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.

DINSTAR 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

DINSTAR

The Gateway to VoIP World

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

Signaling tracking

Media Capabilities

Codecs：G.711a/μ, G.723, G.729A/B, iLBC, G.722, 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

Security

Control of Access Policies

Policy-based Anti-attacks

Call Security with TLS/SRTP

Whitelist & Blacklist

Access Control List

Built-in VoIP Firewall

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

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.

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

TLS mutual authentication

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