PBXpress: Recording VoIP PBX with Dynamic Cluster Technology

PBXpress is a replacement of traditional PBX in the new era - the era of the Internet. All of your phones are connected and provisioned via an IP network, your calls can be delivered to any place on Earth via Internet at a fraction of the usual cost, and all system configuring is done from your web browser! In addition to new exciting features, PBXpress offers the reliability of good old telephony service: in case of any problem with your Internet connection, PSTN fallback is engaged, and the call goes via one of the telco phone lines.

- Reduced hardware costs
- Huge cost savings on long-distance and international phone calls
- PSTN fallback to ensure that calls are always completed
- Support for multiple VoIP/telco providers
- Easy deployment and management
- Scalability up to 500 registered IP phones and 100 concurrent calls
- Ability to record and playback all telephone conversations
- Integrated voicemail, faxmail, auto-attendant, and music on hold
- ACD - automatic call distribution queues
- Solution for networks with firewall/NAT
- Integration with external billing systems via RADIUS
- Dynamic cluster architecture with self-adjustable topology and auto-configuration

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Technical Specifications

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
- Hunt Groups
- Three Way Calling
- Auto Attendant
- Click-To-Dial
- Connection select prefix
- Dialing Rules for every Extension Group
- **Direct Inward System Access (DISA)**

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

Firewall support
- LAN only behind multiple NAT's
- WAN+LAN network bridging

Recording Capacity
a) **F400** - 1020 hours (~6 weeks)
b) **B200** - 1020 hours (~6 weeks)
c) **TE1000** - 17000 hours (~2 years)
d) **TE2000** - 39100 hours (~4 1/2 years)

SIP
- RFC3261 compliant
- Components: SIP registrar and proxy, B2BUA, RTP proxy
- Codecs: G711, G729a, Fax pass-through
- 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 dialling rules for every connection
- Failover routing over multiply SIP servers
- PSTN fallback

Accounting
- External: **RADIUS** Cisco VSA compliant, **ODBC**
- Internal: CDR's, call recording
- Archive management: Optical media

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

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

a) **F400** - 4 FXO
- The FXO module allows terminating analog "Plain Old" Telephone Systems' Lines (POTS).
- Polarity Reversal Detection
- Tone and Pulse Dialling
- CallerID

b) **B200** – Single BRI
c) **TE1000** – Single-span T1/E1
d) **TE2000** – Dual-span T1/E1

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 disconnect 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 dialling rules for every connection
- Failover routing over multiply PSTN connections
- VoIP fallback

Power
- 110 – 240 V
- UPS: compatible with any APC with USB interface