Brismark
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Asterisk™

Open Source IP PBX

The Future of Telephony

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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 as a gateway

It can also be built out as the heart of a media gateway, bridging the legacy PSTN to the expanding world of IP telephony. Asterisk's modular architecture allows it to convert between a wide range of communications protocols and media codecs.

Asterisk as a Feature / Media / IVR Server

Need an IVR? Asterisk's got you covered. How about a conference bridge? Yep. It's in there. What about an automated attendant? Asterisk does that too. How about a replacement for your aging legacy voicemail system? Can do! Unified messaging? No problem. Need a telephony interface for your web site? Ok.

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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.

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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
Alarm Receiver
Append Message
Authentication
Automated Attendant
Blacklists
Blind Transfer
Call Detail Records
Call Forward on Busy
Call Forward on No Answer
Call Forward Variable
Call Monitoring
Call Parking
Call Queuing
Call Recording
Call Retrieval
Call Routing (DID & ANI)
Call Snooping
Call Transfer
Call Waiting
Caller ID
Caller ID Blocking
Caller ID on Call Waiting
Calling Cards
Conference Bridging
Database Store / Retrieve
Database Integration
Dial by Name
Direct Inward System Access
Distinctive Ring
Distributed Universal Number
Discovery **(DUNDi™)**
Do Not Disturb
E911
ENUM
Fax Transmit and Receive (3rd
Party OSS Package)
Flexible Extension Logic
Interactive Directory Listing
Interactive Voice Response (IVR)
Local and Remote Call Agents
Macros
Music On Hold
Music On Transfer:
- Flexible Mp3-based System
- Random or Linear Play
- Volume Control
Predictive Dialer
Privacy
Open Settlement Protocol (OSP)
Overhead Paging
Protocol Conversion
Remote Call Pickup
Remote Office Support
Roaming Extensions
Route by Caller ID
SMS Messaging
Spell / Say
Streaming Media Access
Supervised Transfer
Talk Detection
Text-to-Speech (via Festival)
Three-way Calling
Time and Date
Transcoding
Trunking
VoIP Gateways
Voicemail:
- Visual Indicator for
Message Waiting
- Stutter Dialtone for
Message Waiting
- Voicemail to email
- Voicemail Groups
- Web Voicemail
Interface
Zapateller

Computer-Telephony Integration

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

Scalability

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

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Codecs

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

Protocols

IAX™ (Inter-Asterisk Exchange)
H.323
SIP (Session Initiation Protocol)
MGCP (Media Gateway Control Protocol
SCCP (Cisco® Skinny®)

Traditional Telephony Interoperability

E&M
E&M Wink
Feature Group D
FXS
FXO
GR-303
Loopstart
Groundstart
Kewlstart
MF and DTMF support
Robbed-bit Signaling (RBS) Types

PRI Protocols

4ESS
BRI (ISDN4Linux)
DMS100
EuroISDN
Lucent 5E
National ISDN2
NFAS

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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.

Brismark is a wholly owned subsidiary of MAGSNET LIMITED.

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