SIP IP-PBX Communication Server

DX080/DX160/DX240

PLUG & PLAY

The DX080, DX160 and DX240 bundles Dial-Office software with an embedded communication server. Dial-Office is a high-end SIP PBX telephony software that provides advanced features such as call conferencing, unified messaging, detailed CDR, call monitoring, and much more.

The DX080 supports up to 8 extensions and trunks (IP & PSTN).

The DX160 supports up to 16 extensions and trunks (IP & PSTN).

The DX240 supports up to 24 extensions and trunks (IP & PSTN).

Pre-configured and pre-tested the DX080, DX160 and DX240 are painless to install, and simple to manage. This solution provides high quality voice and supplies end-users with interactive tools that can increase productivity, lower communications costs and enhance customer service.

ENHANCED FEATURES

Dial plan with Permissions

Voicemail to Email

Call Detail Reporting (CDR)

Unified Messaging

Interoperates with VoIP Service Providers

Hunt Groups/Basic ACD

DIALEXIA

FEATURES

Integrated Voicemail

Auto Attendant/IVR

Advanced Call Forwarding

Call Park/Hold

OA&M WEB Application

BENEFITS

SIP Compliant - Interoperable with SIP Devices

Lower Communication Costs with VoIP Termination

Fully Interoperable with Industry SIP Devices

Multi-Branch Communication & Remote Extensions

DX080/DX160/DX240

IP- PBX SPECIFICATIONS

Telephony Features

- 3-way Conferencing
- Basic Automatic Call distribution (ACD)
- Attended/Unattended Transfer
- Call Detail Records (CDR)
- Call Forward - On Busy/On No Answer
- Call Park/Hold/Retrieve, Call Queuing, Call Record,
- Call Relay, Call Routing (DID & ANI), Call Transfer,
- Call Waiting,/Caller ID, Caller ID with Name/Number
- Dial by Extension/Name
- Dial by PIN Permission
- Distinctive Ring (device dependant)
- Hunt Groups
- Music On hold
- Outbound Call Blocking
- Overhead Paging (through external speakers)
- Remote Call Pickup (PARK)
- Remote Office/Worker (NAT Traversal Support)
- Speed Dialing (device dependent)

Key Features

- Auto-attendant (multi-level) with IVR
- Automatic Route Selection (ARS)
- Distribution Lists
- E911 Support Capability (Call provider dependant)
- Email Notification of Voicemail Message
- Interactive Directory listing
- Instant Messaging & Chat (with Dial-Com Softphone)
- Interactive Voice Response (IVR)- Multi-site/location
- Message Waiting Visual Indication
- Message Waiting Stutter Dial tone
- Multiple User Voicemail Greeting (Customizable)
- Call Recording
- Presence Monitoring and User Status
- Unified Messaging
- URI Mapping Engine
- Voicemail to Email
- SIP trunking with ITSP

Management Features

- Auto-restart after Power Failure
- Browser-based Configuration & Management
- Error/Alarm Logs
- Server Logs
- SIP Session Information Logs
- User Settings
- Voicemail Web Interface (remote access)

EMBEDDED SERVER SPECIFICATIONS

General

- CPU: 1.2 GHz processor VIA Eden
- BIOS: AMI BIOS
- System Memory: 512MB DDR2
- Onboard LAN: 10/100Mbps LAN

Audio

- Audio Interface: Line out, Line in, Mic in

Ethernet

- Chipset: RealTek RTL8100B 10/100 Base-T
- Remote Boot ROM: built-in boot ROM function

Mechanical & Environmental

- Power Requirement: +5V @ 4.5A
- Operating Temperature: 0 ~ 60 C (32 - 140 F)
- Operating Humidity: 0% - 90% relative humidity, non-condensing
- Size (W x H x D): 170 x 124 x 58 mm
- Weight: 940g

PROTOCOL STANDARDS

- DTMF - Out of Band (RFC 2833)
- Locating SIP Servers (RFC 3263)
- Offer/Answer model for SDP Codec Negotiation (RFC 3264)
- Reliable Provisional Responses (RFC 3262)
- RTP/RTCP (RFC 1889)
- SDP (RFC 2327)
- SIP Event Notification (RFC 3265)
- SIP using UDP & TCP transports (RFC 3261)
  - Refer Method (RFC 3515)
  - Referred-By Header (RFC 3891)
  - Replaces Header (RFC 3892)
- Symmetric Response Routing (RFC 3581)
- Voicemail message indication - MWI (RFC 3842)

DEVICE INTEROPERABILITY

- Cisco (IP Phones, Digital Gateways)
- Polycom (IP Phones)
- Linksys (FXS, IP Phones)
- Mediatrix (FXS, FXO, Digital Gateways)
- Aastra (IP Phones)
- Quintum (FXS, FXO, Digital Gateways)
- Grandstream (FXS, FXO, IP Phones)

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