List of Features

System Application Services

All the sipXcom application services are allocated to specific server roles. Using the centralized cluster management system each role can be instantiated on a dedicated server or several (all) roles can be run on a single server. Configuration of all services and participating servers is fully automatic and Web UI based.

- SIP Session Router, optionally geo-redundant and load sharing
- Media server for unified messaging and IVR (auto-attendant) services
- Conferencing server based on FreeSWITCH
- XMPP Instant Messaging (IM) and presence server (based on Openfire)
- Contact center (ACD) server
- Call park / Music on Hold (MoH) server
- Presence server (Broadsoft and IETF compliant resource list server for BLF)
- Shared Appearance Agent server to support shared lines (BLA)
- Group paging server
- SIP trunking server (media anchoring and B2BUA for SIP trunking & remote worker support)
- Call Detail Record (CDR) collection & processing server
- Third party call control (3PCC) server using REST interfaces
- Management and configuration server
- Process management server for centralized cluster management

SOA Architecture / Business Process Integration using Web Services

- Web Services SOAP interfacefor key administrative functions
- Web Services REST interface for user portal functions and third party call control
- All components centrally managed using XML RPC
- Google Web Toolkit (GWT)

Core Calling Features (Telephony Features)

- Transfer (consultative & blind)
- Call coverage
- Call hold / retrieve
- Consultation hold
- Music on Hold for IETF standards compliant phones
- User-specific MoH files
- MoH music from an external streaming source
- Admin or user configurable Busy Lamp Field (BLF) presence and softkeys
- New: Shared lines (BLA) for boss - secretary applications (Polycom only)
- Uploadable music file
- 3-way / 5-way video and voice conference on the phone
- Call pickup (global and directed call pickup)
- Call park & retrieve
- Hunt groups
- Intercom with auto-answer (bi-directional)
- SIP URI dialing
- CLID (Calling Line Identification)
- CNIP (Calling party Name Identification Presentation)
- CLIP (Call Line Identification Presentation)
- CLIR (Call Line Identification Restriction)
- Per gateway CLIP manipulation
- Call waiting / retrieve
- Do not Disturb (DnD)
- Forward on busy, no answer, do not disturb
- Multiple line appearances
- Multiple calls per line
- Multiple station appearance
- Outbound call blocking - Calls from phones to PSTN numbers, or classes of numbers, can be blocked based on:
    - The destination of the call; for example, when a user or device cannot initiate an international long distance call.
    - The source of the call; for example, when a lobby phone can only initiate calls to internal numbers.
- Click-to-call
- Redial
- Call history (dialed, received, missed)
- Auto off-hook / ring down
- Incoming only
- Configuration of individual Speed Dial softkeys
- Auto-generation of directory information

E911 Emergency Response

- Internal notification using email and SMS

Remote Branch office support

- Centralized deployment: Branch only provides phones and optionally PSTN gateway for failover, reduced WAN BW consumption or E911 calls
- Distributed deployment: Branch provides full call server with SIP site-to-site dialing between offices
- Branch office locations can be defined in the mgmt UI with a postal address
- Users, phones, gateways, SBC, and servers can be assigned to a branch location
- A PSTN gateway can be available for calls that originate in a specific branch only or for general use
- Source routing allows call routing based on location (branch local calls are routed through local gateway preferably)
- Branch postal address automatically proliferates to user's office address
- Survivable branch configuration possible with Audiocodes gateways SAS functionality (auto-configured)
- Certain sipXcom services can be deployed in the branch as part of the cluster (e.g. conferencing)

Enterprise Instant Messaging (IM) and Presence

- XMPP based IM and presence server based on Openfire
- Supports XMPP standards based clients
- Auto-configuration of user's IM accounts
- Auto-configuration of IM user groups
- Personal group chat room for every user auto-configured
- Federation of phone presence with IM presence
- Customizable "on the phone" presence status message
- Dynamic call routing based on user's presence status
- Message archiving and search for compliance (pending)
- Server-to-server XMPP federation
- Optional secure client connections
- Client-to-client file transfer
- Group chat rooms
- XMPP search
- Integration of user profile information and avatar (pending)

Personal Assistant IM Bot

- My Buddy Personal Assistant feature
- Dynamic call control using IM
- Dynamic conference management using IM
- Unified messaging management using IM
- Call history / missed calls
- Call initiation using corporate dialplan
- Corporate directory look-ups

Presence and IM Federation

- Server side federation with other public XMPP IM systems
- Allows group chat sessions across systems
- Allows message archiving (if enabled) across systems
- User self-administration of credentials for other IM systems

Fixed Mobile Convergence (FMC) Application

- 3rd Party FMC application with the following functionality:
  - Enterprise number dialing
  - System call-back saves on wireless toll charges
  - Corporate directory look-ups
  - Call history
  - Presence sharing
  - IM

Web Conferencing & Collaboration

- Commercial options available through eZuce's view**me** and view**me** Cloud products

User Self-Control (User Web Configuration Portal)

- Every user on the system gets access to a personal Web user portal for self-management and control
- Management of unified messaging (voicemail)
- Configuration of unified messaging preferences
- Time based find-me / follow-me
- Flexible configuration of call forwarding
- Management of personal profile data including avatar
- Personal call history

- Personal phone book, speed dial and presence management
- Click-to-call
- Individual phone management
- Personal auto-attendant
- Management of personal IM account
- Personal MoH music upload and preferences

Superior Voice Quality

- Peer-to-peer media routing for best quality (media not routed through the sipXcom server)
- Unmatched voice quality with lowest delay and jitter
- Support for any codec supported by the phone or gateway (including video)
- Support for HD Voice (Polycom and other phones)
- Codec negotiation (no transcoding required)
- Conferencing, auto-attendant and voicemail support HD voice w/ transcoding if necessary

User Management

- Create a user, provision a phone and assign a line in only three clicks - easy!
- Numeric or alpha-numeric User ID
- User PIN management (UI or TUI)
- Aliasing facility (numeric and alpha-numeric aliases)
- Extension and alias uniqueness assurance
- Management or auto-assignment of user's IM ID and display name
- Automatic IM buddy list creation based on user groups
- Granular per user permissions
- Call permissions:
    - 900 Dialing
    - International Dialing
    - Long Distance Dialing
    - Mobile Dialing
    - Local Dialing
    - Toll Free Dialing
- System permissions:
    - User has voicemail inbox
    - User listed in auto-attendant directory
    - User can record system prompts
    - User has superuser access
    - User allowed to change PIN from TUI
    - User can use Microsoft Exchange VM
    - User has a personal auto-attendant
- Custom permissions as defined by the admin
- Supervisor permission for groups (e.g. Call Center supervisor)
- Management of user contact record (user profile)
- Comprehensive profile data
- Work and home address
- In-building location information
- Assistant information
- Support for avatar including support for gravatar
- SIP password management for security
- User groups with group properties
- Per user call forwarding (follow me)
    - To local extension, PSTN number, or SIP address
    - Based on user or admin defined time schedules
    - Parallel or serial ring
    - Allows definition of ring time before trying next number
    - Allows several forwarding destinations
    - Follow-me configuration using user portal
- Extension pool with automatic assignment
- Per user Caller ID (CLID) assignment

Dial Plan

- Easy to use GUI based dial plan manipulation
- Time-based dialing rules with different admin defined schedules
- Rules based least cost routing
- Dynamic call routing based on user's IM presence status
- Directly route to voicemail on IM status DND
- Dynamically add forwarding destination based on phone number in custom presence status
- Automatic gateway redundancy and fail-over
- Specific E911 routing
- Permission based rules
- Prefix manipulation
- Dialplan templating for international dial plans
- Built-in support for U.S., German, Swiss, and Polish local dial plans
  (Any other local dial plan can be added as a plugin)

- Specify internal extension length
- Specific rule for site-to-site call routing between SIP systems
- Redirector plugins - any imaginable dial rule can be added as a plugin

Internet Calling

- Ability to configure SIP URI based call routing to other domains
- Specific SBC selection for call routing
- Configuration of native NAT traversal w/ optionally redundant media anchoring if necessary
- Media anchoring supports voice and video for any codec

Directory, Softkeys, Speed Dial

- Automated generation of directory information per user or per user group
- Support for complete contact information and user profile, including avatar
- Crreation and Management of many different directories (per user, per user group, per location, etc.)
- Upload of contacts from GMail and Outlook
- User management of directory information
- Automated provisioning of directory information into user's phones
- Allows adding contacts to the directory from a .csv file (Excel)
- User configurable speed dial (internal / external numbers, SIP URIs)
- Speed dial generated server side and backed up
- Auto-provisioning of speed dial to phones
- User configuration of Busy Lamp Field (BLF) to monitor presence of other users or phones (e.g. attendant console)

PSTN Trunking

- Unlimited number of PSTN gateways and trunk lines
- Supports most SIP compliant gateways (e.g. Audiocodes, Mediatrix, Sangoma, Patton, etc.)
- Gateways can be in any location
- Gateway selection per dialing rule
- Source routing of calls so that calls can be routed through a local gateway to save WAN bandwidth
- DID
- Local DID per gateway
- DNIS
- CLIP Management
    - User CLIP
    - Gateway default CLIP
    - Prefix stripping / appending
- Per gateway CLIR
- Automatic Route Selection (ARS)
    - Implemented with XML-formatted mapping rules.
    - Mapping values re-write SIP URLs to specify the next hop or destination for a SIP message that has been received by the Communications Server component.
    - Direct messages to different SIP/PSTN trunk gateways, either on premise or at a remote premise location, based on any portion of SIP URL or E.164 number.
    - Route messages to commercial SIP/PSTN service providers, which reduces or eliminates the need for on-premise trunk gateways.
- Least-cost routing (LCR)
- Automatic failover if unavailable
- Automatic failover if busy
- Inbound FAX support
- Mixing of PSTN and SIP trunks with least cost routing

SIP Trunking

- Basic SIP trunking gateway w/ NAT traversal
- Remote worker support w/ near-end and far-end NAT traversal and auto-detection
- ITSP templates for simplified configuration
- Interop (not certified) with the following ITSPs:
    - BT (UK)
    - AT&T
    - Bandwidth.com
    - CBeyond
    - Bandtel
    - CallWithUs
    - Eutelia (Italy)
    - LES.NET
    - SIPcall (Switzerland)
    - Vitality
    - VOIPUser (UK)
    - VOIP.MS
    - Appia

- Easy configuration templates exist for the above ITSPs

- Many other ITSPs are compatible, see ITSP interop in Wiki

- SIP interop with Nortel CS1000 R6
- SIP call origination & termination
- Branch office routing
- Proxy to proxy interconnect using ACLs
- Least-cost-routing (LCR)
- Mixing of PSTN trunks with SIP trunks
- TLS support for secure signaling
- Route header for flexible call routing through an SBC
- Flexible rules for SBC selection (route selection)
- Support for Skype for Business SIP trunking

Integration with Microsoft Active Directory and Exchange

- Synchronization with Microsoft Active Directory
  - Using LDAP interface
  - On demand or automatically based on a schedule
  - Graphical query design combines ease of use with flexibility
  - Allows preview of records to be imported
- Dialplan integration with Microsoft Exchange voicemail server
  - Allows mixed environment with groups of users on Exchange or the sipXcom VM server
  - Permission based selection of VM server per user or user group
  - Automatic dialplan routing to Exchange VM
- Enables sll speech based Exchange capabilities

Supported Softclients

- Combined SIP / XMPP clients:
  - Counterpath Bria professional
  - Jitsi

- Provisioning server for automated mass deployment
- Automated SIP and XMPP account setup
- Call recording
- Supports BLF (workgroups)
- Scheduled to support BLA
- Automatic user profile and directory management
- XMPP clients:
  - Pidgin
  - Google Talk
  - Trillium
  - Spark
- SIP clients:
  - 3CX softphone

Analog Lines (FXS)

- Supports any SIP compliant FXS gateway
- FAX support
- Analog cordless phone support
- Supports analog Polycom speakerphones
- Plug & play management of FXS gateways from Audiocodes and Grandstream

Performance

- Unlimited number of simultaneous calls (voice, HD voice, video) - only depends on LAN/WAN bandwidth
- 54,000 BHCC, 120,000 BHCC two-way redundant (depends on server HW)
- Up to three-way redundant configuration using cluster mgmt Web GUI
- Up to 10,000 users per dual-server HA system
- Tested up to 10,000 IM users
- 450 simultaneous calls through the SIP trunking gateway require < 20% CPU on dual core system
- Up to 500 simultaneous conferencing ports per server
- Up to 300 media server ports for unified messaging (supports 15,000 users)
- Automatic time distribution of re-registration and subscription events

High Availability

- Optionally fully redundant call control system
- Geo-redundant SIP session manager
- Based in DNS SRV (no cluster required)
- Load balance under normal operating conditions

- Geographic dispersion of redundant systems
- Real-time synchronization of state information
- Automatic recovery after server failure
- Reports on load distribution

Call Detail Records collection and reporting

- Call State Events (CSE) collected for all signaling activity
- Processing of CSEs into CDRs
- All data stored in a database at all times
- Flexible report generation using Jasper Reports, built-in
- Supports redundant call control
- Determines and records call type information
- Internal / external calls
- Calls to specific sipXcom services
- Collates call legs
- Historic Call Detail Record reporting in real-time
- Additional reports using call type info
- Monitoring of currently active (on-going) calls
- Export of active and historic CDRs to Excel (.csv file)
- Direct database access for reporting application (e.g. Crystal Reports, Jasper Reports)
- SOAP Web Services access to CDR data
- Individual call history per user in the user portal

Security

- All outbound calls authenticated
- Secure user password management
- DoS attack prevention
- HTTPS secure Web access
- TLS based signaling for SIP trunks
- HTTPS secures non-SIP communication between sipX components.
- HTTPS secures communications between sipX components and admin and user consoles.
- Secure channel for retrieving messages from voicemail repository.
- HTTP digest authentication for SIP signaling, as specified in RFC 2617, is used for authentication challenges between SIP endpoints and sipX components.
- HTTP digest implementation supports MD5.

System Administration Features

- Browser based configuration and management
- Several admin accounts
- Notification when new version or patches are available
- GUI based software upgrade
- GUI based certificate management
- LDAP integration
- Integration with Microsoft Exchange 2007 for voicemail and Active Directory
- SOAP Web Services interface
- CSV import and export of user and device data
- Administration of Instant Messaging (IM) and Presence settings
- Integrated backup & restore
- Scheduled backups
- Diagnostics
    - Display active registrations
    - Display job status
    - Status of services
    - Snapshot logs for debugging
    - Logging (customizable log levels, message log per service)
    - Display active calls
- Domain Aliasing
- Support for DNS SRV
- Support for DNS NAPTR based call routing
- Automatic restart after power failure
    - Single sipXcom application can start all other application processes associated with starting up sipXcom, including dependent processes that must be started in particular order.
    - Configured from browser interface
- Login history report (successful and unsuccessful)
- Automated testing of network services (DHCP, DNS, NTP, TFTP, FTP, HTTP) for proper configuration

Plug & Play Device Management

- Auto-discovery of phones & gateways on the LAN
- Auto-registration of Polycom phones simplifies installation
- Plug & play management of phones

- Plug & play management of PSTN gateways
- Auto-generation of phone / gateway config profile
- Auto-pickup of profile by phone / gateway
- Centralized management of all the parameters
- Centralized backup and restore of all the configs
- Auto-generation of lines by assigning users to devices
- Device group management & properties
- Firmware upgrade management

Unified Messaging (Voicemail)

- Integrated unified messaging system
- Localized per user by installing language packs
- Number of voicemail boxes only limited by disk size (tested up to 10,000)
- Performance tested up to 300 simultaneous calls (ports) on dual core server
- IMAP back-end connection
- Acts as an IMAP client into MSFT Exchange and other compatible email systems
- User manageable credentials for IMAP federation
- Properly controls MWI on the phone when message is "read" using the email client
- Browser based user portal for unified messaging management
- RSS feed for new messages
- Message Waiting Indication (MWI)
- User configurable distribution lists
- Group and system distribution lists
- Unified Messaging:
- Email notification of new voicemail messages
- Forwarding of message as .wav file
- Supports several parallel notifications
- IMAP client into Exchange
- Per user selectable templates for email format used when forwarding voicemail
- Manage folders: Folders for message organization
- Manage greetings: Multiple customizable greetings
- Operator escape from anywhere
- Remote voicemail access using a phone
- SOA Web Services (REST) access to messages and greetings
- Unlimited number of inboxes
- Auto-removal of deleted messages

Personal Auto Attendant

- User configurable personal auto-attendant for every user on the system
- Up to 10 individual forwarding choices (keys 0 through 9)
- User can record greeting that corresponds with key configuration
- Individual zero-out to a personal assistant or receptionist
- Individual selection of language based on installed language packs
- Personal greeting

Auto Attendant Features

- Unlimited number of auto-attendants
- Dial by extension and name
- Night and holiday service
- Special auto-attendant
- Transfer on invalid response
- Nested auto-attendants (multi-level)
- Fully customizable actions:
    - Operator
    - Dial by Name
    - Repeat Prompt
    - Voicemail login
    - Disconnect
    - Auto-Attendant
    - Goto Extension
    - Deposit Voicemail
- Uploadable custom prompts
- Configurable DTMF handling

Presence Server Features

- Compatible with Broadsoft or IETF implementations
- Centralized management of resource lists for dialog events
- Busy Lamp Field (BLF) feature based on presence
- Used to support shared lines (BLA)
- Presence federated with IM presence to show "on the phone" status

- Support for 3rd party Attendant Consoles (such as Voice Operator Panel)

Hunt Groups

- Unlimited number of hunt groups
- Serial and parallel forking (rings sequentially or at the same time)
- Configurable ring time per attempt
- Enable / disable user call forwarding rules while hunting
- Flexible configuration of destination if no answer

Call Park Server

- Unlimited number of park orbits
- Visual indication on the phone of the state of the park orbit using the presence server (BLF)
- Music on park
- Uploadable music file
- Configurable call retrieve code
- Configurable call retrieve timeout
- Automatic park timeout with configurable time
- Configurable park escape key
- Allow multiple calls on one orbit

Group Paging Server

- Integrated group paging server
- Unlimited number of paging groups
- Supports regular SIP phones using auto-answer
- Supports dedicated in-ceiling devices (SIP)
- Configurable paging prefix

Conferencing Server

- Voice conferencing server that can run on the same sipXcom server or on dedicated hardware
- Support for voice conferencing
- Each user on the sipXcom system can have a personal conference bridge
- Recording of conference calls
- Dynamic conference controls from the user's Web portal (user portal)
- Dynamic conference control using IM
- Participant entry / exit messages
- Roll call
- Mute, isolate, disconnect, invite
- Association of personal conference bridge with personal group chat room
- Automatic migration of group chat to a voice conference using the @conf directive
- Support for HD Audio and transcoding if necessary
- Support for up to 500 ports of conferencing, dependent on hardware
- Configurable DTMF keys for conference controls using the TUI
- A sipXcom IP PBX system can have more than one conference server if more capacity is needed
- All conferencing servers and services centrally managed and configured
- Conferencing based on FreeSWITCH

Call Queueing (ACD)

- ACD server collocated or on a different server hardware
- Several (unlimited) queues per server
- Several lines per queue
- Support trunk lines (many calls per line) or single call per line
- Dedicated overflow queues or overflow to hunt group or voicemail
- Configurable call routing scheme per queue:
  - Ring all
  - Circular
  - Linear
  - Longest idle
- Agent presence monitor using presence server
- Separate welcome and queue audio
- Call termination tone or audio
- Configurable answer mode
- Agent wrap-up time
- Auto sign-out of agents if calls are not answered
- Configurable maximum ring delay
- Configurable maximum queue length
- Configurable maximum wait time until overflow condition
- Unlimited number of agents per queue

sipXcom Managed Devices

Almost any SIP compatible phone works with sipXcom if configured manually (i.e. by logging into the phone's Web interface to configure it one phone at a time). The following devices are plug & play managed automatically and centrally by sipXcom:

- Polycom SoundPoint all models (IP 301, 320, 330, 430, 450, 501, 550, 560, 601, 650, 670)
- Polycom SoundStation IP 4000, 6000, 7000 SIP
- Polycom VVX phones (300/310, 400/410, 500, 600, 1500)
- Audiocodes gateways MP112, MP114, MP118, MP124 FXS
- Audiocodes gateways FXO and PRI/BRI
- Counterpath Bria Professional

sipXcom Managed Devices (Community supported)

Community supported means that the phone plugin for plug & play management is provided as is. These phone plugins are provided and maintained by community members. Some system functionality might not be implemented or supported.

- Aastra 53i, 55i, 57i
- Snom 300, 320, 360, 370 up to firmware 7.x
- Grandstream BudgeTone, HandyTone
- Grandstream GXP2000, GXP1200, GXP2010, GXP2020
- Grandstream GXV3000 Video Phone
- Hitachi IP3000 and IP5000 WiFi phones
- Cisco ATA 186/188
- Cisco 7960, 7940, 7912, 7905
- Cisco 7911, 7941, 7945, 7961, 7965, 7970, 7975
- ClearOne MaxIP Conference Phone
- LG-Nortel LG 6804, 6812, 6830
- Nortel video phone 1535
- Linksys ATA 2102, ATA 3102
- Linksys SPA8000
- Linksys SPA901, SPA921, SPA922, SPA941, SPA942, SPA962
- Nortel 1120 / 1140 SIP
- G-Tec AQ10x, HL20x, VT20x

Required Hardware

- Intel / AMD x86 compatible server
- Min RAM 4 GB or more
- Linux operating system (RHEL, CentOS)
- 64 bit versions available
- No special HW required, sipXcom uses external gateways

Installation and Upgrades

- Automated installation from CD ISO for OS and sipXcom IP PBX application
- Graphical configuration wizard for system configuration after installation
- Self-signed Certificate generation (allows installing a signed certificate if desired)
- GUI based upgrade management from the admin Web interface
- Standard Linux package management (e.g. up2date and yum)
- Optional auto-configuration of DNS, DHCP, NTP, FTP, TFTP, HTTP servers
- Designed so that no Linux admin skills are required for installation and configuration

Centrally Managed sipXcom Distributed System (cluster)

- Automated installation and configuration of a distributed system with specific server roles
- Automated and central configuration of a high-availability redundant sipXcom system
- Allows for dedicated server hardware for conferencing, voicemail, ACD Call Center, and Call Control
- All configuration for remote servers is centrally generated and distributed securely

SIP Implementation

This is probably quite an incomplete list. In any case, sipXcom IP PBX is fully SIP standards compliant.

- RFC 3261 Session Initiation Protocol using both UDP and TCP transports
- Advanced call control using RFCs
    - RFC 3515 Refer Method
    - RFC 3891 Referred-By header
    - RFC 3892 Replaces header
- Provide for consultative and blind transfer and third party call controls
    - Blind transfer (Unannounced) to a different phone without speaking to the other phone prior to transfer.
    - Consultative transfer (announced) to a different phone without speaking to the other phone prior to transfer.

    - Consultative transfer (announced) to a different phone after speaking to the other phone prior to completing the transfer. (Consultative transfers require a SIP phone that supports this feature)
- RFC 3263 Locating SIP Servers - use of DNS SRV records for call routing control and server redundancy.
- RFC 3581 Symmetric Response Routing (rport)
- RFC 3265 SIP Event Notification - for phone configuration and
- RFC 3842 Voice mail message waiting indication (MWI)
- RFC 3262 Reliable Provisional Responses
- RFC 2833 Out-of-band DTMF tones
- RFC 3264 Offer/Answer model for SDP for Codec Negotiation
- RFC 2617 HTTP Authentication: Basic and Digest Access Authentication
- RFC 3327 Path header
- RFC 3325 P-Asserted identity
- RFC 4235 An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)
- RFC 4662 A Session Initiation Protocol (SIP) Event Notification Extension for Resource Lists
- RFC 2327 SDP: Session Description Protocol
- RFC 3326 The Reason Header Field for the Session Initiation Protocol (SIP)
- Early media (SDP in 180/183)
- Delayed SDP (SDP in ACK)
- Re-INVITE: Codec change, hold, off-hold
- Route/Record-Route header fields
- Configurable RTP/RTCP ports
- Configurable SIP ports
- BLA support

- RFC 3680: A Session Initiation Protocol (SIP) Event Package for Registrations
- RFC 3265: Session Initiation Protocol (SIP)-Specific Event Notification
- draft-ietf-sipping-dialog-package-06
- draft-anil-sipping-bla-02

XMPP Compliance

- RFC 3920: XMPP Core
- RFC 3921: XMPP IM
- XEP-0030: Service Discovery
- XEP-0077: In-Band Registration
- XEP-0078: Non-SASL Authentication
- XEP-0086: Error Condition Mappings
- XEP-0073: Basic IM Protocol Suite
- XEP-0004: Data Forms
- XEP-0045: Multi-User Chat
- XEP-0047: In-Band Bytestreams
- XEP-0065: SOCKS5 Bytestreams
- XEP-0071: XHTML-IM
- XEP-0096: File Transfer
- XEP-0115: Entity Capabilities
- XEP-0004: Data Forms
- XEP-0012: Last Activity
- XEP-0013: Flexible Offline Message Retrieval
- XEP-0030: Service Discovery
- XEP-0033: Extended Stanza Addressing
- XEP-0045: Multi-User Chat
- XEP-0049: Private XML Storage
- XEP-0050: Ad-Hoc Commands
- XEP-0054: vcard-temp
- XEP-0055: Jabber Search
- XEP-0059: Result Set Management
- XEP-0060: Publish-Subscribe
- XEP-0065: SOCKS5 Bytestreams
- XEP-0077: In-Band Registration
- XEP-0078: Non-SASL Authentication
- XEP-0082: Jabber Date and Time Profiles
- XEP-0086: Error Condition Mappings
- XEP-0090: Entity Time
- XEP-0091: Delayed Delivery
- XEP-0092: Software Version
- XEP-0096: File Transfer
- XEP-0106: JID Escaping
- XEP-0114: Jabber Component Protocol
- XEP-0115: Entity Capabilities
- XEP-0124: HTTP Binding
- XEP-0128: Service Discovery Extensions
- XEP-0138: Stream Compression
- XEP-0163: Personal Eventing via Pubsub
- XEP-0175: Best Practices for Use of SASL ANONYMOUS