New

- 2.4-inch LCD display with backlight
- 2 SIP lines，13 programmable keys
- Optimized noise reduction, HD sound quality
- 2.4G&5.8G Wi-Fi module is optional
- One-stop remote address book management
- Simple appearance combines elegance and functionality

TLS&SRTP | OpenVPN | HD audio | Five-party conference | DSS 32 | Address book

Explore the ST780 series- Unified communication

ST780 series phone, simple design, intuitive operation, redefine enterprise IP communication, designed for small and medium-sized enterprises to create efficient and intelligent communication experience.

Brightening characteristic

Efficient communication: Dual SIP lines, efficient call management.
Clear display: 2.4-inch backlit LCD screen for a striking and clear visual experience.
Simple operation: 13 programmable keys, simplify the operation process, improve work efficiency.
Smooth collaboration: Support for 5-party conference calls, so that team collaboration is more smooth.
Powerful: Cover all advanced call functions to meet diverse office needs.
Security: OpenVPN, TLS&SRTP encryption technology is integrated to ensure call security, so that every conversation is at ease.
Convenient management: The advanced LDAP technology is integrated to achieve integrated remote address book management, simplifying contact access and synchronization, and improving communication efficiency.

Free extension

Seamless connection: DSS expansion disk, speed dial, real-time monitoring of user status.
Flexible configuration: PoE power supply, 2.4G/5.8G WiFi, free desktop space, flexible deployment.

Advanced audio experience

HD sound quality: Advanced noise reduction technology to create a clear call experience like face to face.

Phone Features
- 2 SIP lines
- Multi-language selection (Chines/ English/ Turkish/ Korean/ Russian/ Traditional Chinese/ French/ Italian/ German/ Portuguese/ Polish/ Thai/ Hindi/ Arabic/ Japanese/ Spanish/ Vietnamese)
- Caller ID with display name
- Custom DSS key, BLF monitoring, hotline, speed dial
- Handle/ hands-free/ headphone mode
- Call hold, mute, blind transfer, inquiry transfer
- DND, call forwarding, call waiting
- Multicast, intercom, text message, keypad lock, emergency call
- Direct IP call, redial, callback, auto answer
- Five-party conference, line sharing, dialing rules
- 10 types of built-in ringtones/ Upload custom ringtones
- Manually or automatically synchronize network time
- Action URL & Action URI

Audio
- High-fidelity sound quality: HD handle，HD hands-free
- Broadband codec: G.722, Opus
- Narrowband codec: G.711(A/μ), G.723.1, .729AB, G.726, iLBC
- DTMF: In-band transmission（In-band) ,
Out-of-band transmission (RFC 2833)，SIP INFO
- Full-duplex hand-free
- Acoustic echo cancellation (AEC)
- Voice activity detection (VAD)
- Comfort noise generation (CNG)
- Packet loss compensation (PLC)
- Background noise detection (BNE)
- Automatic gain control (AGC)
- Dynamic adaptive RTP jitter buffer up to 300 ms

Address Book
- Blacklist
- XML/LDAP address book
- Contact smart search
- Contact export/ import
- Local contacts (1000 entries can be stored)
- Call history: dialed/ received/ missed/ forwarded calls (1200 records in total)

IP PBX Features
- BLF, voice message
- Anonymous call, anonymous call rejection
- Call park, call pickup
- Server-based recording function

Configuration and Maintenance
- Web configuration, Phone configuration
- Automatic deployment: Support FTP/ TFTP/ HTTP/ HTTPS/ SIP PnP/TR-069
- BIN/ CFG configuration
- Contact configuration export/ import
- CSTA remote control
- Network packet capture and diagnosis
- CSTA remote control
- Network packet capture and diagnosis
- Custom/ full restore factory settings
- System firmware upgrade

Network
- Support IPv4 / IPv6 / IPv4&IPv6
- IP address allocation mode: Static IP / DHCP
- Virtual Local Area Network: VLAN
- Data link layer protocol: CDP / LLDP
- SIP v1 (RFC2543), v2 (RFC3261)
- Support redundant servers
- Support ICE, STUN (NAT session traversal）
- Support UDP/ TCP/ TLS transmission
- Support SRTP encryption protocol
- SIP trust serve (anti-attack mechanism）
- HTTP/HTTPS
- TR-069 network management protocol
- SNTP automatically synchronizes network dates and times
- QoS: QoS: 802.1p/Q tagging (VLAN), Layer3/ ToS/ DSCP
- HTTPS certificate management
- Support configuration file encryption
- Support information verification mechanism
- RTCP-XR, VQ-RTCPXR
- OpenVPN, IEEE802.1X
- Wi-F (2.4G&5G) optional (ST780W)

Phone Interface
- 1 x RJ-9 (4P4C) handle interface
- 1 x RJ-9 (4P4C) headset interface
- 2 x RJ-45 10/100M adaptive Ethernet port ST780/P, ST780W
  2 x RJ-45 10/100/1000M adaptive Ethernet port ST780G
- DC power interface, USB2.0 interface
- USB Type C for DSS 32

Product Specifications
- Phone main screen: 2.4-inch (132x64) LCD display with backlight
- Information indicator: LED red light
- Phone keyboard:
- 2 SIP line keys (with LED lights, programmable)
- 4 customizable soft keys
- 12 standard telephone number keys
- 6 function keys headset, voice message, mute, call forwarding, redialing, hands-free
- 5 navigation keys
- 2 volume up/down adjustment keys
- Color: business gray
- Installation method: Desktop/ Wall-mounted
- Desktop size: 187mm*184mm*147mm
- Phone power consumption：2~3W
- PoE: IEEE 802.3af/IEEE 802.3at (ST780P and ST780PG)
- Working environment humidity: 10~95%
- Working temperature: -10~50° C (+14~122° F)

Carton Package
- Packing list：
- IP phone：ST780 series
- Standard Ethernet cable, handle curve
- Base, handle and Quick start guide
- Power adapter: input AC 100~240V,
  Output DC 5V/ 1000mA
- Quantity/ box: 10units
- Gross weight/ carton: 9.5 kg
- Color box size：211 mm×195 mm×100 mm
- Carton size：535 mm×410 mm×230 mm

Model	Network	PoE	DSS 32	Power adapter
ST780	10/100M	×	√	√
ST780P	10/100M	√	√	Optional
ST780G	10/100/1000M	√	√	Optional
ST780W	10/100M&WiFi	√	√	√