IP SIP Phone

The Intelligent Communication Device For Next Generation Network

Qunantier's IP SIP Phone is feature-rich yet cost competitive Ethernet Phone, which provides the voice data convergence solution with state-of-the-art technology. With high-quality voice, the IP SIP Phone can work perfectly with standard-based IP telephony system of SIP Protocol Stack.

The IP SIP Phone users can experience both the familiar operation of legacy hi-end PSTN Phone and intelligence of an IP network device with the LCD display, which can provide all the necessary information when proceeding a calling activity.

The IP SIP Phone is the most powerful network device for next generation networks.

- ***Support 2-Concurrent Call***
- ***Caller ID / Call Waiting Capability***
- ***Music On Hold***
- ***One Touch Speed Dialing***
- ***Address Book***
- ***CID Call Back Function***
- ***Call Conference***
- ***LAN / PC Interface***
- ***Customizable Ring Tone***

QUANTIER

IP SIP Phone Technical Specification

CPU	Agere Single Chip Solution (ARM940T-100MIPS, DSP 100MIPS/100MHz)	
Flash	2Mbytes	
SDRAM	16Mbytes	
Display	2 x16 Characters LCD	
Keypad	12 dialing buttons (0~9, *, #), 9 fixed function buttons, 20 DSS-key with 2 colors LED Buttons, 2 volume buttons.	
RJ-45	Dual 10/100 Mbps Ethernet ports.	
RJ-11	One connects to Handset, another to Headset.	
Power Adapter	AC: 100~240V, 47~63Hz DC: 9V, 1.1A	
Environmental	Operation temperature	0~40°C
	Storage temperature	0~70°C
	Relative Humidity	95% Max, non condensing
	Shock	Up to 75cm (30 in.) drop depending upon package.
Dimension	228mmx201mmx115mm	

Protocol

- SIP call signaling – RFC3261 with backward compatible with RFC2543
- RTP/RTCP(RFC1889/RFC1890)
- SDP (RFC2327)
- Capacity exchange based on SDP (RFC3264)
- NAPTR for SIP URI lookup (RFC2915)
- E.164 Number and DNS (ENUM, RFC2916)
- DHCP / PPPoE (optional) for automatic IP address assignment/Static IP assignment
- SNTP/TFTP for batch provision
- IEEE 802.1Q VLAN
- Support IP ToS and 802.1P Class-of-Service

Voice Processing

- Codec: G.711 (a-law and u-law), G.723.1(A) and G. 729A, G.729AB.
- Acoustic Echo Cancellation
- VAD (Voice Activity Detection)
- CNG (CNG for G.711 as well)

Features

- Mute
- Volume adjustment
- 2-Color-LED Message indication
- DTMF Relay: Support in-band RTP voice mixing and out-band DTMF over RTP (RFC2833).
- Hand free Speaking
- Loud Speaker Capability
- Headset Interface
- Support 2-concurrent Call
- Address Book Capability
- Caller ID / Call Waiting
- Last Number Redial
- Tone Generator / DTMF Generator
- Music On Hold Capability
- Call forwarding
- Call Hold
- Call transfer
- Call rejection
- Call screening
- Call conference
- Auto redial
- Speed dialing
- Missed Calls Indication
- Received Calls Indication
- Dialed Numbers Indication
- Multiple outbound Proxy, Registrar and Redirect server support, up to 3 different service domain
- NAT & firewall support by STUN or pre-configured NAT Gateway port mapping
- Customizable Ring Tone
- Support SNMP

* Specifications subject to change without notice

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