Virtual Session Border Controller WebRTC

Main characteristics

- WebRTC support
- SBC for installation within different infrastructures (on a physical server, virtualized or in the cloud)
- Flexible licensing
- Security features for VoIP network protection
- Log routing
- Compatibility with SIPREC* recorders
- High level of active availability/standby*

* *Requires additional licensing.*

Applications

- Virtualized SBC in private or public cloud
- Compatible with virtual machine migration needs
- May be installed on hardware of client's choice
- SIP trunk deployment in the cloud
- PBX protection in the cloud
- Intelligent routing with route failover
- Allows agents to be distributed over the Internet
- Internal network protection in registering extensions remotely

Overview

vSBC One W is the new version of Khomp's Session Border Controller software and may be installed in a variety of infrastructures, whether on a bare metal server, virtualized or in the cloud. vSBC One W is a powerful tool specifically developed to control traffic in telephony operations.

A SBC is a piece of equipment placed between different networks for the purposes of protecting and connecting VoIP traffic. The vSBC One W has security mechanisms in place to detect malicious behaviors and traffic sources, call encryption, normalization of error packets, and prevent traffic from unauthorized SIP packets in the VoIP network. It offers NAT and transcoding capabilities, offering, together with security features and connection between different VoIP networks, translating between network protocols and codecs.

vSBC One W software can be installed within various scenarios and offers flexibility and economy in its deployment. It can operate in physical servers, virtualized environments, such as VMWare®, or in cloud environments such as Azure, AWS, Huawei, Oracle and Google Cloud. With vSBC One W, it is possible to have a robust SBC that offers advanced routing and security resources and can be accessed from anywhere.

Securing your VoIP network

A secure VoIP network must have features that prevent unauthorized access, malicious attacks, or call interception. These situations could seriously compromise operations, jeopardizing the success of your business. vSBC One W was developed with VoIP network security in mind, offering features such as call encryption, prevention of the interception of signaling and audio, and topology obfuscation, which prevents external devices from accessing the corporate VoIP network structure.

The vSBC One W also provides, as an additional feature, Registration Routing. Registration Routing is an indispensable feature for those working remotely outside the corporate network, but that need to connect and make calls using the corporate telephony infrastructure.

WebRTC

The vSBC One W offers brand-new support for WebRTC, a protocol designed by Google to allow direct connection of audio and video between Internet browsers. The advantages provided by WebRTC include platform and device independence and security due to the required use of encrypted connections and the fact that it does not require specific software that needs maintenance. Additionally, since it is a highly integrable protocol, WebRTC can easily be used to enhance the user experience within other technologies and applications.

Registration Routing

With the arrival of WebRTC, the way in which business handl Registrations has changed. Now, in addition to the support of WS and WSS protocols and the ICE/STUN, we also allow the finalization of registers in vSBC. This means that it is possible to configure parameters so that the vSBC is the software that creates and maintains registrations on an extension line. vSBC also monitors extensions, as well as imports/exports extensions in mass, in order to facilitate administration and controls.

It is now also possible to forward registrations (routing) in order for them to be finalized in another NAP and allow the use of vSBC features during all routing stage (receiving, forwarding and finalization). Through this configuration, it is even possible to use vSBC to receive registrations and forward them to other vSBCs, which will then perform finalization. Additionally, it is possible to change the protocol that will be used after forwarding. For example, a registration arriving to the vSBC via WSS can be configured to be forwarded to another NAP via UDP. This routing of records also offers route prioritization.

vSBC also offers support for Websocket and Websocket Secure media protocols, as well as SRTP, DTLS and DTLS-FB audio protocols. For each SIP protocol, users can choose which media protocol to use, which allows them to limit the protocols accepted in NAP, SIP Invite, and registration.

Management and diagnostics tools

Extension Management: Now that registrations can be finalized on vSBC One W, it is now possible to register, import, export and monitor extensions directly on the interface.

Call Trace: Displays the flow of an SIP call, presenting detailed call information and all SIP messages exchanged, which assists in analyzing issues. The information is sent to Insight by vSBC, which is responsible for handling this data and assembling the Call Trace graph.

Netconsole: Makes it possible to create a system using a remote machine in order to debug potential issues in the operating system.

Routing with route failover

Obtain increased control of expenses related to telephony fees through the configuration of routes per prefix or by developing loyalty among VoIP telephony operators, which allows calls to be directed to the operators that offer the best cost-benefit for each call, thereby reducing costs related to fees. Besides the cost, vSBC One W allows you to configure automatic route overflow by day or time. In addition to reducing costs, this allows calls to be directed to another number after business hours, for example.

Route failover is an important feature for those who cannot afford to be without telephony services in their network. It is implemented using routes together with the monitoring of the call destination's VoIP server. If the VoIP server does not respond to monitoring commands, vSBC One W looks for another compatible route.

Technical specifications

Security
- Access to web interface with password
- Access via HTTPS protocol
- Access control - ACL (whitelist and ban list)
- TLS and WSS SIP protocols
- SRTP and DTLS media protocols
- Network topology obfuscation
- Protection against malformed packets
- Fraud prevention: call blocking by destination and origin
- DoS/DDoS protection

VoIP Resources
- Proxy SIP fallback
- Monitoring of NAPs (Network Access Points) or Keep Alive via SIP OPTIONS
- Monitoring of registered extensions
- DTMF send mode selection: In band, Out band - RTP (RFC 2833), Out band - SIP Info
- Manipulation of destination number (To) and source number (From)
- Addition, removal and retransmission of headers
- Transcoding (conversion between G.711, G.729, G.722 and Opus codecs)
- Compatibility with the SIPREC standard in recording
- SIP-I

Interoperation
- Fax Interoperation (T.38 with fallback to G.711)
- IPv4 to IPv6
- RTP with conversion between UDP, TCP, SRTP and DTLS
- SIP trunking
- Direct routing for Microsoft Teams, interoperability both with and without media bypass

Virtualization systems (Hypervisor)
- VMWare®
- KVM®
- XenServer®

Cloud environments
- Amazon Web Services
- Google Compute Engine
- Microsoft Azure
- Huawei Cloud
- Oracle Cloud

QoS (Quality control)
- DiffServ - RFC 4594 (classification and management of traffic)

Call routing
- LCR - least cost routing
- Routing based on source, destination, time, and prioritization
- Scripted routing
- Consulting of portability database
- Fallback for failed routes
- Failover retry based on cause of failure
- Route profiling
- Load balancing

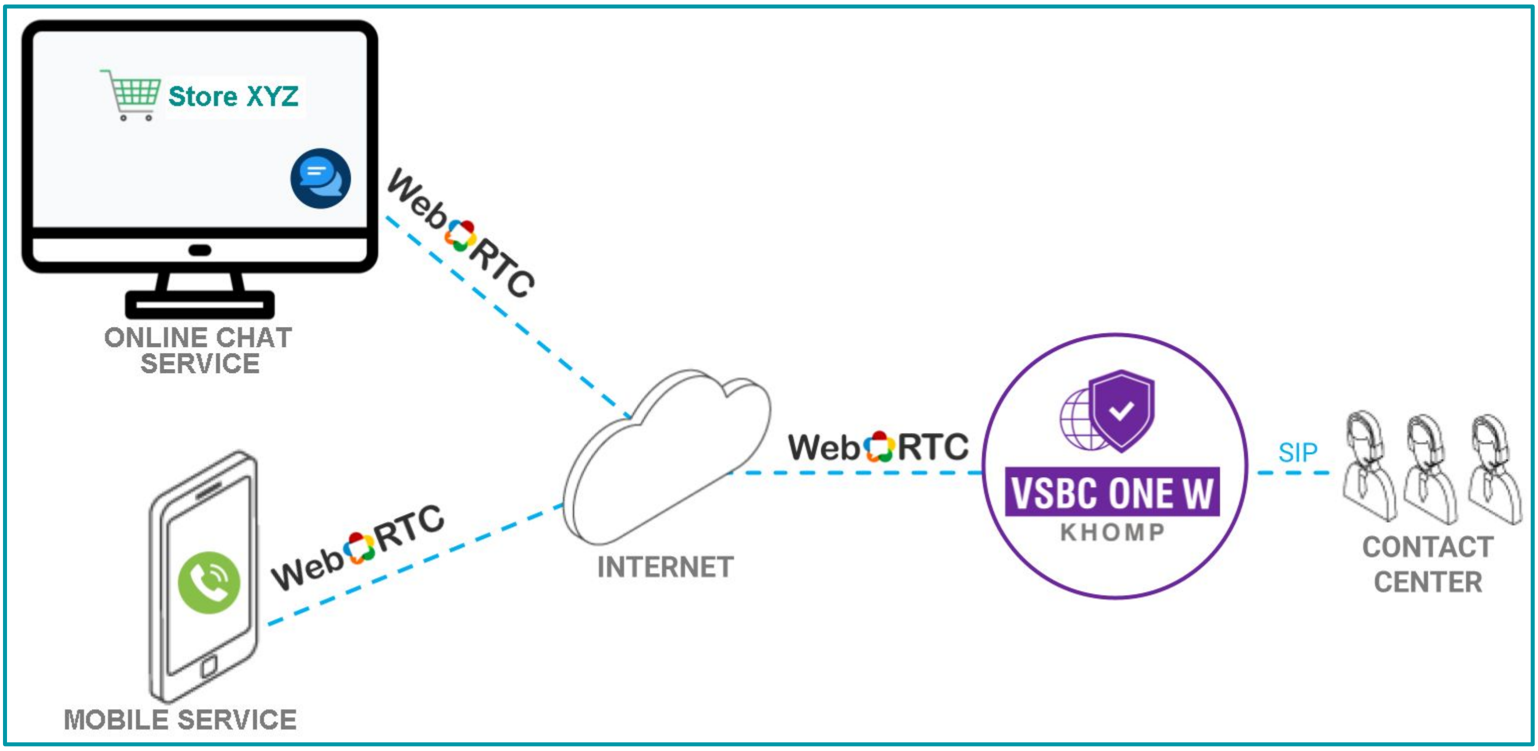
Registration routing
- Configuring of registration by NAP
- Sending of registration
- Finalization of registration
- Forwarding of registration
- Prioritization-based routing

NAT traversal
- Interconnection between different networks
- External IP configuration
- STUN
- ICE

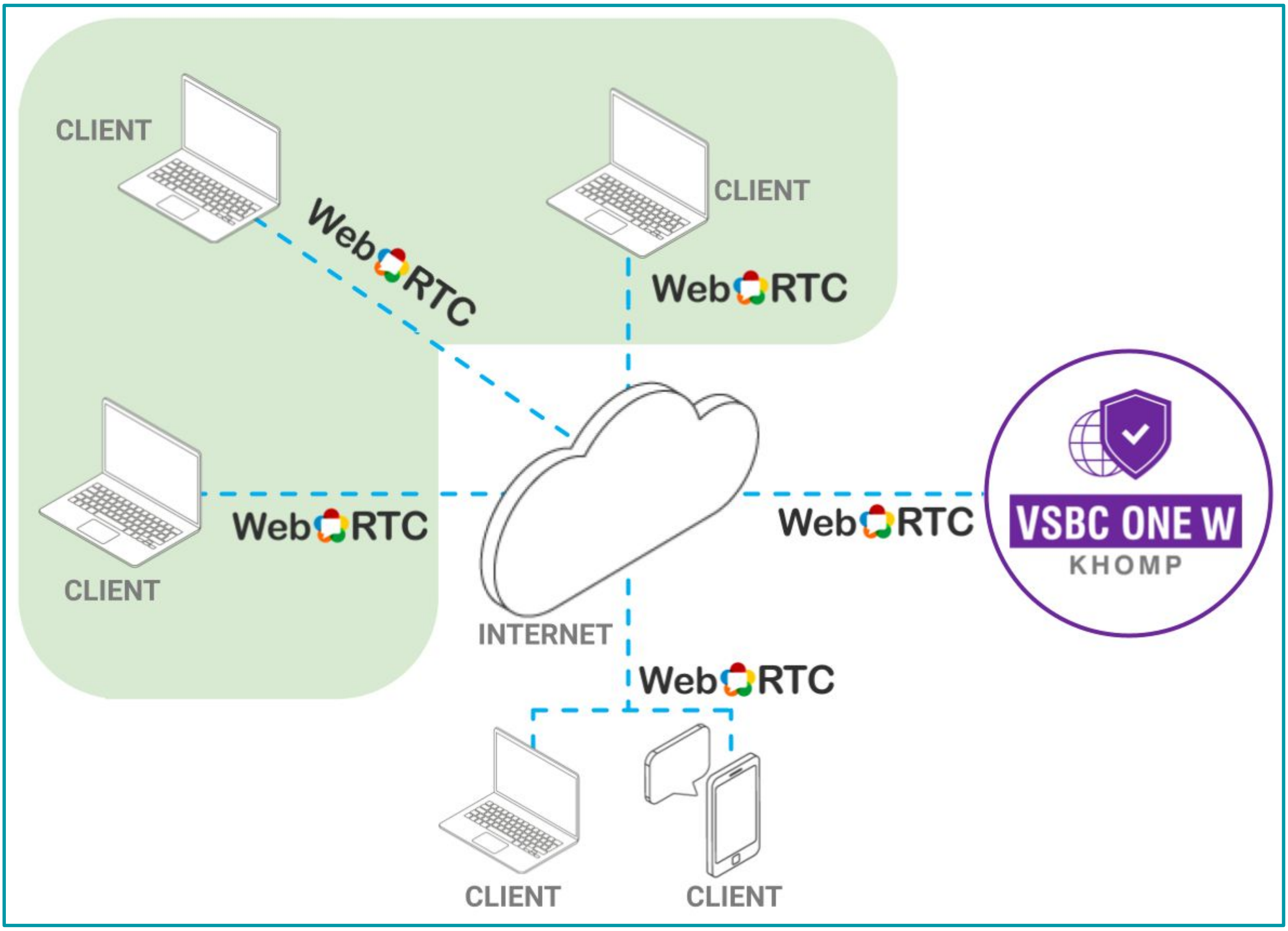
Supported codecs
- G.711 a-law/µ-law
- G.729A
- G.722
- Opus
- DVI
- GSM

Other functions
- HA - high availability
- DR - Disaster Recovery
- Auditing of configuration and recovery
- Limiting of concurrent calls per network
- MOS call quality statistics
- Provisioning (configurations for exporting and importing)
- Configuration, monitoring, administration and diagnosis via Web
- CLI (Command Line Interface) tool
- Generating of signaling and system logs
- Customizable CDR
- Interface access control for users of different access levels
- SNMP support
- Use of RADIUS protocol for accounting (billing)
- Packet Capture
- Test calls

Suggested application



Caption: *Standard arrange for use and Click-to-Call E-Commerce Integration.*



Caption: *Connection between distributed Clients and Agents on the Internet.*

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