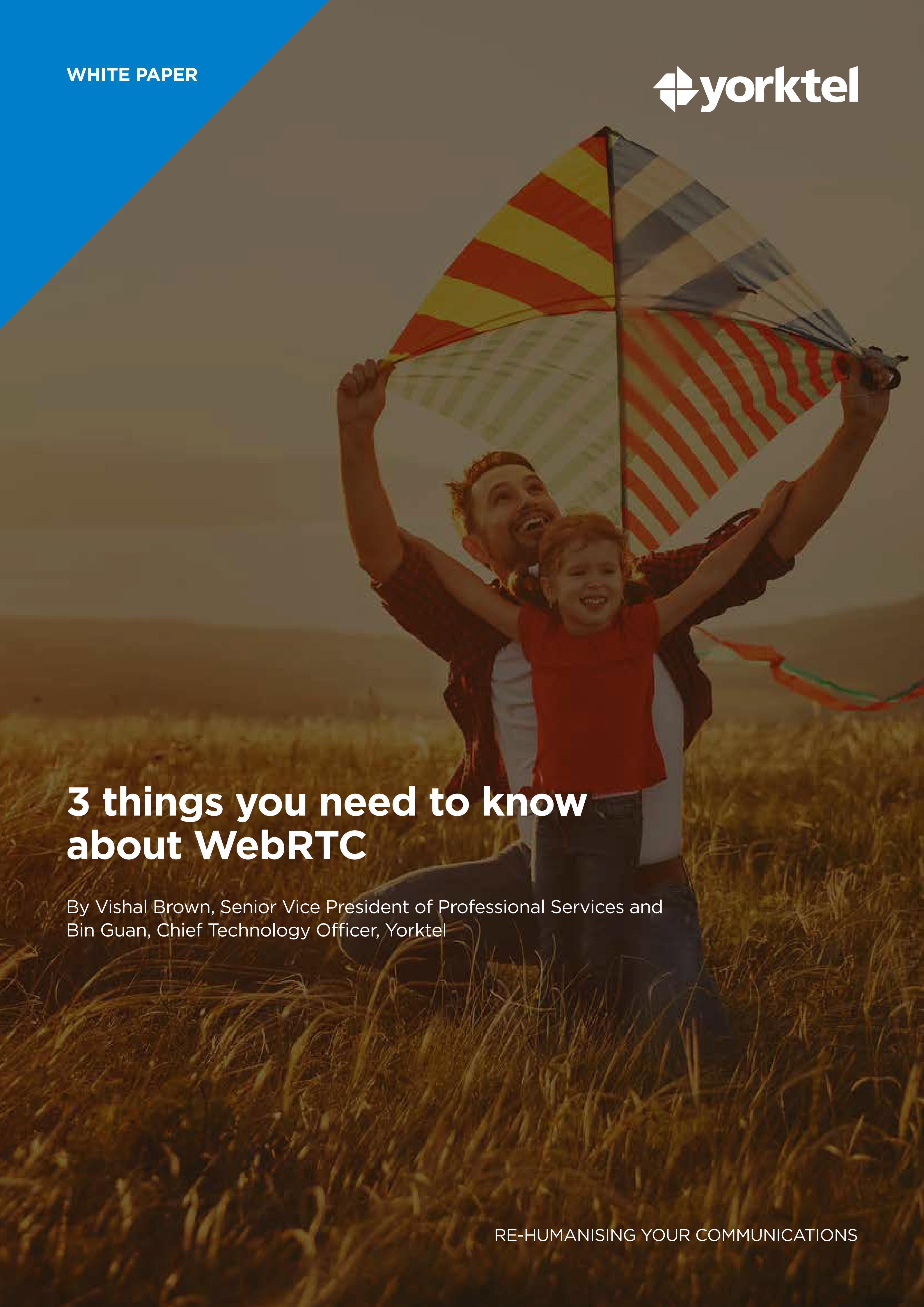

WHITE PAPER

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3 things you need to know about WebRTC

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RE-HUMANISING YOUR COMMUNICATIONS

WebRTC IS AN API (APPLICATION PROGRAMME INTERFACE) STANDARD THAT ENABLES **REAL-TIME VOICE, TEXT AND VIDEO COMMUNICATIONS CAPABILITIES**. IT IS ABOUT PUTTING REAL TIME CAPABILITIES INTO A STANDARD BROWSER WITHOUT THE NEED FOR DOWNLOADS, PLUGINS OR FLASH.

The video communications industry has had a major shakeup. Understanding WebRTC (real-time communication) is essential to understanding how this trend will affect your organisation.

As the video communications space continues to evolve, there is a term we're hearing more and more about: WebRTC. Similar to what happens with other new technologies or platforms, the marketing hype surrounding the term makes it difficult to discern what it really is and what are the benefits it offers. In this article, we'll explore three things enterprise executives and IT decision makers need to know about WebRTC, including what it is, what's keeping it from the mainstream, and what its impact could look like on the video communications space.

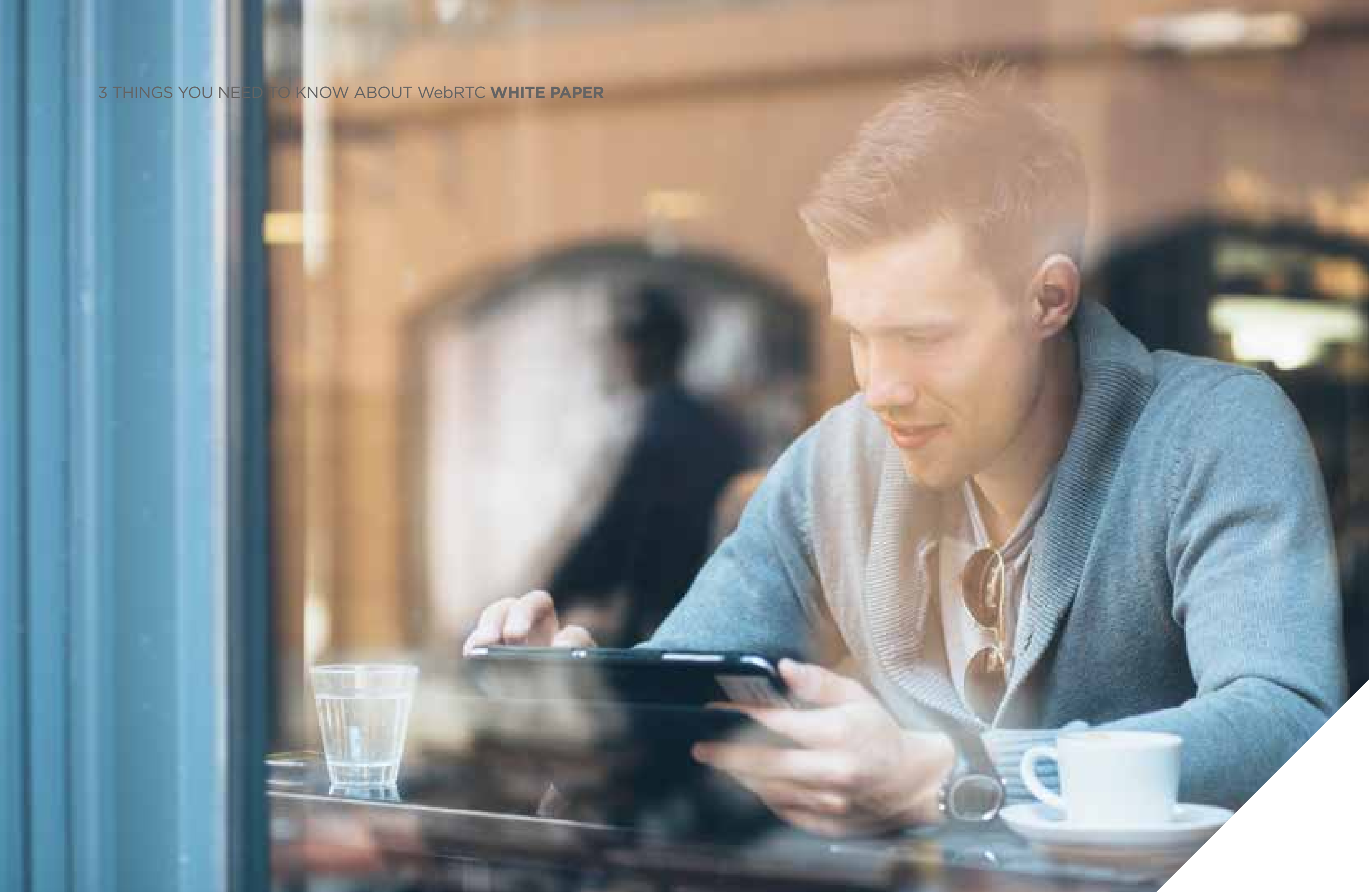
1. WHAT IS WebRTC (AND WHAT IS IT NOT)?

WebRTC is an API (application programme interface) standard that enables real-time voice, text and video communications capabilities. It is about putting real time capabilities into a standard browser without the need for downloads, plugins or Flash. We need to delve deeper into this interface standard to understand how it's different from well-known consumer and enterprise video platforms such as Skype, WebEx, or Cisco Jabber. Following are a few of WebRTC's distinguishing characteristics:

a. **It's browser based.** Unlike traditional apps that are installed locally on a computer or server, with WebRTC web applications have direct API access to the microphone and camera in a device using standard Javascript API calls. WebRTC therefore eliminates the need for a plugin or download, making it device, platform, and OS agnostic.

b. **It's inherently secure.** Unlike traditional security applications that often require users to "accept" and install updates manually, many of the leading browsers automate that step. So, the next time you launch your favourite browser, you're opening the latest version of the browser — including all the latest bug fixes — without any extra steps on your part. As such, WebRTC being browser based supports only SRTP (secure real-time transport protocol) by default, which uses encryption and authentication to minimise the risk of denial of service attacks. The main premise for this decision is the fact that a call is private at all times, not an option that someone should have to select.

Video managed services providers like Yorktel that have already invested in the development of private cloud infrastructures built upon a foundation of many leading manufacturers are in a good position to address this upcoming spike in the adoption of cloud services.



c. **It uses advanced audio and video codecs.** Despite the ever increasing availability of bandwidth, the growing trend of remote workforces combined with the demands of high definition video and audio communication exceeds many corporations' bandwidth capabilities, causing dropped packets and a choppy voice and video experience. In a WebRTC environment, advanced codecs enable more efficient use of video and audio packets, which results in clearer communication even in circumstances that would render traditional communication environments useless. Additionally, it supports variable bit rates, constant bit rates, and even narrowband communications.

d. **It promotes stronger session authentication.** Traditional video communications environments rely on third-party relay media servers to manage sessions in order to traverse firewalls. WebRTC establishes a peer-to-peer reliable session even through NAT's (Network Address Translation), therefore eliminating the need for third-party communications servers. This has traditionally been a problem for SIP-based VoIP systems.

2. WHAT'S SLOWING WebRTC FROM ACHIEVING MAINSTREAM ADOPTION?

Before we'll see any widespread use of WebRTC in the enterprise, there are a couple of obstacles that must be addressed. First, settling on the right standards has been a major factor impeding WebRTC's progress. For audio, the IETF (Internet Engineering Task Force) ratified the Opus codec in 2012, making it an MTI (mandatory-to-implement) codec for WebRTC. What's nice about that decision is that Opus is a free and open standard. When it comes to the video side of WebRTC, there are two standards vying for the MTI position: VP8 and H.264. VP8 is owned by Google and created by On2 Technologies as a successor to VP7. It's currently supported by Chrome, Firefox, and Opera browsers, and proponents of VP8 like the fact that while Google owns the codec, it has irrevocably released all the VP8 patents under a royalty-free public licence. The H.264 codec, on the other hand, has been around for 10 years, and it's been the industry standard for much of that time, giving it an implementation advantage over VP8, which has only been around for two years. Two strikes against the H.264 codec, however, are the fact that developers are required to pay a royalty fee to use it, and it uses more bandwidth than VP8. The same group that developed the H.264 codec, however, released a high efficiency video coding (HEVC) codec (which is also being called H.265), which provides double the data compression ratio compared with H.264.

One week before the IETF met to make a final decision on the MTI video codec, Cisco added another point in H.264's favour by announcing it would pay for all H.264 licensing. So, after several sessions and hearing multiple views on the VP8 vs. H.264 debate, the IETF's decision was: No Decision. As of press time, there's no definitive date that the IETF is committing to make its decision.

Another factor keeping WebRTC from moving forward are the disagreements among the major stakeholders, such as Apple, Cisco, Google and Microsoft. While Google is clearly in the VP8 camp, Apple is aligned with H.264. Although Microsoft hasn't said anything about WebRTC (which some speculate is due to the fact that WebRTC could potentially cannibalise some of Microsoft's products, such as Skype), the fact that its browser does not support VP8, speaks for itself.

Until these industry giants can get on the same page (which will probably be dictated by the IETF's eventual video codec decision), the industry is still largely in a wait-and-see mode. Once the standards are ratified, however, we'll start to see movement in this space and then real progress can start to occur.

As enterprises are more easily able to extend video conferencing to remote workers via a myriad of mobile platforms, the need for burstable bandwidth, computer processing, and storage will become in higher demand, which will further drive the adoption of true cloud services.

3. WHAT VIDEO COMMUNICATIONS WILL LOOK LIKE IN THE NEXT 18 MONTHS

Shortly after the governing body makes its decision, we'll start to see the big players mentioned earlier (i.e. Apple, Cisco Google, and Microsoft) making the necessary changes to their products to play nice with this video communications standard. At the same time all this is happening, we'll see other companies in the video communications space developing products such as media gateways designed specifically to enable legacy video communication infrastructures to accommodate WebRTC-based communications. Startup businesses and greenfield video communication projects will have an advantage in that they won't require investments in these third-party servers.

During this transition, it's also important to keep in mind two adjacent trends happening that will play an important role in the growth of WebRTC: remote workforces and BYOD. WorldatWork estimates that 16 million U.S.-based employees work at home at least one day a month, a number that increased almost 62% between 2005 and 2010 and continues to grow each year. Video communication has been a key enabler behind this trend. As WebRTC removes many of the current restrictions associated with video conferencing, work will more naturally become less about the 'where' and more about the 'what.' BYOD is important to note, too, because as more employees use their own personal smartphones, tablets, and laptops for work, it lessens the video conferencing capital expense burden on employers, which again contributes to the adoption of video becoming a first choice for real-time communications. In addition to the reduced expense, WebRTC addresses the security concerns many companies currently have, which is another plus in WebRTC's corner.

As enterprises are more easily able to extend video conferencing to remote workers via a myriad of mobile platforms, the need for burstable bandwidth, computer processing, and storage will become in higher demand, which will further drive the adoption of true cloud services. Video managed services providers like Yorktel that have already invested in the development of private cloud infrastructures built upon a foundation of many leading manufacturers (e.g. Avaya/Radvision, Cisco, Polycom) are in a good position to address this upcoming spike in the adoption of cloud services. While some managed services providers within the video communications space may be nervous about the shakeup happening within the market, Yorktel is excited to finally be on the verge of witnessing the democratisation of video. With any major disruptive technology, there's always a certain learning curve required to make the required business changes. But, make no mistake about it, those who embrace these inevitable changes and prepare ahead of time are setting themselves up for exciting times ahead.

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Logistics planning in Geneva, chat with CIO in Oslo, presentation in Milan & soared high into the sky.

All on the same day. Face to face.

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Yorktel is a leading global provider of video, cloud and managed services for some of the world's large enterprise, government and healthcare customers. Founded in 1985 and with offices throughout the US and EMEA, Yorktel enables customers to successfully integrate video into their operations – from video conferencing to streaming, video event production, to digital signage.

Our core expertise in video communications is in our roots, not an add-on service – making us a trusted partner not only to our clients, but also to our channel partners. Some of the world's largest enterprises, government agencies and healthcare institutions rely on us to take them through their visual collaboration journeys, from initial consult and design to integration and managed services.

The numerous industry awards and certifications, including the international standard for information security management, ISO 27001 are testament to our commitment to excellence. Our goal is to enable and facilitate collaboration – person to person regardless of device or location.

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