Genesys WebRTC Service

Genesys WebRTC Service provides a simplified, secure channel of communication including voice, video, and data over the Web to further enhance the customer engagement experience.

Creating a Compelling Contact Center Experience Using Video

An organization's website is the most significant source of company visibility, revenue generation and more often than not, the first point of direct customer or prospect impression. It therefore makes sense to fully leverage the company's Internet presence and streamline a visitor's web experience in order to maximize business opportunities and revenue potential.

The multiple communication channels integrated into the website define the scope and limitations of how a customer can interact with your business – typically by initiating a text-based chat session or filling out a standard email form. Often, this web interaction can be associated with a telephone interaction using legacy means. Alternatively, you may have integrated off-the-shelf Internet voice or video solutions into the website in an effort to add additional communication channels, reduce the overall number of toll-free phone calls and help to drive down costs. In this scenario, website visitors typically download a third-party voice/video application for a call to work. Unfortunately, third-party voice/video tools can often add extra complexity and increase a visitor's frustration and engagement time considerably – leading to a downturn in customer satisfaction and a potential loss of business.

Lightweight, Convenient, Standards-Based Communications

With the advent of the WebRTC (Web Real-Time Communications) standard, lightweight voice and video services can now be implemented directly into HTML 5 browsers to normalize both ends of an Internet-initiated conversation. This eliminates the need for additional software or plug-ins installed on the customer's web-enabled device.

The Genesys WebRTC Service provides a simplified, secure channel of communication including voice, video, and data over the Web to further enhance the customer engagement experience. Contacting a company representative or service agent is seamless for partners, prospects, and customers, as they can quickly initiate a WebRTC voice or multimedia video call via their browser directly from your company website. This keeps customer frustration to a minimum and maintains a seamless level of interaction to enrich their overall web experience.

Streamlining the Customer Experience and Reducing Contact Center TCO

The Genesys WebRTC Service is quickly and easily deployed with zero disruption to the existing contact center and customer service operations. WebRTC helps to reduce the contact center's cost of ownership and adds additional levels of flexibility into how agents can be provisioned and managed. The Genesys WebRTC Service integrates with the Genesys SIP Contact Center solution, leveraging the Genesys routing and cross-channel contact infrastructure to enable a robust, scalable, and flexible virtual customer service solution across the entire organization.

For customer service agents, WebRTC-initiated calls are identical to the regular IP/SIP (Session Initiation Protocol) voice and video calls they already handle, without the need for specialized user interfaces, controls or processes. Therefore, your contact center can quickly and fully support WebRTC calls with no additional development or comprehensive agent training.

BENEFITS

- Website visitors experience seamless integration of multi-channel communications, including real-time, voice only and face-to-face video calls
- Customers can initiate contact with a company representative or customer service agent directly from the website and without the need to install an additional third-party application
- Integrates with Genesys SIP solutions to leverage existing routing and cross-channel customer service infrastructures
- Use effective, real-time video calls and/or video clips for a differentiated customer service that enhances the customer experience
- Cut communication costs and improve TCO by reducing the need for dedicated 800# line subscriptions
- Eliminate the incompatibility concerns, complexity and cost of third-party video call conferencing software
- Collect contextual data from a visitor's web sessions and forward it to the Genesys SIP-enabled contact center as attached data
- Separately identify WebRTC initiated sessions for performance cost impact analysis reporting

"WebRTC is one of the most disruptive internet technologies for the contact center we've seen in years"

Slava Zhakov,
CTO, Genesys Labs

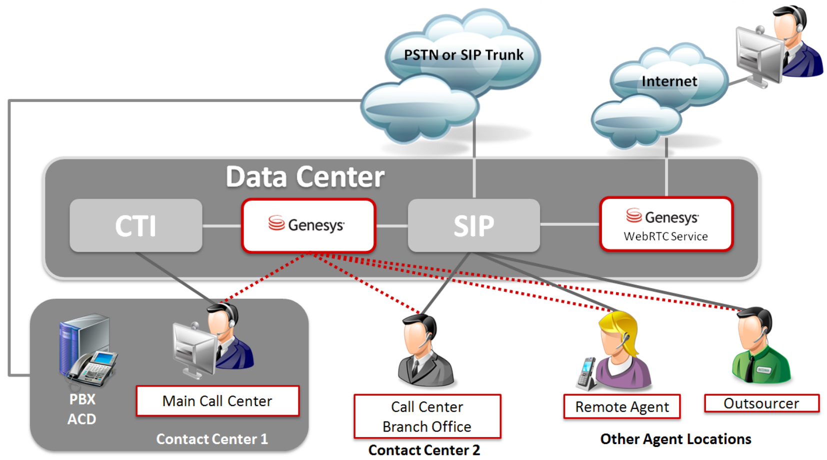

Fig 1. Genesys WebRTC Service can be quickly deployed and easily managed to support Contact Center operations

Technical Specifications

Communications

- Support for incoming calls from the web
- Two-way audio-only calls
- Two-way audio and video calls
- Support for transitions between audio-only and video/audio sessions
- Transcoding provided on supported media types in both directions

Compatibility with Genesys Solutions

- Genesys SIP Contact Center Solutions
- Genesys Active Call Recording

Supported web browsers

- Google Chrome

Additional WebRTC-enabled browsers when available

Deployment Modes

- Dedicated server / Shared server with other Genesys components
- Multiple load-balanced instances for scalability

Security

- Support for HTTPS, SIP TLS, TLS-FIPS, and secured connection to Configuration Server
- Encrypted RTP traffic according to IETF recommendations (SRTP)
- Support for configuration to use SRTP when outside the trusted area
- Support for Client Side Port Definition (CSPD), allowing for customization of the:
  - > SIP-side RTP port
  - > Web-side RTP port
  - > Transport address and port for a client-side connection
- Support for the use of multiple NICS, to separate public and private interfaces when both are used (DMZ deployment)
- Firewall traversal with support of ICE

Supported Codecs

- G.711 A-law and μ-Law audio format
- G.729 audio
- H.264 profiles up to and including Profile 3.1 on SIP/RTP side
- VP8 video
- Real-Time media transcoding whenever required
- RTCP AVPF (*partially supported for the current release)

KEY FEATURES

- Click-through voice and video calls directly from the Web
- Software-only, standards-based, zero foot-print integration—no plug-ins, downloads, or installs
- WebRTC APIs for application development, customization, and enhancement
- Contextual user data can be associated with the calls
- TCO reduction from replacing costly dedicated 800# lines
- Easy deployment with minimum disruption to existing customer service operation
- Secure communications with signaling/media encryption and firewall traversal
- Full integration with Genesys SIP virtual customer service solution

About Genesys

Genesys is a leading provider of customer service and contact center solutions. With more than 3,000 customers in 80 countries, Genesys software directs more than 100 million interactions every day from the contact center to the back office, helping companies deliver fast, simple service and a highly personalized cross-channel customer experience. Genesys software also optimizes processes and the performance of customer-facing employees across the enterprise.

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