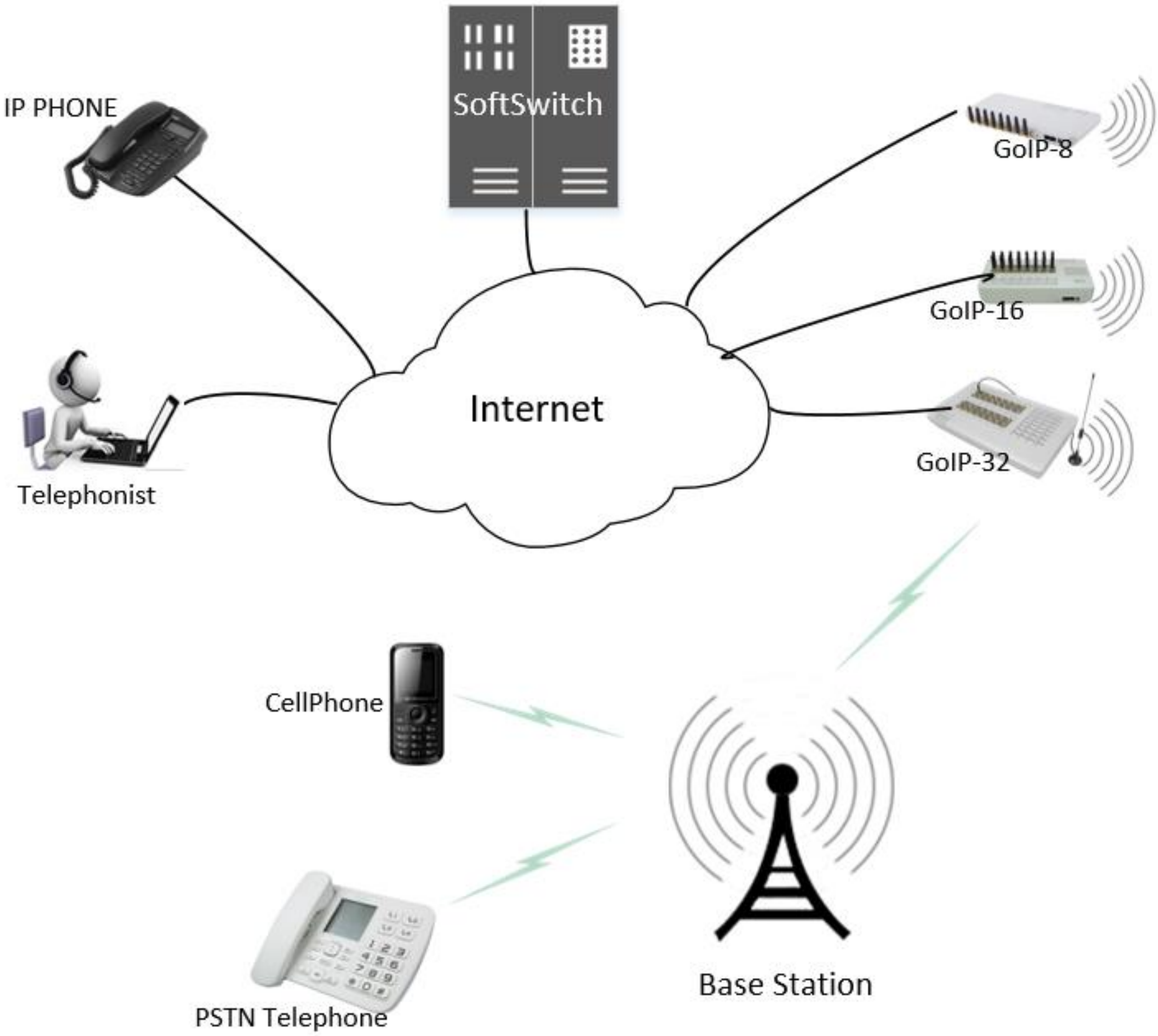
HYBERTONE
零海拔科技



VoIP GSM Gateway GoIP-1 HyberTone's GoIP-1 is a VoIP GSM Gateway for call termination (VoIP to GSM) and origination (GSM to VoIP). It is SIP&H.323 based and compatible with Asterisk, Trixbox, 3CX, SIP Proxy Server, VoipBuster. It can enable to make 1 call simultaneously from IP phones to GSM networks and GSM networks to IP phone.

Key Features	Call from VoIP to GSM or GSM to VoIP
	Bulk SMS
	Auto check balance and recharge
	Remote SIM Control
	Outbound call from voip
	Inbound call from gsm
	Remote management
Software Features	VoIP Protocols: SIP V2, H.323 V4
	Audio Codecs: g711(alaw/ulaw), g729, g723.1
	DTMF format: SIP INFO, RFC2833, Inband
	Packet Loss Concealment
	Programmable Jitter Buffer: Fixed, Dynamic, Adaptive
	Network Connection: DHCP, PPPOE, Static IP
	STUN Server
	Support English, Chinese
	Support Signaling encryption
	Support media encryption
	Support Media NAT Traversal
	Support Remote control
Hardware Features	GSM Channel: 1
	Power Adapter: DC 12V/500MA
	CPU: ARM9/300MHz
	RAM: 16MB
	Flash: 4M
	GSM Frequency: 850/900/1800/1900MHz
	Max power consumption: 5W
	Weight: 0.5KG (Including AC/DC Adapter)
	Size: 230mm (L) x 180mm (W) x 70mm (H)
	Operating Temperature: 0 - 45℃
	Working Humidity: 10% - 90% non condensation
Free Software	SMS Server
	SIM Server (Sim Bank Scheduler Server)
	Relay Server
	Remote Access