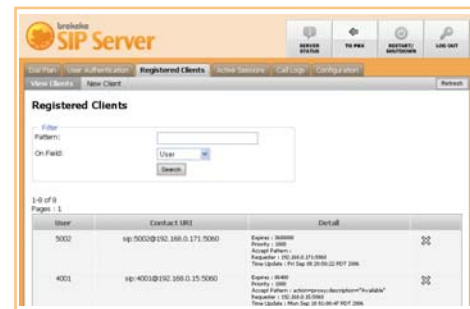


Outline

Brekeke SIP Server is a software proxy and registrar server. Supporting industry-standard protocol SIP (Session Initiation Protocol), Brekeke SIP Server is used to register SIP UAs, authenticate VoIP calls, then route the calls between user agents. Brekeke SIP Server is an ideal product to create an scalable and reliable SIP communication network for your business and organization.

Key Benefits

- **Multiple OS Support**
- **Intuitive Web-based Management System**
- **Support for Media Stream over NAT Traversal**
- **Multiple-Domain Support**
- **Highly Interoperable with third party products and services**
- **Customizable to meet unique needs through Dial Plans & Plug-ins**
- **Flexible Scalability**



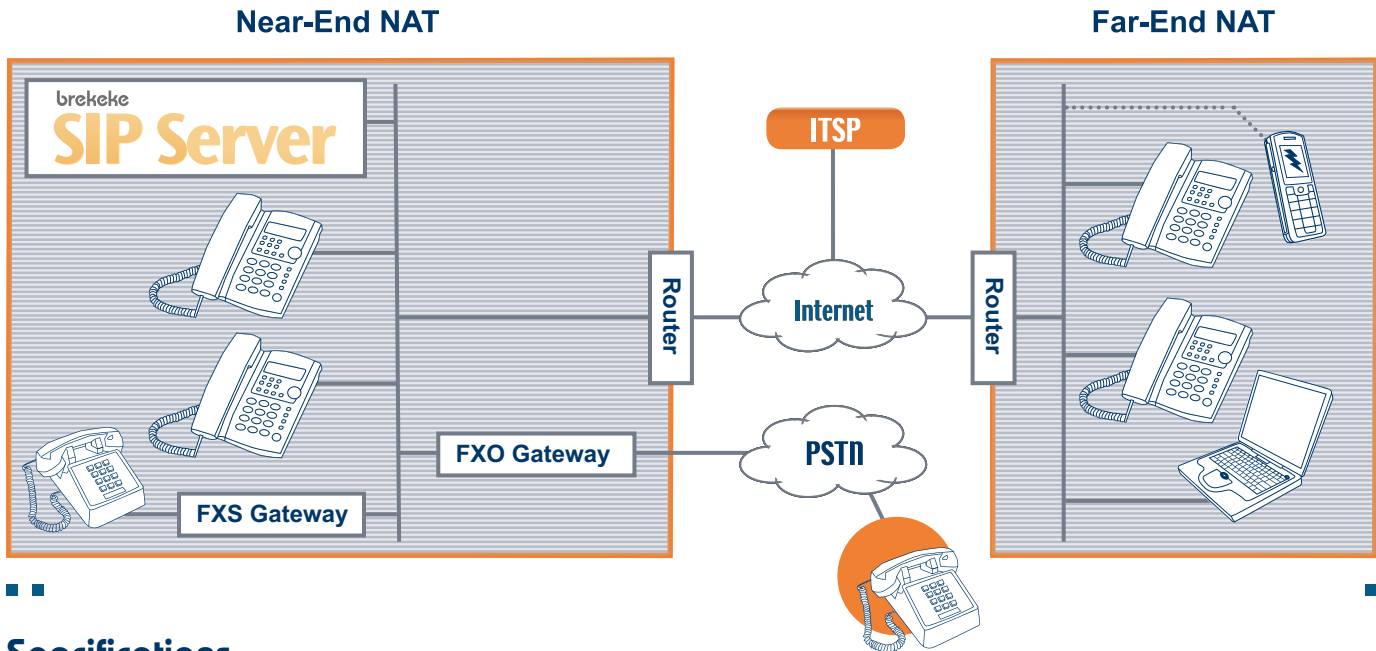
Sample view of Brekeke SIP Server

Features

- **Registrar Service**
Brekeke SIP Server receives REGISTER requests from SIP UAs and updates its database appropriately. Using the registrar function, you can receive calls from any SIP UA using your unique SIP-URI.
- **Call Routing**
Brekeke SIP Server will route SIP requests from a SIP UA, or other server, to the most appropriate SIP-URI address based on its registrar database and Dial Plan. Brekeke SIP Server supports SIP redirect feature which allow servers to redirect a request back to SIP UA.
- **Authentication**
By specifying authentication settings on REGISTER or INVITE requests, you can limit calls that go through Brekeke SIP Server. Authentication Plug-in is available for the users who wish to use an existing user directory service.
- **NAT Traversal**
Brekeke SIP Server enables SIP UAs behind the NAT to talk with other SIP UAs, including video over NAT traversal. Using NAT-enabled firewalls ensures the security that you want to retain, while giving users the ability to make media calls over the internet between different networks.
- **Dial Plan**
With a Dial Plan you can use regular expressions to define matching or filtering rules for headers and IP addresses in the SIP packets. Brekeke SIP Server's dial plan increases compatibility between SIP compliant products and provide added capability for creating complex call routing.
- **Upper/Thru Registration**
Upper/Thru Registration allows for easy configuration of parallel users of pre-existing or other SIP servers. With this feature, you can take advantage of SIP communication through ITSP lines or third party SIP Servers.
- **Multiple-Domain Hosting**
Brekeke SIP Server can host multiple domains on one server install. This feature allows user to manage multiple domains under one server setup.
- **Session Management**
The real-time session management is available through GUI. View session status or manually terminate the active calls.



Sample Network Structure



Specifications

■ Protocol for Signaling	SIP (RFC3261 Standard)
■ Protocols for Delivery of Media	RTP, RTCP (When using NAT Traversal)
■ Routing Methods	Register database or Dial Plan
■ NAT Traversal	Proprietary method
■ Maximum Number of Concurrent Sessions	Unlimited
■ Administration	Web-based

Brekeke SIP Server Plug-in

■ Accounting Plug-in	Controlling sessions based on Accounting information or collecting call log data for accounting purposes (ex., radius accounting)
■ Authentication Plug-in	Authenticating SIP UAs using existing or customized method (ex., radius authentication)
■ Dial Plan Plug-in	Creating Dial Plan to work with existing or customized programs and services

Operating Environment

■ Operating System Supported	Windows, Linux, Mac OS X, Solaris
■ Java	JDK 1.4 or later
■ Memory	256 MB minimum

Contact Us



Brekeke Software, Inc.

4 West 4th Avenue, Suite 501, San Mateo, CA 94402-1614 U.S.A.

TEL: +1 650-401-6636 FAX: +1 650-401-6629

URL: <http://www.brekeke.com/>

email: sales@brekeke.com sip:info@sip.brekeke.com

