



Open Source IP PBX

The Future of Telephony





What is Asterisk

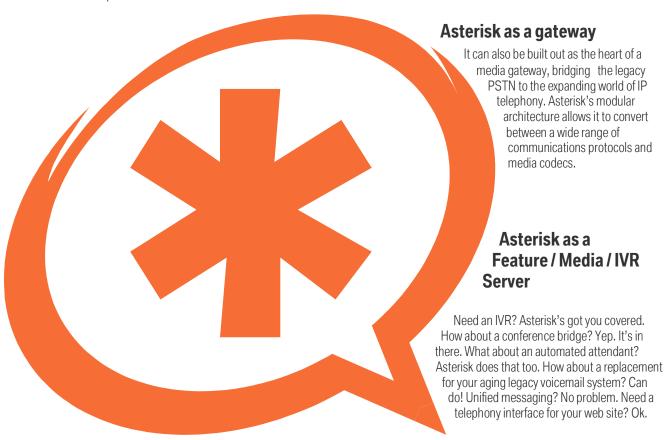
Asterisk is the world's leading open source telephony engine and tool kit. Offering flexibility unheard of in the world of proprietary communications, Asterisk empowers developers and integrators to create advanced communication solutions...for free.

Asterisk® is released as open source under the GNU General Public License (GPL), and it is available for download free of charge. Asterisk® is the most popular open source software available, with the Asterisk Community being the top influencer in VoIP.

Asterisk as a switch (PBX)

Asterisk can be configured as the core of an IP or hybrid PBX, switching calls, managing routes, enabling features, and connecting callers with the outside world over IP, analog (POTS), and digital (T1/E1) connections.

Asterisk runs on a wide variety of operating systems including Linux, Mac OS X, OpenBSD, FreeBSD and Sun Solaris and provides all of the features you would expect from a PBX including many advanced features that are often associated with high end (and high cost) proprietary PBXs. Asterisk's architecture is designed for maximum flexibility and supports Voice over IP in many protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.







Asterisk in the Call center

Asterisk has been adopted by call centers around the world based on its flexibility. Call center and contact center developers have built complete ACD systems based on Asterisk. Asterisk has also added new life to existing call center solutions by adding remote IP agent capabilities, advanced

skills-based routing, predictive and bulk dialing, and more.

Asterisk in the network

Internet Telephony Service Providers (ITSPs), competitive local exchange carriers (CLECS) and even first-tier incumbents have discovered the power of open source communications with Asterisk. Feature servers, hosted services clusters, voicemail systems, pre-paid calling solutions, all based on Asterisk have helped reduce costs and enabled flexibility.

Asterisk everywhere

Asterisk has become the basis for thousands of communications solutions. If you need to communicate, Asterisk is your answer.

Supported Platforms

Asterisk® is primarily developed on GNU/Linux for x/86 and runs on GNU/Linux for PPC along with OpenBSD, FreeBSD, and Mac OS X. Other platforms and standards-based UNIX-like operating systems should be reasonably easy to port for anyone with the time and requisite skill to do so.

Supported Hardware

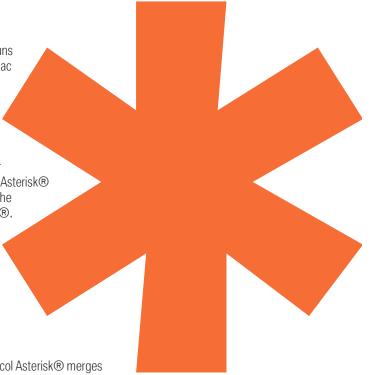
Asterisk® needs no additional hardware for Voice over IP. For interconnection with digital and analog telephony equipment, Asterisk® supports a number of **hardware** devices, most notably all of the hardware manufactured by **Digium®**, the creator of Asterisk®.

Supported Protocols

Asterisk® supports a wide range of protocols for the handling and transmission of voice over traditional telephony interfaces including H.323, Session Initiation Protocol (SIP), Media Gateway Control Protocol (MGCP), and Skinny Client Control Protocol (SCCP).

Using the Inter-Asterisk eXchange (IAX™) Voice over IP protocol Asterisk® merges voice and data traffic seamlessly across disparate networks. The use of Packet Voice allows Asterisk® to send data such as URL information and images in-line with voice traffic, allowing advanced integration of information.

Asterisk® provides a central switching core, with four APIs for modular loading of telephony applications, hardware interfaces, file format handling, and codecs. It allows for transparent switching between all supported interfaces, allowing it to tie together a diverse mixture of telephony systems into a single switching network.





Features

Asterisk-based telephony solutions offer a rich and flexible feature set. Asterisk® offers both classical PBX functionality and advanced features which interoperates with traditional standards-based telephony systems and Voice over IP systems

Call features

ADSI On-Screen Menu System Caller ID Blocking Alarm Receiver Caller ID on Call Waiting Append Message Calling Cards Authentication Conference Bridging Automated Attendant Database Store / Retrieve Blacklists Database Integration Blind Transfer Dial by Name Call Detail Records Direct Inward System Access Call Forward on Busy Distinctive Ring Call Forward on No Answer Distributed Universal Number Call Forward Variable Discovery (DUNDiTM)

Do Not Disturb Call Monitoring Call Parking E911 **ENUM** Call Queuing

Call Recording Fax Transmit and Receive (3rd Party OSS Package) Call Retrieval Call Routing (DID & ANI) Flexible Extension Logic Call Snooping Interactive Directory Listing Call Transfer Interactive Voice Response (IVR) Call Waiting Local and Remote Call Agents Caller ID

Macros

Music On Hold Time and Date Music On Transfer: Transcoding - Flexible Mp3-based System Trunking - Random or Linear Play VoIP Gateways - Volume Control Voicemail: **Predictive Dialer** - Visual Indicator for Privacy

Message Waiting Open Settlement Protocol (OSP) - Stutter Dialtone for Overhead Paging Message Waiting Protocol Conversion - Voicemail to email Remote Call Pickup - Voicemail Groups Remote Office Support - Web Voicemail Roaming Extensions Interface Route by Caller ID Zapateller SMS Messaging

Computer-Telephony Integration

AGI (Asterisk Gateway Interface) Graphical Call Manager Outbound Call Spooling **Predictive Dialer** TCP/IP Management Interface

Scalability

Spell / Say

Streaming Media Access

Text-to-Speech (via Festival)

Supervised Transfer

Three-way Calling

Talk Detection

TDMoE (Time Division Multiplex over Ethernet) Allows direct connection of Asterisk PBX Zero latency Uses commodity Ethernet hardware Voice-over IP Allows for integration of physically separate installations Uses commonly deployed data connections Allows a unified dial plan across multiple offices



Codecs	Protocols	Traditional Telephony Interoperability	PRI Protocols
ADPCM G.711 (A-Law & µ-Law) G.722 G.723.1 (pass through) G.726 G.729 (through purchase of a commercial license) GSM iLBC Linear LPC-10 Speex	IAX™ (Inter-Asterisk Exchange) H.323 SIP (Session Initiation Protocol) MGCP (Media Gateway Control Protocol SCCP (Cisco® Skinny®)	E&M E&M Wink Feature Group D FXS FXO GR-303 Loopstart Groundstart Kewlstart MF and DTMF support Robbed-bit Signaling (RBS) Types	4ESS BRI (ISDN4Linux) DMS100 EuroISDN Lucent 5E National ISDN2 NFAS



Architecture

Asterisk is carefully designed for maximum flexibility. Specific APIs are defined around an advanced, central PBX core system. The advanced core handles the internal interconnection of the PBX, cleanly abstracted from the specific protocols, codecs, and hardware interfaces from the telephony applications which allows Asterisk to use any suitable hardware and technology available now or in the future to perform its essential functions - connecting hardware and

Loadable module APIs

Four APIs are defined for loadable modules, facilitating hardware and protocol abstraction. Using this loadable module system, the Asterisk core does not have to worry about details of how a caller is connecting, what codecs are in use, etc.

Channel API

The channel API handles the type of connection a caller is arriving on, be it a VoIP connection, ISDN, PRI, Robbed bit signaling, or some other technology. Dynamic modules are loaded to handle the lower layer details of these connections.

Application API

The application API allows for various task modules to be run to perform various functions. Conferencing, Paging, Directory Listing. Voicemail, In-line data transmission, and any other task which a PBX system might perform now or in the future are handled by these separate modules.

Codec Translator API

Loads codec modules to support various audio encoding and decoding formats such as GSM, Mu-Law, A-law, and even MP3.

File Format API

Handles the reading and writing of various file formats for the storage of data in the file system.

Items handled by core internally

PBX Switching

The essence of Asterisk, of course, is a Private Branch Exchange Switching system, connecting calls together between various users and automated tasks. The Switching Core transparently connects callers arriving on various hardware and software interfaces.

Application Launcher

Launches applications which perform services for uses, such as voicemail, file playback, and directory listing.

Codec Translator

Uses codec modules for the encoding and decoding of various audio compression formats used in the telephony industry. A number of codecs are available to suit diverse needs and arrive at the best balance between audio quality and bandwidth usage.

Scheduler and I/O Manager

Handles low-level task scheduling and system management for optimal performance under all load conditions.





Using these APIs, Asterisk achieves a complete abstraction between its core functions as a PBX server system and the varied technologies existing (or in development) in the telephony arena. The modular form is what allows Asterisk to seamlessly integrate both currently implemented telephony switching hardware and the growing Packet Voice technologies emerging today. The ability to load codec modules allows Asterisk to support both the extremely compact codecs necessary for Packet Voice over slow connections such as a telephone modem while still providing high audio quality over less constricted connections.

The application API provides for flexible use of application modules to perform any function flexibly on demand, and allows for open development of new applications to suit unique needs and situations. In addition, loading all applications as modules allows for a flexible system, giving administrators the ability to design the best suited path for callers on the PBX system and modify call paths to suit changing communication needs.

About Digium

Digium®, Inc., the Asterisk company, is the original creator and primary developer of Asterisk, the industry's first open source telephony platform. Digium provides quality hardware and software products, including AsteriskNOW™, the complete open source software appliance; Asterisk Business Edition™, the professional-grade version of Asterisk; the AA50, the Asterisk Appliance™ hardware-based telephony solution; and Switchvox, a complete turn-key IP PBX solution, to enterprises and telecommunications providers worldwide. Digium also offers a full range of professional services, including consulting, technical support, and custom software development. All of Digium's commercially offered products come with the Exceptional Satisfaction Program™ (ESP), the only 100% customer satisfaction guarantee in the open source telephony world today.

Used in combination with Digium's telephony interface cards, Asterisk offers a strategic, highly cost-effective approach to voice and data transport over IP, TDM, switched and Ethernet architectures. Digium's offerings include VoIP, conferencing, voicemail, legacy PBX, IVR, auto attendant, media servers and gateways, and application servers and gateways.

About Brismark

Established as a Privately held Company in 2008, Brismark (Pvt) Limited is a Technology Distribution and Marketing company serving wide variety of corporate, government, educational, health and public service sectors.

Initially, the Company's focus is on providing Communication, Software and Consulting Services to the diversified industry segments. Leveraging on the experience gained from successfully serving this market, we offer turnkey implementations for IP-PBX, IVR, Call Center, Software, and Consulting.

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