

This is a list of the current features based on release 1.2.7.1

Asterisk™ Features

Asterisk-based telephony solutions offer a rich and flexible feature set. Asterisk offers both classical PBX functionality and advanced features, and interoperates with traditional standards-based telephony systems and Voice over IP systems. Asterisk offers the features one would expect of a large proprietary PBX system such as Voicemail, Conference Bridging, Call Queuing, and Call Detail Records.

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 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 through [Digium](#))
- 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
- MFC-R2 (Not supported by Digium. However, a patch is available at [Voip-info.org](#))

PRI Protocols

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



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