



XENTINO FLEXY 1823 IP PBX NEXT GENERATION IP PBX PHONE SYSTEMS

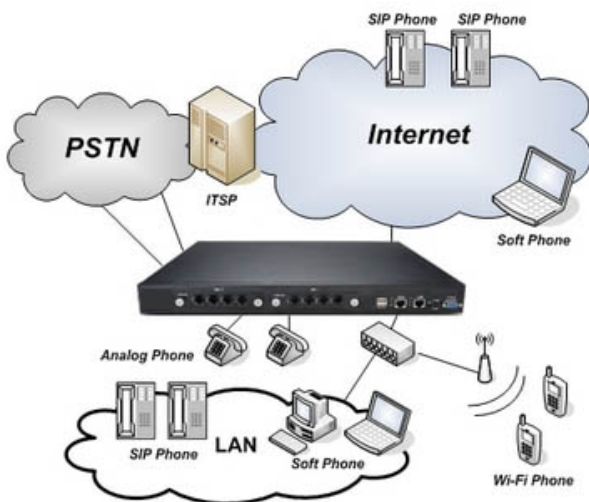
Xentino FLEXY 1823 is designed and optimized for SMB enterprise daily communications with easy configuration and cost advantages. Designed as a highly reliable, stable, and open telephony solution, the FLEXY 1823 IPPBX System provides the essential features of telephony which are required for around 200 users on their telecommunication needs.



The FLEXY 1823 is an embedded system with built-in SIP Proxy/Registrar Server and NAT traversal that make it perfect for SMB enterprise usage. The FLEXY 1823 provides not only basic call functions like Call Forward, Call Transfer on traditional PBX System, but also many advanced functions including Meet-Me Conference, Call Parking, and Voice Mail System with E-mail Notification and Attachment etc.

These advanced functions are impossible on traditional PBX. The FLEXY 1823 provides IP-based communications, 3-layer IVR, Meet-Me Conference, Call Parking, Voice Mail System with E-mail Notification and Attachment, and Web GUI Management. The FLEXY 1823 also provides 2 PCI slots that can insert FXO or FXS module cards, so enterprise can seamlessly integrate with existing traditional PBX. With the FLEXY 1823, standard SIP phones such as Xentino IP-Phones can be easily integrated in your office.

FLEXY 1823 is established by a very stable embedded system platform. All the System Configurations, Voice Mail and Call Detail Record are stored in the flash disk. Administrators could backup the system configuration file to their own computer through web GUI. If the system is damaged, administrator only have to restore the configuration file to a new FLEXY 1823. No more re-configuration is needed and minimizes the maintenance effort.



Supported Standards

RFC 3261, RFC 3311, RFC 3515
RFC 3265, RFC 3892, RFC 3361
RFC 3842, RFC 3389, RFC 3489
RFC 3428, RFC 2327, RFC 2833
RFC 2976, RFC 3263, RFC 3264

SIP Registrar

- Static/Dynamic Registration
- Configurable Expiry Time
- MD5 Authentication
- Handle Loose RFC-Compliant Phones
- Resilient Message Retry Mechanism
- Seeding Historical Registrations

SIP Proxy

- Stateful Proxy Server
- NAT Traversal for Clients
- Inter-Proxy Call Hand-Off
- Outbound Proxy behind NAT Device

PBX Features

- Supported Codec G711, G723.1, G726, G729
- Support Call Hold, Call Waiting, 3-way Call Conference with Feature Phones
- Built-in in-line Call Transfer
- Unconditional, Unavailable, Busy Call Forward
- Per-Calling-Number Forward and Rejection
- Group-Based Call Pick-up
- Call-Parking
- Multi-Room Meet-Me Conference
- Call Privilege Grouping
- Support T.38 FAX
- Support T.30, T.38 Fax Pass-Through
- On-line Conversation Recording With Feature Phones (Optional)
- Caller ID Detection
- Echo Cancellation(G.168 support)
- FXO Interface for PSTN Inbound/Outbound
- FXO Disconnection Tone Detection
- FXO/FXS Trunk
- Trunk Hunting
- Busy Call Back
- Life-line Priority Call
- In-Band/RFC2833/SIP-INFO DTMF Translation
- Inter-PBX SIP Trunking
- Intra-PBX Stackable Trunking Over Ethernet

- ITSP Account Sharing for Extensions
- Interoperable with Cisco CallManager, CCME, IOS SIP Gateway, Unity, CUE, 79XX, ATA Wired/Wireless Phones
- Music on Hold
- Direct Line to Extension / DID to Extension
- Direct Line by Called Number / DID by Number
- Direct Line by Group / DID by Group
- Blacklist of Number Patterns

Auto Attendant

- 3-Layer IVR
- Holiday Setting
- Configurable Greeting prompts
- Music on Ringing Extensions
- Forward to Voice Mail on No-Answer

Voice Mail

- User Authentication by PIN
- Multilingual(English, Japanese, Mandarin)
- Multi-Folder Archive
- Fast-Forward/Rewind/Undelete
- MWI Notification
- E-mail Notification and Attachment (Unified messaging)
- Personal Reception on Unavailability and Busy
- Voice-mail Forwarding
- Reply Call or New Call in Voice-mail Menu

Storage

- Built-in 20GB Hard Disk Drive for Voice-Mail
- *Support Removable USB 2.0 HDD for Voice-Mail and System Configurations Backup*

Meet-Me Conference

- Multiple Rooms with Configurable Number and PIN
- Up to 18 Parties per Conference (allow 8 Transcoding Parties)
- Music on First Dial-in Party
- Hot Key for User to Leave the Conference
- Hotkey for Admin to Manage the Conference

NAT

- Auto NAT Discovery and Traversal
- Built-in STUN Client
- RTP Proxy
- RTP Port Range Designation

Administration

- Web-Based Configuration
- Flat System Event Syslog
- Flat Call Detail Record (CDR)
- Extension Status Display
- TFTP Server and TFTP Repository Maintenance
- Network Time Protocol Time Synchronization
- Real Time Clock Setting
- DHCP Server with Multiple Partitions, per-MAC IP Binding, List of options
- Configurable Time Zone
- Firmware Upgrade through Web Interface and console

Network Management

- DHCP/PPPoE/Static IP on WAN
- Support MAC Clone on WAN
- Static Routes for LAN
- Firewalling on Predefined Services
- Virtual Server
- NAT for Outbound Traffic from LAN
- QoS Queuing Mechanism for VoIP and Data Traffic
- DNS Forwarder and Dynamic DNS
- SNMPv2
- Bandwidth and/or Call Admission Control
- Intelligent Bandwidth and DSP Trade-off

Feature Cards (optional)

- 4-port FXO Card
- 4-port FXS Card
- 4-port ISDN BRI TE Card
- 8-port FXO Card
- EST58800 Security Acceleration Card

Maximum Capacity

- 100 Extension Registrations
- 100 Voice-Email Accounts
- 30 Concurrent Calls

