VoiceCon
VoIP SIP IP PBX
VOI-9200

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VoIP SIP IP PBX with 2 FXS and 2 FXO Gateways

Integral VoIP Solultion
IP telephony is gaining popularity while traditional private branch exchange (PBX) and key systems are in decline. With a compound annual growth rate exceeding 35 percent, IP PBXs are projected to become an integral part of VoIP solution deployments for business operations.

In this light, LevelOne is announcing the launch of an IP PBX specially designed to assist Small and Mid-sized Business to provide a secure SIP-enabled VoIP solution that allows them to take advantage VoIP technology.

Feature Rich, Secured and Easy to Manage
The LevelOne VOI-9200 is a feature rich, highly secured and advanced IP PBX that integrates traditional PBX services and VoIP services into one device.

The VOI-9200 IP PBX features a built-in Auto Attendant for internal and external attendant functions, a built-in Voice Mail server as well as all class 5 traditional PBX functions.

The IP PBX is easy to install and is SIP compatible. It allows for the registration of up to 30 extension numbers that is managed by a flexible routing plan to assist users to choose the lowest telephony service provider and route. It further supports NAT-T (NAT Transversal) to assist IP-phones to pass through NAT systems, reducing connectivity errors.

An outstanding feature of the IP PBX is its ability to manage conference calls and conference meetings in different locations, regardless the telephone system participants are using. PSTN, ITSP and internal lines are centrally managed and diverted to the locations where the call needs to be transferred. It also allows for a centralized control to set-up and maintain all IP phones registered on the IP PBX.

Bridge between PTSN and IP Networks
The VOI-9200 also acts as bridge that connects all branches and sites to the PSTN network, either using the service provider's VoIP infrastructure or a VoIP gateway on the premises.

User Friendly Web Based Management

This IP PBX is also remotely managed via the Internet. Fault monitoring, performance management, configuration, new extension registration and changes to current settings are all control locally or remotely via user friendly web browser interface.

VOI-9200

Key Features

- Advanced Telephony features with built-in voice mail and conference calling
- Supports a variety of popular Telephony codecs (G723.1, G726, G729a/G729b, G711 A/u-law)
- Flexible scalability and reduce cost of ownership
- Complies with IETF SIP v2 RFC3261 standard
- Intelligent routing for inbound and outbound calls
- 30 extensions, 30 voicemail accounts, 10 concurrent calls
- Web-based user-interface for easy configuration

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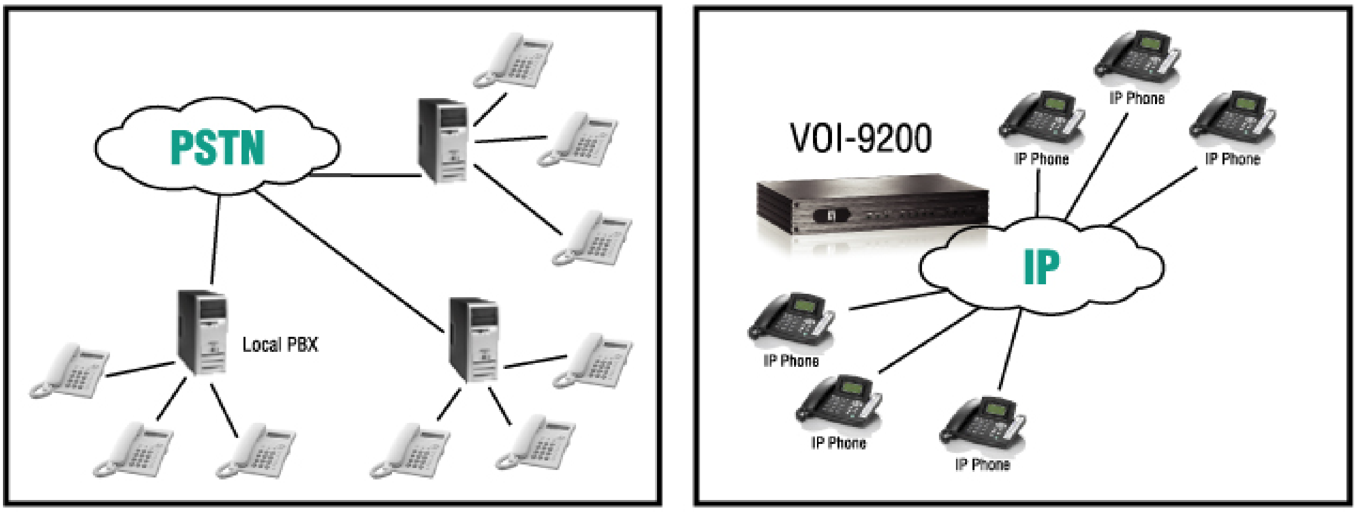
Product Diagram

Convergence transforms telephony



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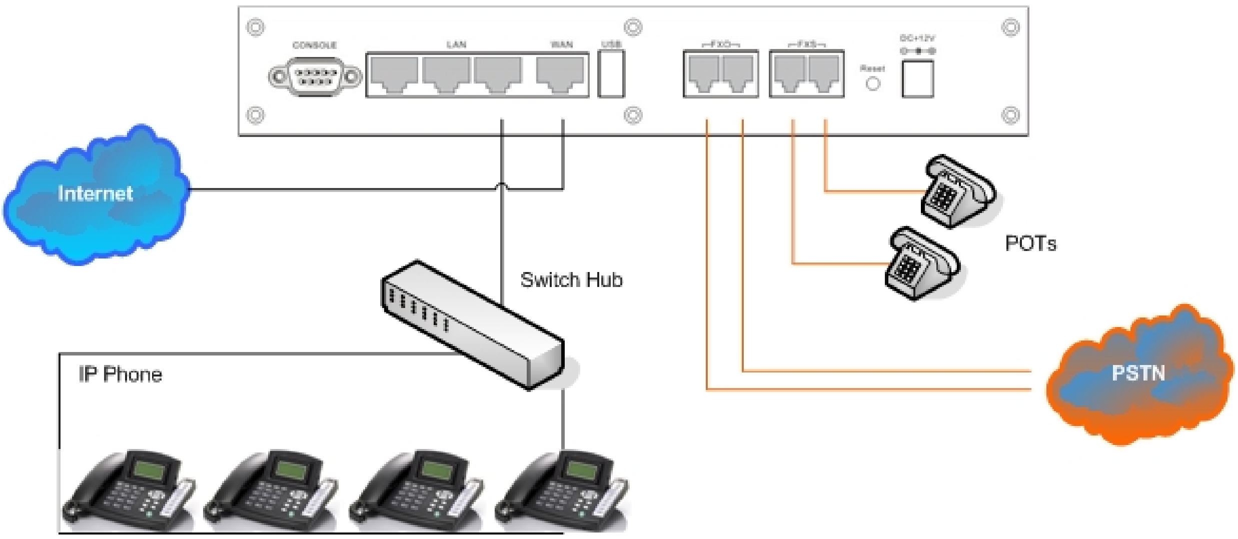

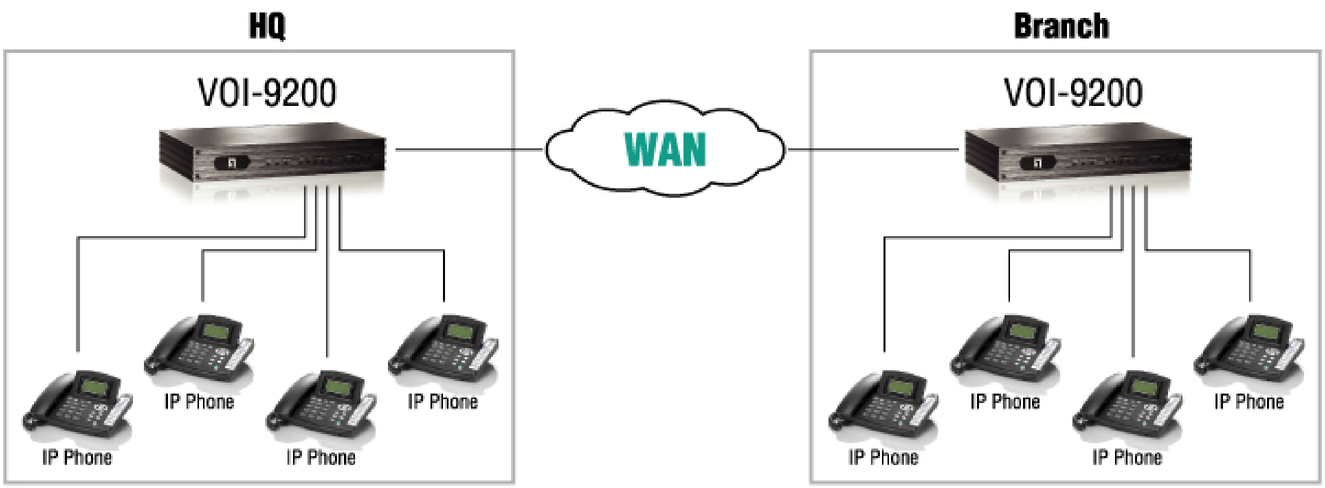

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Technical Specification

Hardware Specifications

Ports
4 LAN ports
1 WAN port
Internal USB2.0 port
External USB2.0 port
2 FXS
2 FXO

Fax Support
T.38 compliant

Codec Support
G.723.1, G.729, G.711 (µ/A-Law), G.726 ADPCM

Echo canceller
G.168-2002 compliant with programmable echo tail of up to 128ms

Integrated VoIP Products
Seamless integrated with VOI-7000 / VOI-7100 IP Phone and VOI-800x VoIP Gateway

IP PBX Features.

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
- Proxy server
- Call-based MD5 authentication
- NAT traversal for clients
- Outbound proxy with or without WAN
- Inter-proxy call hand-off

PBX Features
- 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
- Auto-attendant
- Voice mail system
- Call privilege grouping
- FXO interface for PSTN

Inbound/outbound
- FXO disconnection tone detection
- FXO hunt group
- Caller ID detection
- Echo cancellation
- In-band/RFC2833/SIP-INFO DTMF translation
- Support 5 SIP trunk
- Intra-PBX stackable trunking over Ethernet
- FWD/Vonage account sharing for extensions
- Interoperable with Cisco CallManager, CCME, IOS SIP gateway, Unity, CUE, 79XX, ATA
- Call admission control for wired/wireless phones
- Music on hold
- Direct line
- Outbound 900/0204 blocking

Auto Attendant
- Configurable Greeting
- Key to reach operator
- Timeout interval and timeout action
- Music on ringing extensions
- Forward to voice mail on no-answer

Voice Mail
- User PIN
- 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 voicemail menu Storage
- Support USB 2.0 storage media

Meet-me Conference
- Multiple rooms with configurable number and PIN
- Music on first dial-in party
- Hot key to leave conference

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
- NAT for outbound traffic from LAN
- DNS and dynamic DNS
- QoS queuing mechanism for VoIP and data traffic

Maximum Capacity
- 30 extension registrations
- 30 voicemail accounts
- 10 concurrent calls with RTP
- 4 analog calls

NAT
- Auto NAT discovery and traversal
- Built-in STUN client
- RTP proxy
- RTP port range designation

Relational Provision
- Logical partition/relation on users and trunks
- Logical provision on outgoing and incoming calling search scopes
- Rich dial-plan expressiveness through route patterns
- Object-oriented provisioning paradigm

Ordering Information

VOI-9200: VoIP SIP IP PBX w/ 2FXS and 2FXO