Transition to Cloud Video Conferencing

How IMS positions conferencing service providers (CSPs) to address market trends over alternative approaches

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Overview

Video conferencing is finally coming of age. Cisco forecasts that business video conferencing will grow six-fold over the forecast period between 2011 and 2016. End users, with ubiquitous connectivity and increasing choice across a growing number of mobile devices and video calling and conferencing clients, are ready to push the limits of what's possible. This trend is expected to boost demand for cloud video conferencing.

A variety of video conferencing solution alternatives exist to fulfill this growing demand. Historically, business video conferencing infrastructure was largely based on multipoint control unit (MCU) architectures. These products were designed as customer premise equipment for the meeting room and executive desktop. Video conferencing is increasingly one of many services packaged in modern Enterprise Unified Communication (UC) offerings, while Over-The-Top (OTT) solutions offer consumer-oriented video calling and conferencing on a best-efforts basis. More recently, a new approach based on scalable video coding (SVC) has emerged that pushes more of the processing from the network center to the client.

This white paper is about video conferencing solutions based on the IP Multimedia Subsystem (IMS) and its benefits—a solution explicitly developed to address the needs of telecommunications service providers offering multiple services on a large scale to many enterprise customers and consumers. IMS also delivers the interworking between the "islands" of video conferencing capabilities across today's MCU, Enterprise UC, or OTT offerings. By service providers offering cloud video conferencing services based on IMS, their enterprise customers can reduce costs, while improving ubiquitous video collaboration with third-party participants and remote workers.

This paper reviews the recent growth in hosted audio conferencing, trends in hosted video conferencing, compares video conferencing deployment architectures, encoding options, and video distribution technologies, and describes how the IMS architecture is the best approach to position CSPs to generate revenues through cloud-based multimedia conferencing service offerings to Enterprise customers and consumers.

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Hosted Conferencing Trends

According to Wainhouse Research, worldwide audio conferencing is forecasted to grow steadily at an 11 percent compound annual growth rate (CAGR), as measured by minutes of use (MOU) and shown in Figure 1. Comparing geographies, European and U.S. markets are relatively mature, with many enterprise users consuming audio conferencing services, whereas APAC markets are still evolving and expected to grow faster with respect to MOU and equipment sales.

Modern CSPs offer conferencing collaboration services using audio, video and web technology. While audio conferencing remains the dominant revenue generator for CSPs, and will continue to grow over the next five years, revenues from video conferencing services are growing much faster, driven by user acceptance of video calling and conferencing, improved mobile broadband, cost reductions, and the ability to participate in video sessions outside of the enterprise video conferencing meeting room. According to Cisco, video conferencing traffic will expand at a 48 percent CAGR, or by six times, between 2011 and 2016.[1] This growth is illustrated in Figure 2, which shows a forecast of business users employing three different video conferencing usage models.

Approaches to delivering enterprise video conferencing

Enterprises have a growing list of approaches to fulfill end-user demand for video conferencing services, such as traditional MCU video conferencing products, layered video switching products using SVC (Scalable Video Codec extension to the standard H.264 video codec specification), a video conferencing component added to their existing UC solution or even OTT consumer-class video conferencing services.

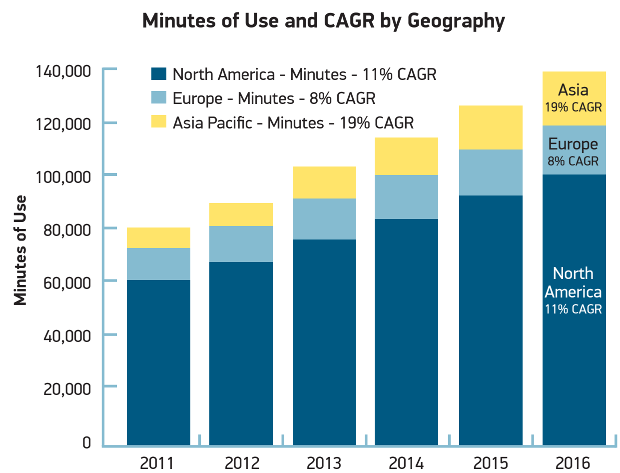

Figure 1. *Audio conferencing minutes of use and forecasts by geography (Source: Wainhouse Research)*

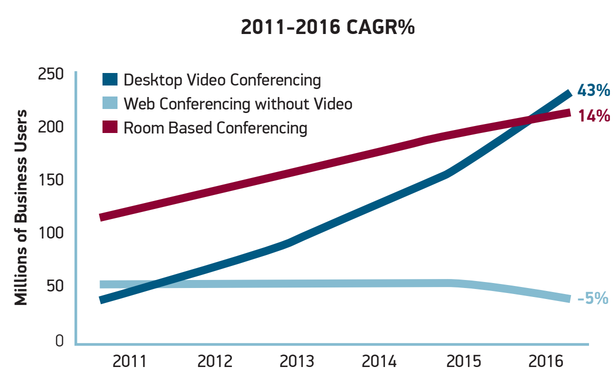

Figure 2. *Business desktop video conferencing (including mobile and tablets) is expected to grow significantly. (Source: Cisco)*

However, advances in IMS infrastructures are allowing CSPs to deliver business-quality video conferencing services using cloud-based, multi-tenant infrastructure. IMS video conferencing is optimized for service provider needs, while providing technical and economic benefits over alternative video conferencing solution approaches (and at the same time integrating with those existing solutions), as will be described in this whitepaper.

Meeting New Market Demands

End users are waiting for a video conferencing experience that is ubiquitous, cost effective and can be delivered with high quality and a guaranteed service level. At the same time, cloud computing is gaining momentum as a substitute for traditional IT infrastructure:

- *Improved Unified Communications (UC):* End users want their UC solutions to extend beyond the Enterprise to interconnect customers, partners and suppliers, in order to make calls more collaborative and productive. Today's "islands" of video conferencing capabilities need to become better connected (e.g. more readily enable inter-domain and cross-vendor connectivity), seamless, ubiquitous and easier to use.
- *Over-The-Top (OTT) Integration:* OTT voice and video calling began in the consumer domain, but the rise of the influence of Consumerization on enterprise technology has created demand from enterprise users to ask IT to support OTT offerings along with their enterprise UC systems.

Cloud Computing—New Approaches for Addressing Market Demands

Leveraging an on-demand model for providing computing resources, cloud computing offers service providers more flexibility in addressing video conference market demands. In "Have it your way" fashion, CSPs can choose from different levels of service, ranging from video media processing resources in the cloud accessible and "rented" to service providers, to complete "video conferencing as a service" offers that are sold direct to enterprise customers and end users.[2]

For the enterprise customer, the key economic benefit is a flexible operating expense to rent video conferencing services on demand from the cloud, rather than purchasing and operating their own enterprise video conferencing equipment. In addition, cloud providers will manage interoperability complexity, which will make conferencing easier to use and increase video conferencing minutes of use.

While many services today are offered over the public cloud and delivered using a best efforts Internet, two-way conversational video calling and conferencing services will require high-performance IP networks with low latency. Hence, emerging cloud video conferencing services, as discussed in this whitepaper, will need to be delivered using private clouds engineered to deliver high bandwidth with low latency—a key consideration in video architecture design.

The Business Challenges for Service Providers

Service providers looking to offer hosted video conferencing currently face several critical business issues:

- *Expensive capital equipment:* Today's video conferencing solutions can cost hundreds of thousands of dollars to buy and just as much to manage and maintain.[3] CSPs need a solution architecture that can achieve scale economies for multiple users and services, as lowering cost per unit can influence demand elasticity through price reductions to the end user.
- *Missing features for hosted offerings:* Key capabilities critical to service providers, like multi-tenant and cloud-based delivery, are not getting addressed because many of today's solutions are designed and sold as enterprise premise equipment. Adaptations that enterprise vendors make to cater to service providers render the solution a round peg in a square hole.
- *Inadequate UC strategies:* Limited adoption of open APIs by vendors is creating interoperability issues.
- *Lack of integration:* With a disparate range of video conferencing technologies on the market today, users of OTT solutions, such as FaceTime or Skype, may not be able to join an MCU-hosted conferencing call.
- *Poor audio-only economics:* Video conferencing platforms are not cost effective in supporting audio only participants.

- *Service flexibility:* While operators want cost-effective, hosted video conferencing solutions, some also would like the same infrastructure to support both other conferencing services (audio and web) and other non-conferencing services, such as video messaging, multimedia targeted advertising or ringback tones.
- *Deployment flexibility:* Operators are also interested in cloud deployment models. The scope of cloud services could range from platform solutions like Media Processing as a Service (MPaaS) for conferencing application developers, to complete Conferencing as a Service (CaaS) offerings to the enterprise.

Hosted Audio Conferencing Architecture Trends

Adoption of VoIP Audio Conferencing and Next-Generation Architectures

Over the past decade, PSTN-based audio conferencing solutions were replaced with Voice Over IP (VoIP) and next-generation network architectures, resulting in a migration away from TDM circuits carried by E1/T1 lines to voice streams transferred as data packets over IP networks. Capital price per port for VoIP systems was significantly lower, partly because they have up to 10x port density compared to a TDM audio bridge of the past. For ongoing operational costs, next-generation VoIP conferencing architectures require less power, space and cooling. More importantly, VoIP platforms can connect directly with an IP carrier network using Session Initiation Protocol (SIP) trunking, at a fraction of the cost-per-minute, compared to legacy TDM trunking interconnection. An IP-based conferencing solution is also naturally suited to service any IP-based endpoint such as an IP Soft Phone on a PC, an IP phone or emerging Voice over LTE (VoLTE) devices.

Main Components: Integrated VoIP Conferencing Solution

The expanding number and variety of VoIP endpoints can connect directly to an IP-based conferencing solution, which includes application servers, IP media servers and media gateways, as shown in Figure 3 and described in the following:

- *Application Servers* (features and applications) host the conferencing call flow and service logic; provide database access and Interactive Voice Response (IVR) dialog logic; support billing and provisioning; and coordinate with Web clients for real-time conference control. An IP-based architecture, using the Session Initiation Protocol (SIP) for call control, delivers to service providers the business model flexibility that was inconceivable with traditional audio bridge equipment and TDM networks.

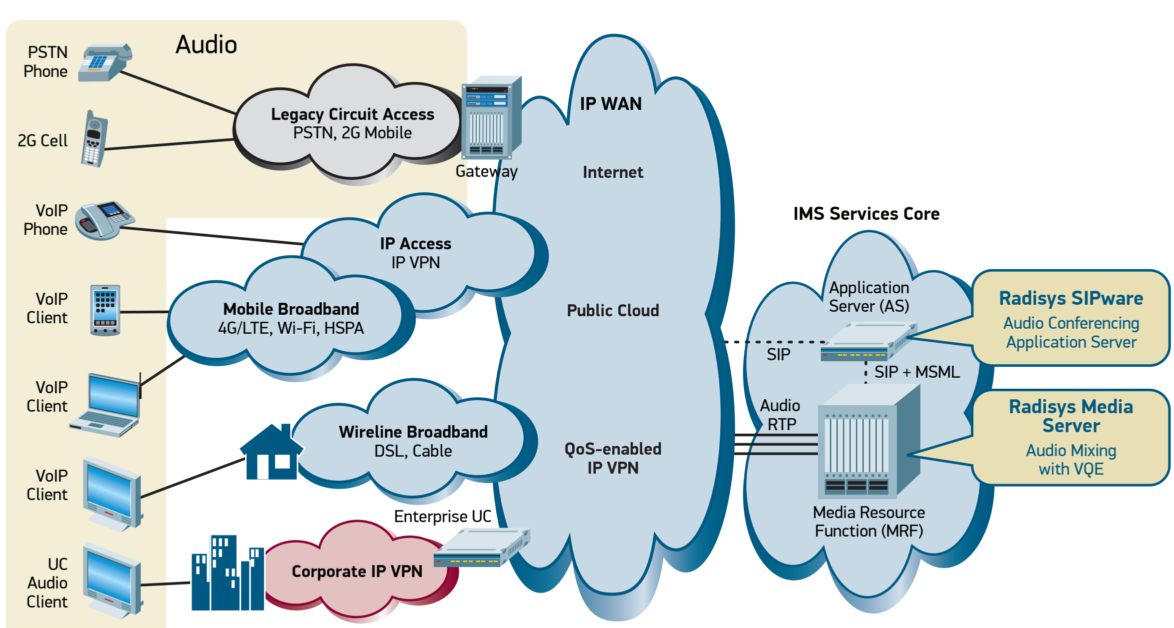

Figure 3. *IMS architecture delivering hosted VoIP audio conferencing with a media gateway for PSTN access*

- *IP Media Servers* (media packet processing) connect to an IP network and are controlled by the application server. They perform media processing functions such as media bridging, mixing and format transcoding, acoustic echo cancellation, noise reduction, announcements, IVR, recording and playback, automatic speech recognition (ASR), text to speech (TTS), fax handling and video processing. The role of the IP Media Server in IMS architecture is delivered by the IMS Media Resource Function (MRF).
- *Media Gateways* (line and trunk cards) are deployed at the border of circuit-based access networks, like the PSTN or 2G cellular networks, and the IP core. Many customers continue to consume audio conferencing services with circuit-based devices—for example, a 3G mobile phone uses circuit radio access to the network. So the primary function of a media gateway is to convert media circuits on one side into Real-time Transport Protocol (RTP) over IP media packet streams on the other side. Some CSPs might own and operate their own gateway equipment, connecting to a carrier's circuit network with TDM trunks, but increasingly, it is the large network carriers that host the gateway equipment, with the CSP interconnecting with the carrier using cost-efficient SIP trunking.

These architectures drove a major disruption in the infrastructure costs of the conferencing industry. Today, all leading conferencing service providers have adopted VoIP technology and distributed hosted architectures for the delivery of their audio services. CSPs that continue to offer hosted audio conferencing services using legacy TDM audio bridge technology are at a cost disadvantage compared to modern VoIP-equipped providers.

Notably, VoIP-based architectures also drove simplifications on how business users consumed audio conferencing services. Previously, audio conferencing services required the assistance of a human operator to schedule and orchestrate the conference. Today, this is called Event Conferencing. Earlier audio bridge equipment had to support the scheduling and reservation of audio bridge resources as required by audio conferences.

Radisys is the Industry's #1 Choice for IP-based Conferencing Solutions

#1

Radisys has been a pioneer in introducing IP-based cost efficiencies and enhanced capabilities to the conferencing industry. Radisys continues to be a leading supplier of IP Media Server products and technology, and helped accelerate the adoption of VoIP technology in the industry. Radisys supplies media servers to eight of the top 10 conferencing service providers around the globe. Radisys is also uniquely positioned with a solution portfolio allowing service providers to build their own conferencing applications based on their competencies and business strategy, or deliver a turnkey solution that supports customization for unique SP capabilities and integration with OSS / BSS.

Many of Radisys' CSP customers have developed their own conferencing application servers, which control the feature-rich capabilities of the Radisys media servers, including extensive Voice Quality Enhancement (VQE) capabilities. In addition, Radisys works with independent application developers and enterprise collaboration vendors to provide differentiated, economical, IP-based audio and video conference media mixing for their solutions.

For CSP customers looking for complete turnkey IMS conferencing solutions, Radisys also offers SIPware, a conferencing application server family pre-integrated with the Radisys media server. SIPware supports both reservationless and event VoIP audio conferencing today, including flexible APIs for integration with existing provisioning, billing, web portal and network management systems. SIPware application layer support for video conferencing will be available in 2013.

Next-generation network architectures, using IP call control and VoIP audio mixing, facilitated the introduction of user-friendly Reservationless Conferencing. It enabled subscribers to set up conference calls on their own dynamically, have an "always on" meeting number or uniform resource identifier (URI), as well as schedule a traditional Meet-Me conference call at a specific time. Subscriber PINs and passwords allowed easy, instant access to services. By eliminating the hassle of pre-scheduling conference calls through a human operator, the consumption of audio conferencing services soared, and is arguably the key driver for the large growth in hosted audio conferencing usage over the past decade.

Technology Trends for Hosted Video Conferencing

The same technology advances that enabled service providers to drive audio conferencing use and revenues to double-digit, year-over-year increases are now set to disrupt the status quo in video conferencing. Video services are still largely the domain of enterprise-class MCU equipment. While video MCU technology has evolved from ISDN to H.323 to SIP, with similar improvements in video quality to support capabilities such as telepresence, the architectures supporting video conferencing services are still largely delivered by enterprise-class MCU equipment. Why? This difference in architectural evolution is possibly explained by:

- *Video is more complex than audio,* due to many variants in standards and approaches. For this reason, many successful vendors continue to own the entire video conferencing ecosystem (MCU gear, along with video terminals and endpoints, and umbrella management systems). This strategy ensures a vendor's solution works end-to-end, but it often limits the ability for one vendor's solution to interwork with another's, or for a video conferencing platform to also be economically used in the delivery of other services.

- *Business quality cloud-based video conferencing is still in its infancy.* Certainly, many Fortune 500 businesses regularly use HD video conferencing and telepresence in their day-to-day business today, but the MOU for video conferencing is still a small fraction of the huge volume of audio conferencing around the globe today. One factor holding back mass adoption is the expense, given that video is bandwidth-hungry, and bandwidth has historically been quite expensive. Moreover, many video systems are still based on a reservation-based model, a complexity that is also a barrier to user consumption. As video usage grows, competition will increase and drive the availability of more cost-effective and easy-to-consume solutions.

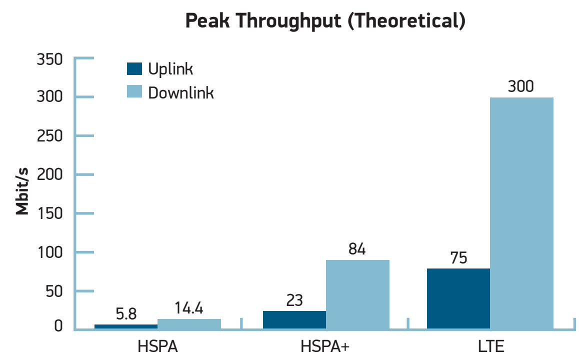

Figure 4. *Peak throughput of 3G and 4G mobile networks*

LTE with Wi-Fi: Satisfying the Bandwidth Requirements for Mobile Video Conferencing

The telecom industry is currently in the middle of a significant mobile infrastructure upgrade from 3G to the latest 4G/LTE technology. LTE provides much higher throughput than its predecessor, High Speed Packet Access (HSPA) used in 3G networks, namely four times more downlink and almost eight times more uplink as shown in Figure 4. Furthermore, LTE has better cell edge performance, improved latency and lower cost per gigabyte, while servicing more users with a higher quality of service (QoS).

As more and more users call in to video conference calls from their mobile devices, it is 4G/LTE that will finally provide the network bandwidth required. Not only does 4G/LTE deliver significant bandwidth improvements over 3G, but the technology also reduces network latency, which together provide the network requirements essential for supporting real-time mobile video services.

Good Wi-Fi service can support wireless connectivity to video conferencing services today. But it is LTE that will provide mobile wireless connectivity with high bandwidth and low latency, and is the catalyst for the significant growth projections for video conferencing services to smartphone, tablet and laptop devices.

SIP Trunking

Another significant trend supporting increased video collaboration services is SIP trunking. The migration of enterprise locations using ISDN or T1 trunks for interconnection to broadband IP connections with SIP signaling reduces the barriers, cost and complexity of transmitting video for enterprise users in their offices or via VPN.

Video Network Deployment Architectures

The industry has increasing consensus that LTE technology, combined with small cell technology for spectrum reuse, and Wi-Fi offload techniques at the network edge, along with SIP trunking and even the prospect that large enterprises will deploy IMS and peer their IMS core with their service providers' networks, will together deliver high bandwidth with low latency essential for video conferencing services. However, the approach to deliver video conferencing services over broadband IP networks to enterprise users and consumers is less obvious. Let us explore the considerations around hosted video conferencing service delivery in more detail, starting with a comparison of video network deployment architectures.

Multi-Codec MCU

The large majority of video conferencing solutions deployed today are based on Video Multipoint Control Unit (MCU) architectures. Most offerings continue to share many similarities with audio bridge equipment in the past, in that the conferencing application and signaling software is tightly integrated with video media processing hardware in a single integrated network element, although some MCU vendors are increasingly offering cloud-based offerings. MCU products today offer HD 1080p (or higher) video resolution and telepresence, with feature-rich user interfaces, including far-end camera controls, flexible multi-pane display and integrated data sharing. MCU solutions are also known as multi-codec, because they can support transcoding and transrating between different video and audio codecs, along with backward compatibility with legacy ISDN or H.323 video conferencing equipment.

The challenge is that MCU equipment is expensive, especially for connecting lower resolution endpoints into a conference—or even more cost-prohibitively, connecting audio only users into a meeting. MCU equipment is difficult to manage, hence many Enterprise customers outsource the management of their IP networks and MCU infrastructure to telecom service providers (another industry segment experiencing huge growth). MCU equipment can't be shared with other services, as MCU architectures are optimized for video conferencing only. MCUs from one vendor are often difficult to integrate with other vendors, or with third-party endpoints.

Enterprise Unified Communications (UC) Platforms

Leading Enterprise UC vendors offer video conferencing with other desktop communication applications including presence, chat, mail, calendar and audio/web conferencing. These platforms improve productivity across the enterprise facilitating collaboration with others using the exact same UC systems and clients, while providing flexibility to tune the UC system to unique enterprise business processes and requirements.

The challenge with UC systems is that the Enterprise customers also need to talk to third-party customers, partners and suppliers. It is the connection of "islands" of video conferencing functionality that is a recurring theme and challenge, and one of the key benefits of the IMS approach.

Over-The-Top (OTT)

Many of us have experienced the ease of use and benefits of OTT video calling and conferencing services, offered by Apple FaceTime, Skype, or Google Talk. The simplicity of these offerings prove to consumers what is possible, and demonstrate the benefits of visual collaboration.

	H.264 AVC	H.264 SVC	VP8	H.265
Strengths	• High interoperability • Large installed base • Low bandwidth • Minimized endpoint processing requirements	• High resilience to packet loss • Minimizes centralized processing costs • Minimized latency	• Similar to H.264 AVC performance • Royalty free • Adopted by WebRTC standards and initiatives (IETF and W3C)	• Roughly half the bandwidth compared to H.264 AVC • Extensions to H.265 could also support layered codec approach • Enables Ultra HD video
Weaknesses	• Requires centralized multi-codec mixing • End-to-end latency • Susceptible to packet loss	• More processing power required in endpoints • Higher bandwidth requirements • Small installed base • Low interoperability with other SVC or encoding standards • Proprietary implementations	• Small installed base • Limited codec and client implementations	• Not standardized yet (planned for 2013) • Requires more compute power

Table 1. *Comparison of Video Encoding Options*

Unfortunately, OTT offerings are inconsistent in quality and reliability, as they work over a best-effort network, the Internet, resulting in limited adoption for serious Enterprise-class communications. OTT clients also suffer from the "islands" of functionality problem. While OTT is proven to scale in subscriber base, OTT video conferencing has significant limitations in terms of the number of participants in a single conference, limiting its applicability for large enterprise-class video conferencing services.

Notable about OTT, or other "free" conferencing services, is that paid conferencing continues to grow, as enterprise customers continue to value consistent levels of high service quality, backed by service level agreements (SLAs).

Video Encoding Options

Early video conferencing solutions started with ISDN networks, then H.323, and more recently with H.263 video communication and encoding standards. For the purposes of this whitepaper, we'll instead explore the strengths and weaknesses of today's front-running video encoding approaches.

H.264 AVC

H.264 Advanced Video Coding (AVC) is by far the most common video encoding standard used in new conferencing deployments and video content delivery today. The result is high interoperability between MCU and SIP video client solutions across vendors. It has a large installed base and low bandwidth requirements, with minimized processing requirements in the endpoint devices.

H.264 often requires centralized multi-codec video mixing architectures requiring lots of processing power, which can make multi-codec H.264 systems expensive. These centralized processing systems can also increase end-to-end latency. H.264 video is also susceptible to packet loss.

H.264 SVC

The H.264 standard defines many profiles to support different market needs for video quality, latency, computational requirements and bandwidth. A subset of the H.264 profiles is considered scalable because parts of the video stream can be removed in a way that allows the receiving end to still decode and display the video.

Essentially, an H.264 SVC video stream consists of a base layer, along with different levels of enhancement layers (for example various bitrate, frame rate, and quality layers). This characteristic allows video conferencing architecture based on layer switching. The centralized layer switch might send only a base layer to a small screen mobile device with a low bandwidth connection. In contrast, a desktop or room system might receive the base layer with all the enhancement layers, to render an HD-quality video of the same conference.

	SIP	HTTP	WebRTC	OTT/Proprietary
Strengths	• Industry standard for IP-based telephony • Most commonly used in two-way conversational video services	• More commonly used for one-way video streaming • Allows adaptive bit rate switching	• Support from Internet industry to add two-way communications into emerging HTML5 browsers • No browser plugins	• Economic • Simple to use • Basic capabilities
Weaknesses	• Limited to VoIP and telecom services • Requires client-side SIP terminals	• Not suitable for two-way conversational video • Requires plug-ins • Incompatible HTTP streaming implementations	• Standards development in progress • Relies on browser implementations	• Proprietary • Limited interoperability between OTT vendors • Poor scalability for large conferences

Table 2. *Comparison of Video Distribution Technologies*

Layer switching requires less computational power than multi-codec transcoding, resulting in an economic benefit of lower price per port. Lower computational requirements make this approach attractive for cloud video conferencing service delivery. Layer switching with H.264 SVC also offers higher resilience to packet loss. In summary, H.264 SVC works well and economically with other H.264 SVC endpoints from the same vendor.

The downside with H.264 SVC is that this approach still has a small (but growing) installed base of proprietary implementations, resulting in low interoperability with other systems or even other SVC implementations. The "Achilles heel" of the H.264 approach is the expense of the gateways to overcome these interoperability problems. It also requires more processing power in the endpoints compared to H.264 AVC.

VP8

VP8 is a relatively new video codec with similar performance to H.264 AVC. It is gaining rapid industry interest, as it is a royalty free codec adopted by Web Real Time Communications (WebRTC) standards and initiatives within the IETF and W3C. However, it still has a small installed base with limited codec and client implementations.

H.265

H.265 (also known as HEVC, High Efficiency Video Coding) is still relatively new. While it hasn't been standardized yet in the industry (expecting completion in 2013), it will only require half the bandwidth of H.264 AVC, while supporting future Ultra HD video resolutions (up to 8K x 4K screen size and up to 300 frames per second). Extensions to H.265 will also support layer switching, similar to SVC extension for H.264. When it does arrive in vendor solutions, H.265 will require even higher levels of video media processing power in centralized infrastructure and endpoint devices.

Video Distribution Technologies

SIP

Session Initiation Protocol (SIP) is the industry standard for IP-based telephony. It is most commonly used in two-way conversational interactive voice and video services, but also supports one-way streaming. Deployments to date are limited to VoIP and telecom services, and it requires client-side SIP terminals.

HTTP

HTTP is the most commonly used video streaming protocol for wireline and mobile networks. It also allows adaptive bit rate switching. Unfortunately, HTTP is not suitable for two-way conversational services like video conferencing. HTTP also requires plug-ins for incompatible vendor implementations of adaptive bit rate streaming, such as HTTP Dynamic Streaming (Adobe®), HTTP Live Streaming (Apple®), and HTTP Smooth Streaming (Microsoft®).

WebRTC

WebRTC has growing support from the Internet industry to add real time communications (RTC) into emerging HTML5 browsers, hence, no browser plugins will be required. Standards development is still in progress at IETF and W3C. The power of WebRTC is that it piggybacks on the ubiquity of browsers on virtually every computing device. In addition, WebRTC related codecs are royalty free.

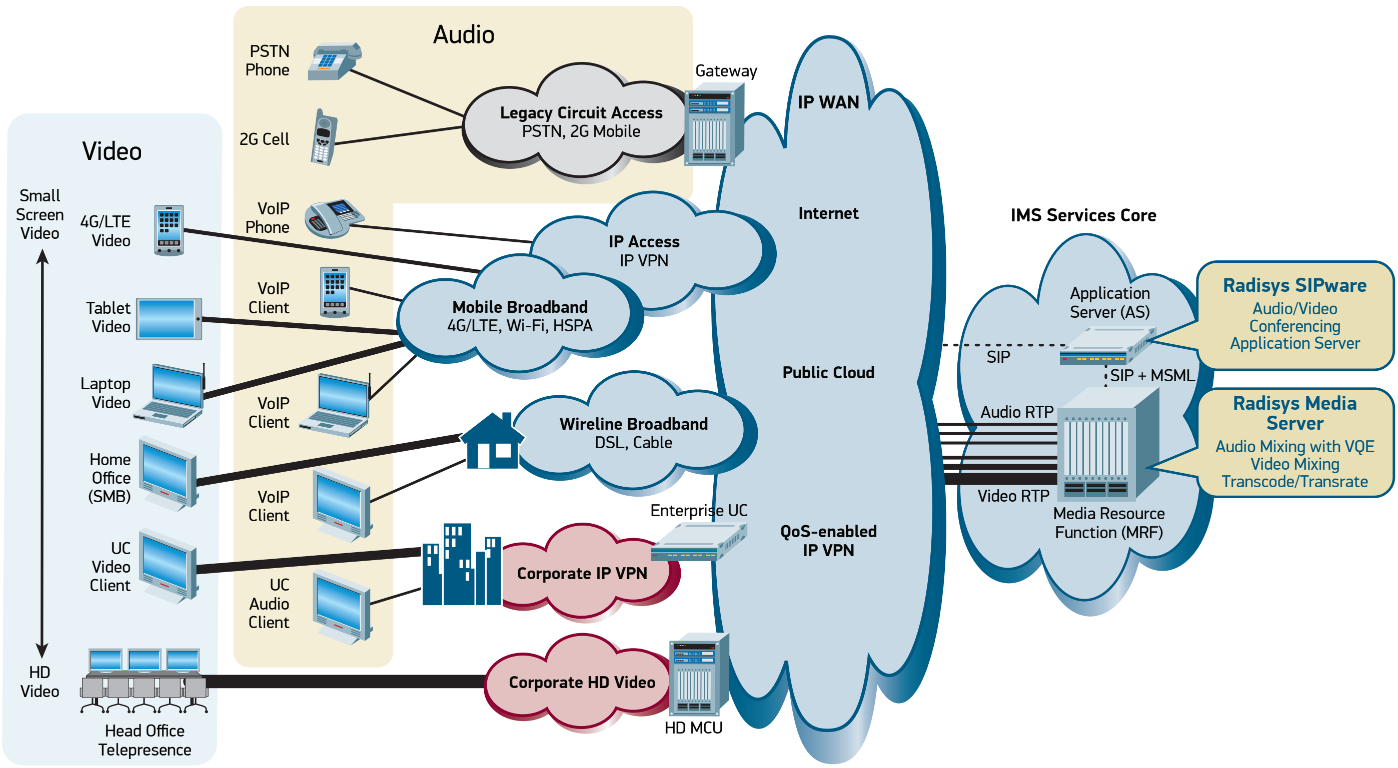

Figure 5. *The same IMS architecture used for audio conferencing also supports hosted video conferencing*

OTT/Proprietary

OTT solutions offer economic, simple-to-use clients for basic interactive video calling and conferencing between the same clients. However, their proprietary nature limits interoperability with other OTT clients or other video conferencing architectures. In addition, the mesh nature of connecting multiple OTT clients to achieve a conference call doesn't allow scalable video conferencing, while also consuming a great deal of network bandwidth.

IMS: Solving the Requirements for Cloud Video Conferencing Service Delivery

Over time, next generation VoIP architecture principles have evolved and broadened into IP Multimedia Subsystem (IMS) standards expressly developed to deliver any one-way streaming and two-way conversational service to any access network technology. As a result, IMS infrastructure communicates natively with VoIP endpoints, or with legacy 2G cellular or PSTN endpoints through media gateways that provide backward compatibility.

Similar changes and benefits are in store for cloud video conferencing using the exact same IMS architecture, shown in Figure 5. In some cases, the addition of video conferencing capabilities may only require a simple software upgrade to existing IMS audio conferencing deployment.

IMS is the best video network deployment architecture to drive the reusability and scalability essential to bring together the islands of competing video codec standards and conferencing architectures, and drive interoperability across:

- Devices
- Codecs
- Protocols
- Network Architectures
- Services

	Video MCU	OTT	Enterprise UC	IMS
Strengths	• Highest HD video • Telepresence • Bundled end-to-end solutions • Feature-rich capabilities • Integrated data sharing • Comprehensive management features	• Simple to use • Economic • Early adoption driving demand for video • Consumer focus	• Integration of video conferencing with other desktop applications • Excellent inside the Enterprise • Configurable to unique enterprise needs	• Reusability • Scalability • Multi-service • Multi-codec • Multi-protocol • Multi-tenant • Management
Weaknesses	• Expensive • Dedicated platform (not multi-service) • Limited multi-tenant (designed for Enterprise) • Difficult to connect third-party endpoints	• "Islands" of functionality • Lacks enterprise-class features • Inconsistent quality • Limited management • Limited HD	• "Islands" of functionality • Limited HD • Enterprise needs to manage the UC	• Integration scope between many third-party endpoints and IMS • IMS burdened with compatibility across diverse set of access networks

Table 3. *Comparison of Video Network Deployment Architectures*

Table 3 contrasts IMS with other video network deployment architectures.

Technological Advancements

Although the IMS architecture was designed from the beginning to support multiple services, including video conferencing, there are several other technical aspects at play. Foremost, both wireless and wired broadband IP networks now support the network bandwidth and address the stringent latency requirements for personalized two-way video. Early on, 2G and 3G wireless lagged behind wired broadband networks, which provided sufficient bandwidth via Ethernet directly to the Enterprise, and DSL or cable modem services to the home. Recently, wireless broadband has caught up with Wi-Fi. Today, LTE will bring similar improvements to mobile wireless broadband, which greatly boosts network bandwidth with the low latency. There's no doubt that LTE delivers a much better mobile video calling and conferencing experience than 3G. IMS was designed to work together with all of these access technologies.

Also needed to make video conferencing a reality was smartphone and tablet innovation, which provided front-facing cameras, computing power and improved broadband speeds and reliability—along with mobility.

Deep inside the network, video codecs are also evolving, H.261 followed by H.263, to the current H.264 AVC codec that supports 1080p and higher in telepresence systems. H.264 AVC has progressed with scalable video coding extensions, which effectively "switches" video layers through the network, but with the downside of using up more bandwidth compared to alternatives that utilize transcoding and transrating instead. New codecs, such as H.265 and VP8 are emerging as well. MRF platforms in the IMS networks are also evolving over time to support transcoding and transrating between all these video coding technologies.

IMS Conferencing Flexibility Will Drive Mass Market Adoption

An IMS architecture will make video conferencing easier to use by enabling ubiquitous meet me reservationless services, a key feature that has the potential to dramatically increase both business and consumer usage minutes. IMS can achieve this objective through its cost efficiencies, in that IMS service elements are designed to be shared across many services (not just video conferencing), access networks and customers. Sharing resources for a lower overall cost structure, along with addressing the interoperability challenges, will improve mass-market adoption of video conferencing.

IMS multimedia conferencing solutions also support other user-friendly features, including continuous presence (CP) and voice-activated switching (VAS)—see Video Conferencing Modes sidebar.

Video Conferencing Modes

Radisys MRF equipment simultaneously supports two modes of mobile video conferencing: continuous presence (CP) and voice-activated switching (VAS). This flexibility is important due to the wide assortment of tablets, smartphones and legacy mobile phones, their corresponding screen capabilities, and the actual mobile bandwidth condition at each caller's location.

Well-suited to a desktop or tablet with a relatively large screen, CP video conferencing takes video streams from multiple video conference participants and renders them on a single screen, providing an immersive experience. Unlike an SVC approach, an IMS-based solution achieves transcoding, transrating and video mixing in the MRF, such that a single video RTP stream (comprising the multi-pane CP view) traverses the network between the MRF and each video participant endpoint device.

For mobile devices with a limited screen size, CP may be impractical, since the panes displaying participants may be too small to see clearly. A better approach for small-screen devices is VAS because the user only sees the active speaker (one video stream) accompanied by his/her name displayed via a text-overlay feature. Depending on who is speaking, the MRF switches from one video-enabled participant to another. Audio-only participants can also be added into the same conference mix. Leading MRFs support VAS-video, CP-video and audio-only participants in the same conference mix, with full transcoding and transrating as required.

Figure 6. *Continuous Presence (CP)*

Figure 7. *Voice-Activated Switching (VAS)*

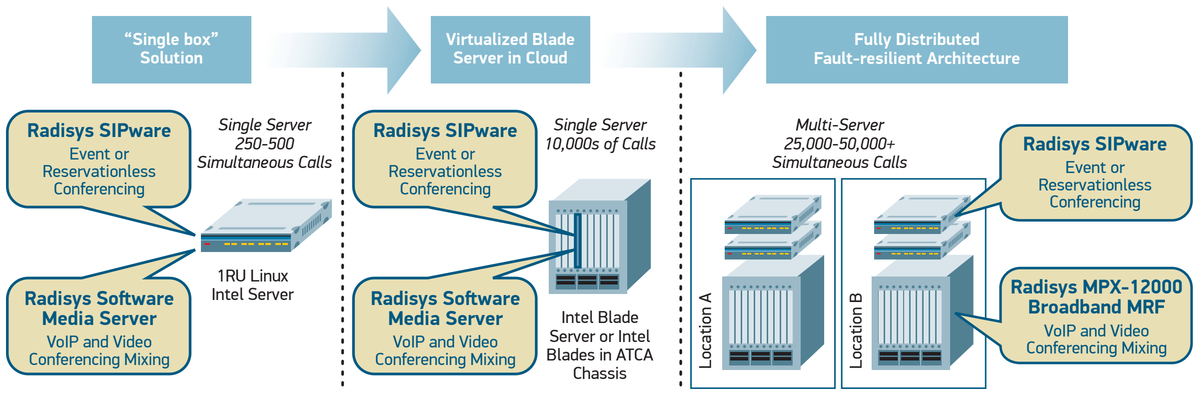

Figure 8. *Radisys conferencing solutions deliver cost-optimized scalability*

Applying IMS Multimedia Conferencing Solutions to Address Key Business Challenges

Problem #1: *Expensive Gear*
MCU solutions using proprietary platforms are costly and don't scale easily.

Solution: *Cloud Architectures Using Scalable, Low-cost Commodity Hardware*
Audio and video conferencing solutions based on IMS architecture can scale from low-cost Intel multi-core commodity servers on virtualized platforms, suitable for cloud infrastructure deployments, up to high-capacity conferencing-optimized MRF equipment, as shown in Figure 8.

Problem #2: *Connecting "Islands" of Video Conferencing Capabilities*
Enterprise users are increasingly benefiting from integrated UC solutions across the enterprise, from solutions and vendors like Microsoft Lync, Cisco or Avaya.These products offer a productive collaboration experience for participants within the company, but what about connecting customers, partners or other third-party participants? With video, there is no equivalent to the telephony world's e.164 number to identify and connect users between enterprises. While federation is one technical solution, it's operationally challenging and costly to create many-to-many federated relationships. Enabling inter-domain connectivity is a key challenge for existing enterprise UC solutions, but more generally for the IP communications industry.

Solution: *Video Interworking Gateway*
IMS was designed from the outset to provide a common service delivery core for multiple access technologies, enterprise UC solutions and device technologies. IMS multimedia conferencing solutions, hosted by a service provider in the cloud, will evolve to deliver increasing integration capabilities with Enterprise UC environments. In fact, the hosted video conferencing service provider will have an opportunity to offer video interworking gateway services (Figure 9), so video participants in company A (using Microsoft Lync) could enjoy a video conference with participants in company B (using a Cisco video conferencing solution). Since participants in a video conference (or a mixed video and audio conference) meet in the cloud, many of the challenges of trying to identify and connect to end points and conference bridges in the enterprise are eliminated.

Problem #3: *Lack of OTT Integration*
What about users using Apple FaceTime for video calls and conferencing? How would you integrate a Skype participant into a call with Microsoft Lync participants? The "islands" of capability problem is not isolated to enterprise UC solutions, but it also includes OTT client participants as well.

Solution: *Video Interworking Gateway*
Video interworking gateways built around an IMS cloud infrastructure can be used to integrate OTT clients with other clients, including the Enterprise UC clients mentioned earlier.

Problem #4: *Economics for Low-Bandwidth Connections with Enterprise MCU*
Enterprises may already own video MCU equipment that delivers a superior HD immersive experience for the executive team with room-based systems or a few HD desktop participants. Enterprise telepresence equipment is ideal for room and HD video conferencing, but it's expensive for connecting lower-bandwidth mobile devices to the same conference mix. Enterprises also want seamless management of this connectivity.

Solution: *Extending Enterprise MCU Using IMS in the Cloud*
IMS cloud video conferencing is complementary with Enterprise MCU deployments, while delivering cost savings and benefits to the Enterprise CIO. The cloud video conferencing provider will use its IMS to conference together third-party customer, partner and remote participants using different codecs, clients, devices or access technologies. The Enterprise MCU will continue to support HD video requirements for executives and room-based systems across the Enterprise IP VPN. Integration between the IMS and the Enterprise MCU equipment will be achieved through Telepresence Interoperability Protocol (TIP) and other interoperability forums including OVCC (Open Visual Communications Consortium).

Problem #5: *Poor Audio-only Economics*
While video conferencing is growing rapidly, the large majority of conferencing minutes continues to be audio conferencing, which is higher than video and web conferencing combined. Today's MCU video conferencing solutions are too expensive to support audio-only conferencing. For a CSP, this means maintaining two sets of infrastructure to address two different markets.

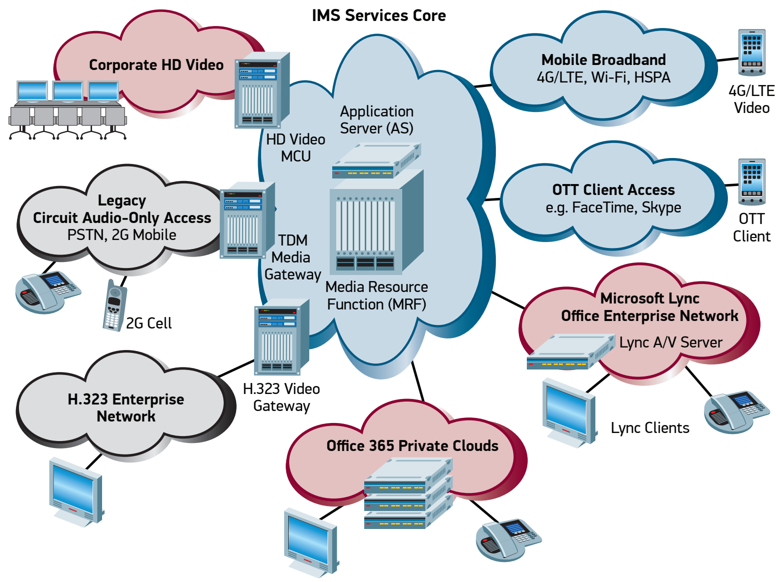

Figure 9. *IMS architecture delivering video interworking capabilities*

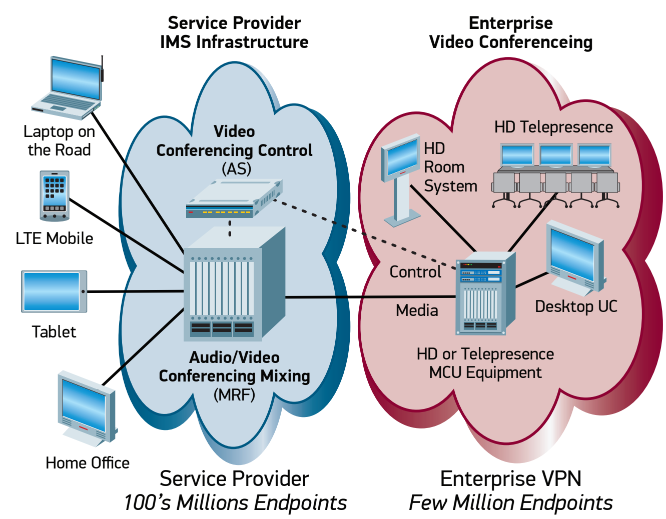

Figure 10. *IMS architecture extending Enterprise MCU conferencing to the masses*

Solution: *Single Optimized Platform for any Communications Service*
IMS architecture supports both video and audio conferencing along with a myriad of non-conferencing services such as VoLTE, Rich Communications Suite (RCS), Video Ringback Tones, and other revenue-generating Value Added Services (VAS)—all on the same cost-effective architecture.

Unlike a multi-codec MCU product which is specifically designed for video conferencing only, the separation of application logic and media processing in an IMS architecture facilitates reuse. The MRF can be reused and shared across multiple application servers, often from different vendors. It is the reuse and multiservice qualities of IMS that drive the superior economics of a cloud conferencing approach based on IMS.

Radisys Solutions for Cloud Video Conferencing

Radisys is the leading supplier of embedded wireless infrastructure solutions for the telecommunications industry.

Radisys' Trillium® LTE wireless protocol software addresses LTE Femtocells (Home eNodeB) as well as the Evolved Packet Core (EPC) Mobility Management Entity (MME), Serving Gateway (SWG), and Evolved Packet Data Gateway (ePDG). These standards-based Trillium LTE wireless protocols allow customers to rapidly develop LTE infrastructure to compete for early design wins in the dynamic LTE marketplace.

Radisys offers a variety of Intel-based platform options for building LTE, EPC and IMS network elements, including its award-winning T-Series ATCA and NEBS-compliant Rack Mount Servers.

Radisys media servers have a proven track record in supporting IP-based media processing in broadband wireline and 3G networks and are already working in 4G / LTE deployments. A natural progression is the cloud, where Radisys offers a software media server for Intel multi-core architectures and virtualized environments. Radisys software media server can be deployed on the Radisys Intel platforms mentioned above, commercial off-the-shelf (COTS) rack servers, or COTS bladed servers.

Radisys introduced the award-winning MPX-12000 to satisfy the IMS multimedia conferencing requirements of the future. It offers increased capacity and Voice Quality Enhancement (VQE) features for audio conferencing, making it an ideal choice to serve the role of media resource function (MRF) in an IMS multimedia conferencing system. The system has the horsepower needed for High Definition hosted video conferencing with both voice-activated switching (VAS) and continuous presence video (CP). The MPX-12000 supports VoLTE and LTE mobile endpoints with automatic transcoding and transrating, along with adaptive bandwidth and media quality control mechanisms.

The Radisys SIPware Reservationless Conferencing Application solution is a next-generation approach for delivering cost-effective, feature-rich, multi-tenant conferencing. Based on a pure IP implementation, it can be readily deployed network-wide through SIP-enabled media gateways or using cost-efficient SIP trunking to other VoIP carrier networks. The solution offers service providers a complete, off-the-shelf solution for delivering profitable conferencing services with VoIP economics. When combined with the Radisys SIPware Event Conferencing solution, supporting up to several thousand participants in a single event conference, conference service providers have a complete hosted conferencing solution built on cloud-computing architectures and technologies. This market-leading platform will support video conferencing features in 2013.

Conclusion

The market and technology for high quality conferencing services is changing rapidly. Video conferencing will be the growth engine for the next decade—extending audio and web conferencing to enable richer meetings. And coupled with the proliferation of OTT, UC and IMS clients, and even browser based meeting access, new applications for conferencing and collaboration will emerge.

Historically, enterprises have worked directly with vendors to purchase MCU infrastructure for deployment within their private IP VPNs, or with service providers who manage this MCU infrastructure on premise or in the network. However, enterprises will have the option of contracting services from CSPs who are building cloud video conferencing infrastructures based on IMS in order to deliver improved economics and interoperability for video conferencing services of the near future.

When deciding which architectural path to follow, CSPs will closely consider cost, scalability, flexibility and reliability; and for all these criteria, IMS is a better solution than adopting an MCU solution for large-scale multi-tenant requirements. IMS multimedia conferencing solutions address the concerns of users, particularly ease-of-use, interworking with unified communications (UC) solutions and multiple service options, including OTT. Addressing user expectations and responding to industry trends is vital for CSPs looking to increase video conferencing minutes and revenue growth.

Today, Radisys is a leading supplier of VoIP audio conferencing solutions and MRF media processing for video conferencing services, with a roadmap to deliver IMS video conferencing application solutions to help cloud operators capture upcoming video conferencing business opportunities.

References

[1] Source: Cisco VNI Service Adoption Forecast, 2011-2016, pg. 3, http://www.cisco.com/en/US/solutions/collateral/ns341/ns525/ns537/ns705/ns827/white_paper_c11-481360.pdf

[2] For further information this topic, please read "Cloud Media Processing," Radisys, May 2012 http://www.radisys.com/2012/cloud-media-processing/

[3] Source: Robert Mullins, "Blue Jeans Network Undercuts Cisco Video Pricing," March 29, 2012, http://www.informationweek.com/telecom/unified-communications/blue-jeans-network-undercuts-cisco-video/232700452.

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