin.com/sipproxy to download a free trial license for the Basic Interaction SIP Proxy. Trial licenses for BCM and Load balancing versions available upon request.

*Permanent license must be purchased after Basic version trial license expires.
Order permanent licenses at http://www.inin.com/sipproxy.

INTERACTIVE INTELLIGENCE®

Interactive Intelligence offers unified business communications solutions for contact center automation, enterprise IP telephony, and business process automation, based on our open standards, all-in-one software suite. More than 4,000 organizations worldwide currently benefit from our on-premise solutions and cloud-based Communications as a Service (CaaS) offerings, including value-added services for software, hardware, implementation, consulting, support and education.

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