- **Up to 80 concurrent calls, 50-attendee conference**
- **Integrated 4 PSTN Trunk FXO Ports Plus 2 FXS Ports**
- **Comprehensive features for unified communication**
- **High level of security protection (SRTP, TLS & HTTPS)**

The CLIXXO IPX22K-UC is an IP PBX appliance designed to bring enterprise-grade unified communications and security protection to all levels of businesses at an unprecedented price point without any licensing fees, costs-per-feature, or recurring fees. The CLIXXO IPX22K-UC enables enterprises to unify multiple communication technologies, such as comprehensive voice, fax, calling, conferencing, video/audio surveillance, data tools, security surveillance, mobility, and facility access management into one commonly managed or accessible network.

With an advanced hardware platform and software functionalities, the CLIXXO IPX22K-UC can support up to 500 registered users and offer effortless setup and deployment via the web-browser user interface. Besides auto-discovery of diverse endpoints and auto-provisioning, the CLIXXO IPX22K-UC series offers a set of comprehensive features, including customizable call-routing, multi-level IVRs, call queues, auto-attendant, call detail records (CDR), multi-site peering, voicemail/fax forwarding to email and more.

Supports up to 500 users, 80 SIP trunk accounts, up to 80 concurrent calls, 50 conference attendees

Integrated 4 PSTN trunk FXO ports, 2 analog telephone FXS ports with Lifeline capability

Supports up to a limitless-level IVR (Interactive Voice Response)

Built-in call recording server; recordings accessed via web user interface

Supports call queue for efficient call volume management

Built-in Call Detail Records (CDR) for tracking phone usage by line, date, etc.

Support voicemail and fax forwarding to email

Integrated LDAP and XML Phonebooks, flexible dial plan

Zero configuration provisioning of Mainstream SIP endpoints

Highest level of security protection using SRTP, TLS and HTTPS encryption

Hi-speed network ports with Integrated NAT router and built-in firewall

Multi-language auto-attendant to efficiently handle incoming calls

1.5GHz ARM Quad-core processor, 1GB DDR RAM, 8GB EMMC Flash

Unique Selling Points

- **Hi-Interoperability with Network**

CLIXXO IPX22K-UC has the super NAT network adaptability. In the system deployment, the remote SIP extension registered to the CLIXXO IPX22K-UC need not any NAT traversal setting.

- **Excellent Compatibility**

Without NAT traversal setting, CLIXXO IPX22K-UC could be compliant with other mainstream SIP endpoints or components with changeable IP addresses, which effectively reduces complexity of configuration.

- **Flexible Resource Allocation**

CLIXXO IPX22K-UC optimizes system resource utilization and system efficiency via stochastic algorithm, effectively minimizing hitting over processor resource and improving reliability in any scenarios.

- **High User-Friendliness**

CLIXXO IPX22K-UC leverages auto clip intelligent inbound routing mechanism. With call records, CLIXXO IPX22K-UC can intelligently match inbound call number with historic called one in auto clip. Moveable extension, call forwarding, DND, etc. are available.

- **Multiple High-Security Modes**

Multiple security mechanisms in CLIXXO IPX22K-UC are available, including password, ACL, data filtering, etc. Besides, outbound routing, DISA, conference, voice mail and other applications support PIN code setting to customize dynamic firewall.

- **Flexible Surveillance**

CLIXXO IPX22K-UC adapts flexible multiple-layer monitoring modes to protect privacy at maximum level and ensure high-level of security and reliability in most conditions.

Technical Specifications

Interfaces	
Analog Telephone FXS Ports	2 ports (both with lifeline capability in case of power outage)
PSTN Line FXO Ports	4 ports
Network Interfaces	Dual 10/100 RJ45 ports
NAT Router	YES
Peripheral Ports	USB, TF
LED Indicators	Power/Ready, Network, PSTN Line, USB, TF
Reset Switch	YES
Voice/Video Capabilities	
Voice-over-Packet Capabilities	LEC with NLP Packetized Voice Protocol Unit, 32~128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer
Voice and Fax Codecs	G.711 A-law/U-law, G.722, G.723.1 5.3K/6.3K, G.726, G.729A/B, GSM, AAL2-G.726-32; T.38
Video Codecs	H.264, H.263, H263+
QoS	Multiple Layers
Signaling & Control	
DTMF Methods	In Audio, RFC2833, and SIP INFO
Provisioning Protocol & Plug-and-Play	TFTP/HTTP/HTTPS, auto-discovery & auto-provisioning of various IP endpoints with no Configuration
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS, LADP
Disconnect Methods	Call Progress Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect, Busy Tone
Security	
Media Encryption	SRTP, TLS, HTTPS, TELNET with Fail2ban, Whitelist, Blacklist, alerts and more to protect against attacks
Physical	
Universal Power Supply	Output: 12VDC, 2A; Input: 100 ~ 240VAC, 50 ~ 60Hz
Dimensions	186mm L x 108mm W x 30mm H
Weight	Unit weight 0.83kg, Package weight 1.1kg
Environmental	Operating: 32 ~ 113ºF / 0 ~ 45ºC, 8 ~ 90% (non-condensing); Storage: -4 ~ 185ºF / -20 ~ 85ºC
Additional Features	
Multi-Language Support	English for Web UI; Customizable IVR/voice prompts for English, British English
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF
Polarity Reversal/Wink	Yes, with enable/disable option upon call establishment and termination
Call Center	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/ busy level, in-queue announcement
Customizable Auto Attendant	Unlimited layers of IVR (Interactive Voice Response)
Maximum Call Capacity	Up to 50 even in SRTP encrypted
Conference Bridges	Up to 25 simultaneous PSTN or IP participants
Call Features	Call park, call forward, call transfer, DND, ring/hunt group, paging/intercom etc.