SBC8000 Session Border Controller

Overview

A Session Border Controller (SBC) is a SIP/IP network core element that deployed for next generation networks (NGN) to support peer/access connection by SIP/IMS trunk and provide interworking between incompatible signaling messages or media flow from endpoint devices and SIP applications. In particular SBC supports SIP/IMS network connection, sip endpoint remote registration, NAT traversal, signaling modification, media control, flexible call routing, codec transcoding, billing and QoS management.

Controller SBC8000 is a new software-based SBC. It provides very rich features for medium and large enterprise SIP network connection and supports various security mechanisms, call routing based on control policies, carrier-grade high availability(HA) and debug tools for service providers and telecom operators.

SBC8000 can be installed on VM/cloud-based system or third-party hardware appliances. Users can install SBC8000 in X86, ARM architecture and any cloud-based platforms for many different SIP implementations with advantages of VM and cloud platforms.

Key Features

- Up to 10,000 concurrent call sessions, 5,000 media transcoding and 100,000 SIP registrations
- Operate on physical server, virtual machine and public cloud
- Intelligent bandwidth limit and dynamic blacklist
- Cross-network, NAT traversal and high availability(HA)
- SIP over TLS, SRTP
- Compatible with different codecs: G.711A/U, G.723.1, G.729A/B, iLBC, AMR, OPUS
- Flexible call routing
- Broad compatibility with IMS network
- VoIP firewall, anti-attacks and core network protection
- Call recording

Deployment Platforms

Hardware Architecture: X86/ARM

Virtual Machine: VMware, Fusionsphere, FusionComputer, KVM

Cloud Deployment: Alibaba Cloud, Amazon Cloud, Baidu Cloud, Huawei Cloud, Telecom Cloud, etc.

Capabilities

Concurrent Calls
- Up to 10,000 SIP sessions

Transcoding
- Supports 5,000 transcoding calls

CPS for Call
- 800 calls per second at maximum

Registrations
- Up to 100,000 SIP registrations

CPS for Registration
- 800 Registration per second

Media Capabilities

Codecs：G.711a/μ, G.723, G.729A/B, iLBC, G.726, AMR, OPUS

Silence Suppression

Voice Activity Detection(VAD)

Comfort Noise Generator(CNG)

Echo Cancellation: G.168 with up to 128ms

RTP/RTCP

Voice Interrupt Protection

Adaptive Dynamic Buffer

Adjustable Gain Control

Automatic Gain Control (AGC)

FAX: T.38, Pass-through

DTMF: RFC2833/Signal/Inband

Call Control Features

Dynamic Load Balancing and Call Routing

Flexible Routing Engine

Routing Based on Caller/Called Prefixes

Regular Express

Call Routing Base on Time Profile

Call Routing Base on SIP URI Call

Routing Base on SIP Method

Caller/ Called Number Manipulation

VoIP

SIP 2.0 Compliant, UDP, TCP, TLS,

SIP Trunk (Peer to peer) SIP Trunk (Access)

SIP Proxy Registrations: Up to 3,000

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 Header Manipulation

SIP Malformed Packet Protection

Multiple Soft-switches Supported

QoS (ToS, DSCP)

NAT Traversal

Security

Prevention of DoS and DDoS Attacks

Control of Access Policies Policy-based Anti-attacks

Message format detection and processing

UDP-Flood Anti-attacks TCP-Flood Anti-attacks

Call Security with TLS/SRTP

Whitelist & Blacklist Access Control List Built-in VoIP Firewall

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

NTP

Multiple languages support

SNMP

Remote Web and Telnet