2014 WebRTC State-of-The-Market Report

How professionals are using WebRTC now and plan to use WebRTC in the coming year.

April 2014

By: Larry Hettick
Editorial Director and Senior Research Fellow
Webtorials

Sponsored by:

AVAYA The Power of We™ — Sonus — GENBAND

Introduction and Key Findings

WebRTC has been the center of notable industry focus for the past 18 months, with active proponents and skeptics alike. This State-of-the-Market Report provides quantitative insights and qualitative analysis on how IT professionals are using WebRTC today and how they plan to use WebRTC in the coming year.

WebRTC is a platform that provides web application developers the ability to write real-time multimedia applications (such as voice calls and video chat), without requiring software plug-ins, downloads or installs. It enables real time applications that are developed in the browser using JavaScript Application Programming Interfaces (APIs) and HTML5. WebRTC is supported by browsers from Google, Mozilla and Opera; however, it is not currently supported by Microsoft's Internet Explorer or Apple Safari. The World Wide Web Consortium (W3C) is developing the WebRTC Application APIs, while Internet Engineering Task Force (IETF) is working on associated protocols.

Among the key findings:

- 43% of respondents are using WebRTC or plan to use it within the next 12 months
- 43% of respondents have heard of it, but have no plans to use WebRTC
- 90% of those surveyed believe WebRTC has the potential to improve contact center services
- 67% viewed WebRTC as a potential solution for external video communications
- 85% of those surveyed see WebRTC and SIP (session initiation protocol) solutions as complementary
- Near-term support for WebRTC was greatest among small businesses and larger enterprise

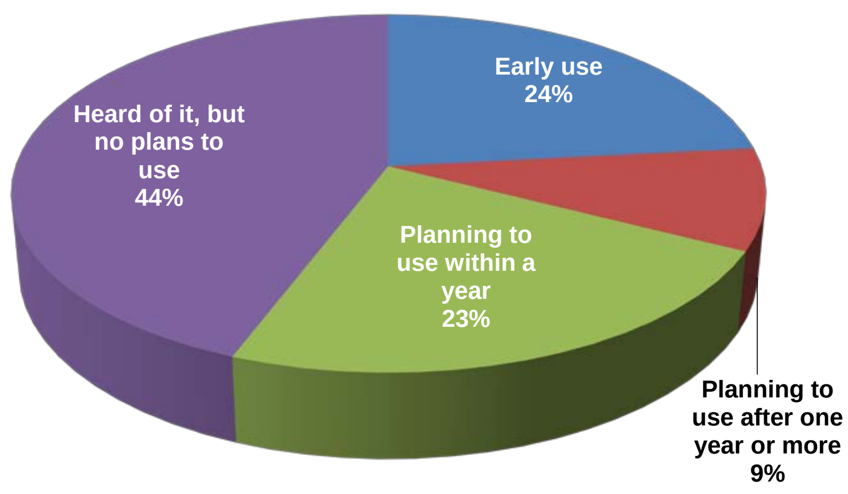
The survey, conducted from December 2013 through January 2014, asked 210 IT professionals about WebRTC plans and perceived trends. Of these, 22 respondents' results were removed from the sample because they did not demonstrate a qualifying understanding of the technology. Data was collected to highlight the perceived benefits and barriers to introducing WebRTC, and the results also show who IT managers will seek out for WebRTC support and services.

Respondents were well balanced by company size: about 24% answered for small business, while another 24% came from organizations with 1,000 to 1,999 employees. About 16% of responses represented organizations with 100 to 999 employees; 17% from organizations with 10,000 to 49,999; and 19% came from organizations with 50,000 or more employees. Geographically, 57% of the survey sample came from the U.S., 19% came from Europe and 16% from Asia, with the balance coming from other regions.

WebRTC Deployment Plans: Yes and No

Validating both WebRTC proponents and skeptics alike, the number of respondents who plan to use WebRTC and those who don't plan to use it soon are almost equally split. Counting those already use or plan to use WebRTC: some 24% already use it, another 23% plan to use WebRTC within a year, for a total of 47%. Meanwhile, another 44% have heard of WebRTC but have no plans to use it. Another 9% are planning to use WebRTC after a year. Conclusion: a slight advantage for WebRTC proponents when counting all who use or plan to use it in the future.

Figure 1: WebRTC Deployment Stage



At what stage do you consider your company or organization to be in the deployment of WebRTC?

The Link between WebRTC, UC and Collaboration Technologies

To understand correlations between support for WebRTC and other technologies, we asked about responding companies' current support for collaboration technologies including VoIP and session border controller (SBC) deployments, voice and video conferencing, Google Apps, and BYOD (bring your own device.)

- Respondents to this survey have a higher percentage for using VoIP and SBCs than those of the general business population: just over 96% of those who completed the survey are using VoIP today, while 72% are using SBCs today and another 13.6% are evaluating SBCs.

- Voice and video conferencing use was also high among survey respondents. Nearly 97% or those who completed the survey said their organizations use voice conferencing, while about 82% use video for intra-company conferencing and 84% use video for external or intercompany communications.

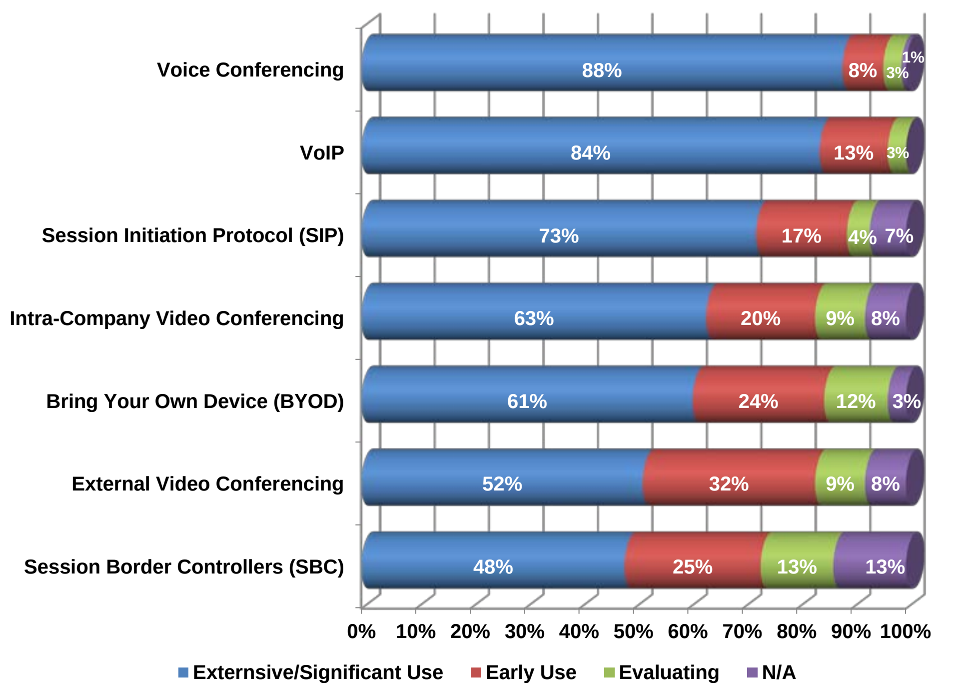
- Nearly all respondents represent companies that encourage BYOD, with 97% offering BYOD support.

- Just over 80% of organizations represented in the survey use and support Google Apps

Admittedly, these high-use percentages of collaboration technologies among survey respondents are atypical of the general business population[1]. However, the data does suggest that current VoIP and SBC users, voice and video conferencing users, BYOD supporters, and Google Apps users are more likely to *understand* WebRTC, because only those who demonstrated a basic understanding of WebRTC were invited and encouraged to complete this survey; nearly 30% of respondents who did not demonstrate a working understanding of WebRTC were excluded from the survey results

"For those suppliers who offer or plan to offer WebRTC solutions, the existing adoption of collaboration tools is a leading indicator to qualify potential WebRTC clients."

Figure 2: Collaborative Technology Deployment



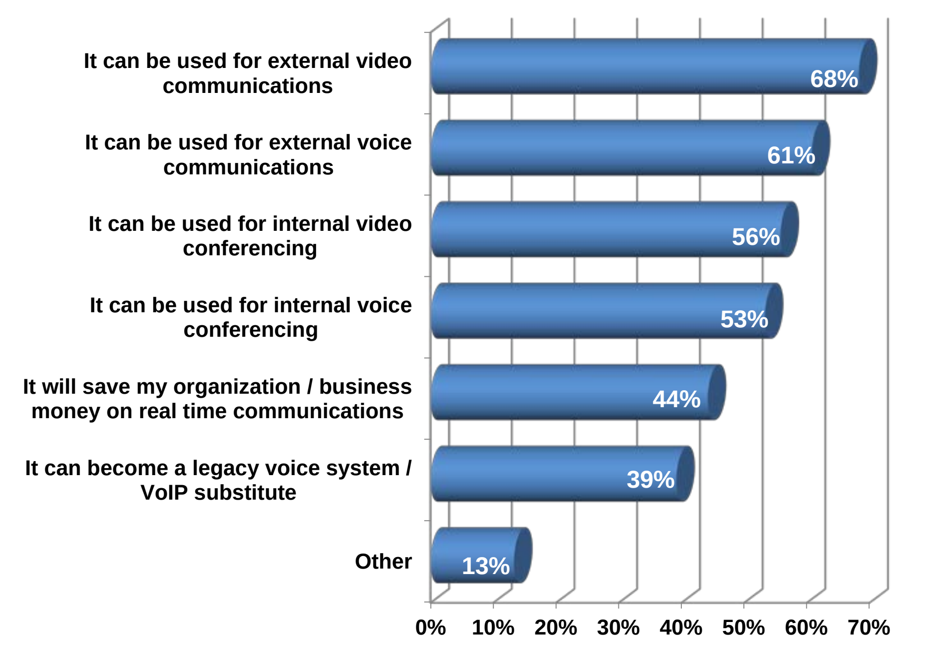
At what stage do you consider your company or organization to be in the deployment of these services?

[1] While these percentages are somewhat atypical of the general business population, they are not at all surprising in that the surveyed base consisted of the Webtorials community which tends to attract thought-leaders and early adopters.

WebRTC Uses and Benefits

Validating what may be obvious to most informed observers: voice conferencing and video conferencing remain very important to organizations -- whether using WebRTC or not. What is also noteworthy is that respondents believe that WebRTC is now or will become very valuable when used for video communications: 68% rated *external* video communications as an important WebRTC application, while 56% rated video communications using WebRTC as important for *internal* conferencing. Similarly, over 60% noted that WebRTC is important for external voice communications, while 53% rated it as important for internal voice conferences.

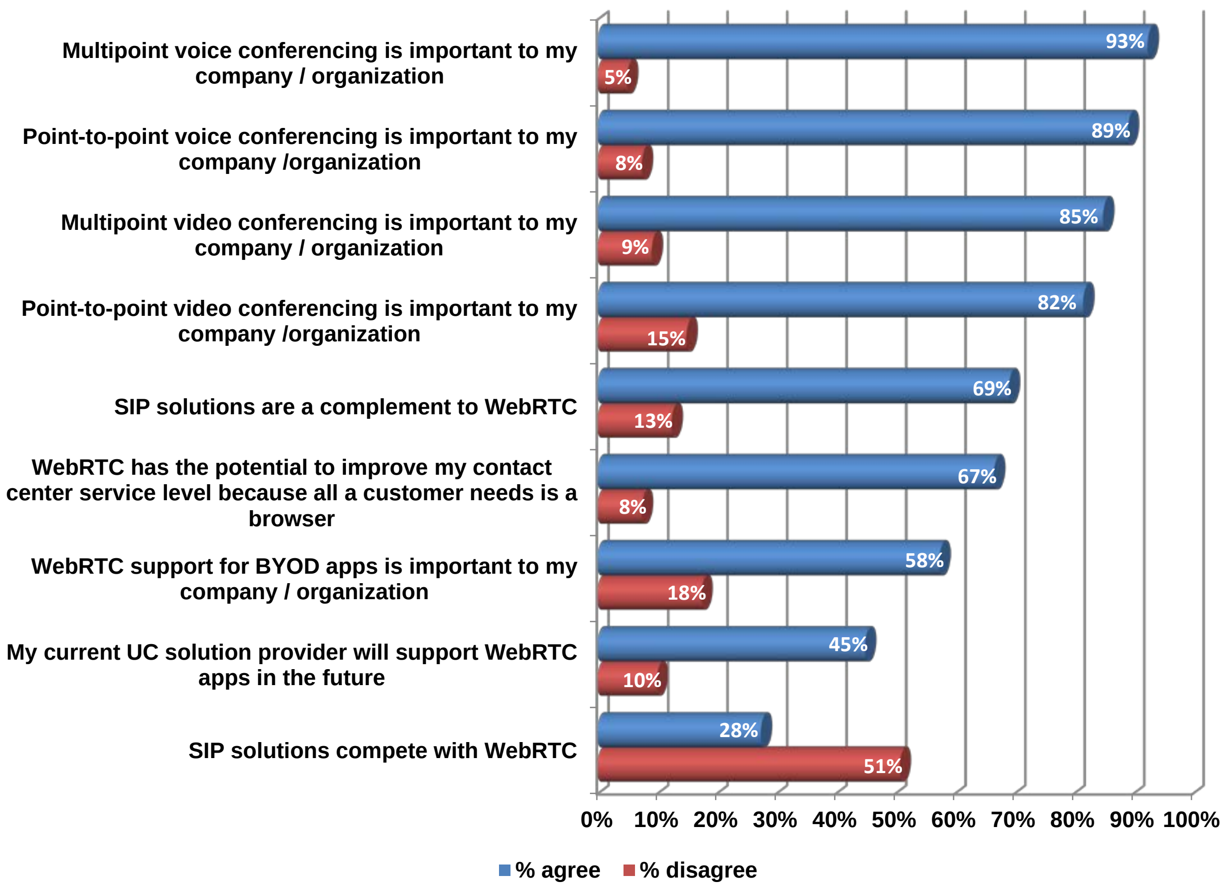
Figure 3: WebRTC Uses and Benefits



Contact center opportunities rated even higher than video communications when ranking WebRTC importance as a collaboration and communications platform: 90% of respondents believe that "WebRTC has the potential to improve contact center service levels because all a customer needs is a browser."

Of those who are using collaboration technologies today, fewer than 40% expect that WebRTC can substitute for traditional voice or VoIP; rather, 85% believed that SIP-based communications are complementary with WebRTC. Only 42% of respondents were looking for WebRTC to save their organization money on real time communications.

Figure 4: User Opinions on Communications and Collaboration Platforms

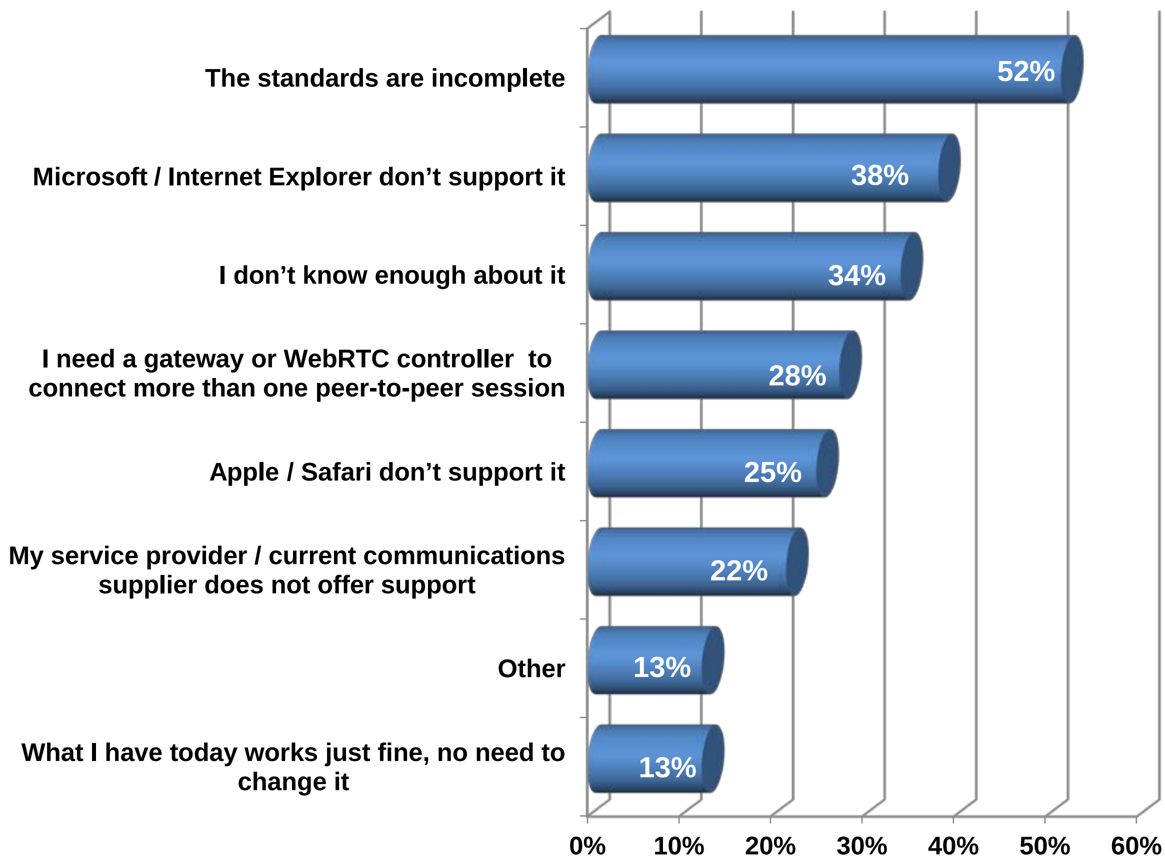


Please indicate the extent to which you agree or disagree with each of these statements.

The Problem With WebRTC Is …

The biggest concern among respondents about WebRTC is that standards are not yet complete, with 52% identifying this as an issue to overcome. Second to incomplete standards is the fact that Microsoft does not support WebRTC; however, as discussed below the lack of Microsoft support is not insurmountable. Despite the fact that survey respondents were pre-qualified as having a basic understanding about WebRTC, 34% said they don't know enough about it, so this is also a potential roadblock to deployments.

Figure 5: Issues with WebRTC



What is *not* a problem among respondents is the perceived support among infrastructure suppliers: only 22% noted that their service provider or current communications system supplier does not offer WebRTC support. The need to use a gateway or controller when connecting more than one peer-to-peer session was also not seen as an huge impediment, with only 28% identifying this as a problem. Interestingly, fewer than 40% expect to use a cloud-based WebRTC gateway service, suggesting that most who will deploy intend to use a premise-based gateway solution.

Is Limited Browser Support or Limited Mobile OS Support a WebRTC Barrier?

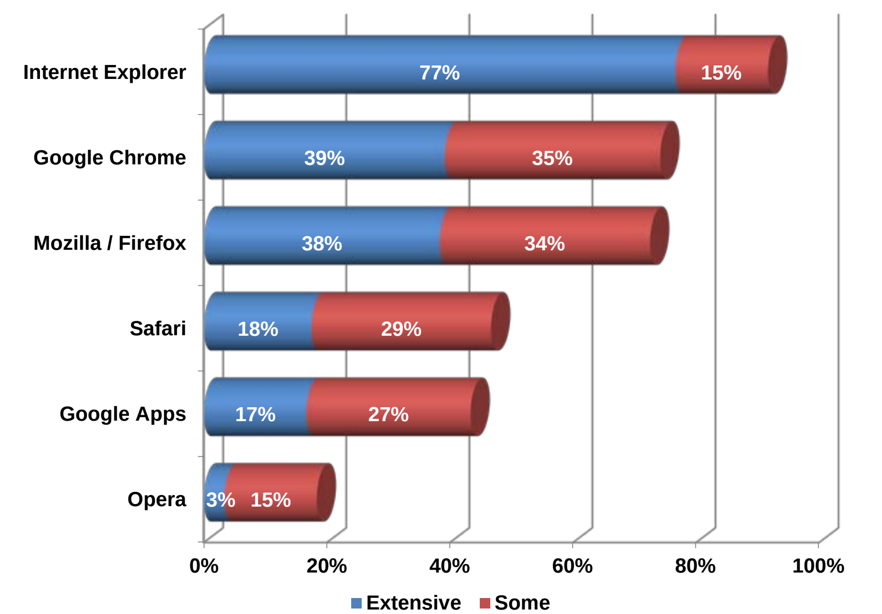
Much ado has been made about the fact that WebRTC is not currently supported by Microsoft's Internet Explorer or Apple's Safari web browsers: 38% of respondents consider that today's lack of support is a barrier to using WebRTC.

While Internet Explorer is still the prime choice (supported by 98% of users), other browsers are also being used and supported by about three-quarters of responding organizations. For example, 74% of companies support Google Chrome and more than 73% support Mozilla Firefox—two browser options that sustain WebRTC applications today. The lack of support by Apple's Safari seems a lesser concern as a WebRTC barrier to entry, with only about 47% of responding IT organizations supporting Safari browsers. When looking at the WebRTC support results for those who also support Chrome or Firefox in addition to IE vs. those who only support IT, the demographics were not materially different.

One important note about the response to Microsoft's WebRTC support plans: nearly 40% of those asked were unwilling to offer an opinion on Microsoft's support, reinforcing the need for Microsoft to be clearer on their strategy.

Most companies surveyed use a Microsoft Windows, PC-based infrastructure for employees (evidenced by their support for Internet Explorer), so they can control browser support for B2B applications should IE support WebRTC, or by supporting Chrome and Firefox internally. However, the lack of Apple Safari support may come back to bite businesses who need WebRTC for B2C (consumer) communications because Safari users may not otherwise have access to WebRTC apps since the Safari web browser is commonly used by Apple computers. And when it comes to mobile web browsers, the lack of Safari support on mobile devices can be an even greater barrier, given the dominance of Safari as the leading mobile web browser platform.

Figure 6: Browser Usage



To what extent does your company use and support each of these browsers?

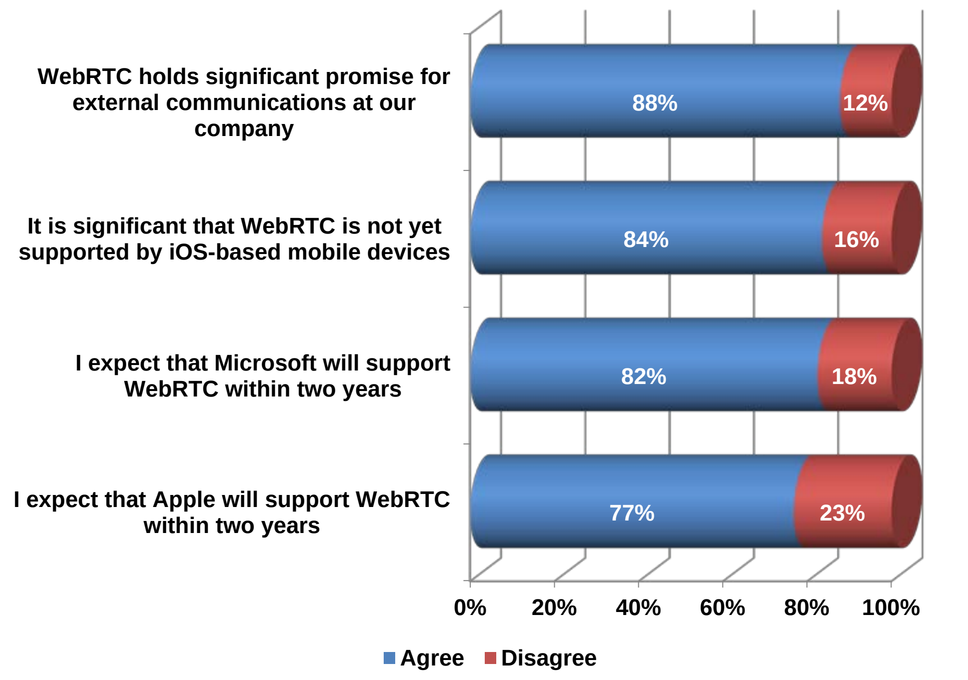
Who Should Support WebRTC?

Microsoft and Apple do not support WebRTC today, but more than three-quarters of respondents with an opinion on the topic believed these both companies would support WebRTC within the next two years: 82% expect Microsoft to support it, while 77% believe Apple will do so.

In a tip of the hat to Google's Android operating systems, 76% consider it significant that Android-based operating systems support WebRTC, while 84% also rate it significant that Apple mobile devices do not offer WebRTC support.

Given that 97% of all companies surveyed offer some level of BYOD support, and that 88% expect WebRTC to be used for external communications, pressure may build for Apple iOS to support WebRTC on its mobile device operating system: 77% agree that WebRTC support for BYOD apps is important to their company or organization.

Figure 7: Who Should Support WebRTC?



Session Border Control (SBC) support for WebRTC is also important: just over 60% of those surveyed expect their SBC to support WebRTC applications. However, answers were less conclusive about the need for separate WebRTC gateways used to provide interoperability between WebRTC and other applications platforms: 44% agreed that a separate gateway was ok, while 26% were not comfortable with a separate gateway and 30% were undecided. It's likely that many of the respondents are still forming an opinion on how best to deploy a WebRTC solution. The technology is still new so many respondents are only beginning to understand the platform choices and technology trade-offs between separate gateways and SBC-based WebRTC support.

Summary

WebRTC awareness is strongest among those who are already using unified communications and collaboration tools, and with 56% of respondents using or planning to use WebRTC-enabled applications, demand is surprisingly high among this group. Mid-tier business users are less likely to be enthusiastic about WebRTC; however, this is not surprising given that large enterprise usually have more IT staff to support while the smaller business is often more willing to try new technologies. Video communications in general and external video communications in particular represent the largest opportunity for using WebRTC, with contact center applications coming in as a close second to video. BYOD apps that use WebRTC are also important, with 77% of responses rating these as important.

Although respondents are somewhat concerned about the need for Microsoft support, half of respondents are optimistic that Microsoft will eventually support WebRTC, and many have alternatives by using other browser and mobile OS options, most notably those supported by Google. The bigger issue to overcome is the need for complete WebRTC standards, suggesting that: a) survey respondents have a realistic understanding of standards progress, and b) industry leaders need to work harder to complete standards that can be implemented.

Infrastructure suppliers who provide session border controller functionality should note that respondents expect SBCs to provide WebRTC support. Gateway support to provide interoperability and multi-participant sessions is an acceptable alternative for most, and most of these features are planned as a premise-based solution rather than as a cloud-based federation.

WebRTC stakeholders still need to work on educating the public about WebRTC. While the survey answers suggested that most who responded to the survey do have some understanding about WebRTC, some IT professionals have little or no knowledge on the subject. In particular, UC solution providers need to communicate their WebRTC strategies clearly; 45% of those who have a UC solution in place today did not know if their solutions provider had plans to support WebRTC, and 30% of those who have deployed SBCs did not know if their SBC provider would offer WebRTC support.

All in all, WebRTC is riding a wave of optimism: 88% of those surveyed believe it holds significant promise as platform for external communications.

About Larry Hettick

Larry is the Editorial Director and a Senior Research Fellow at Webtorials. A thirty-year telecom veteran who has managed products for service providers and infrastructure supplies, he has provided industry analysis focusing on Unified Communications for the last 15 years. Before joining Webtorials, he spent a decade working with Current Analysis, where he remains a contributing analyst. Hettick also authors Network World's bi-weekly VoIP and Convergence Newsletter.

About the Webtorials® Editorial/Analyst Division

The Webtorials® Editorial/Analyst Division, a joint venture of industry veterans Steven Taylor and Jim Metzler, is devoted to performing in-depth analysis and research in focused areas such as Metro Ethernet and MPLS, as well as in areas that cross the traditional functional boundaries of IT, such as Unified Communications and Application Delivery. The Editorial/Analyst Division's focus is on providing actionable insight through custom research with a forward looking viewpoint. Through reports that examine industry dynamics from both a demand and a supply perspective, the firm educates the marketplace both on emerging trends and the role that IT products, services and processes play in responding to those trends.

For more information and for additional Webtorials® Editorial/Analyst Division products, please contact Jim Metzler at jim@webtorials.com or Steven Taylor at taylor@webtorials.com.

The primary author of this study was Larry Hettick.

Published by Webtorials Editorial/Analyst Division
www.Webtorials.com

Division Cofounders:
Jim Metzler *jim@webtorials.com*
Steven Taylor *taylor@webtorials.com*

Professional Opinions Disclaimer
All information presented and opinions expressed in this publication represent the current opinions of the author(s) based on professional judgment and best available information at the time of the presentation. Consequently, the information is subject to change, and no liability for advice presented is assumed. Ultimate responsibility for choice of appropriate solutions remains with the reader.

For editorial and sponsorship information, contact Jim Metzler or Steven Taylor. The Webtorials Editorial/Analyst Division is an analyst and consulting joint venture of Steven Taylor and Jim Metzler.

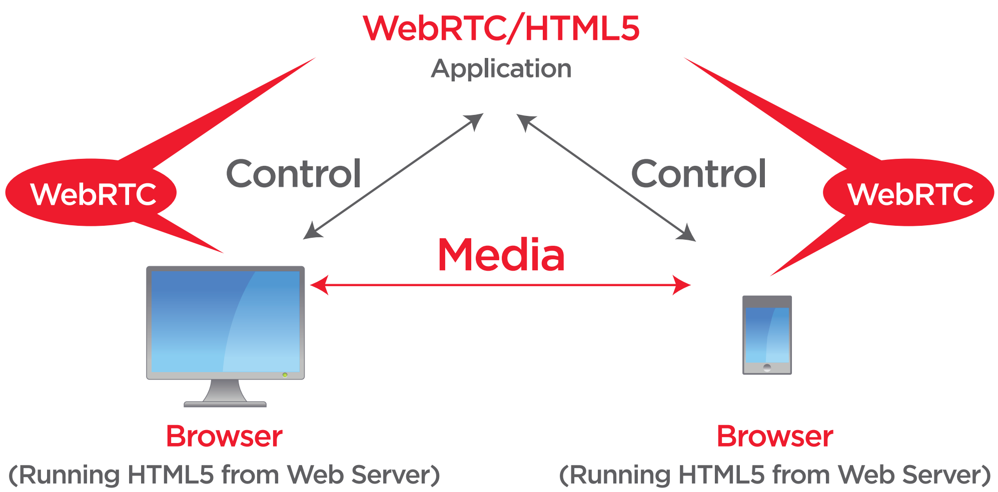
The

FUTURE is WebRTC

Real-time communications through a Web browser could disrupt telecom markets soon. But it also offers plenty of promise for innovators.

By (left to right)
John Yoakum, *Consulting Engineer*;
Harvey Waxman, *formerly with Avaya*;
and Alan Johnston, *Distinguished Engineer, External Standards, (All Avaya)*

For a game-changing platform, WebRTC's advantages can feel subtle. Starting this year, WebRTC—the RTC stands for real-time communications—builds interactive high-fidelity voice, video, and data interchange into standard compliant Web browsers. It does this by handling all of the grunt work necessary to enable the fast and easy creation of real-time communications-enabled apps. >>



WebRTC makes it incredibly easy for businesses to add one-click access for voice, video, chat, and more to their websites.

For instance, WebRTC makes it easy for developers to manage, blend, and use multiple multimedia channels, including microphone audio, camera video, screen grabs from your app or media player, etc. WebRTC also ensures that the audio and video quality remains good by adapting to network conditions. In addition, users running WebRTC can "talk" to each other without the media going through a hosted or cloud server first, as the media can flow directly between browsers.

While many of these things are accomplished today via current VoIP technologies, it takes more development effort and very specific expertise. If you are a developer building for four different Web browsers on three different operating systems, that's like creating 12 different applications. Not with WebRTC, which lets you write once and run everywhere with a common, yet application-specific user experience. As an end user, no downloads of third-party plug-ins such as Adobe Flash, Apple QuickTime, or Microsoft Silverlight are needed. In addition, all media flows and data are encrypted end-to-end.

Possibilities Are Endless

"So what?" you, the end user, might say. I already have Web-based video sharing, online games, and video and audio conferencing applications. True. But to create those applications needed millions of developer-hours. WebRTC, in conjunction with HTML5, makes it MUCH easier and faster to create communications-enabled apps. How much easier?

"When you think about that, [with] just a few JavaScript APIs and as little as 15 lines of JavaScript code in an HTML page, you can create a simple one-to-one videoconferencing solution," Hugh Finnan, a director of product management for Google, said at the inaugural WebRTC Conference in November 2012. "This has the potential to be as important to the Web as HTML was in the beginning."

Any webpage can easily be enhanced to include interactive media, while mobile developers can use WebRTC for apps if that is desired. Just as app stores enabled millions of developers to create and market their own apps, the supremely democratic WebRTC is expected to unleash the creativity of millions of Web developers embracing real-time multimedia interactions for the first time.

The possibilities are endless. One can envision an application that lets musicians play together in real-time via their browsers in high fidelity. Or an enterprise could extend an existing Web-based document repository to allow for direct voice or video interactions between users. That's because WebRTC both enables new ways to create rich, yet simple ways to drive user interactions AND new ways to extend the functionality of existing systems. For example, WebRTC makes it incredibly easy for businesses to add one-click access for voice, video, chat, and more to their websites.

The Challenges

There are several things that could slow down WebRTC's progress. As of this writing, the major browser makers had not yet settled on a standard video codec. As an emerging standard, WebRTC is obviously very young. A lot can happen. It could fragment into several not-always-interoperable flavors, as some VoIP technologies have done. Enterprises may find it hard to let fully encrypted WebRTC media cross the edge of their networks until new technologies assure such flows pose little threat. In addition, regulatory requirements—especially in business—may provide challenges to adoption. One of the most interesting characteristics of WebRTC is that communications industry players, including carriers, will likely have far less control over its use and adoption than they historically have enjoyed.

Today, many enterprises are actively engaged in deploying SIP-based solutions. This infrastructure is not going to go away anytime soon, and in fact is likely to grow. It is why the standards bodies that developed WebRTC have taken great care to enable several models of interaction between WebRTC applications and SIP environments. While many WebRTC applications will not require any SIP infrastructure, when interactions are needed between WebRTC users and enterprise SIP users, it's good to know that these two worlds have been designed to co-exist.

Avaya has already welcomed WebRTC as an opportunity. Avaya has key members of the standards bodies creating the WebRTC standard. We also have a number of R&D staff working on numerous projects and additional prototypes under development in our Avaya Labs. All the opportunities that are suddenly available to Web developers are readily available to Avaya. We are well-positioned to create innovative uses of WebRTC while also interworking it with SIP systems to help solve enterprise-specific WebRTC concerns. ●

John Yoakum *is a Consulting Engineer and champion of emerging innovation and disruptive technologies at Avaya. He has distinguished himself as an innovator and visionary with numerous patents, publications, presentations, and standards contributions over the course of his professional career at Motorola, Nortel, and Avaya.*

Harvey Waxman *formerly with Avaya started his career with Bell Laboratories and has over 25 years of experience.*

Alan Johnston *is a Distinguished Engineer at Avaya and a contributing author of more than a dozen Internet Engineering Task Force (IETF) standard specifications, including the SIP specification. Besides serving as an adjunct instructor at Washington University in St Louis, Alan has authored several best-selling technical books, including* WebRTC: APIs and RTCWEB Protocols of the HTML5 Real-Time Web.

Although WebRTC is generating a very high level of interest and excitement among enterprises and consumers, it will be operating in an environment that has already adopted, and continues to adopt, SIP in a big way, both in the enterprise and service provider market segments. In order for WebRTC to succeed, it cannot function as a separate communication island, but rather must interwork seamlessly with existing SIP-based networks to support real-time voice, video and data. Two important factors IT departments need to consider for this are: security and interworking.

Enterprises cannot implement new products or services that make them vulnerable to cyber attackers, so they need to be judicious about network security when WebRTC browsers start communicating with their networks. Opening such an application presents the possibility of the threat of a malicious web application's taking over a user's browser and directly communicating with a contact center – a scenario IT managers find it critical to avoid.

Consider a traditional SIP-based contact center that now has to support WebRTC clients. By hijacking unsuspecting WebRTC clients, cyber attacks can cause excessive bandwidth and CPU utilization at the contact center, filling up of storage disks with useless data and causing damage to system and IT network configuration. This can result in service revenue loss, network and disk storage repair and rebuild costs, and a host of other negative consequences for the contact center.

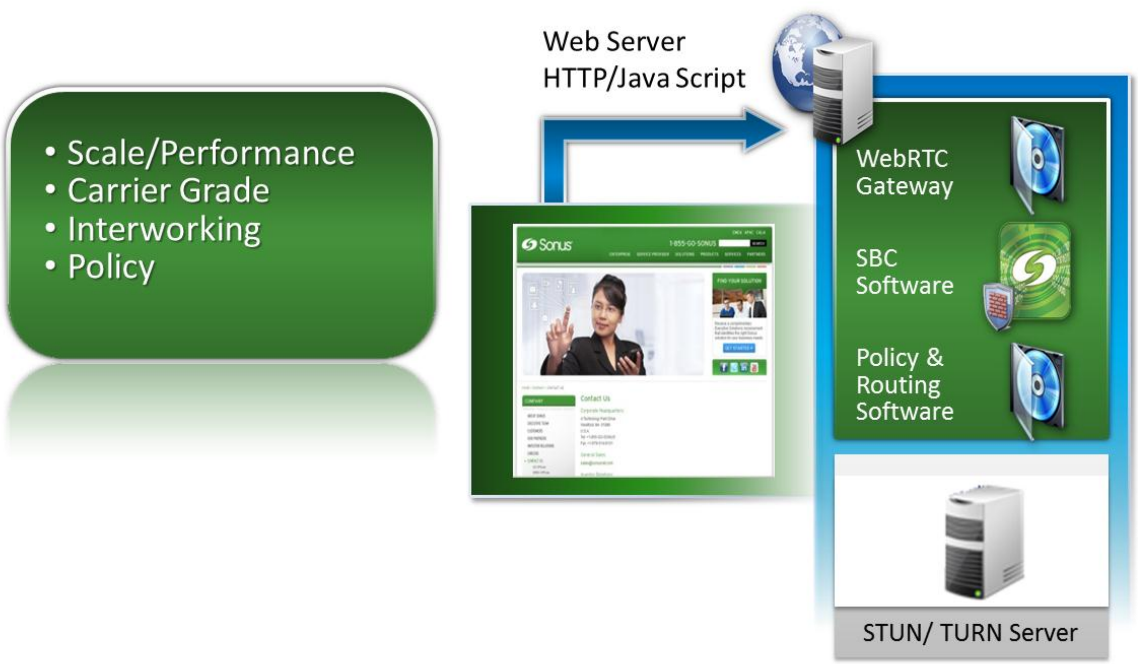
A security gateway, such as a session border controller deployed at the edge of the enterprise network, can provide the functionality necessary to protect the enterprise from WebRTC-initiated cyber attacks. By monitoring traffic, the SBC can recognize threats to the network and shut them out.

At the same time, interworking is another important part of a comprehensive WebRTC game plan. Every enterprise – and consumer – is at a different stage in adopting the latest and greatest technology, and playing nice with old and new is necessary for WebRTC to be adopted widely.

Interworking is required for both signaling and media (voice/video) communication. Media interworking presents a greater challenge because of the plethora of standards (codecs, protocols) for video and voice communications. H.323, SIP and WebRTC video communication islands all must interwork for widespread adoption of UC video conferencing to happen. At the same time, video transcoding, trans-rating and tran-sizing must be done in real time, while policy control effectively manages the quality of service and network bandwidth for sessions, since UC video communication can involve large volume of data.

Sonus WebRTC Solution

Sonus offers a comprehensive WebRTC solution for both enterprises and service providers that includes a WebRTC Gateway, a portfolio of SBC products, and the field proven Routing and Policy Engine.



Sonus Solution Elements

- **WebRTC Gateway**
  - Communicates session details to browser via JavaScript Client
  - Communicates with SBC via SIP when WebRTC to SIP required
- **SBC Software**
  - Media services (OPUS/G711, VP8)
  - ICE (STUN/TURN) for non-WebRTC endpoints
  - Interop issues: SRTP/RTP, SIP etc.
  - Plus SBC feature set
- **Policy & Routing Software**
  - Policy & Routing, inclusive of internet information

Sonus Solution Benefits

Sonus WebRTC Gateway	Sonus SBC	Sonus Policy & Routing Engine
▪ Carrier grade WebRTC implementation	▪ Secure way to communicate between WebRTC and non-WebRTC users ▪ PSTN ▪ IP Telephony ▪ SIP ▪ Unified Communications ▪ Routing, Security, Transcoding and Interworking	▪ Intelligent policy based routing will offer increased flexibility and more granular service options

Sonus WebRTC Collaboration Lab

The WebRTC Collaboration Lab in the Sonus Richardson, Texas office showcases Sonus WebRTC Solution and Applications, and offers a unique lab environment for prototyping and building proof of concept applications. This helps reduce development time and costs, and speeds up time to market for WebRTC applications. It also helps customers evaluate the performance and capabilities of the Sonus Solution prior to actual deployment in their networks.

Please contact a Sonus Sales Representative (www.sonus.net) for more information.

Sonus North American Headquarters

4 Technology Park Drive
Westford, MA 01886
U.S.A.
Tel: +1-855-GO-SONUS

Sonus APAC Headquarters

1 Fullerton Road #02-01
One Fullerton
Singapore 049213
Singapore
Tel: +65-68325589

Sonus EMEA Headquarters

Edison House
Edison Road
Dorcan, Swindon
Wiltshire
SN3 5JX
Tel: +44-0-1793-601-400

To learn more, call Sonus at 855-GO-SONUS
or visit us online at www.sonus.net

Microsoft Partner
Gold Communications

Voice
Unified Communications
Business Productivity Solutions
Midmarket Solution Provider

GENBAND

Leveraging the Complete Potential of WebRTC

The Impact of WebRTC

The development of WebRTC is expected to be a disruptive force in the telecommunications industry and has the potential to be as significant as SIP was a decade ago. Much as browsers revolutionized access to email, WebRTC promises to do the same for voice, video and collaboration. WebRTC delivers Real Time Communications (RTC) natively in a web browser. Practically speaking, any user with a WebRTC-compliant browser can access two-way voice, messaging, video and video conferencing services without having to download a client or plug-in.

GENBAND, a leading developer of multimedia and cloud communications solutions, is committed to better enabling WebRTC-based communications. GENBAND solutions include:

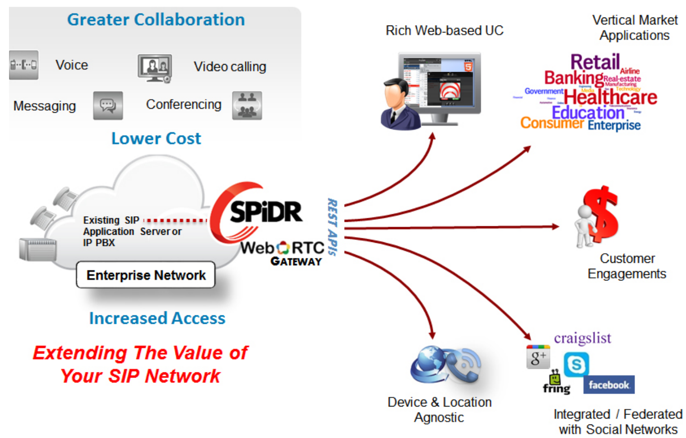
- SPiDR™ WebRTC gateway — provides access to collaboration solutions using RESTFul APIs (designed for web developers) reducing the barriers to communication-enabling applications.
- SMART OFFICE Premium Web "Client" — leverages the power of WebRTC to deliver full-featured Unified Communications and collaboration services with just a web browser.

SPiDR WebRTC Gateway

SPiDR provides a bridge between traditional SIP-based communication networks and the web. GENBAND made it easy and cost effective for enterprises with existing SIP networks to elegantly add SPiDR to their network and securely enable access to new web-based services, without disrupting existing services or solutions.

SPiDR sits at the edge of the enterprise network and provides open, web-centric Rest APIs that allow application developers to securely expose the organization's rich communications services—including voice, video, corporate address book, call history, instant messaging and collaboration. SPiDR's media interworking and application interworking engines make it easy to connect the web to existing IP clients and applications. It also normalizes conversations and media between web users and traditional SIP devices – there is no need to start over or lose existing investments. Additionally, SPiDR simplifies the steps and the investment required to deliver an entirely enhanced, web-based, user experience.

Unlocking New Business



While SPiDR has the ability to deliver a full suite of communications services into any web-centric experience, it can just as easily be used to create a simple communications path for customers connecting to the organization. With SPiDR, every connected device is a portal on the enterprise network and can leverage the reach and capabilities of the web.

SPiDR simplifies deployment of WebRTC-based services including:

- Communication-enabling existing applications such as contact centers, web pages, web-based ERP and CRM solutions, social networks and e-commerce.
- Delivery of feature-rich, secure unified communication services without installing a client – using just a web browser. Ideal for today's BYOD initiatives.

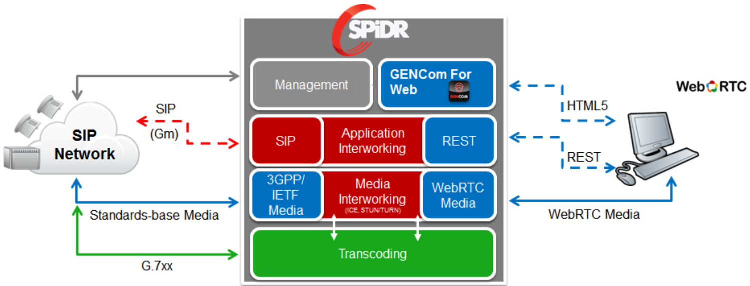
The Real Power of RESTful APIs

GENBAND embraced the power of RESTful APIs to help enterprises create more compelling experiences for real time communications. Far from a new idea, GENBAND is emulating the approach used by successful web applications, such as Twilio, Twitter and Facebook and consciously made its REST APIs simple so developers do not need telecom domain knowledge. RESTful APIs make communication enablement as simple as cutting and pasting a few lines of code into web applications. APIs are included to provide:

- User identity, association, authentication/authorization
- Call set-up and management
- Access to user data such as call logs, address books
- Messaging
- Presence (publish, subscribe, fetch, permit, deny)

By embracing standard web development tools like HTML5 and RESTful APIs, GENBAND makes it far easier for a traditional web developer to incorporate real time services without having to learn SIP. Some have suggested that simply adding SIP over web sockets to an SBC offers a similar set of application tools but in practice this model dramatically shrinks the pool of potential developers. Fewer developers means higher costs and fewer innovations.

WebRTC Gateway Functional Layers



GENBAND is actively fostering innovation with a new developer web portal that includes generic APIs and specific use cases that expose more services and service enablers. Independent developers and entrepreneurial organizations are already creating an increasing array of compelling use cases. GENBAND is committed to staying ahead of the curve and moving WebRTC to the next level.

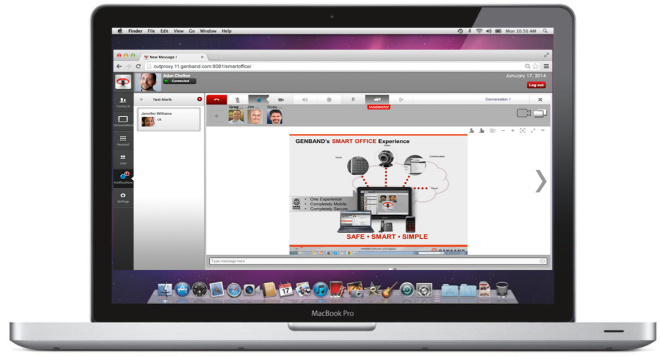
SMART OFFICE: Improving the User Experience with WebRTC

GENBAND's SMART OFFICE is a powerful set of Unified Communications (UC) and collaboration solutions that blurs the line between being in the office and the office being wherever you are. SMART OFFICE services are already enabled on IP phones, PCs, Macs and smartphones; WebRTC makes it even easier to make a user's business identity – their phone number – device independent. Users can dynamically access their communications from any WebRTC enabled device and engage in voice calls, messaging, video conferencing or screen sharing from the location and device of their choice.

SMART OFFICE is the first commercially available UC solution that includes a WebRTC platform. It was designed based on today's reality that communications tools need to be developed for a BYOD and mobile workforce. It is ideal for businesses, governments, health care organizations and educational institutions.

SMART OFFICE can be deployed in existing PBXs or greenfield environments. It is powered by GENBAND's EXPERiUS™ Application Server and SPiDR™ WebRTC gateway and can be deployed in a centralized datacenter to support multiple sites, as a cloud-based service or a combination of both.

SMART OFFICE Video Conference



Secure Access

One of the most important considerations for enterprise providers is to protect the integrity of their networks. GENBAND's SPiDR gateway provides interworking services that create a natural demarcation point between the enterprise network and the service providers. Additionally, GENBAND's QUANTiX family of Session Border Controllers (SBCs) adds a second layer of security while supporting a wide range of interoperability tools to integrate IP PBXs and other network elements. QUANTiX's scale and ability to be centrally managed with the SPiDR gateway assures that security is never compromised.

SAFE, SMART, ROBUST

GENBAND's SPiDR gateway is one of the most cost effective tools an enterprise can use to communication-enable their business tools. It extends your reach to more customers as you extend your employee's access to each other and key business tools. And SPiDR is more than just a tool for developers, it's also the foundation for a new generation of GENBAND's UC services; GENBAND's SMART OFFICE. We are committed to delivering the experiences enterprises need, from turnkey services to APIs to developer forums. Complete communications solutions from one market leading organization.

www.genband.com 1-866-GENBAND