Deltapath frSIP® Unified Communications Core

Why buy a phone system when you can buy a Unified Communications Platform?

Traditionally phone systems are in a world of its own, closed system, transmit voice only, independent network, locked in by a single vendor, and expensive to maintain.

Today, Deltapath's frSIP® Unified Communications Platform is an open standard system, transmits video and voice, integrates with your business applications, can be run on a shared network, supports multiple vendors, and is extremely easy and economical to maintain.

The frSIP® UC Core fits into the traditional communication network of desktop telephone, mobile phones, voice mail, email, fax, and the Polycom video conferencing , in making a messaging environment a truly unified business communications environment.

Mobility

With your employees roaming from office to office and frequently on the road, you need a platform that allows them to stay connected virtually everywhere.

- Roam to any hot desk phone with your extension login at any branches; continue to receive and make calls as if you are at your home country.
- Pair your mobile with your desk phone and stay in touch with customer.
- Make VoIP calls on your GSM mobile while you have no Internet.

Availability and Survivability

The frSIP® UC Platform comes with a decade of IP Telephony design and development experience. You can easily stack 2 units of frSIP® UC Switches to form a cluster; allowing the secondary to take over the primary if it fails for whatever reason including power outage.

Deltapath's proactive remote monitoring and maintenance service combining with the built-in self-diagnostics and self-healing tools gives you 99.999% availability. Together with a proven track record of Global 1000 companies deploying our platform across 25 countries and 65 major cities, you can be confident to let us power your organization's communication infrastructure.

Conferencing & Collaboration

Traditionally, Conference Bridge was a separate and privileged system reserved for upper management use. Today, we offer it as a built-in standard feature, allowing your employees to host an ad-hoc conference or invite participants to a meet-me bridge. You can conference with a mixture of audio, video* and desktop content*.

*Requires Polycom® RMX® conference bridge

Manageability

- Incoming call notification and control on your desktop.
- Take full control from hold and transfer to conversation recording and call history at your finger tips.
- Change your status and follow-me profile at anytime.
- Personalize your phone to treat each caller differently base on your rules.

HD Voice™

Polycom HD Voice is a technology feature that enables life-like voice clarity and highfidelity sound for more productive conversations over the phone.

- Twice the Clarity of ordinary Phone calls
- Natural life-like conversations
- Hear every word without repeating

Optimized For Polycom

Deltapath's frSIP® Unified Communications solution is designed and built base on Polycom's end-points specifications. Hence, you get the most out of the Polycom devices with our true native support.

Since 2002, Deltapath has been testing various brands of IP phones to bundle with its solution. Out of all its candidates, Polycom won our hearts base on price/value, reliability, look and feel, and functionalities. Polycom was also a true winner in sound quality, especially its speaker phone quality even on its entry level models, breaking through the misconception of voice over IP equals bad sound quality. With Polycom's High Definition voice technology, VoIP is more crystal clear than ever.

Features and Benefits

Increased productivity

Integrate with your core business applications such as CRM and ERP systems to place phone calls to your customers with a click of a button.

Get frSIP® to notify your applications for screen pop ups or panels to display vital information before you pick up the call.

Messaging

E-mail and voicemail are both common tools of communication. Now we even bring you video messaging to your e-mail, unifying all three tools under one platform. In addition, users on the road without e-mail access can be configured to receive voicemail / videomail SMS notification.

Presence

Learn about someone's status before you call them! Why waste your time ringing someone who is already engaged on the line?

Cost Savings

Direct Cost Savings	Indirect Cost Savings
Voice and Video over IP Ready – no extra hardware/ software needed	Reduced cabling cost - Integrated management of data and voice networks under one physical network
Free extension to extension calling across all branches	Increase employee productivity with click to dial, transfer by name, and presence information with simple mouse click
Free long distance to the city where you have presence	No special skills for adds, moves and changes
Reduced equipment and monthly line rental cost by using SIP Trunk to your local phone company	Reduce number of missed calls and messages by pairing desk phone with the mobile
Connect to VoIP IDD service providers that give much lower rates than traditional PSTN IDD providers	Reduce monthly leased circuits cost by sharing voice and data over one single pipe
Add a free phone line to every employee's home office	Improve customer satisfaction with less waiting time and better call distribution process
Continue to enjoy free calls using your mobile without the Internet	Intuitive and universal dialing plan - no more dialing 9 to get an outside line; no need to remember prefixes when calling a colleague at the other branch
Free roaming by utilizing your smart phones to make and receive calls for free via Wi-Fi	No special setup required for video calls

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Features Highlight

- High Performance Multi-Threaded Core Signaling Engine
- PBX Alarm Event Logging
- Multi-task Handling Engine
- Intelligent Call Routing Engine
- Nano Second Call Detail Records granularity
- Built-in Session Border Controller for phones behind NAT
- Dual Cluster Mode: Local Cluster with IP Take Over and Geographically Separated Cluster
- Browser-based interface for configuration and management with CSV and real-time batch export, modify, and import tool and multi-level searching
- Call Permission Control per user
- Foreign peer monitoring with automatic trunk fail-over
- Handles multiple Direct-Inward-Dial number ranges
- Configurable outbound caller ID (auto resolve, hidden, custom, default, relay original on forward) on a per user basis
- Multiple extensions sharing one single phone
- Multiple phones sharing one extension
- Directory service – dial by name
- User customizable RSS web feeds on deskphone
- Corporate, group, and personal directories on web, desktop, and IP phone.

Interactive Voice Response System (IVR)

- Customizable auto attendant greetings.
- Unlimited level and number of Auto Attendants
- Record IVR voice prompts on phone or computer.
- Playback of voice prompts on phone or computer.
- Customizable IVR menus in browser interface
- Support for error prompts: invalid option, no input, and multiple invalid options, and disconnect.
- Configurable default behavior if no input received.
- Automatically transfer a caller to an IVR after the callee hangs up.
- Customizable LCD Display to show the purpose of call base on what voice menu the caller selected

Unified Messaging

- Support for video and voice message and greeting recording.
- Support for recording in high definition voice (G722, Siren™ 7 and Siren™ 14) voice message and greeting
- Voicemail and Videomail to E-mail in (wav/mp3/wmv/mov)
- E-mail server independent
- Press 0 to operator during greeting announcement
- Support for multiple time zones
- Mailbox setup wizard for new users upon first login
- E-mail integration:
  - ✓ Automatically mark a voicemail message as read when a user views the voicemail e-mail
  - ✓ User can remove his/her copy of a voicemail from the PBX by clicking a DELETE button in the email message.
- Automatically delete unread or read voicemail after a specified age.
- Support for HD voicemail recording and prompts
- Folders for message organization
- Configurable caller ID and envelope information
- announcement
- Customizable personal greetings
- Remote voicemail access
- Send fax from Switchboard (PDF, PNG, TIFF)
- Fax to E-mail
  - ✓ Personal DID fax numbers
  - ✓ Automatically convert incoming faxes to PDF and dispatch by e-mail
  - ✓ Record and notify failed fax attempts
  - ✓ Fax log data stored in database for retrieval
- Fax on demand

Meet-me Audio Conference Bridge

- Supports wideband and narrowband codec mixing
- Maximum Concurrent Participants: T256: 256
- Participants; T512: 512 Participants
- Audio alert and name announcement when entering and leaving a conference
- Individual volume and mute control for each participant
- Administrative control for the conference host
- Dedicated private conference access mode
- Public conference bridge access mode
- Configurable user limits for each conference
- Outbound calling to conference attendees
- Record conference calls
- Bridge status overview
- Conference host control via Switchboard.
- Room access by time validity.

DISA Remote Dial-in

- Allow users to dial back into the system from their mobile and make calls as if they are at the office.
- Caller ID and Password Authentication
- Automatically retrieve permission profile upon login
- Integrated with billing module for automatic billing
- Missed call call-back
- Continue to perform call recordings even on your mobile

SMS Facility

- Send SMS to any GSM numbers from Switchboard
- Provide voicemail and videomail notification via SMS
- Receptionist can send messages to users via SMS

Multiple language support

- Web interface
- IVR prompts
- E-mail Templates

Manager / Assistant Functionality

- Shared Line Appearance
- Shared line hold and pickup from multiple shared devices
- Multiple managers per assistant
- Multiple assistants per manager
- Push-to-talk intercom
- Secretary Barge-In
- Secretary Barge-and-Conference

User Roaming/Hot Desking (hotelling)

- Local and inter-branch roaming
- Simple sign-in and sign-out via IVR.
- Automatically charge your calls to your extension
- Remote and automatic log out facilities.
- Automatic download of phone book and user settings upon sign-in.

Network and Administration

- Web based interface
- IPV4
- IPV6 Ready
- DHCP
- Static routes
- Virtual IP (WAN Port Forwarded)
- Default route
- Stackable to form local cluster with IP take-over
- Console DB9 interface
- System level management
- User management
- SNMP monitoring

API

- XML Based API for call control & phone book retrieval

Microsoft Integration

- Synchronize and Publish Microsoft Active Directories records to frSIP® Switchboard and IP phones.
- Click to dial from Microsoft Outlook, Word, Excel, and Powerpoint.
- Make and receive calls to/from Microsoft OCS and Microsoft Lync via SIP.

Unified Communications Switch Specifications

Unified Communications Switch	
Form Factor	19” Rack Mountable 2U High
Dimensions	445 (W) x 430 (D) x 88 (H) mm
Operating Platform	64-bit Embedded Linux platform on flash memory powered by Intel® Core™2 Quad Processors
Number of Stations	T256/T256-S: Up to 1,500 devices T512/T512-S: Up to 3,000 devices
Total Number of Concurrent Calls	T256/T256-S: 256 Calls T512/T512-S: 512 Calls
Busy Hour Call Completions	T256/T256-S: 100,000 T512/T512-S: 200,000
Signaling Engine and Protocol	High Performance Multi-Threaded Core Signaling Engine Session Initiated Protocol (TCP & UDP) (RFC3261 Compliant)
Quality of Service	Type of Service Class of Service Differentiated Services Code Point
Availability	99.999%
Data Storage	T256/T256-S: 20,000 Hours of Voicemail and Voice Recording Storage or 4,000 hours of Videomail and Video Recording Storage T512/T512-S: 30,000 Hours of Voicemail and Voice Recording Storage or 6,000 hours of Videomail and Video Recording Storage Solid-State Drive (SSD)
Disaster Recovery	Front side access compact flash for system backup and recovery
Clustering	Local Cluster: Active/Hot Standby (IP Take Over; Must within the same subnet) Geographically Separated Cluster: Active Active (Able to be two different subnet)
Codecs	Auto Codec Negotiation Supporting G722.1C (Polycom® Siren™ 14) G722.1 (Polycom® Siren™ 7) G722 G711A G711U G729A (Pass thru) iLBC H.264 (Video)
RTP	Configurable RTP Paths (Forced via PBX or End to End)
DTMF	Inband and Out of Band (RFC2833)
Faxing	T.30 & T.38 with built-in fax receiver

Unified Communications Switch Specifications

Unified Communications Switch (continued)	
Quad Band Built-in SMS Gateway (Optional Module)	GSM 900/1800 GSM 850/1900 Front access SIM Card Holder and 50 Ohm SMA Antenna Connector
Ethernet Interface	6 x RJ45 Connectors supporting 10/100/1000 Half/Full Duplex
10BASE-T Cable Support	EIA Categories 3, 4, or 5 unshielded twisted-pair (UTP) (2 or 4 pair) up to 328 ft (100m)
100BASE-TX Cable Support	EIA Category 5 UTP (2 pair) up to 328 ft (100m)
1000BASE-T Cable Support	EIA category 6 UTP (recommended), Category 5E UTP, 5 UTP (2 pair) up to 328 ft (100m)
Console	DB9
Electrical	1 + 1 Hot Swappable Redundant Power Supplies Auto Resume After Power Failure
Status and Alerts	10 status LEDs + 1 audio alert
Operational input Voltage Ranges	Auto Sensing 90-132 VAC minimum 200-264 VAC maximum
Operational Input Current Range	7.5A (at 100 VAC nominal) 3.8A (at 240 VAC nominal)
Operating Temperature	0° to 40° C
Operating Humidity	10%~90% non-condensing
Cooling System	7 Auto Sensing Fans; 1 Internal, 2 Inlets (Side), 4 Outlets (Rear)
Non Operating Shock (with original packing)	350G, 2ms
Certification and Compliance	CE, FCC Class A, RoHS, CCC
Preferred Conference Infrastructure:	Polycom® RMX® Series Conference Platforms
Preferred IP Phones:	Polycom® SoundPoint® Series IP Phones Polycom® Business Media Phones
Preferred Video Conferencing Systems:	Polycom® HDX® Series Video Conference Systems
Supported Soft Clients:	Polycom® Telepresence m100 Linphone Counterpath® X-Lite and eyebeam
Supported Mobile Client:	frSIP® Mobile for Apple® iOS devices frSIP® Mobile for Android devices Polycom® Realpresence Mobile
Supported Wireless End Points	Polycom® DECT and Wifi Communications
Other Supported End Points	Cisco® IP Phones with SIP Support

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Gateway Specifications

Gateway (PRI/BRI/FXO/FXS)	
Form Factor	19" Rack Mountable 1U High, except MP11X Series
Dimensions	M800 Series: 320 (W) x 345 (D) x 45 (H) mm MP11X Series: 220 (W) x 172 (D) x 42 (H) mm
Codecs	Auto Codec Negotiation Supporting G711A G711U G729A
Fax Support	T.38 compliant (real time fax), Automatic bypass to PCM
Trunks	MP Series: Choice of 4, 8 FXO Ports for CO Trunks or 4, 24 FXS Ports for Analogue Devices M800 B Series: 4 BRI S/T ports M800 ET Series: Choice of 1 (30 channels) or 2 (60 channels) T1/E1/PRI Spans with 4 FXS Ports for Analogue Devises
PSTN Coding	E1 A-law T1 µ-law
Echo Cancelation	Enhanced (Carrier Grade) Echo Cancellation: G.165 and G168-2002, with 32, 64 or 128 msec tailsize
PRI Signaling Protocols (M800 ET Series)	ETSI/Euro ISDN, ANSI NI2 AND OTHER VARIANTS(DMS100, 5ESS), VN3, VN4, VN6
BRI Signaling Protocols (M800 B Series)	Net 3 ETSI EuroISDN (TBR3), VN4/6, QSIG
Analogue Signaling (MP Series)	Loop Start, Reverse Battery, Battery Disconnect
Switching	VoIP to circuit, and circuit to circuit (Tandem/TDM) switching
Maximum Call Rate	MP Series: 1,800 Calls / Hour / Gateway M800 B Series Series: 7,500 Calls / Hour / Gateway M800 ET Series: 7,500 Calls / Hour / Gateway
LAN Interface	MP Series: 10/100 BASE-TX, RJ45 M800 Series: 10/100/1000 BASE-TX, RJ45
Quality of Service	DiffServ, TOS, 802.1 p/Q VLAN tagging
Models Available	37501-M800-V-1ET4S 1 Port E1/T1 with 4 FXS Voice Gateway 37501-M800-V-2ET4S 2 Port E1/T1 with 4 FXS Voice Gateway 37501-M800-V-4B-12L-P 4 Port BRI Gateway 37501-MP114/4O 4 Port FXO Analogue Gateway (Non Rack Mountable 37501-MP118/8O 8 Port FXO Analogue Gateway 37501-MP112/2S 2 Port FXS Analogue Gateway (Non Rack Mountable) 37501-MP114/4S 4 Port FXS Analogue Gateway (Non Rack Mountable) 37501-MP124/24S 24 Port FXS Analogue Gateway with 24 Port Patch Panel and RJ21 Cable

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