SBC300 Session Border Controller

Overview

SBC300

The SBC300 provides rich SIP-based services such as safe network access, robust security, system interworking, 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 network, embedded Linux operating system, SBC300 delivers high capability while achieves low power dissipation.

It supports 50 concurrent SIP sessions and transcodes up to 50 concurrent calls, and allows encrypted sessions via TLS and SRTP. The session border controller supports transcoding of G.729, G.723, G.711, G.726, iLBC, AMR and OPUS. Besides, it also supports WebRTC, allowing desktop and mobile browsers to initiate and receive calls to/from your SIP service over websocket and WebRTC completely transparently.

Key Features

- Supports up to 50 SIP sessions and 50 transcoding sessions
- Support WebRTC2SIP to turn WebRTCclient into a phone with audio capability
- SIP trunks & flexible routing rules for accessing IMS
- Embedded VoIP firewall, prevention of DoS and DDoS attacks
- Bandwidth limitation and dynamic white list & black list
- QoS, static route, NAT traversal
- Import & export of remote upgrade and configuration data
- User-friendly web interface, multiple management ways

Physical Interfaces

Ethernet Ports:

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

1* 10/100/1000M Base-T (Admin)

1* USB

1* TF Card Slot

Serial Console 1* RS232, 115200bps, RJ45

Media Capabilities

Voice, FAX support Codecs: G.729, G.723, G.711, iLBC,AMR, OPUS 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: DC 12V

Power Consumption: 10W

Operating Temperature. 0 ℃ ~ 45 ℃

Storage Temperature: -20 ℃ ~80 ℃

Humidity: 10%-90% Non-Condensing

Dimensions (W/D/H): 227×147×39mm

Unit Weight: 1.5 kg

Capabilities

Concurrent Calls

Supports up to 50 SIP sessions

Transcoding

Supports up to 50 transcoding calls

Registrations

Maximum SIP registrations: 1000 CPS

for Registration

20 Registration per second

SIP Trunks

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