贝克通信

智守通信，一呼百应

SBC Gateway

SBC1000

The Becke SBC1000 product provides small and medium-sized telecom operators with a rich suite of security, access, interconnection, and routing solutions for their SIP networks.Services include policy management, signaling flow control, QoS, and media processing.

The device employs a multi-core processor, a non-blocking gigabit switching network, and an embedded Linux operating system, achieving high performance while maintaining excellent stability.Low power consumption.

The device supports 500 concurrent sessions and 200-channel voice media transcoding, and supports WebRTC, SIP over TLS, and SRTP encrypted sessions, ensuring security and reliability. The SBC1000 supports multiple encoding standards including G.729, G.723, G.711a/u, G.726, AMR, OPUS, and iLBC.Media encoding and decoding.

Highlights

- ☑ Supports 500 concurrent sessions and 200-channel voice media transcoding.
- ☑ Standard SIP protocol and flexible routing rules, perfectly compatible with IMS system.
- ☑ Provides a VoIP firewall to protect the core network from attacks.
- ☑ Supports intelligent bandwidth limiting and dynamic blacklists
- ☑ Supports cross-network and NAT traversal, adaptable to various networking environments.
- ☑ Supports SIP over TLS and SRTP encrypted sessions, ensuring security and reliability.
- ☑ Multiple encoding standards: G.711A/U, G.723.1, G.729A/B, iLBC, AMR, OPUS
- ☑ A user-friendly web-based user management interface that offers multiple management methods

Detailedparameters

VoIP Protocols	SIP v2.0, UDP, TCP, TLS,RFC3261
	SIP trunking operating modes: Peer/Access
	B2BUA user agent
	SIP call flow control
	SIP registration flow control
	SIP registration packet attack dynamic scanning detection
	SIP abnormal call attack dynamic scanning detection
	SIP header field transformation
	SIP header field pass-through
	SIP packet redundancy mechanism
	QoS service
	NAT traversal
Management andmaintenance	Web-based visual configuration and maintenance
	Data backup/restore
	Web firmware upgrade
	Built-in network packet capture function
	System Log
	Operation Log
	Call detail records
	NTP automatic time synchronization
	Multilingual switching
	NMS Centralized Management Platform
	SNMP
	Remote Web and SSH
Physical Specifications	Ethernet interface:5 x 10/100/1000M Base-T Ethernet interfaces
	1 USB 2.0 port (reserved)
	1* SD card slot (reserved)
	Serial port:1* RS232, 115200bps
	2*E1 interfaces (reserved)
physical properties	Power supply: Redundant power supply, 100-240VAC, 50-60 Hz
	Power consumption: 15W
	Operating temperature: 0ÿ ~ 45ÿ
	Storage temperature: -20ÿ ~ 80ÿ
	Humidity : 10%-90%non-condensing
	Dimensions (W/D/H): 226×146×39mm
	Weight: 2.5kg
	Certifications: CE, FCC

Voice characteristics	Speech encoding: G.711a/ÿ, G.723 , G.729A/B, iLBC, G.726, AMR, OPUS
	Comfort noise generation (CNG)
	Voice activity detection (VAD)
	Echo cancellation (G.168), maximum 128ms
	RTP/RTCP
	Voice interruption protection
	Adaptive dynamic buffering
	Quality monitoring
	Fax: T.38 and Passthrough
	DTMF modes: RFC2833/Signal/Inband
Safety	DOS/DDOS attack defense
	Access control policies
	Policy-based IP and SIP attack prevention
	Malformedmessagedetectionandprocessing
	UDP-Flood Attack Defense
	TCP-Flood Attack Defense
	TLS & SRTP encrypted sessions
	Caller ID and callee number
	ACL control
	VoIP firewall
performance	Call concurrency: Supports up to 500 concurrent calls
	Concurrent transcoding:Supports 200-channel media encoding/decoding
	Call CPS: 25
	Number of registered users: 5000
	Register for CPS: 25
	IMS account registration: 3000
	Call routing:A maximum of 1024 call routes can be added
Call processing	Flexible routing rules
	Redundant route backup/selection
	Control call routing by time period
	Routing by calling/called number prefix
	Match routes by SIP URL domain
	Match routes according to SIP request method
	Change of calling and called numbers
	IP trunk redundancy and load balancing
	Regular expressions for numbers