The perfect voice quality to anywhere with FREE cost.

This standalone IP Phone takes advantages of a range of new technologies , allowing its users to experience superlative voice communications over IP network. Providing conveniences and functionalities communication strategy, for successes of your business.

stX1101

SIP Phone

NTEGRATOR

Today's Integration Tomorrow's Solution

Network Protocol

- SIP v2(RFC3261)
- IP/TCP/UDP/RTP/RTCP
- IP/ICMP/ARP/RARP/SNTP
- TFTP Client/DHCP Client/ PPPoE Client
- Telnet/HTTP Server

Tone

- Ring Tone
- Ring Back Tone
- Dial Tone
- Busy Tone
- Programming Tone

Phone Function

- Volume Adjustment
- Speed dial, Phone book (140 sets)
- Flash
- Call History
- Voice mail / Incoming call Indicator

Codec

- G.711: 64k bit/s (PCM)
- G.723.1: 6.3k / 5.3k bit/s
- G.726: 16k / 24k / 32k / 40k bit/s (ADPCM)
- G.729A: 8k bit/s (CS-ACELP)
- G.729B: adds VAD & CNG to G.729

IP Assignment

- Static IP
- DHCP
- PPPoE

Voice Quality

- VAD: Voice activity detection
- CNG: Comfortable noise generator
- LEC: Line echo canceller
- Packet Loss Compensation
- Adaptive Jitter Buffer Security

SIP Server

- Registrar Server (three SIP accounts)

Security

- HTTP 1.1 basic/digest authentication for Web setup
- MD5 for SIP authentication (RFC2069/ RFC 2617)

QoS

- ToS field

Call Function

- Call Hold
- Call Waiting
- Call Forward
- Caller ID
- 3-way conference

NAT Traversal

- STUN
- Outbound Proxy

Configuration

- Web Browser
- TFTP/FTP
- Keypad

DTMF Function

- In-Band DTMF
- Out-of Band DTMF
- SIP Info

Firmware Upgrade

- TFTP
- HTTP

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