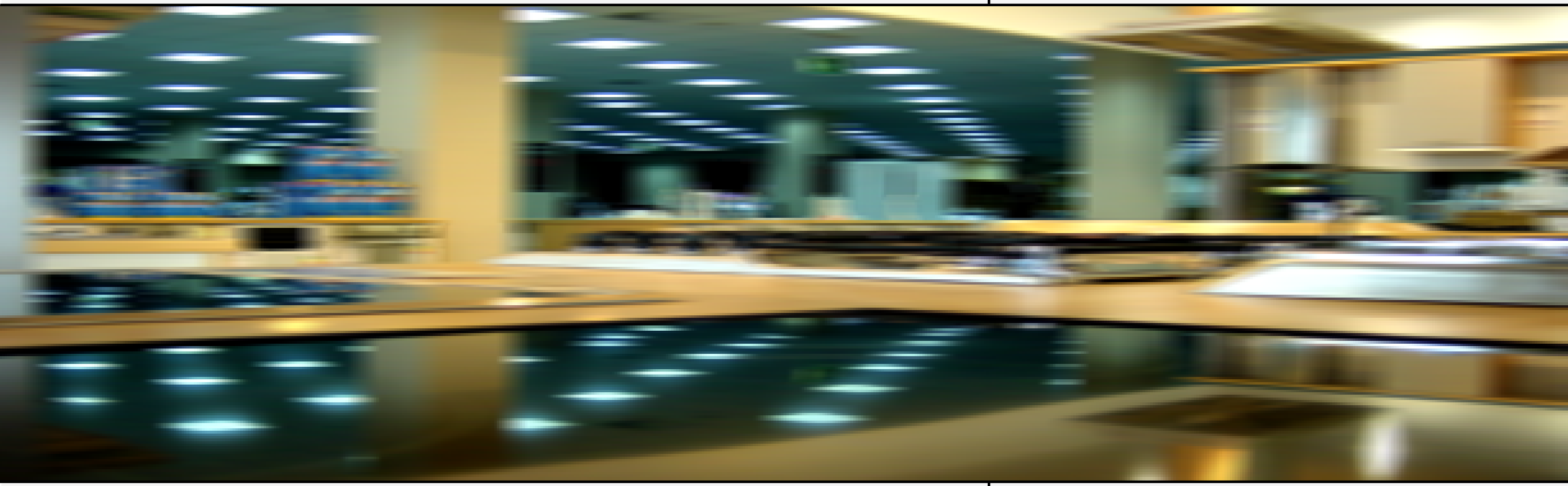
PORTA ® ONE

PortaSIP



Product Brochure

www.portaone.com

PortaSIP

PortaSIP allows IP Telephony Service Providers to deliver communication services at incredibly low initial and operating costs that cannot be matched by yesterday's circuit-switched and narrowband service provider PSTN networks.

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PortaSIP Overview

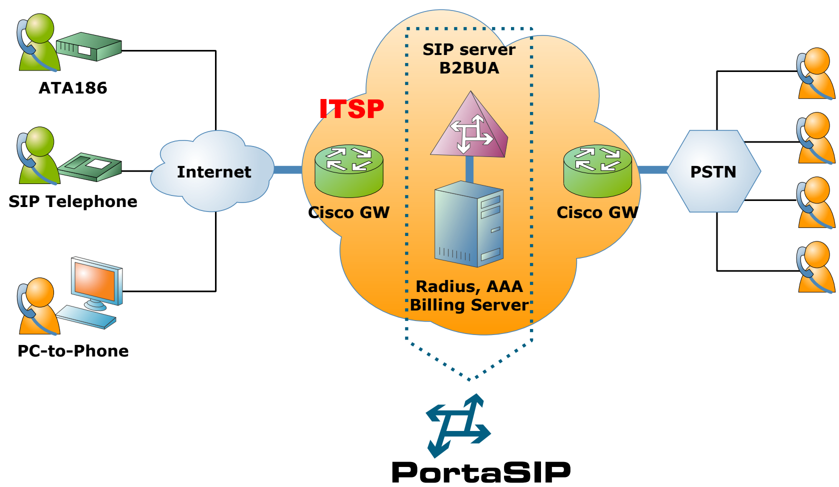
The PortaSIP solution provides carriers with an open, SIP-based architecture to support enhanced voice applications such as hosted IP PBX service, unified messaging, conferencing, and IP call center, as well as a whole new class of services that integrate telephony with the web, e-mail and other communications portals.

With PortaSIP, services that combine elements from telephony and other web applications such as e-mail, messaging, the Internet and video streaming can be created easily and with a shorter planning/implementation cycle, allowing increased market responsiveness.

The PortaSIP Solution, consisting of:

- **PortaSIP Extension for PortaBilling100**
- **SIP Proxy Server**
- **B2BUA**

provides a platform to deliver innovative, high-margin services that can be deployed over IP, as well as cable, POTS, Broadband, and ISDN networks.



A Typical PortaSIP Network Architecture

PortaSIP supports:

1. Any SIP Endpoint (network element which can terminate a SIP session)
2. Universal Numbering Plan, PBX Extensions, and other numbering schemes
3. Customer Dialing Rules & Abbreviated Dialing
4. Clustering of gateways and other servers

PortaSIP API allows 3rd party vendors to access all of the important features of the system (Message Waiting Indication, Directory, etc.).

The **Customer Dialing Rules** & Abbreviated Dialing feature works with both SIP and H.323, with adequate support of either SIP Server or Cisco TCL IVR. A PortaBilling customer can set up Dialing Rules as an international prefix, outside prefix, direct number (e.g. 911) or abbreviated dialing for his accounts. More than one dialing rule can be assigned to each account.

The system allows "**clustering**" of gateways and other servers to "roll over" in case of capacity constraints or failure. This allows several devices to be used in tandem, with one overflowing to the next, if the number of calls being handled exceeds the limitations of a current device, or if a device fails.

PortaSIP Applications

The PortaSIP solution brings the simplicity and universality of IP to PSTN by providing an open services architecture with distributed intelligence, which generates greater earnings than traditional voice services. It provides a platform for the delivery of enhanced business and residential communications services that are extensible and can be independently branded, including:

- IP PBX/Centrex
- Unified/Instant Messaging
- Presence/Mobility
- Multi-Party Conferencing
- Voice-enhanced e-commerce
- Web Call Centers

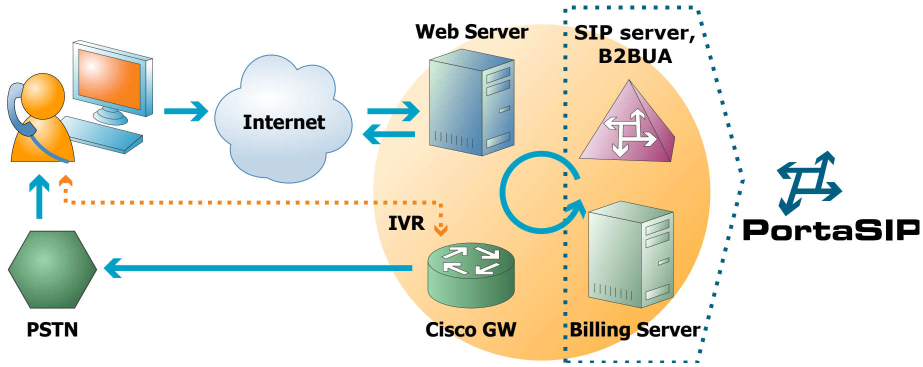
Voice-enhanced e-commerce

A website can feature click-to-dial links that establish a session between the end-user and the website organization. A travel agent's website, for example, could offer a toll-free service whereby a prospective customer can speak to an agent, who then guides the user through a series of pages, perhaps showing video clips of potential holiday destinations. The customer can complete booking forms online while talking to the agent, and may also have the opportunity to buy currency or rent a car from a partner organization. Human interaction can be used to enrich the e-commerce experience.

This kind of service could be a part of a value-added web-hosting service offered by a service provider, or it could be developed in-house by the company's IT department.

Web call centers

A similar idea can be used in the provisioning of web call centers. A web page may pop up when a particular number is called (with SIP, it is just as easy to direct a user to a web page as to a telephone). Agents can step in if the user requests help. SIP can support IVR-type (Interactive Voice Response) functionality, navigating users through options and providing auto-responses for common requests.



When a user is connected directly to a customer service agent, the user's details are visible to the agent, who can then give the appropriate response. They can also view the same sets of data on their respective screens. Any number of types of communication between the two parties can be supported: voice, e-mail, IM or video-conferencing.

Why Use PortaSIP?

Cutting-edge PortaOne technology is helping service providers launch the next generation of broadband communications. The PortaSIP Architecture, including PortaBilling100 platforms, Sip Proxy Server and B2BUA, provides evolutionary, packet-based platforms for delivering multiple voice and data applications over IP networks. Unmatched in its scalability and versatility, PortaSIP supports a revolutionary migration path for future service creation and delivery in the New Public Network. Our commitment to providing complete solutions for service providers' success is further extended by PortaOne's comprehensive portfolio of professional and technical support services.

Benefits for the company

PortaSIP reduces the total costs of owning telephony, exploiting a company's own IP infrastructure and, when combined with presence, instant messaging, location and mobility applications, increasing the efficiency of communications.

PortaSIP improves company productivity by enabling new, converged applications that allow the company's software to interact better with people in ways they prefer. PortaSIP allows companies to respond more effectively to customers, while customers enjoy personalized service that is tailored to their needs and preferences.

In the short term, PortaSIP can provide rapid return on investment by reducing the demand on PSTN services and WAN bandwidth, eliminating unnecessary calls and routing calls more cost-effectively.

Benefits of using PortaSIP

- Tight integration with the billing engine for call authorization and accounting
- Built-in features allowing NAT traversal

- Solid support for debit accounts – pre-paid business model
- Much improved stability, performance and correctives of the B2BUA component
- **high-performance SIP Proxy/Registrar**, one of the fastest and most flexible available on the market
- ability to stack unlimited number of PortaSIP servers, when lowest-level PortaSIP looks simply like a user agent for highest ones
- ability to distribute load among multiple PortaSIP servers running in parallel, allowing for **unlimited** number of users served
- built-in support for redundancy to provide high-availability of service
- support for secure digest authentication for both downlink and uplink SIP connections;
- web-interface for configuring PortaSIP settings
- original support for securely authenticating inbound SIP calls from Cisco gateways (Cisco IOS has no support for secure SIP authorization);

PortaSIP Components

PortaBilling100

http://www.portaone.com/solutions/billing

PortaBilling100 is an extremely flexible, robust, state-of-the-art customer management, mediation and billing platform that enables providers of IP Telephony services to launch, price and provision VoIP services instantly.

PortaSIP provides centralized control of all network components via a user-friendly PortaBilling100 web interface, which performs accounting functions by collecting and correlating billing information from various network elements in the service delivery path. It enables carriers to turn these new enhanced services into billable revenue immediately using a flexible platform for converting call detail records (CDR) into basically any format. PortaBilling100 also provides valuable call traffic data for generating reports on capacity utilization, resource availability, and calling patterns by time, trunk, subscriber and call type.

SIP Server

http://www.iptel.org/ser
SIP Express Router (SER) is a high-performance, configurable, cost-free SIP (RFC3261) server. It can act as a registrar, proxy or redirect server. SER features an application-server interface, presence support, SMS gateway, SIMPLE2Jabber gateway, RADIUS/syslog accounting and authorization, server status monitoring, FCP security, and so on. Web-based user provisioning, serweb, is also available.

SER's performance allows it to deal with operational burdens such as broken network components, attacks, power-up reboots and rapidly growing user population. SER's configuration ability meets the needs of a whole range of scenarios, including small-office use, company PBX replacements, and carrier services.

The SIP Server supports SIP nodes, such as SIP telephones or an ATA186 with both Global IP addresses and NAT addresses located behind the NAT device (e.g. Cable/DSL router). Regardless of the IP address (global or NAT address behind a firewall), a SIP node is able to exchange 2-way voice traffic with any other device on the network (static global IP, dynamic global IP, or any other device behind a NAT firewall).

B2BUA

http://www.vovida.org/
The Back-To-Back User Agent (B2BUA) is a Session Initiation Protocol (SIP) based logical entity that can receive and process INVITE messages as a SIP User Agent Server (UAS). It also acts as a SIP User Agent Client (UAC), determining how the request should be answered and how to initiate outbound calls. Unlike a SIP proxy server, the B2BUA maintains the complete call state and participates in all call requests.

Integrating B2BUA with PortaSIP ensures that every call made between endpoints (off-net, on-net, etc.) is authorized, authenticated, and billed. The system is also able to provide "Debit" billing (i.e. to disconnect the call if the account balance falls below zero).

The following major changes were made to Vovida B2BUA for use with PortaSIP.

1. Better FreeBSD support, including:
   - fixes to threading code to better support FreeBSD user-space POSIX threading package;
   - various compilation fixes.
2. Enhancements for Radius stack, including:
   - fixed previously broken support for Vendor-Specific Attributes (VSA);
   - added support for SIP-specific Radius attributes.
3. Fixes for SIP stack, including:
   - fixed error in retransmission code which caused retransmission timeout not to be reported to upper layer, thus leading to "hanged" sessions that were never reported back to originated UA and never cleaned up from memory.
4. Radius accounting sent by B2BUA has been made more Cisco-like, particularly:
   - a number of VSA h323-attributes added to the accounting: h323-remote-address, h323-call-type, h323-call-origin, h323-session-protocol, h323-disconnect-reason, call-id, h323-setup-time, h323-connect-time, h323-disconnect-time, h323-voice-quality;
   - B2BUA extended to map SIP error code into h323-disconnect-reason, using table similar to that used on Cisco gateways;
   - B2BUA now sends Stop accounting in all cases when authorization is completed successfully, even if call did not go through for some reason (e.g. request cancelled, SIP error reported by UA, network error, etc.)

5. B2BUA now uses a secure Proxy-Authorization method to authorize sessions.
6. B2BUA now has the ability to limit session duration using information obtained in the authorization "accept Radius message" (h323-credit-time).
7. B2BUA was extended to allow optional call forwarding when a call is forcibly disconnected due to exhaustion of the credit limit.
8. Fixed bug in B2BUA whereby SDP body was stripped from provisional 18x messages, thus preventing in-band alerting from going through to the originating party.
9. B2BUA was extended to allow it to handle calls in two directions: from the proxy to the remote UA (originate mode) and from the remote UA to the proxy (answering mode).
10. A number of stability fixes were applied to B2BUA:
    - session state kept in memory for 2 minutes after call termination to prevent delayed requests and replies from causing race conditions which, in some cases, could result in crashes due to the multi-threading nature of B2BUA;
    - an incoming request with call-id that does not match any existing session other than INVITE is simply ignored;
    - fixed bug whereby a delayed message received after a call is already terminated but not yet cleaned up causes assertion failure.

What is SIP?

Session Initiation Protocol (SIP) is an application layer control (signaling) protocol for creating, modifying and terminating sessions with one or more participants. These sessions include Internet conferencing, telephony, presence, events notification, and instant messaging.

SIP enhances the ability to integrate voice with Internet-based services – making it the natural choice of service providers who want to develop and install new applications in less time and at lower cost than traditional telephony services.

SIP invitations are used to create sessions, and they carry session descriptions that enable participants to agree on a set of compatible media types. SIP makes use of proxy servers to help route requests to the user's current location, authenticate and authorize service users, implement provider call-routing policies, and provide other features to users.

Like other VoIP protocols, SIP is designed to address signaling and session management functions within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

SIP provides the ability to:

- Determine the location of the target endpoint—SIP supports address resolution, name mapping, and call redirect.
- Determine the media capabilities of the target endpoint—via the Session Description Protocol (SDP), SIP determines the "lowest level" of common

services between the end points. Conferences are established using only those media capabilities that can be supported by all endpoints.

- Determine the availability of the target endpoint—if a call cannot be completed because the target endpoint is unavailable, SIP determines whether the dialed party is already on the phone, or did not answer within the allotted number of rings. It then returns a message indicating why the target endpoint was unavailable.
- Establish a session between the originating and target endpoints—if the call can be completed, SIP establishes a session between the endpoints. SIP also supports mid-call changes, such as the addition of another endpoint to the conference, or the changing of a media characteristic or codec.
- Handle the transfer and termination of calls—SIP supports the transfer of calls from one endpoint to another. During a call transfer, SIP simply establishes a session between the transferee and a new endpoint (specified by the transferring party) and terminates the session between the transferee and the transferring party. At the end of the call, SIP terminates the session among all parties.

SIP Components

A SIP-based network is made up of several components:

- **SIP user agent**—Any network endpoint that can originate or terminate a SIP session; this might be a SIP-enabled telephone, a SIP PC client (for example Netmeeting), or a SIP-enabled gateway.
- **SIP proxy server**—A call-control device that provides various services, such as the routing of SIP messages between SIP user agents.
- **SIP redirect server**—A call-control device that provides routing information to user agents when requested, giving the user agent an alternate uniform resource identifier (URI) or destination user-agent server (UAS).
- **SIP registrar server**—A device that stores the "logical" location of user agents within a domain or sub-domain; an SIP registrar server stores the location of user agents and dynamically updates data via REGISTER messages.
- **SIP location services**—An additional functionality that can be used by proxy, redirect and registrar servers to identify (via a unique URI) and "logically" locate user agents within the network.
- **Back-to-back user agent**—A call-control device that provides routing similar to a proxy server, but allows centralized control of network call flows. This device allows SIP networks to replicate certain traditional telephony services that require centralized knowledge of device state, such as call park and pickup; this component is always dialogue state-aware.
- **SIP-aware network devices**—Devices that recognize the SIP protocol and allow the network to function more efficiently; this type of device might be a firewall or Network Address Translation (NAT) device that allows SIP traffic to traverse network borders, or a load-balancing switch that enables requests to SIP servers to be handled more efficiently.

SIP vs H.323

- H.323 is not scalable enough to meet the growing needs of IP telephony subscribers. PortaSIP ensures interoperability among standards and provides end-to-end connectivity throughout the network. This allows IP Telephony providers to offer value-added services, and demonstrates the power of IP-based communications.

- PortaSIP gets rid of proprietary protocols and enables a more distributed model of telephony.
- PortaSIP Interoperability, with firewall and Network Address Translation technology (NAT), allows subscribers to use SIP telephones, whereas this is not possible with H.323 technology.