WHYCOREIP UCX

The Communications Hub

CoreIP Unified Communications

User adoption and deployment headaches due to a myriad of different client side applications has been a challenge for organizations deploying unified communications systems. CoreIP alleviates user adoption issues with the innovative CoreIP UCX Web client. A simple web link that is easy to deploy on desktop or mobile computing devices that consolidates all communication channels into a single, visual, easy-to-use communications hub.

Unified presence combined with instant messaging and a unified inbox makes collaboration much more engaging and immersive. With the same application in use across all devices it promotes end-user adoption, efficiency and productivity. CoreIP UCX Web client ensures broad adoption by making unified communications easy and familiar for end-users, helping organizations be more collaborative, social and productive.

Why choose Unified Communications for your enterprise? Here are several examples of why users like UC.

UCX - Features

- Enterprise Communications: all of the legacy PBX functionality that users expect available across devices today's users want to use. Voicemail, Unified Messaging, Conferencing, Collab¬oration, Instant Messaging all available to the user on their choice of open standards devices and software.
- Enterprise Instant Messaging & Group Chat: powerful IM that is highly available, group chat, and compliant archiving. Integrated phone/PC presence make chat more productive & collab¬orative, often eliminating the need for separate tools. End-to-end encryption ensures security and compliance for any shared content.
- Presence-Based Communication: establishes a user's personal availability and willingness to communicate. With rich presence, users can make more effective communication choices. A user's presence can include additional information such as activity, location and status. Unified presence includes states for "On/Off the Phone" visible to all IM users.
- Dynamic Conference Management: Each user on the system has a personal conference room and bridge. CoreIP enables users to invite conference participants with a single click. Easy to manage voice conferences with built-in dynamic conference management, including invi¬tations, conference recording, muting, kicking and locking conferences.
- Unified Messaging and Call History: Unified Messaging and voicemail are integrated into the CoreIP UCX client. Users can access and play new voicemail messages directly from the main window of the client, or from a traditional desk phone. Provides a complete call history including received calls, missed calls, and placed calls.
- Ready To Integrate:Highly scalable with small and large systems Origin uses advanced interfaces making it extremely manageable so administrative and user experience is consistent.

Your Communications,
Your Way.

Tailored to your needs

Vendor lock-in costs businesses handsomely. By utilizing hardware and software that complies with SIP standards for communications, CoreIP allows customers to tailor their communications to their existing infrastructure. Standards allow data networks to be built using the hardware and software needed for the job, CoreIP does the same for communications.

Servers, Gateways & Phones

Built to run on standard server hardware or under modern virtual server systems, CoreIP fits easily into your existing IT infrastructure. Operate it from your Data Centers or from any number of Cloud Providers. Whether connecting to traditional PSTN lines or to SIP Trunks standards based gateways allow you to centralize or distribute your trunking needs.

Often the single largest cost in traditional systems, allowing our customers a choice of standards based end- user devices ensures that deployment costs are reasonable and remain reasonable over the life of the system.

100% Indigenous (Designed & Developed in India)

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Understanding the Needs of Customers

If you're still using a more traditional route when it comes to your phone system, isn't it time to look into a progressive and easy to manage VoIP phone system? Along with additional features like call forwarding, holding, diverting and so on, CoreIP UC can operate wherever an IP Network is available. This makes it a cheaper medium for long-distance communication. Analog phone systems can only provide connectivity in localities where there is physical wiring and high call rates make long distance calls quite expensive.

CoreIP understands the important role that communication plays, in building and maintaining your business relationships. How effectively you communicate with customers, resellers, key suppliers and business partners can be the difference between business won and business lost. CoreIP Unified Communications Business Solutions are streamlined, easy to use applications that can be tailored to meet the needs of almost any business. The solution is able to integrate contact centre functionality with unified messaging, rich presence, FAX over mail, Voicemail over email, mobility, desktop telephony applications and Integration with your ERP or CRM applications.

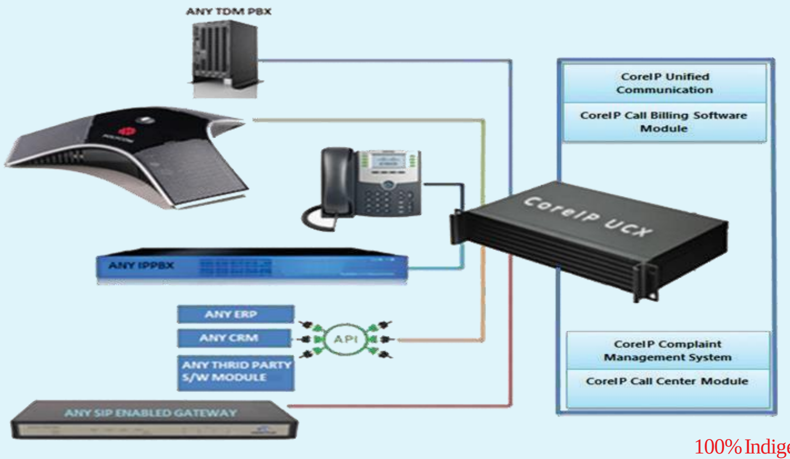
Flexibility for the modern VoIP user

The ability to use an analog phone on VoIP opens up a world of flexibility for its users. It gives almost anyone the ability to take advantage of the benefits of a VoIP phone system without investing in specific VoIP equipment. However, it's important to note that your phone's capabilities will be limited based on the phone you use. That is why a majority of VoIP users eventually make the switch to an IP, VoIP specific phone.

As you can see, switching to VoIP doesn’t require a complete overhaul of your current equipment. A majority of it can be programmed to work on your new, Internet based system. If you're interested in using your analog phone with your VoIP phone service, then CoreIP provides solutions to your system.

Open Platform

The term open platform means the ability to connect with any third party devices with the standard technological platform and ability to integrate with any third party software through Application Programming Interface (API). CoreIP UCX is developed to connect any third party SIP end point like CISCO, AVAYA, ALCATEL, POLYCOM, MATRIX, GRANDSTREAM and many more. Through APIs, CoreIP UCX also connects any other software platform like any third party NMS, Billing software (H8, TALLY), CMS, ERP, CRM etc.



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CoreIP Unified Communication Exchange Server

- SIP Session Router, optionallygeo-redundant and load sharing with non blocking
- Media serverfor unified messaging and IVR (auto-attendant) services
- BHCC per server 300,000
- Conferencing serverbased on FreeSWITCH
- XMPP Instant Messaging (IM) and presence server (based on Openfire)

SOA Architecture / Business Process Integration using Web Services

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

Enterprise Instant Messaging (IM) and Presence

XMPP based IM and presence serverbased 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 everyuser 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 searchfor compliance (pending)
Server-to-server XMPP federation
Optional secureclient connections Client-to-client file transfer
Group chat rooms XMPP
search
Integration of user profile information and avatar (pending)

High Availability

Optionally fully redundant call control system
Upto5000 user in single server
server is considered in duplication.1+1Server considered for redundancy, switchover from primary to secondary sever without manual intervention in case of failure of primary server/CPU.
Geo-redundant SIP session manager Based in
DNS SRV (no cluster required)
Load balance under normal operating conditions
Geographicdispersion of redundant systems
Scalability minimum 25000 user per instance
Real-time synchronization of state information
Automatic recovery after server failure
Reports on load distribution

- 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
- 30 days of call recordings, stored in the HDD

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 corporatedialplan
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

Plug & Play Device Management

Auto-discovery of phones & gateways on the LAN
Auto-registration of 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

- Contact center (ACD) server
- Call park / Music on Hold (MoH) server
- Presence server (Broadsoft and IETF compliant resource list server for BLF)
- Third party call control (3PCC) server using REST interfaces
- Management and configuration server
- Process management serverfor centralized cluster management

Remote Branch office support

Centralized deployment: Branch only provides phones and optionally PSTNgateway for failover, reduced WAN BWconsumption 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 branchlocation

A PSTN gateway can be available for calls that originate in a specificbranch only or for general use

Source routing allows call routing based on location (branch local calls are routedthrough 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 CoreIP services can be deployed in the branch as part of the cluster (e.g. conferencing)

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
Presencesharing IM

Internet Calling

Ability to configure SIP URI based call routing toother domains
Specific SBC selection for call routing
Configuration of native NATtraversal w/ optionally redundant media anchoring if necessary
Media anchoring supports voice and video for any codec
Softphone licenses are offered on a chargeable basis and are provisioned according to the customer's specific requiremen

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Technical Specifications of UCX

User Self-Control (User Web Configuration Portal)

Everyuser 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
Personalphone 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

SIP Trunking

Basic SIP trunking gateway w/ NAT traversal
Remoteworker support w/ near-end and far-end NAT traversaland auto-detection
ITSP templates for simplified configuration
Interop with the following ITSPs:
Tata (Tatateland TTSL),
Airtel,
Vodaphone,
Reliance,
BSNL
MTNL
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
CLIP Management for incoming call
Branch office routing
Proxy to proxy interconnect using ACLs
Least-cost-routing (LCR)
Mixing of PSTN trunks with SIP trunks
TLS support for securesignaling
Route header for flexible call routing throughan SBC
Flexible rules for SBC selection (route selection)
Support for Skype for Business SIP trunking

Hardware Specs

1U/2U Rack Mountable Server
Single power supply
8 core CPU
16 GB RAM
500 GB Hard Disk storage

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
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 directoryfrom a .csv file (Excel)
User configurable speed dial (internal / external numbers, SIP URIs)
Speed dial generated serverside 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
DOD
Local DID per gateway DNIS
CLIP Management for incoming call
User CLIP
Gateway default CLIP Prefix stripping / appending 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. messages to different SIP/PSTN trunk gateways, either on premise or at a remotepremise 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
Automaticfailoverif busy Inbound FAX support Mixing of PSTN and SIP trunks with least cost routing

Centrally Managed CoreIP Distributed System (cluster)

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

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 voicemailinbox
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
Linux/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 avatarincluding support for gravatar
SIP password management for security
User groups with group properties Per user call forwarding (follow me)
To localextension, PSTN number, or SIP address
Based on user or admin defined time schedules
Parallel or serial ring
Allows definition of ring time beforetrying next number
Allows several forwarding destinations
Follow-me configuration using user portal
Extension pool with automatic assignment
Per user Caller ID (CLID) assignment
Call billing software is a built-in
Voicemail and IVR software is a built-in

Security

All outbound calls authenticated
Secureuser 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.

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Technical Specifications of UCX

VoIP Protocols

Protocol:
TLS / SRTP
SIP v2.0 (UDP/TCP),RFC3261
SDP,RTP(RFC2833), RFC3262, 3263,3264,3265,3515,2976,3311
RTP/RTCP, RFC2198, 1889
RFC4028 Session Timer
RFC3266 IPv6 in SDP
RFC2806 TEL URI
RFC3581 NAT, rport
H.264 or latest
Primary/Backup SIP Server
Outbound Proxy
DNS SRV/ A Query/NATPR Query
SIP Trunk
Early Media/Early Answer
NAT:STUN, Static/Dynamic NAT

Voice & FAX

G.711A/U law, G.722, G.723.1, G.729A/B, G.726
Silence Suppression
Comfort Noise Generation(CNG)
Voice Activity Detection(VAD)
Packet loss (PLC)
Echo Cancellation(G.168), with up to 128ms
Adaptive (Dynamic) Jitter Buffer
Programmable Gain Control
T.38/Pass-through
Modem/POS
DTMF mode: Signal/RFC2833/INBAND
VLAN 802.1P,802.1Q,MPLS
Layer3 QoS and DiffServ

Network

Network Protocol:
TCP
UDP
IP,
RTP/RTCP,
HTTP/HTTPS, ARP, ICMP,
DNS,DDNS, DHCP, NTP, TFTP, TELNET,SSH, PPPoE, STUN.
Static/dynamic ARP
IPv4/IPv6,
static/dynamic IP
DHCP Client/PPPoE

About CoreIP

To introduce ourselves we would like to call us a Technology company. A company that helps global organizations implement optimized/customised Telecom/IT solutions with simplicity, which is an absolute must for any organization for its successful growth. Our scope of expertise includes Unified Communications, Security solutions, Networking solutions and a lot Core IP's vision of the digital future is driven by the needs and aspirations of our customers and consumers around the country who are in need of latest technology every day. We share their dream to live a fuller life by providing ways of working smarter and enjoying the rewards of technological advanced solutions. As we move forward together with our customers into the uncharted future of the 21st century with the prospect of future technologies and systems yet to be thought of, Core IP's standards are still firmly grounded in the philosophy of company's directors. CoreIP promises to deliver best-in-class products and services, enhanced by extensive in house research and development capabilities.

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