open communication.

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eZuce openUC V4.4 Features Listing

openUC Communications & Collaboration System

openUCTM is the easiest to use and most powerful Enterprise communications solution available. It communications enables organizations and connects people in new ways. Simple and engaging, it enables access from anywhere, allowing you to have your office where you are.

As a Service Oriented Architecture (SOA), its components are independent and can be distributed. Additionally, each component can be used stand-alone.

openUC is a very capable, highly scalable and distributed system that offers redundancy independent of topology or geography. The following are its main components:

1) SIP Session Manager, an optionally geo-redundant and load-sharing call control system loaded with features

2) Instant Messaging & Presence application enabling presence based communications and a new user experience

3) Configuration and Administration Web GUI application that makes it simple to use for end users and administrators

4) Unified Messaging application that takes care of all your offline messaging needs

5) Conferencing application for a high user impact and immersive work environment designed for collaboration

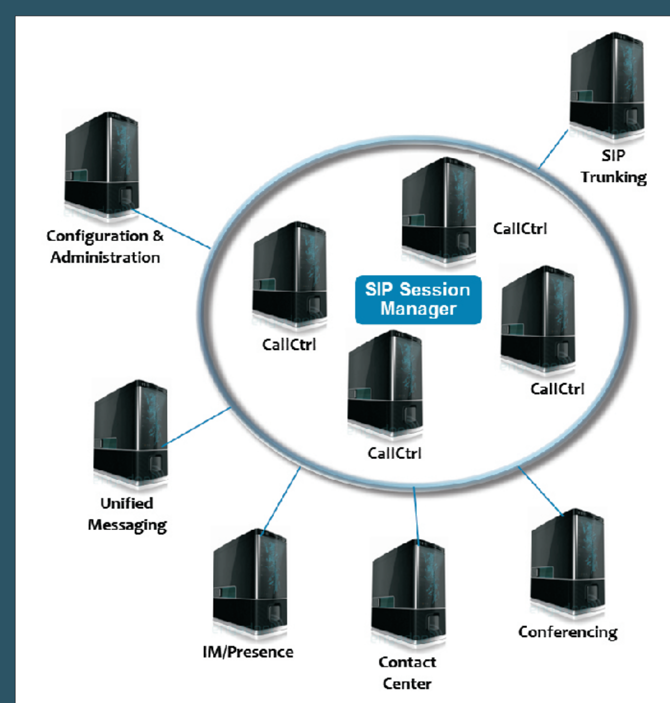
6) Contact Center application for many call queuing needs in customer support, sales, and help desk applications

7) Basic Media Services applications offer paging, call park, music on hold, auto-attendant and other media features

8) SIP Trunking application for resilient, secure and scalable connections to the outside world

9) Business Process Integration that enables Social Business application integration

Other independent services and applications included in openUC are a SIP presence and resource list server (RLS) for line state presence, a shared appearance agent server for shared lines (BLA), a group paging server, a call detail record (CDR) collection & processing server, a third party call control (3PCC) server using REST Web Services interfaces, a process management server for centralized cluster management, a device auto-provisioning server, and an alarm and notification server.



Required Hardware and Operating System

- eZuce openUC runs on standard Intel servers with no additional hardware required
- Red Hat Linux Enterprise 5 or CentOS 5 operating system, 32 bit and 64 bit editions
- Minimum of 4 GB RAM, 120 GB disk. Server hardware redundancy based on customer requirements
- PostgreSQL database for configuration and CDR data

openUC is delivered as a single ISO that automatically installs all of the above required software with minimal information required.

Note – Some features are dependent upon other SIP components such as phone clients and gateways.

1. SSOA SIP Session Manager

A new kind of Session Initiation Protocol (SIP) system, built on the power and imagination of software. SIP Service Oriented ArchitectureTM (SSOA) integrates communications with Information Technology (IT), it communications-enables your business and it provides an all new presence-based user experience. openUCTM offers communications for the way you work, by creating an immersive experience that unites the PC, the phone and the browser into an always-on productivity tool.

eZuce SSOA Architecture

- Distributed and geo-redundant SIP Session Manager combined with an Enterprise Instant Messaging and presence server based on the XMPP industry standard
- Load-sharing and self-healing architecture
- Centrally managed cluster for scalability and resiliency
- Uniform dialplan with extension mobility and number portability
- Strict separation of SIP signaling from media, offering better voice and video quality and scale
- Presence enabled communications for a new user experience
- Business process integration (BPI) using Web Services API for social business integration
- Software only IT application that is easy to manage ad saves cost

Core Telephony System Features

- Transfer (consultative & blind)
- Call coverage
- Call hold / retrieve
- Consultation hold
- Music on Hold for IETF standard compliant phones
- User specific MoH files
- Admin or user configurable Busy Lamp field (BLF) presence and soft-keys
- Shared lines (BLA) for boss - secretary applications
- Up-loadable music files
- 3-way / 5-way video and voice conference on the phone
- Call pickup (global and directed call pickup)
- Group call pickup using hunt groups
- Call park & retrieve
- Hunt groups
- Intercom with auto-answer (bi-directional)
- SIP URI dialing for direct Internet calls
- 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

Remote Branch Support

- Centralized deployment: Branch only provides phone / PC / mobile clients and optionally PSTN gateway(s) for failover and reduced WAN BW consumption or E911 calls
- Distributed deployment: Branch provides full call server with SIP site-to-site dialing between offices and a global dial plan
- Centralized management: Branch office locations can be defined in the administration system with full flexibility. Users, phones, gateways, SBCs, and specific system services 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 with full failover
- Source routing allows call routing based on location (branch local calls are routed through local gateway preferably)
- Alternative survivable branch configuration possible with Audiocodes gateway's SAS functionality (auto-configured)
- Certain openUC services can be deployed in the branch as part of the cluster (e.g. conferencing) for scale or redundancy
- Branch nodes can be redundant

Flexible Dialplan

- Easy to use GUI based dial plan manipulation
- Time-based dialing rules with different administrator 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 destinations based on user's presence status
- Automatic gateway redundancy and fail-over
- Specific E911 call routing
- Permission based rules
- Prefix manipulation
- Dialplan templating for international dial plans
- Flexible internal extension length
- ISN dialing based in ITAD numbers (see freenum.org for more information)
- ENUM support for public and private ENUM based call routing
- Specific rule for site-to-site call routing between SIP systems
- Redirector plugins allow any imaginable dial rule to be added as a plugin, including based on database lookups

Enterprise Performance

- Unlimited number of simultaneous calls (voice, HD voice, video). System capacity only depends on available LAN / WAN bandwidth
- 54,000 BHCC (SIP) per server node (15 cps per node). System scales linearly by adding distributed server nodes
- Up to release 4.4 openUC supports up to three-way redundant configurations with seamless load-sharing at the transaction level and all nodes centrally managed
- Release 4.6 and newer allows distributed configurations up to several hundred load-sharing nodes, centrally managed, for a seamless distributed system
- Up to 10,000 users per dual-server redundant system,

Redefining High Availability and Resiliency

- Optionally fully redundant call control system with load-sharing and redundancy at the transaction level that provides a geo-redundant SIP session manager
- True load-sharing based on the priority policy set in the DNS infrastructure for SIP DNS SRV records
- Load-sharing between branch and centralized data center location based on local branch DNS policy
- No dropped or failed calls upon a server outage. Media does not traverse the server and signaling auto-recovers through dynamic failover to an alternate node
- Real-time synchronization of transaction state information between all nodes using a distributed data replication mechanism
- Uniform user, credential, permission and call routing information available to all nodes (global dial plan and centralized user management)
- Automatic recovery after server failure. Node coming back in seamlessly enters load-sharing mode with other participating nodes
- Reports on load distribution between servers

Enterprise Level Security

- All outbound calls authenticated to prevent toll fraud or unauthorized calls
- Authentication and authorization based on user's permissions sustained during node outage (e.g. survivable branch scenario)
- Secure user password management with auto-generated SIP passwords for maximum security
- DoS attack hardening
- HTTPS secure Web access for administrators and users
- TLS based signaling for SIP trunks
- TLS connection to clients, including clients for remote user's
- Optional secure IM client connection using TLS
- Web Services API over secure connection and authenticated using user credentials
- Certificate management for self-signed and third party signed certificates. Auto-generates CSR requests

Superior Voice Quality

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

Direct Internet Calling

- Ability to configure SIP URI based call routing to other domains
- Specific Session Border Controller (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
- IM / Presence support through firewall

Call Detail Records (CDR)

- Call State Events (CSE) collected for all signaling activity
- Background 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, such as for internal / external calls and calls to specific openUC services
- Collates call legs into single CDRs
- Historic Call Detail Record reports available in near 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 applications (e.g. Crystal Reports, Jasper Reports)
- SOAP / REST Web Services access to CDR data
- Individual call history per user in the user portal
- Can connect to external call accounting system

2. SSOA Presence and Instant Messaging

Always on experience puts the user in charge with presence based communications. The phone, the PC and the browser come together giving the end user control over when and how to communicate with others. openUC does away with the isolation of the old phone system, integrating presence and IM with voice and video communications for an all-inclusive, immersive, and always-on experience.

Enterprise Instant Messaging & Presence

- Unified presence as the lynchpin for all communications
- Standard XMPP based IM and presence capability that scales to enterprise requirements
- Supports XMPP standard based clients, including third party and open source clients on different platforms (Windows, Mac, Linux, smartphones)
- Auto-configuration for ease of administration as part of the openUC cluster
- Pre-population of user's IM groups and roster based on organization structure simplifies the deployment
- Personal group chat room for every user with easy escalation from group chat to a conference call
- Unified presence that federates phone presence with calendar availability and user defined IM presence (calendar available in release 4.6)
- Customizable "on the phone" presence status message allows for dynamic call routing based on user's presence status
- Message archiving and search for compliance and legal discovery
- Server-to-server federation allows to connect into the global XMPP network of federated users
- Optional secure client connections using TLS
- Client-to-client file transfer and desktop sharing (client feature)
- Integration of user profile information and avatar

MyBuddy for Anywhere Access

- MyBuddy personal assistant feature for every user allows for dynamic call control and mobile presence awareness using IM
- Dynamic conference management using IM
- Unified messaging management using IM
- Enables Fixed Mobile Convergence (FMC) application for mobile workers
- Call history / missed calls directly to the user's mobile phone
- Call initiation using corporate dialplan from anywhere
- Corporate directory look-ups from anywhere

Presence & IM Federation

- openUC allows participation in and federation with the global XMPP based network for presence and Instant Messaging
- Server side federation with other public IM systems including:
  - Yahoo, AIM, MSN
  - ICQ, IRC
  - IBM Sametime
  - Facebook IM, MySpace IM
  - SIP / SIMPLE
- Server-to-server federation with Google Talk using XMPP

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- Federation with Microsoft OCS R2 and Lync 2010 using XMPP
- Allows group chat sessions across systems
- Allows message archiving (if enabled) across systems
- User self-administration of credentials for other IM systems establishes the necessary security and privacy

SIP Line State Presence

- Line state presence server based on SIP/SIMPLE
- Compatible with Broadsoft and IETF standard implementations
- Centralized management of resource lists for dialog events
- Busy Lamp Field (BLF) feature based on line state presence
- Supporting shared lines (BLA) for boss secretary applications
- Federated with IM presence to show "on the phone" status in the user's unified presence status
- Line state presence as required by Attendant Consoles

Social Business Integration

- Using REST based Web Services, openUC is ideally suited to presence enable third party applications
- Social Business integration including unified presence, avatar and other user profile information

Fixed Mobile Convergence (FMC)

- FMC application enabled via Instant Messaging with the following capabilities offered to mobile users:
  - Enterprise internal number dialing
  - System call-back saves on wireless toll charges
  - Corporate directory look-ups
  - Call history and missed calls
  - Presence sharing & Instant Messaging
  - Conference management with entry / exit alerts
- Server-to-server (XMPP) federation with GTalk allows using GTalk client on smart phones
- Dynamic conference bridge control for mute / unmute, include / isolate, invite, disconnect

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3. openUC Configuration and Administration

openUC includes a comprehensive Web based configuration and management application, built using the latest Service Oriented Architecture (SOA) and Web 2.0 user interface. This makes openUC the easiest communications and collaboration system to manage yet.

Directory Server and LDAP Integration

- openUC allows for integrated user management using an existing directory service over an LDAP interface
- Microsoft Active Directory integration
- LDAP bind authentication and authorization for centralized password management and single sign-on
- Upload of user profile information with a flexible mechanism that allows mapping LDAP schema fields to openUC profile fields
- Scheduled re-sync with the corporate directory
- Global address book (GAL) synchronization with the directory server
- TLS based secure connections over LDAP

User Management

- User management with ease: Create a user, provision a phone and assign a line in only three clicks
- Numeric or alpha-numeric user IDs
- User PIN management (Web 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
- Calling permissions:
  - 900 Dialing
  - International Dialing
  - Long Distance Dialing
  - Mobile Dialing
  - Local Dialing
  - Toll Free Dialing
  - Forward Calls External
- System permissions:
  - User has voicemail inbox
  - User listed in auto-attendant directory
  - User can record system prompts
  - User has superuser access
  - User is allowed to change PIN from TUI
  - User can use Microsoft Exchange 2007/10 VM
  - User has a personal auto-attendant
  - Permission to subscribe to presence
- Custom permissions as defined by the administrator
- 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 maximum security
- User groups with group properties
- Per user call forwarding (find me / 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
  - Find me / follow me configuration using the user portal
- Extension pool with automatic assignment
- Per user Caller ID (CLID) assignment
- Per user Caller ID blocking
- IM notification settings for conference entry and exit messages and alerts when someone is leaving a voicemail message

User Self-Control

- 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 and missed calls
- Personal phone book, speed dial and presence management
- Allows contact upload from GMail and Outlook
- Click-to-call
- ACD presence and call center supervisor capabilities
- Individual phone management
- Personal auto-attendant
- Dynamic conference management w/ click-to-conference
- Management of personal IM account
- Personal MoH music upload and preferences

Directory, Speed Dial, Softkeys

- Automated generation of directory information per user or per user group including a global address book (GAL)
- Support for complete contact information and user profile, including avatar
- Creation 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)
- User configuration of Shared Lines (BLA)

Centrally Managed Cluster

- Automated installation and configuration of a distributed system with specific server roles
- Automated and centralized configuration of a high-availability redundant 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

Installation & Upgrades

- Automated installation from CD ISO for OS and openUC application components
- Graphical configuration wizard for system configuration after installation
- Certificate generation (allows installing a signed certificate if desired)
- GUI based upgrade management from the administrator 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 administrator skills are required for installation and configuration

System Administration Features

- Browser based configuration and management
- Several administrator accounts
- GUI based certificate management
- LDAP / AD integration
- 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 & active calls
    Display job status
    Status of services
    Snapshot logs for debugging
    Logging (customizable log levels, message log per service)
- Domain Aliasing
- Support for DNS SRV
- Support for DNS NAPTR based call routing
- Automatic restart after power failure
- Server statistics (integrated graphs and SNMP)
- Login history report (successful and unsuccessful)
- Automated testing of network services (DHCP, DNS, NTP, TFTP, FTP, HTTP) for proper configuration
- Downloadable test tool to run network services test
- Domain administration
- Installation and management of language packs for localization

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 configuration profile
- Auto-pickup of profile by phone / gateway
- Centralized management of all the parameters
- Centralized backup and restore of all the configurations
- Auto-generation of lines by assigning users to devices
- Device group management & properties
- Firmware upgrade management

Managed Client Devices

- Polycom SoundPoint all models
- Polycom SoundStation all models
- Polycom VXX-1500 video phone
- Audiocodes gateways MP112, MP114, MP118, MP124 FXS
- Audiocodes gateways FXO and PRI/BRI
- Counterpath Bria Professional
- IsCoord softphones
- Nortel 1210, 1220, 1230, 1235 with SIP firmware
- Nortel 1100 Series SIP with some limitations
- eZuce Communicator client
- Pidgin and Adium IM clients

Managed Devices (Experimental)

- 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
- Counterpath X-lite
- IPDialog SIPTone V
- LG-Nortel LG 6804, 6812, 6830
- Nortel video phone 1535
- Linksys ATA 2102, ATA 3102
- Linksys SPA8000
- Linksys SPA901, SPA921, SPA922, SPA941, SPA942, SPA962
- G-Tec AQ10x, HL20x, VT20x
- SIP Communicator softphone

Note: For phones in the experimental category eZuce does not offer commercial support. There can be some firmware issues that prevent all features to work.

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Alarms and SNMP MIBs

- SNMP Traps can send alerts to network monitoring systems for immediate attention
- All alerts can also be e-mailed or sent to SMS devices
- Alerting includes ability to alert on-site staff of emergency number (911) being dialed.
- Alarm groups allow administrators to make sure appropriate staff is notified.

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4. openUC Unified Messaging

A new way of collaboration requires a truly unified inbox for all offline communications. openUC can seamlessly and transparently integrate with many different email systems, including Microsoft Exchange, Zimbra mail, Open-Xchange, Google Mail, and other IMAP based email systems.

Unified Messaging (Voicemail)

- Integrated unified messaging system with all the necessary features required to replace legacy voicemail systems, such as Nortel CallPilot, Meridian Mail, Octel, Cisco Unity, Avaya Messaging and similar
- Support for 10,000+ inboxes, limited only by disk space
- Mailbox size only limited by available disk space. System requires about 1MB of disk per minute of voicemail recording.
- Performance tested up to 400 simultaneous calls (ports) on dual core server hardware
- Unified messaging service can run on dedicated server hardware or co-located with other openUC components, centrally managed
- Message store can be NFS mounted
- Localized per user by installing language packs
- 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
- New message alerts with or without the message attached as a wav file
- Supports several parallel notifications to different email accounts
- Per user selectable templates for email format used when forwarding voicemail to accommodate smartphone message formats
- Manage folders: Folders for message organization
- Manage greetings: Multiple customizable greetings
- Operator escape from anywhere
- Remote voicemail access using a phone
- Auto-removal of deleted messages
- Daily report on disk usage sent to admin

Personal Auto Attendant

- User configurable personal auto-attendant for every user on the system gives caller a choice of where to go
- 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

Unified Messaging Integration

- IMAP integration can be used for email systems that support all the necessary IMAP primitives (e.g. Microsoft Exchange)
- IMAP back-end connection: Acts as an IMAP client into MSFT Exchange and other compatible email systems
- Voicemail messages automatically appear in the user's inbox
- User manageable credentials for IMAP federation
- Properly controls MWI on the phone when message is "read" using the email client

Nortel Legacy System Integration

- Replacing an aging legacy voicemail system with openUC unified messaging is possible in a fully integrated configuration
- System interconnection using TDM Audiocodes gateway (Nortel CS 1000 release 5.5 and prior)
- System interconnection over SIP trunk (Nortel CS 1000 release 6 and newer)
- Proper call diversion from Nortel legacy phone connected to CS 1000 to openUC unified messaging
- Message Waiting Indication (MWI) back to the Nortel legacy phone (requires Q.Sig support for CS 1000)

Cisco Legacy System Integration

- openUC unified messaging can substitute for expensive Cisco Unity voicemail solution
- System interconnection using SIP trunks into Cisco CM
- Proper call diversion from Cisco legacy phone to openUC unified messaging server
- Message Waiting Indication (MWI) as a SIP subscribe / notify back to the Cisco phone

Web Services Application Integration

- Unified Messaging integration into other applications using Web Services REST interface allows for maximum flexibility
- Integration into Outlook 2010 with Web Services based MWI activation / deactivation
- Works with any email system including Google Mail, Open-Xchange, Zimbra and others
- Used to communications-enable legacy systems

Speech Recognition

- openUC unified messaging supports the MRCP interface for speech recognition and text-to-speech applications
- Speech processors from Nuance, Loquendo, and Voxeo can be connected
- TTS systems from a variety of vendors are supported using the MRCP interface

Note: In release 4.4 openUC only supports speech recognition and TTS as a custom integration project. Please ask us for additional information.

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5. openUC Conferencing

openUC Conferencing provides a personal conferencing facility to every user on the system with sufficient capacity so that scheduling a shared resource is no longer a concern. Easy escalations from chat to call or video and from group chat to conferencing make collaboration a breeze.

Conferencing

- Software based conferencing solution that offers comprehensive and cost-effective conferencing for the whole enterprise
- Voice conferencing server that can run on the same openUC server or on dedicated server hardware
- Support for voice and HD voice conferencing
- Each user on the system can have a personal Meet-Me conference bridge
- Recording of conference calls where the recording is forwarded to the users unified messaging inbox
- 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 escalation of group chat to a voice conference
- 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
- An openUC system can have more than one conference server if more capacity is needed
- All conferencing servers and services centrally managed and configured
- Direct DID number assignment to user's personal conference bridge for direct external entry
- User can control conference bridge through their user portal web interface
- User can also control Conference Bridge via MyBuddy Instant Messaging Chat Bot.

6. openUC Contact Center

openUC offers an integrated contact center solution suitable for informal call center applications, such as help desks and other applications with basic queuing requirements.

Call Center Server (ACD)

- Basic call center (ACD) application built-in with other openUC services and applications
- Scalability to 30 online agents and 50 calls in queue per call center node (server)
- Supports several ACD servers per cluster
- 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 barge in (early termination of welcome message if agent becomes available)
- Agent presence monitor using presence server
- Separate welcome and queue audio
- Call termination tone or audio
- Configurable answer mode
- Agent wrap-up time configurable per queue
- 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
- Statistics:
  - Agent statistics
  - Call statistics
  - Queue statistics
- ACD historic reporting
- Supervisor authorization for agent monitoring
- ACD historic reports for agents, calls, queues
- All reporting stored in database for post-processing if needed

Note: openUC release 4.6 will include a new robust call center application that introduces skills based call routing and removes the currently existing scalability requirements. Contact us for more information.

7. openUC Basic Media Services

Basic media services are features typically offered by a PBX system. openUC offers these features as independent services connected to the SIP Session Manager for maximum flexibility.

Auto Attendants

- Unlimited number of auto-attendants
- Customizable IVR menus
- Dial by extension and name
- Night and holiday service, based on schedule
- 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
  - Go to Extension
  - Deposit Voicemail
- Up-loadable custom prompts
- Configurable DTMF handling

Group Paging

- 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

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 & Retrieve

- 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
- Up-loadable 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
- Users can have a personal park orbit

Music on Hold

- Music on hold server plays music during hold times
- Per user or per user group configurable music files
- Allows external music source connected over a sound card (legacy system)

Analog Lines (FXS)

- Supports any SIP compliant FXS gateway
- FAX support with FAX machine connected to FXS gateway (gateway should support T.38)
- Analog cordless phone support
- Supports analog legacy Polycom speakerphones
- Plug & play management of FXS gateways from Audiocodes, Grandstream, and Cisco

FAX Server

- Automated inbound FAX to email support (FAX receive)
- DID assignment to FAX service for personalized FAX number

Authorization Codes

- Authorization codes allow users to make privileged calls from any phone entering a secret code
- Codes can be assigned to any user
- IVR service allows easy authorization

8. Flexible Trunking

openUC incorporates external gateways for both PSTN and SIP trunking. This offers unprecedented flexibility, allowing customers to choose gateways from different manufacturers and deploy them anywhere in the network where needed. Centralizing all trunk access into a data center can save significant operating cost as trunk utilization increases. Allowing for trunk access to remain local to a branch can improve public safety and E911 calling.

PSTN Trunking

- Unlimited number of PSTN gateways and trunk lines, connected to the IP network
- Supports any SIP compliant gateway (e.g. Cisco, Audiocodes, Patton, and others)
- Gateways can be in any location, connected to the routed network (no NAT between openUC servers and gateways)
- Flexible gateway selection per dialing rule
- Source routing of calls so that calls can be routed through a local gateway to save WAN bandwidth
- DID management for incoming and outgoing calls
- Local DID range per gateway
- DNIS
- CLIP Management
  - User CLIP
  - Gateway default CLIP
  - Prefix stripping / appending
- Per gateway CLIR
- Automatic Route Selection (ARS)
- Least-cost routing (LCR)
- Automatic failover if unavailable or busy. Call attempts failover to the next available gateway specified in the dialing rule without user impact
- FAX support (gateway to support T.38)
- Mixing of PSTN and SIP trunks for least cost routing
- Plug n' play auto-configuration of Audiocodes trunk gateways (Mediant 600, 1000, and 2000) significantly simplifies gateway configuration and maintenance

SIP Trunking

- 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
- Interoperability with many ITSPs. Please ask us for more information
- SIP interoperability with Nortel CS1000 R6 and higher
- SIP interoperability with Cisco CM
- SIP call origination & termination
- Branch office routing
- Proxy to proxy interconnect using ACLs
- Least-cost-routing (LCR)
- Mixing of PSTN trunks with SIP trunks for least cost routing and failover
- 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

Intoducing your new enterprise communications system.

Sounds easy? Now it is!

9. Business Process Integration

The future is about a cohesive communications experience that is able to incorporate different clients, applications and workflows. openUC is explicitly built to communications enable other applications and integrate into existing business processes. openUC is all about integration, enabling a Social Business experience.

Microsoft
Microsoft Active Directory

- Synchronization with Microsoft Active Directory using the LDAP interface, on demand or automatically based on a schedule
- User profile synchronization with support for the user's complete profile
- Graphical query design combines ease of use with flexibility mapping AD schema fields to openUC; allows preview of records to be imported
- LDAP bind authentication and authorization into Active Directory
- TLS based secure connection to Active Directory

Microsoft Exchange

- Dial plan integration with Microsoft Exchange 2007/2010 voicemail server, allows mixed environment with groups of users on Exchange or the openUC Unified Messaging server
- Permission based selection of Unified Messaging server per user or per user group
- Automatic dialplan routing to Exchange VM enables use of speech based Exchange 2007/2010 capabilities

Microsoft Lync 2010

- SIP trunk interoperability between openUC and Lync 2010 allows end points on either system to call each other
- Call flow interoperability for call transfer

Microsoft Office 2010

- Communications-enables Outlook 2010 offering an alternative to Lync 2010 with much of the same user experience
- Adds presence to Outlook 2010 as a substitute for Lync 2010
- Adds social network connector to Outlook 2010 enabling address book and full user profile sharing with openUC, including user's avatars
- Offers full unified messaging integration into Outlook 2010
- Click-to-call with automatic phone number discovery in received emails
- Outlook 2010 integrated scheduling of conference calls with click-to-join button in invite message
- Outlook 2010 integrated dynamic conference call management with entry / exit messages and full conference experience controls
- Additional calling features integrated with Outlook 2010
- Direct launch of IM conversation from within Outlook 2010
- Counterpath softphone as an Outlook plugin allows voice, video and click-to-dial, enables call recording on the desktop

Google
Google Mail

- Google Mail with Outlook 2010 as the client and openUC provides an alternative to Microsoft Exchange / Lync 2010
- Google contact upload into openUC

Google Talk

- Google Talk interoperability for Instant Messaging and presence over XMPP
- Google Talk client on Smartphones can be used for a fixed mobile convergence experience (FMC) using openUC

10. Public Safety and E911

Nothing is more important that the safety of your employees. openUC's flexible emergency call routing and on-site notification capabilities can be used to create a system that gives internal personnel and emergency responders the information they need to locate people in need.

Enhanced 9-1-1 Support

- Flexible on-site notification via e-mail, SMS and SNMP traps
- Allows administrator to create device Emergency Response Locations (ERLs) as locations and add phones to the locations.
- Push Emergency Line Identification Number (ELIN) out T1/PRI or select local gateway for 911 dialing.

11. Standards Compliance

openUC is 100% standard compliant. We guarantee that no tricks are played with proprietary 'enhancements' that hamper interoperability. openUC is part of the foundation for a globally interoperable communications infrastructure.

SIP & XMPP Compliance

- RFC 3261 Session Initiation Protocol using both UDP and TCP
- Advanced call control using RFCs
- RFC 3920: XMPP Core
- RFC 3921: XMPP IM
- A large number XMPP XEP standards
- RFC 3515 Refer Method
- RFC 3891 Referred-By header
- RFC 3892 Replaces header
- Provide for consultative and blind transfer and third party call ctrl
- 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 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