XENTINO

VERSO 1212 IP PBX

NEXT GENERATION IP PBX PHONE SYSTEMS

Xentino VERSO 1212 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 VERSO 1212 IPPBX System provides essential features of telephony which are required for SMB users on their telecommunication needs.

The VERSO 1212 is an embedded system with built-in SIP Proxy/Registrar Server and NAT traversal that make it perfect for SMB enterprise usage. The VERSO 1212 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 VERSO 1212 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 VERSO 1212 also provides 4 POTS ports with either FXO or FXS interface, so enterprise can seamlessly integrate with existing traditional PBX. With the VERSO 1212, standard SIP phones such as Xentino IP Phones can be easily integrated in your office.

VERSO 1212 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 VERSO 1212. No more configuration is needed and minimizes the maintenance effort.

Feature	Benefit
A.P. Embedded IP-PBX	Built in WiFi provides easier installations without the need for messy cabling. 802.11 b/g
Full Function PBX	Rich in function & features for smoother transition from traditional PBX to IP PBX.
Emergency Live Lines	In the event of a power failure “Emergency Live Lines” takes over to enable calls for dialing out.
Voice Mail	Easy retrieves Voice Mail messages via handset or IP/PSTN trunks.
Web Interface	Easily configured via simple and intuitive web interface configuration pages.
4 VoIP Trunks	Allows selection of the ideal connection rate via multiple profile registration.
12 User with IP or Analog phone	Supports up to 12 users (IP or Analogy handsets).
Built-in NAT Router	Allows an office network to share a single IP network on the internet.

Features		
Auto Attendant	• 10 unique auto attendants • Time schedule options for business & holiday hours	• Customized Auto Attendant • Configurable Auto Attendant Flow
VoiceMail	• 100 minutes of storage capacity • Messages can be re-played, saved, deleted or forward to other extension.	• Off site message retrieval • Message Waiting Indicators • Date & time stamping
Site to Site Calls	• Connects up to 32 IP PBX sites • Seamlessly transfer calls between IP PBX	• Calls between IP PBX over the IP are free – no external SIP server required.
Remote Users	• Up to 3 end-devices per user • Remote users connect to IP PBX via Intranet or Internet.	• Remote plug-n-play with G-Tek IP Phones. • Works with SIP v2.0 compliant IP Phones
Security	• System backup and restore	• Password protected: - access & configuration - reset to factory default -admin & user access
Calling Feature	• 3 ways conference G-Tek IP Phones • Configurable dialing plan to dedicated trunk • View status of all trunks & extensions • Endside calling features: - Audited outside line seizure	-Call transfer, forward, hold, pickup, redial, part, retrieve -Caller ID -Dial by name -DND, DID -Flexible dialing plan -Speed dialing
Phone Interface	• 2 FXO (CO) & 2 FXS (Extension) • Max 12 user extensions (2 Analogs / 12 IPs) with voicemail • Support SIP v2.0	• 2 Live line ports for a dedicated analog phone ensures connectivity during power failure
VoIP Protocol	• Session Description Protocol (RFC 2327) • Session Initiation Protocol (RFC 3261) • DTMF transport (RFC 2833) • RTP/RTCP (RFC 3550/3551) • MD5 for SIP authentication (RFC 2069/2617)	• Refer Method (RFC 3515) • Reliability of Provisional Responses (RFC 3262) • Offer/ Answer Model (RFC 3264) • Event Notification (RFC 3265) • Symmetric Response Routing (RFC 3581)
Audio	• G.711 a-law and u-law, G.723.1, G.729 • DTMF tone generation & detection • Integrated Media proxy	• Echo cancellation • 4 VoIP Channels
Network Interface	• One RJ-45 10/100 Base-T WAN port with connectivity, activity & speed LEDs for Internet connection from DSL modem or router	• 3 switch RJ-45 10/100 Base-T LAN ports with connectivity, activity & speed LEDs
Networking	• Build-in company internet & Intranet site • WAN: DHCP client/ Fixed IP/PPPoE supported • LAN: DHCP server supported • QOS supported • Network layer firewall security	• Network service port forwarding capability • Transparency mode supported • NAT (Network Address Translation) Routing table
Web Administration	• Built-in web server for easy configuration via web browser • Auto Configure G-Tek IP Phone	• Quick & Easy firmware upgrading • Built-in online help functions
VoIP Trunking	• Supports both ITSP & Legacy PSTN lines simultaneously • Support up to 4 SIP proxies upper registration • Support DTMF Caller ID	• Apply different dialing rules to different telephone services (ITSP/ PSTN) • Compatible with SIP v2.0 proxies
Electrical Information	• Line voltage: Output 12V DC/ Input 100-240 Volt AC Adapter	• FCC part 15 class B, FCC part 68 & CE
Size & Weight	• Dimensions: 200 L x 150 W x 32 H mm	• Weight: 502 gram