Kandy Link — WebRTC Services for Contact Centers

Creating Better Customer Experiences and Better Value

WebRTC – Changing Customer Engagement

The WebRTC standard, adopted by all of the top web browser providers, makes it easy to deliver rich, multi-media, experiences using only a web browser. Users don't need any extra software or plugins for two-way voice, video or collaboration sessions. Essentially, it means that any user can use almost any device to start a two-way conversation – just by browsing to a website.

It sounds simple enough but until WebRTC came along most browsers only supported text – anything more required site specific software. For contact center managers the adoption of this multi-media standard opens up three new powerful and cost-effective ways to engage with customers and to deploy agents.

1. Create client-less contact agent applications to lower the cost of contact center deployments.
2. Offer click-to-connect services for customers from any web page (voice, video, screen share, etc.) – no need to dial a separate toll free number.
3. Use WebRTC APIs to embed communications into mobile apps.

No Disruptions to Existing Contact Center

Of course, with every technology innovation there comes the challenge of integration. Ribbon's Kandy Link WebRTC Gateway, was created to bridge the divide between the traditional SIP-based communications most contact center use and WebRTC. Kandy Link is designed to look like an incoming SIP trunk or as a SIP endpoint to an existing contact center implementation. Users interact via WebRTC services while the contact center sees each session as a SIP session

Contact center managers can deploy Kandy Link WITHOUT disrupting an existing contact center deployment. That means no expensive upgrades, no extra licenses, no massive professional services engagements or center-wide agent retraining.

Better Value

Traditionally, upgrading contact center functionality requires multiple elements to be upgraded and in many cases licenses added for new functionality. This can be costly or even cost prohibitive. By contrast, Kandy Link has simple use-based licensing so deploying WebRTC-based services is simple and cost effective. Once deployed Kandy Link's services start returning value almost immediately. Cost savings come from:

A. Reducing toll-free traffic and costs.

B. Speeding transactions by providing context to an agent by including the site, page or product/service the customer is viewing.

C. Improving customer experiences – opportunity to use video, co-browsing, etc. to personalize service and improve first call resolution.

D. Simple use-based licensing that is independent of contact center software versions and licensing constraints.

Carrier Grade Availability

Ribbon Communications is a leading provider of service provider software and solutions. The Kandy Link WebRTC Gateway is designed to meet the availability and scale requirements of the world's largest service providers. It's always deployed in high-availability pairs to assure uptime.

Multiple Deployment Options

Kandy Link is deployed in a subscription based, fully managed model. Users can choose the operational deployment model that best fits their use case:

Public Cloud – as a Service

No hardware or capital investment – WebRTC services are delivered from the public Kandy cloud with access to Kandy's entire apps portfolio.

Private Cloud

In network or in a 3rd party data center. Available as software-only running on virtualized environments (VMWare, KVM). Kandy Cloud delivers access to the Kandy apps portfolio.

On-Premise

Kandy Link on-premise for local media and user data (for enhanced privacy, security, and media performance). Kandy Link connects to Kandy Cloud for services such as Mobile Push Notifications (for Apple, Google).

Private or Hybrid Cloud Deployment Environments

- VMware ESXi 5.5 and 6.x environment with RHEL 6.9 as a Guest OS
- Support for RedHat Host OS / KVM and Oracle Linux Host OS / KVM.
- Production deployments are multi-node for high availability support

Key Features

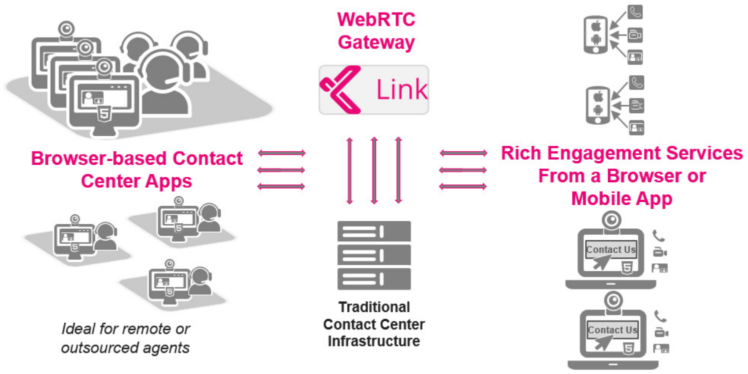
- Industry leading capacity up to 2m BHCA
- Carrier-grade, multi tenant platform
- Anonymous call/Call Me for unregistered users
- Collaboration Services Support
- Persistent user subscriptions
- External notification interface
- Instant Messaging Support (with Ribbon AS)
- Presence support (with Ribbon AS)

Security and Authentication

- Authentication framework for REST requests
    - Anonymous and time-limited token-based anonymous authentication
    - Token-based authentication
    - User authentication
- DTLS to SDES fallback deactivation for incoming calls
- DTLS-SRTP interoperability
- Secure connections for Apple devices
- Disable root user SSH login
- FIPS 140-2 compliance
- Customer defined password policies
- TLS versions and cipher suites configuration
- Certificate validity check using OCSP
- SNMP v3
- Network Topology Hiding

Call Control

- Call control from a client
- Call display for incoming calls
- Call disposition



- Call redirect to multiple service networks
- Peer to data channel support
- Peer to peer call support
- Redundancy and fallback and recovery support for CUCM
- Send and receive custom SIP headers
- Feedback using PLI or FIR Interop Support

SIP Signaling

- Configurable call and data channel decline SIP responsecodes
- Incoming SIP messages via TCP
- RTCP multiplexing support
- SIP Reason header support for BYE messages
- SIP UA interoperability with Cisco UCM
- SSRC insertion in SDP
- Support for AS and Kandy Link software integration with SBC for transcoding

Media

- Early media and RFC 3262 support
- ICE NAT traversal for media and signal for restricted client network
- Trickle ICE support
- Media tiers behind a one-to-one NAT
- Support for DSCP packet marking on the media broker
- Support for TURN and Broker tiers (media tiers) that are behind a one-to-one NAT.
- RTCP multiplexing support
- SSRC insertion in SDP
- Multiple Media Stream Support

WebRTC & APIs

- JavaScript Library (JSL)
- Mobile Software Development Kit (SDK)
- REST API for subscribing to generic SIP events
- WebSocket disconnection and connection detection

Performance

- Fault and Performance Management
- CPU overload controls
- Distributed caching framework
- Glare condition handling
- Installation templates for network configuration
- Statistics and reporting
- Usage KPI reporting