THOMSON

THOMSON ST2022
VoIP Business Phone

TELECOM | CABLE | SATELLITE | TERRESTRIAL

Voice | Data | Video

High Quality Solution

The THOMSON ST2022 offers a high-quality IP solution at a cost effective price making Voice over IP (VoIP) more accessible for businesses.

The THOMSON ST2022 is a VoIP phone that is compliant with SIP/MGCP and can be used with any SIP compliant PBX, Softswitch or or IP Centrex solutions of the market. It addresses the need for a cost effective IP solution, helping to make VoIP a viable, attractive solution for small and large offices environment.

The THOMSON ST2022 includes the main following features:

- 144x32 full graphic display for 3 lines + icons
- Connectivity: Integrated 2 port 10/100 Ethernet switch with PoE and VLAN tagging
- Phone services:
- 2 lines, Call Forward, Call transfer, Call Hold, Call Waiting
- Handsfree
- Alert (missed calls, Message Waiting Indicator)
- Phone book:
  Local (100 entries)
  Remote (HTTP XML)
- Caller ID Display
- Automatic Callback
- 3-ways Conferencing
- Downloadable rings
- Multiple power options: power over Ethernet 802.3af and external power supply adapted (included)
- VoIP Standard: SIP or MGCP
- Voice compression standards: G.711, G.723, G.729ab
- IP addressing: Static or dynamic IP configuration (integrated (DHCP client)
- Quality of Service: ToS DiffServ, 802.1p/Q, VAD, CNG, Packet Loss Compensation, Adaptative Jitter Buffer
- Web browser interface for configuration and firmware (admin & user mode)
- Automatic provisioning system (DHCP, TFTP/HTTP)
- 7 languages

Voice Data Video

TECHNICAL SUMMARY

TELECOM

CABLE SATELLITE TERRESTRIAL

Key Features

- 7 languages (SIP)
- SIP compliant or MGCP
- 144x32 full graphic display
- 2 RJ45 Ethernet switch 10/100
- Handsfree
- Power over Ethernet 802.3af standard
- Navigation keys
- 3 soft keys

Phone sets

Funtion Support

- Multiple call operation (2 lines)
- Alert (missed calls, message waiting) Indicator
- Mute
- Hold
- Transfer
- Forward
- On-hook dialing
- Handsfree
- Call Log (30 entries; incoming/outgoing calls)
- Local Address Book (100 memories)
- Remote Address Book (HTTP XML)
- Downloadable RTTTL Rings
- Caller ID Display
- Volume control (speaker, handset, headset & ringer)
- Date & time display (idle state)
- Call duration timer
- Diall from Call log or Phonebook

Key Pad

- 3 soft keys
- 2 line keys
- 3 speed dial keys
- 1 MWI/Backlit Voice mail key
- Volume up & down
- Menu, OK and Cancel keys
- Phonebook key
- 2 fixed function keys backlit (mute and handsfree)

LED Indicators

Alert Indicator

Interfaces

2 RJ45 autosensing 10/100Mbps
(one for PC and one for LAN)

Physical & Environment

- Mounting: Footstand or wallmount
- AC Adapter included
- Power Input: US, EU and UK power supply available
- Operating Temperature: from 0° to 50° C (32° / 104° F)
- Storage Temperature: from -5° to 80° C (-40° / 158° F)
- Humidity: up to 90% non-condensing

Technical Specifications

SIP Protocol Support

- RFC 3261, SIP: Session Initiation Protocol
- RFC 3264, An Offer/Answer Model with SDP
- RFC 2833, RTP Payload for DMF Digits, Telephony Tones and Telephony Signals
- RFC 3323, A Privacy Mechanism for the Session Initiation Protocol
- RFC 3311, The Session Initiation Protocol UPDATE Method
- RFC 3725, Best Current Practices for Third Party Call Control in the Session Initiation Protocol
- RFC 3515, The Session Initiation Protocol (SIP) Refer Method
- RFC 3265, Session Initiation Protocol (SIP) - Specific Event Notification
- RFC 2976, The SIP INFO Method
- RFC 3262, Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- RFC 3725, Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)
- RFC 3891, The Session Initiation Protocol (SIP) 'Replaces' Header
- RFC 2327, SDP: Session Description Protocol
- RFC4235, An INVITE-Initiated Dialog Event Package for SIP.

MGCP Protocol

- RFC 3435, Media Gateway Control Protocol
- RFC 3149, MGCP Business Phone Packages
- RFC 2327, SDP: Session Description Protocol
- RFC 3264, An Offer/Answer Model with SDP
- RFC 2833, RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals

Audio Code

- G.711 -law/A-law
- G.723.1
- G729ab

Dial Tone Signal Generation

- DTMF (out of band, in Band and INFO in SIP)
- RFC 2833

Internet Support

- IP (RFC 0791), TCP (RFC 0793), UDP (RFC 0768), ARP (RFC 0826) protocols
- DHCP client (RFC 2131)
- SDP
- TFTP
- HTTP
- DNS
- SNTP

Quality of Service

- ToS Diffserv, 802.1p/Q
- VAD, CNG, Packet loss compensation, Adaptative Jitter Buffer

VLAN tagging

- IEEE 802.1Q

Ethernet

- IEEE 802.3 10BASE-T Ethernet
- IEEE 802.3u 100BASE-TX Fast Ethernet

Configuration

IP Number Assignment

- DHCP client or fixed IP

Configuration Support

- Keypad & LCD
- Web browser management with 2 Levels (User and Admin)
- TFTP/HTTP server download- Local & remote warm reboot
- Password protection for configuration

For more information

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