Technical Specification

Call Features

- Automatic Call Distribution: ACD
- Call Forwarding
- Call Waiting
- Calling ID
- Call Hold, Music on Hold
- Call Recording with custom notification
- Unattended Transfer (or "blind transfer")
- Attended Transfer (or "consultative transfer")
- Unconditional Call Forwarding
- Follow Me
- Extension Mobility
- Group and Selective Call Pickup
- Call Parking
- Three Way Calling
- Auto Attendant
- Click-To-Dial
- Connection select prefix
- Dialing Rules for every Extension Group
- Direct Inward System Access (DISA)
- Video Calls
- XML-RPC for third-party integration
- Restricting access to call features by Extension Group
- Admin and user's self-care interfaces
- VoIP Gateways detection and automatic configuration for:
  - Grandstream 4104, 4108
  - Sipura SPA-400
  - MV 370
- SIP UA detection and automatic configuration for:
  - Cisco ATA-186, ATA-188, 7905G, 7912G
  - Sipura SPA-1000, SPA-1001, SPA-2000, SPA-2002, SPA-2100, SPA-3000, SPA-842
  - Linksys PAP2-NA, SPA-901, SPA-921, SPA-922, SPA-941, SPA-942
  - Leadtek BVP 8762
  - IPtel TTphone (ACT)
  - IpDialog SipTone II
  - Grandstream HT286, BT101, BT102, GPX-2000
  - Polycom SoundPoint IP301, IP501, IP601
  - Aastra 480i, 51i
- APC UPS Compatibility

Voicemail - Faxmail

- Unavailable message
- Busy message
- Message forwarding
- Password protected IVR
- IVR and Web message retrieval
- E-mail notification & forwarding
- Receive and store faxes as PDF
- Message Waiting Indicator: MWI

SIP

- RFC3261 compliant
- Components: SIP registrar and proxy, B2BUA, RTP proxy
- Codecs: G711, G729a, Fax pass-through, Fax T38
- Able to register multiple accounts on multiple SIP servers
- Transparent or Substituted CallerID for outbound calls
- Inbound call routing based on originators SIP server, account, called number
- Outbound call routing based on prefix and preference table
- Outbound dialing rules for every connection
- Failover routing over multiple SIP servers
- PSTN fallback

Firewall support

- LAN only behind multiple NATs
- WAN+LAN network bridging

Accounting

- External: RADIUS Cisco VSA compliant, ODBC
- Internal: CDRs, call recording
- Archive management: Optical media

Management

- Multilingual: EN, FR, RU, CZ, NO, SV, SP, IT
- Voicemail: EN, RU
- HTTP/HTTPS access list and password protected
- AJAX Operator's Console allowing drag and drop calls management
- Configuration Backup and Restore: to/from Optical Media OR File via HTTP
- Dynamic cluster architecture with self-adjustable topology and auto configuration

PSTN

All PSTN ports have been approved for use in Australia, Europe, Canada and the United States.

PRI Switch Compatibility

- Network or CPE
- AT&T 4ESS
- EuroISDN
- Lucent 5ESS
- National ISDN type 1
- National ISDN type 2
- Nortel DMS100

CAS & RBS Voice Modes

- A-law, Mu-law, and linear modes supported
- E&M Wink
- Feature Group D
- Ground start (FX0 and FXS) with optional disconnected supervision
- Transparent or Substituted CallerID for outbound calls
- Inbound call routing based called number (DID)
- Outbound call routing based on prefix and preference table
- Outbound dialing rules for every connection
- Failover routing over multiple PSTN connections
- VoIP fallback

The following PSTN cards are compatible with PBXpress:

- Digium TE205P
- Digium TE405P
- Digium TE110P
- Digium TE207P/TE210P/TE212P
- Digium TE407P/TE410P/TE412P

Hardware requirements

- RAM: 512MB minimum
- Hard drive: minimum 20GB
- Minimum CPU speed: 1GHz

PBXpress compatible phone, adaptor or gateway must satisfy the following requirements:

- It must use SIP protocol
- It should have a G729 codec