ABC WEBRTC

WebRTC2SIP Translation

Encryption

Session control

Call routing

NAT traversal

Announcements and recording

Open interfaces

Flexible routing

Load balancing

Software based transcoding

Hardware independent

Could ready

ABOUT FRAFOS

FRAFOS offers highly scalable WebRTC and SBC solutions for VoIP and NGN providers. FRAFOS was established May 2010 in Berlin, Germany, by a team that has strongly contributed to the VoIP revolution in Germany while being part of the Fraunhofer Fokus in the late nineties.

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Combining SIP and WebRTC Services

FRAFOS ABC WebRTC Gateway

With the ABC WebRTC gateway operators deploy a scalable WebRTC to SIP solution that enables VoIP operators and enterprises to extend the scope of their services from SIP and PSTN to browser based applications in a seamless manner. Deploying the ABC WebRTC gateway protects the investment of VoIP service providers and enterprises by enabling them to quickly extend the reach and breadth of their offerings to include rich multimedia directly from any web browser – including voice, video, IM and presence. Following the latest standards the ABC WebRTC gateway provides the needed protocol translation capabilities for the transparent integration of SIP and WebRTC services.

The ABC WebRTC gateway is a software based solution that functions as a virtualized or on-the-box software solution. Beside the ability to translate between SIP and WebRTC protocols, the ABC WebRTC gateway provides VoIP operators with the needed security and access control mechanisms as well as monitoring capabilities, transcoding, rate limiting and signaling and media security using SRTP, DTLS and TLS.

Strong SIP and WebRTC Interoperability

Standards compliant with highly configurable SIP normalization options. The ABC WebRTC gateway can mediate between different flavors of SIP and WebRTC on the SIP transaction and session level as well the transport level.

Secure Border Control

As the interface between end users and internal infrastructure, a high level of secure peering and border control is required from a WebRTC gateway. The ABC WebRTC gateway deploys proven web-based cryptographic protocols that provide encryption for both media and signaling. Further, the ABC WebRTC gateway supports topology hiding on signaling and media layers as well as DoS protection through rate limiting, call rejection and SLA monitoring.

Scalable Design

To accommodate the various needs and performance requirements of operators of all sizes the ABC WebRT gateway was developed to be deployed over different hardware platforms. The ABC WebRTC gateway can be deployed as an integrated piece of software running over a mini router running at an enterprise premises as well as on top of high end hardware platforms supporting the carrier grade requirements of service providers.

Open Application Platform

The ABC WebRTC provides a built-in programmable and open media server platform for supporting recording, announcements and real-time web applications. These applications are highly programmable and offer open interfaces to enable a seamless and rapid integration with the operator’s service delivery platform.

Customizable Policies

Different deployment scenarios will have different requirements in terms of used applications, transport protocols, media codecs, SLAs, SIP signaling and session control. The ABC WebRTC gateway enables operators to customize the behavior of the WebRTC gateway in accordance with their own policies.

Technical Specifications

Supported Platforms

- Linux

WebRTC Features

- Javascript
- SIP over WebSocket
- NAT traversal using ICE, TURN, STUN
- JsSIP support

Media Services

- Routing audio codec including G.711 and OPUS.
- Routing of video codec including VP8
- Dynamic jitter control
- NAT/NAPT on media
- RTP inactivity monitoring
- Codec filtering

Media Applications

- Call recording
- Announcement services
- Software based transcoding (G711u/a, G726, OPUS, iLBC, L16, G722, Speex; on request: G729a, G729a/b, AMR)

Management Capabilities

- GUI based configuration and monitoring
- Secure embedded web-based GUI
- SSH access
- SNMP V2 status and logs
- Local logging of alarms, events and statistics
- REST and XML RPC based open interfaces

Call Routing

- Call blocking and filtering
- Embedded routing engine
- Load balancing
- Peer monitoring and availability detection
- Alternative routing on failure
- Table based routing for LCA

SIP

- Registration passthrough
- Registration caching and offload
- SIP header manipulation
- SIP Back2Back UA

Protocol Support

- UDP, TCP
- Translation between transport protocols
- Per source/destination transport layer mediation
- SNMP, NTP, SSH
- DNS
- RTP, RTCP, SRTP
- WebSocket
- TLS
- DTLS, SDES

Security

- Signaling encryption security using TLS
- Media encryption using SRTP
- Exchange of security keys using DTLS and SDE
- Signaling topology hiding
- Media topology hiding
- RTP DoS protection
- Call rejection under DoS
- Call rate limitation

QoS Control

- Bandwidth limitation and management
- Call admission control per peering partner/trunk

High Availability

- Active/Hot Standby redundancy model

Virtualization

- Amazon cloud
- Virtualization software OVM, KVM ..

Hardware

- Hardware independent