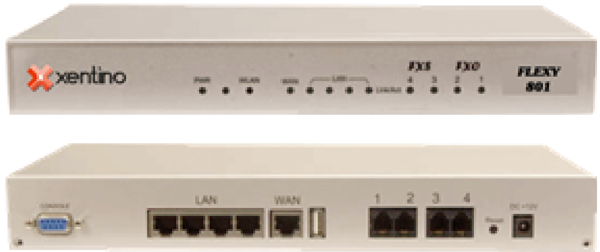
XENTINO

FLEXY 1801 IP PBX

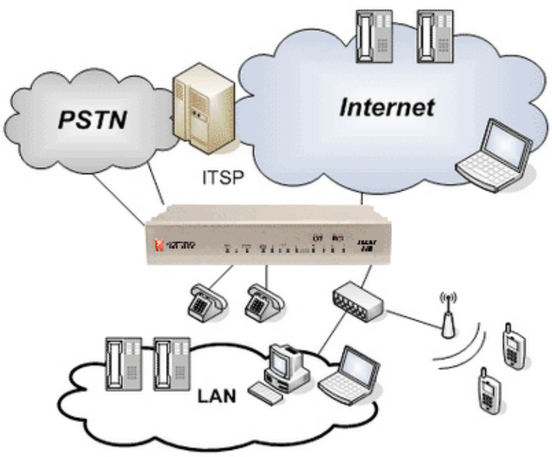
NEXT GENERATION IP PBX PHONE SYSTEMS

Xentino FLEXY 1801 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 1801 IPPBX System provides essential features of telephony which are required for SMB users on their telecommunication needs.



The FLEXY 1801 is an embedded system with built-in SIP Proxy/Registrar Server and NAT traversal that make it perfect for SMB enterprise usage. The FLEXY 1801 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 1801 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 1801 also provides 4 POTS ports with either FXO or FXS interface, so enterprise can seamlessly integrate with existing traditional PBX. With the FLEXY 1801, standard SIP phones such as Xentino IP Phones can be easily integrated in your office.



FLEXY 1801 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 1801. No more configuration is needed and minimizes the maintenance effort.

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FLEXY 1801 IP PBX

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
- Registration Proxy to External Registrars
- Configurable PBX Caller ID
- User Profile
- Handle Loose RFC-Compliant Phones
- Resilient Message Retry Mechanism
- Seeding Historical Registrations

SIP Proxy

- Stateful proxy server
- Call-based MD5 authentication
- NAT Traversal for Client
- Inter-Proxy Call Hand-Off
- Support Outbound Proxy behind NAT Device

PBX Features

- Support Call Hold, Call Waiting, 3-way Call Conference with Feature Phones
- Unconditional, Unavailable, Busy Call Forward
- Built-in in-line Call Transfer
- Per-calling-number Forward and Rejection
- Group-Based Call Pick-Up
- Call-Parking
- Multi-Room Meet-Me Conference
- Auto-Attendant
- Voice Mail System
- Call Privilege Grouping
- Fax Pass-Through
- On-line Conversation Recording with Feature Phones (Optional)
- FXO Module for PSTN Inbound/Outbound
- FXS Module for Analog Phone
- FXO Hunt Group
- Caller ID Detection
- Echo Cancellation(G.168 support)
- In-band/RFC2833/SIP-INFO DTMF Translation
- Inter-PBX SIP Trunking
- Support 5 SIP Trunks
- ITSP Account Sharing for Extensions
- Interoperable with Cisco CallManager, CCME, IOS SIP Gateway, Unity, CUE, 79XX, ATA
- Music on Hold
- Direct Line/DID
- Blacklist of Number Patterns

Auto Attendant

- Configurable Greeting Prompts
- Key to Reach Operator
- Timeout Interval and Timeout Action
- Music on Ringing Extensions
- Forward to Voice Mail on No-Answer

Voice Mail

- User PIN
- Default Language Support: English
- Multi-Folder Archive
- Fast-Forward/Rewind/Undelete
- MWI Notification
- E-mail Notification and Attachment (Unified messaging)
- Personal Reception on Unavailability and Busy
- Voicemail Forwarding
- Reply Call or New Call in Voice-Mail Menu

Storage

- Built-In USB Storage

Meet-me Conference

- Multiple Rooms with Configurable Number and PIN
- Up to 8 Parties
- Music on First Dial-in Party
- Hot Key to Leave Conference

NAT

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

FC CE

Relational Provision

- Logical Partition/Relation on Users and Trunks
- Logical Provision on Outgoing and Incoming Calling Search Scopes
- Rich Dial-Plan Expressiveness thru Route Patterns
- User Privilege Propagation Over Intra-Trunks
- Object-Oriented Provisioning Paradigm

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

Network Management

- DHCP/PPPoE/Static IP on WAN
- LAN IP and Netmask Specification
- 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
- Support SNMPv2

Feature Modules

- 2-Port FXO
- 2-Port FXS
- 1-Port FXO + 1-Port FXS with Emergency Loopback
- 2-Port ISDN BRI TE**

Maximum Capacity

- 30 Extension Registrations
- 30 Voice-Mail Accounts
- 10 Concurrent Calls