G-1212

G-1212 ANALOG (12 FXO)

The G-1212 IP PBX is the ideal small business solution for up to 60 users, which combines full function support of a PBX with the integration of both analog and VoIP telephony. It not only supports the telephony needs of small businesses but also offers compelling features such as: “Never miss a call” - where the system will track a person based on the phone numbers provided; “Automated Attendant” - look more professional by having an automated system answer calls and direct users; “Conference Rooms” - never pay for multiple person conference bridges again, which saves money.

The business phone system has four Ethernet ports, two Foreign Exchange ports for connecting to a phone or fax machine, an echo canceller and functionality for connecting telephone lines and two USB ports, transforming this into a truly powerful and flexible system. The G-1212 IP PBX is a standalone solution that can be installed on any network quickly and easily with the aid of its web browser based configuration that supports wizard based setups.

- Four Ethernet ports (3 LAN, 1 WAN)
- 12 analog phone lines (12 FXO)
- Two analog extensions (2 FXS) for phone or fax
- Two high speed USB ports
- Hardware echo cancellation
- Voicemail to email conversion
- Integrated conference rooms
- Installs simply and quickly
- Supports VoIP out of the box; no upgrades required

POSITRON
Telecommunication Systems

Software Features

Phone Provisioning

- Automatic phone provisioning using:
  - DHCP
  - SIP SUBSCRIBE option (Yealink, Snom)
  - Multicast DNS option (Aastra)
- Manual phone provisioning option
- Interoperability tested with Yealink, Snom Aastra, Polycom and Cisco SPA brands
- Expansion module provisioning (Yealink, Snom, Aastra)
- Custom phone templates

User Provisioning

- User bulk provisioning option
- User template assignment
- Mobile phone dispatch option
- Courtesy phone extension
- Forward external number only (secretary call screening)

Audio Conferencing

- Meet-me style room
- Flexible management of number of conferences
- Participant code access
- Moderator lock
- Music-on-hold per conference

Caller ID

- Caller ID for outgoing calls on SIP lines
- Caller ID for internal calls and remote offices
- Caller ID blocking
- Caller ID on call waiting
- Caller ID prefix and suffix
- Caller ID per user, per area code, global

Call Queues

- Agent log in
- Skill-based routing
- Priority queuing
- In queue music or messaging
- Remote queue membership
- Queue announcements
- Queue visualization
- Caller breakout options

Intercom and Paging

- Zone paging
- Desktop paging
- Extension paging
- Overhead paging

G-1212 Features

Software Features (con't)

Calling Features

- Call park
- Call hold
- Park and page
- Ring groups (ring all, in sequence, audio-out)
- Playback option per ring group
- Follow me with caller announcement option
- Dial by name directory
- Phone number alias
- Group call pickup
- Directed call pickup
- Spy call monitoring
- Direct inward dialing
- Call routing per schedule
- Call routing per IVR selection
- Call routing per caller ID
- Call forward all
- Attended transfer
- Blind transfer
- Busy Lamp Field
- On-demand call recording – 1 call at a time
- Strict call accounting
- Flexible outgoing call routes
- Failover outgoing call routes
- Flexible incoming call routes
- Remote users
- Inter-branch calling
- Point-to-point video calling
- Incoming URI calling
- Audio-in extension
- Routing to user voice mailbox
- Corporate call back (DISA)
- 15 concurrent call channels

Voicemail

- Deposited on plug-in USB memory stick
- Custom voicemail greetings
- Voicemail-to-email
- Voicemail forwarding
- Folders per mailbox
- Voicemail management from user portal
- Caller breakout from voicemail options

Automated Attendant

- Unlimited steps
- Cascaded IVRs
- On-demand time frames
- Time-day-date service
- Voice prompts
- Music on hold (GSM files, audio-in)
- User authentication
- Custom sound manager (GSM)

System Features

System Management

- Web-based (http) administrator interface
- Remote management capability
- Web-based user portal
- Operator console for call handling
- Backup and restore configuration
- PBX and network diagnostics
- Call recording management
- Call detail records management

Trunk Types

- SIP
- Analog
- Music-on-hold per trunk

Hardware Specifications

- 12 FXO RJ-11
- 2 FXS RJ-11
- 4 Gigabit Ethernet RJ-45
- Standard audio input mono jack 1/8
- Standard audio output mono jack 1/8
- Hardware based G.168 echo cancellation chip
- Rack mountable
- Real-time clock with independent power source
- Per-country tone indications

LAN Interfaces

- 1 WAN Ethernet Port
- 3 LAN Ethernet Ports

Codecs

- G.711 u-law & a-law
- H.264

Network Services

- DHCP client and server
- NTP client and server
- TFTP server
- LAN and WAN monitoring

Security

- Blacklist per IP range
- SIP intrusion detection with automatic blocking
- Embedded firewall
- Access control list for SIP registration

SIP Interface

- Version 2.0
- P-asserted indentity header
- Support for a range of source IP on a single trunk
- VoIP provider templates
- NAT traversal for remote extensions
- DTMF modes: RFC2833, SIP info, inband, auto

Storage

- USB interface memory stick

Dimensions

- Width: 12 inches (30.5 cm)
- Depth: 5.25 inches (13.3 cm)
- Height: 1.83 inches (4.6 cm)

Environmental

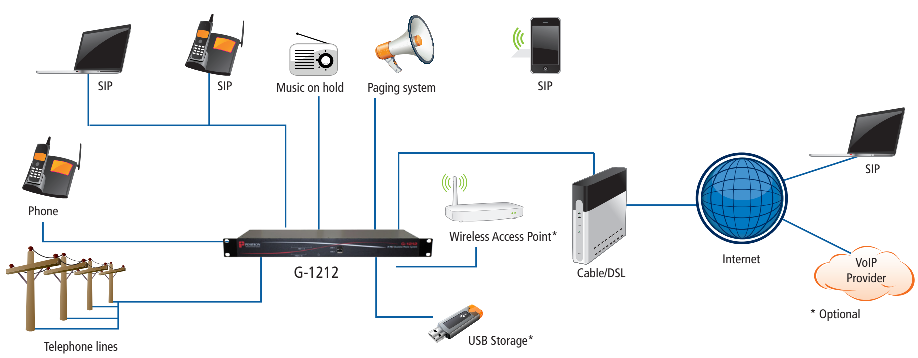
- Operating temperature: 32 to 104°F (0 to 40°C)
- Storage temperature: -4 to 185°F (-20 to 85°C)
- Humidity: 10% - 80% non-condensing

Warranty

- 1 year

Ordering Information

- Product code: 70-00061



Doc#: G-1212 DS-0912

PBX Built In | WEB Management | FXO 12 Port | FXS 2 Port | 10/100 Ethernet | SIP Protocol | VoIP Savings

POSITRON Telecommunication Systems

Positron Telecom • 5101 Buchan Street, Suite 200, Montreal, Quebec, Canada H4P 2R9 • Tel: (514) 664 4719 Toll Free: 1 866 434 3944 • Email: info@positrontelecom.com • www.positrontelecom.com