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Virtual Session Border Controller WebRTC

VSBC ONE W

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Overview

vSBC One W is the latest version of Khomp's software-based Session Border Controller, which can be installed on physical, virtualized, or cloud servers. Designed to manage telephony traffic, vSBC One W provides security against malicious traffic, call encryption, packet normalization, and blocks unauthorized SIP packets. It supports NAT and transcoding, making it easy to connect different VoIP networks and translate between protocols and codecs. It's flexible and cost-effective, working on physical servers, VMWare®, or cloud platforms like Azure, AWS, Huawei, Oracle, and Google Cloud, while offering advanced routing and security features accessible from anywhere.

WebRTC

One of the standout features of vSBC One W is its support for WebRTC, a protocol created by Google that enables direct audio and video connections between internet browsers. The benefits of WebRTC include independence from specific platforms or devices and increased security, since it requires encrypted connections and doesn't rely on additional software that needs maintenance. Plus, being a highly adaptable protocol, WebRTC can easily enhance user experience across various technologies and applications.

Route failover and routing

Gain more control over phone call costs by setting routes based on carrier prefix or VoIP provider preference, sending calls through the most cost-efficient options. vSBC One W lets you set up automatic route switching based on time or day, like redirecting calls after business hours. Route failover guarantees uninterrupted phone service by switching to backup routes if the target VoIP server doesn't respond to monitoring.

Registration routing

With WebRTC integration, vSBC One W now supports WS, WSS, ICE/STUN protocols and can complete registrations directly. It manages and maintains extension registrations, monitors them, and offers bulk import/export. You can also forward registrations to another NAP, using all of vSBC's features during routing. Protocols can be switched mid-routing, for example, from WSS to UDP, while prioritizing routes. It supports WebSocket, WebSocket Secure, SRTP, DTLS, and DTLS-FB media protocols, allowing you to choose and limit accepted protocols for NAP, SIP Invite, and registration.

VoIP network security

A secure VoIP network must have features that prevent unauthorized access, malicious attacks, or call interception. These situations could severely compromise operations, affecting business success. vSBC One W was built with VoIP security in mind, offering call encryption to prevent the interception of signaling and audio, and topology hiding, which ensures external devices can't identify your company's VoIP network structure.

Additionally, vSBC One W includes Registration Routing, a crucial feature for remote workers who need to connect and make calls using the company's telephony infrastructure from outside the corporate network.

Management and diagnostic tools

- **Extension Management**: With the ability to complete registrations directly on vSBC One W, you can now register, import, export, and monitor extensions from the interface.
- **Call Trace**: Shows the flow of a SIP call, providing detailed information and all SIP message exchanges, which helps with troubleshooting. vSBC sends this information to Insight, which processes it and generates the Call Trace graph.
- **Netconsole**: Enables logging to a remote machine to troubleshoot potential operating system issues.

Technical specifications

Security
- Password-protected web interface access
- HTTPS protocol access
- Access control - ACL (allow and block lists)
- SIP TLS and WSS protocols
- SRTP and DTLS media protocols
- Network topology hiding
- Protection against malformed packets
- Fraud prevention: blocking calls by destination and origin
- DoS/DDoS protection

VoIP Features
- SIP Proxy Fallback
- Monitoring of NAPs (network access points) or Keep Alive via SIP OPTIONS
- Monitoring of registered extensions
- DTMF transmission mode selection: In band, Out band - RTP (RFC 2833), Out band - SIP Info
- Manipulation of destination number (To) and origin number (From)
- Addition, removal, and retransmission of headers
- Transcoding (conversion between G.711, G.729, G.722, and Opus codecs)
- SIPREC standard compatibility for recording
- SIP-I

Interoperability
- Fax interoperability (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, with or without media bypass interoperability

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

Call Routing
- LCR - least cost routing
- Routing based on origin, destination, time, and prioritization
- Script-based routing
- Portability database lookup
- Fallback for failed routes
- Failover retry based on failure causes
- Route profiles
- Load balancing

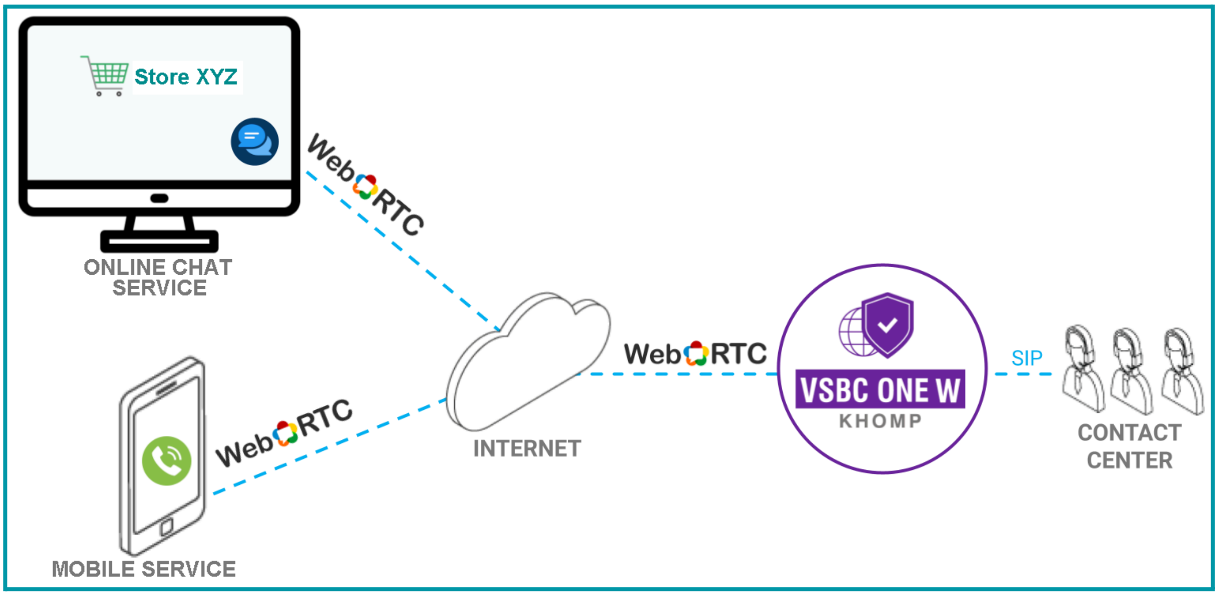
Registration Routing
- Registration configuration by NAP
- Registration sending
- Registration termination
- Registration forwarding
- Routing based on prioritization

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

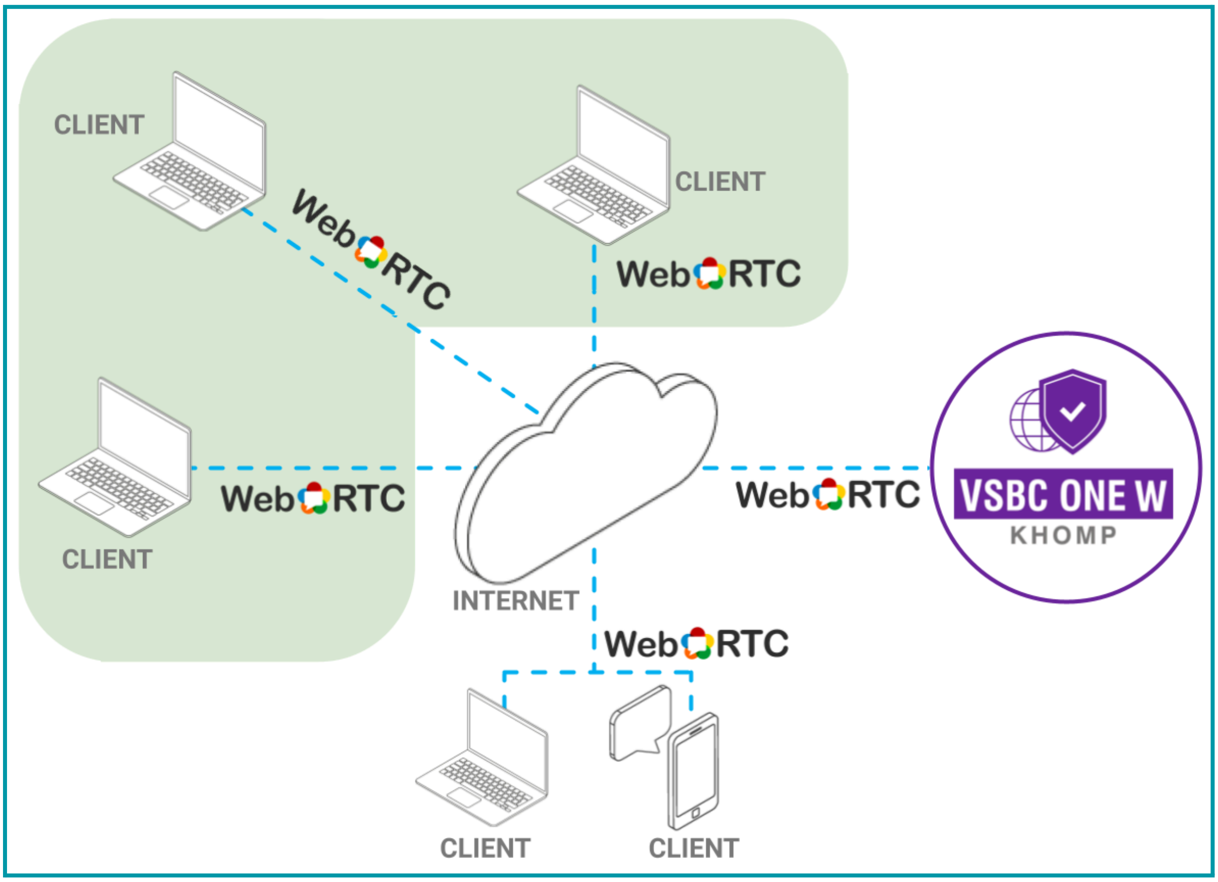
Other Features
- CAPS (call attempts per second)
- HA - high availability
- DR - Disaster Recovery
- Configuration audit and recovery
- Limit on simultaneous calls per network
- Call quality MOS statistics
- Provisioning (configuration import/export)
- Web-based configuration, monitoring, administration, and diagnostics
- CLI (Command Line Interface) tool
- Generation of signaling and system logs
- Customizable CDR
- Access control to the interface for users with different levels
- SNMP support
- RADIUS protocol support for accounting
- Packet capture
- Test call functionality

Supported Platforms
- Baremetal
- OpenStack
- KVM
- VMware
- Microsoft Azure
- Amazon AWS
- Google Cloud
- Oracle Cloud
- Huawei Cloud

Application models



Caption: *Typical use case and Click-to-Call E-Commerce Integration.*



Caption: *Connection with Clients and Agents distributed over the Internet*

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