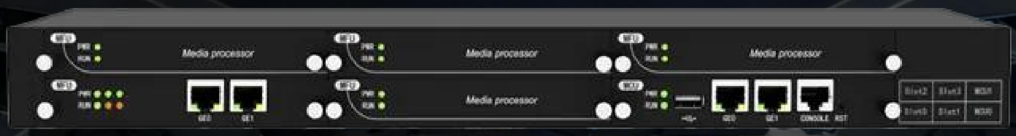
贝克通信

智守通信，一呼百应



SBC Gateway

SBC3000

The Becke SBC3000 Session Border Controller provides rich security, access, and interconnectivity features for the SIP networks of small and medium-sized telecom operators.Services include connectivity, routing/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, while supporting dual hot-swappable power supplies, dual-machine hot standby (HA) and WebRTC, with carrier-grade high reliability.

The SBC3000 single unit supports 2000 concurrent sessions and 1500 channels of voice media transcoding processing, and also supports topology hiding, NAT traversal, and more.It supports secure communication methods such as security protection, SRTP and TLS encryption, and also supports G.729, G.723, G.711a/u, and G.726.The device supports multiple media codecs such as AMR, OPUS, and iLBC, and has routing load balancing capabilities.

Highlights

- ☑ The SBC3000 single unit supports 2000 concurrent sessions and 1500 channels of voice media transcoding processing;
- ☑ Supports dual-machine hot standby for Trunk SBC and WebRTC SBC;
- ☑ Supports WebRTC, voice, and video calls;
- ☑ Standard SIP protocol and flexible routing rules, perfectly compatible with IMS system;
- ☑ Supports topology hiding and security anti-attack strategies to protect the core network;
- ☑ Supports intelligent bandwidth limiting and dynamic blacklists;
- ☑ Supports cross-network and NAT traversal, adapting to various networking environments;
- ☑ Supports SIP over TLS and SRTP encrypted sessions, ensuring security and reliability;

Detailedparameters

VoIP Protocols	SIP v2.0, UDP, TCP, TLS,RFC3261	Voice characteristics	Speech encoding: G.711a/ÿ, G.723 , G.729A/B, iLBC, G.726, AMR, OPUS
	SIP trunking operating modes: Peer/Access		Comfort noise generation (CNG)
	B2BUA user agent		Voice activity detection (VAD)
	SIP call flow control		Echo cancellation (G.168), maximum 128ms
	SIP registration flow control		RTP/RTCP
	SIP registration packet attack dynamic scanning detection		Voice interruption protection
	SIP abnormal call attack dynamic scanning detection		Adaptive dynamic buffering
	SIP header field transformation		Quality monitoring
	SIP header field pass-through		Fax: T.38 and Passthrough
	SIP packet redundancy mechanism		DTMF modes: RFC2833/Signal/Inband
	QoS service	Safety	DOS/DDOS attack defense
	NAT traversal		Access control policies
Management and maintenances	Web-based visual configuration and maintenance		Policy-based IP and SIP attack prevention
			Malformedmessagedetectionandprocessing
	Data backup/restore		UDP-Flood /TCP-Flood Attack Defense
	Web firmware upgrade		SRTP encrypted session
	Built-in network packet capture function		TLS security protection
			Caller ID and callee number
	System Log		ACL control
	Operation Log		VoIP firewall
	Call detail records	performance	Call concurrency: Supports up to 2000 concurrent calls.
	NTP automatic time synchronization		
	Multilingual switching		Concurrent transcoding: Supports 1500 voice transcoding callsconversion
	NMS Centralized Management Platform		Calling CPS: Maximum processing capacity of 150 concurrent requests per
	SNMP		User registration limit: Maximum of 10,000 user registrations
	Remote Web and SSH		CPS Registration: Maximum processing capacity of 200 registration messages per second
Physical Specifications	Ethernet interface: 4 x 10/100/1000M Base-T Ethernet interfaces		SIP trunks: Up to 1024 SIP trunks can be added
			Access/Trunk SBC Dual-Machine Hot Standby (1+1)
	1*MCU: Single master controller		WebRTC SBC Dual-Machine Hot Standby (1+1)
	4* MFU: Service Board		WebRTC Concurrency (Voice): 1500
	1 USB 2.0 port (reserved)	Call processing	Flexible routing rules
physical properties	Serial port:1* RS232, 115200bps		Redundant route backup/selection
	Dual power supply with hot standby: 100-240VAC, 50-60 Hz		Control call routing by time period
			Routing by calling/called number prefix
	Power consumption: 70W		Match routes by SIP URL domain
	Operating temperature: 0ÿ ~ 45ÿ		Match routes according to SIP request method
	Storage temperature: -20ÿ ~ 80ÿ		
	Humidity : 10%-90%non-condensing		Change of calling and called numbers
	Dimensions (W/D/H):437*320*44mm (1U)		IP trunk redundancy and load balancing
	Weight: 6kg		Regular expressions for numbers