

## About Digium

Digium™ is the creator and primary developer of Asterisk™, the industry's first Open Source PBX.

Used in combination with Digium's PCI telephony interface cards, Asterisk offers a strategic, highly cost-effective approach to voice and data transport over TDM, switched, IP, and Ethernet architectures.

Digium solutions reduce the costs of traditional TDM and VoIP implementations through open source, standards-based software and innovative hardware solutions, including legacy PBX, IVR, auto-attendant, and next-generation gateways, media servers, and application servers.

Digium hardware supports traditional voice protocols, including PRI, RBS, FXS, FXO, E&M, Feature Group D, Groundstart, and Loopstart. Data protocols include PPP, Cisco HDLC, and Frame Relay. For packet voice, Asterisk supports IAX (Inter-Asterisk Exchange), SIP, MGCP, Skinny, and H.323 VoIP protocols.

Digium provides a highly refined selection of quality hardware and software products, developed and implemented using innovative engineering techniques (primarily open source development). A full range of professional services complement these product lines, including consulting, technical support, and custom software development services. The open source communications revolution is here, and Digium is leading the way.



### TE405P & TE410P

Quad-Span toggable E1 and T1 card enables per card or per port selection of either T1 or E1 signaling formats.

### T100P & E100P

Single-Span T1 or E1 half-length (available with 2U bracket) PCI card sporting the same features as the Quad-Span T1/E1 cards.

### TDM400P

Quad-Port half-length PCI card. Mix and match FXS and FXO modules to support standard analog or ADSI telephones and regular POTS lines.

### S100I "IAXy"

Single CPE fully featured FXS interface with an Ethernet back-end, speaking the Asterisk-native IAX™ protocol.

For data sheets on the above products, visit *Hardware Products* on the Digium website.

  
digium | Asterisk  
The Open Source PBX

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Say Hello to

# Asterisk

**Say Hello to Asterisk,  
the open source  
telecommunications platform.**

The days of expensive proprietary telecom software are over. Now there's the free, open technology of Asterisk™.

Since Asterisk is Linux-based, it inherits all of the power and stability of the operating system. Linux provides open source alternatives to proprietary applications. Asterisk is the first package to fit all telecommunication needs in a broad variety of environments.

The name Asterisk is derived from the all-inclusive "wildcard" symbol in UNIX®, because the Asterisk platform is providing opportunities for developers worldwide to create solutions which would otherwise be cost-prohibitive or impossible.

Asterisk is global, supporting all sorts of applications, like call centers around the world, IVR platforms in Australia, and university VoIP switches in Europe.



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# Features

## 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
- 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
- Trunking
- 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 & Date
- Transcoding
- VoIP Gateways
- Voicemail
  - Visual Indicator for Message Waiting
  - Stutter Dialtone for Message Waiting
  - Voicemail to email
  - Voicemail Groups
  - Web Voicemail Interface
- Zapatero



## 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 installation
  - Uses commonly deployed data connections
  - Allows a unified dialplan across multiple offices

## Codecs

- ADPCM
- G.711 (A-Law & μ-Law)
- G.723.1 (pass through)
- G.726
- G.729 (through purchase of 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 and 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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# Applications

## PBX

Asterisk allows you to create a PBX that rivals the features and functionality of traditional telephony switches. Other PBXs are expensive, proprietary, and now passé. Asterisk is cost-effective, low-maintenance, and flexible enough to handle all voice and data networking. With Asterisk, Digium hardware, and a common PC, anyone can replace an existing switch or complement a PBX by adding VoIP, voicemail, conferencing and many other capabilities. Asterisk integrates with most standards-based IP telephone handsets and software. Analog phones and ADSI screen phones are also supported. Proven through installations around the globe, Asterisk is solid and stable in SOHO or Enterprise environments.

## Interactive Voice Response (IVR)

Asterisk's flexible IVR capability allows a user to interact with a database using a menu of pre-recorded voice-clips. Using MySQL and other popular databases, Asterisk can interact with the caller through touch tone inputs, record responses, query databases, and utilize AGI scripts to perform specific tasks.

For example, a customer can authenticate a pre-paid calling card with a PIN queried from a database. The Asterisk IVR will give the number of remaining minutes and later disconnect if that customer runs out of time. The spooling feature can allow Asterisk to dial a list of numbers from a database to give warnings during homeland security emergencies. Asterisk IVR allows developers to create a myriad of IVR solutions.

## Auto-Attendant

Asterisk's auto-attendant features include greetings, extended greetings, music-on-hold, voice message forwarding and message appending. Asterisk plays music or pre-recorded messages to customers on hold. Music can be sorted into various folders. Separate auto-attendant feature sets can be used for different situations. The voicemail tree supports directories by department, employee, extension, etc., offering flexibility and allowing a small company to appear large. Unbound by the limits of traditional voicemail, Asterisk can support an unlimited number of simultaneous ports.

## Conference Bridge

The Meet Me Bridge is fully integrated into Asterisk and supports features essential for business conferences, saving the Asterisk user from what was once a huge expense. The Conference Chairperson can select a "listen only" or a "talk and listen" conference. When the Chairperson hangs up the other parties are disconnected. Conferences may be securely accessed only through a pre-defined PIN.

Many companies have also created chat services based on Asterisk Meet Me, with many chat rooms that users can transfer between.

## Media Server

Asterisk augments existing PBXs and Gateways with select features for either PSTN or IP protocols. Acting as an adjunct to a legacy system or soft switch, Asterisk can extend features and functionality by providing voicemail and conferencing services. Asterisk can also retrofit traditional TDM PBXs with VoIP extensions to remote offices which appear as normal extensions of the PBX.

## VoIP and Protocol Gateway

Asterisk's broad support of both traditional TDM and VoIP protocols permits the construction of flexible gateways between different channel types. Using Asterisk, it is not only easy to create many common varieties of protocol converters, translating between T1, E1, PRI, SIP, IAX, GR-303, MGCP, FXS, and many others. It enables you to create more sophisticated gateways and gateways with redundant links. For example, an MGCP to SIP gateway with a PRI backup can be created in case SIP trunks are unavailable—the possibilities are nearly endless.

## VoIP Switch

Asterisk can act as a soft switch in addition to acting as a traditional TDM switch, allowing it to control a variety of devices including phones, gateways, media servers, and other Asterisk servers. It can handle virtually any VoIP protocol, including SIP, IAX, H.323, MGCP, and Skinny.

Asterisk collects call detail records and provides a variety of billing options (including Open Settlement Protocol) and may be configured to carry media (especially useful for SIP+NAT situations) or to have devices send media directly to one another. Asterisk adds extra IP or PSTN capabilities to existing PBXs and Gateways. Asterisk extends features and functionality by providing voicemail and conferencing services when acting as an adjunct to a legacy system or soft switch. Asterisk can also add remote VoIP office extensions to traditional TDM PBXs which appear as normal extensions from the preexisting PBX.

