SBC3000 Session Border Controller

Overview

Controller SBC3000 provides rich SIP-based services such as safe network access, robust security, system interconnectivity, flexible session routing & policy management, QoS, media transcoding and media processing for small-and-medium telecommunication operators. With distributed multi-core processor, rear panel for non-blocking gigabit switching data as well as embedded Linux operating system, the session border controller delivers high capability while achieves low power dissipation.

SBC3000

It is able to process up to 2000 concurrent SIP sessions and transcode 1200 concurrent calls. Meanwhile, it supports dual control boards and dual hot-swappable power supplies, and allows encrypted sessions via TLS and SRTP. Apart form traditional codecs like G.729, G.723, G.711 and G.726, SBC3000 also supports the transcoding of iLBC, AMR and OPUS.

Key Features

- Supports up to 2000 concurrent SIP sessions
- Support WebRTC2SIP to turn WebRTC client into a phone with audio capability
- Support forwarding of 2000 calls on media, and transcoding of 1200 calls on media and fax
- Triple anti-attacking, embedded VoIP firewall, prevention of DoS and DDoS attacks
- Bandwidth limitation and dynamic white list & black list
- VLAN, IPSec, QoS, static route, NAT traversal
- Import & export of remote upgrade and configuration data
- User-friendly web interface, multiple management ways
- 1+1 high availability with two SBCs, dual hot-swappable power supplies

Physical Interfaces

MCU (Main Control Unit): 1

MFU (Main Function Unit): 4

Ethernet Ports:

4* 10/100/1000M Base-T Ethernet ports

1* USB 2.0

Serial Console

1* RS232, 115200bps, RJ45 each MCU

Power Supply: 2

Media Capabilities

Voice, FAX support

Codecs: G.729, G.723, G.711, iLBC, OPUS, G.726, AMR

RTP Transcoding Pass-through fax No RTP detection One-way audio detection

RTP/RTCP, SRTP

RTCP statistics reports

DTMF: RFC2833, SIP Info, INBAND

Silence Suppression Comfort Noise

Voice Activity Detection

Echo Cancellation

Adaptive Dynamic Buffer

Environmental

Power Supply: AC 100-240V, 50-60Hz

Power Consumption: 70W

Operating Temperature. 0 ℃ ~ 45 ℃

Storage Temperature: -20 ℃ ~80 ℃

Humidity: 10%-90% Non-Condensing

Dimensions (W/D/H): 437×320×44mm

Unit Weight: 6 kg

Capabilities

Concurrent Calls

Supports 2000 SIP sessions at maximum

Transcoding

Supports 1200 transcoding calls

CPS for Call

200 calls per second at

maximum Registrations

Maximum SIP registrations: 10000

CPS for Registration

200 Registration per second

SIP Trunk

Unlimited SIP Trunks

Security

Prevention of DoS and DDoS attacks

Control of access policies

Policy-based anti-attacks

Call Security with TLS/SRTP

White List & Black List

Access Rule List

Embedded VoIP Firewall

Call Control

Dynamic load balancing and call routing

Flexible Routing Engine

Call routing base on prefixes

Call routing base on caller/called number regular express

Call routing base on time profile

Call routing base on SIP URI

Call routing base on SIP method

Call routing base on endpoint

Caller/ Called number Manipulation

VoIP

SIP 2.0 compliant, UDP, TCP, TLS

SIP trunk (Peer to peer) SIP trunk (Access)

SIP Registrations

B2BUA (Back-to-Back User Agent)

SIP Request rate limiting SIP registration rate limiting

SIP registration scan attack detection

SIP call scan attack detection

SIP anti-attack

SIP Header manipulation

SIP malformed packet protection

Multiple Soft-switches supported

QoS (ToS, DSCP)

NAT Traversal

Maintenance

Web-bases GUI for Configurations

Configuration Restore/Backup

HTTP Firmware Upgrade

CDR Report and Export Ping and Tracert

Network Capture

System log

Statistics and Reports

Multiple language support

Centralized management system

Remote Web and Telnet