

About RADVISION

RADVISION (NASDAQ:RVSN) is the industry's leading provider of market-proven products and technologies for unified visual communications over IP and 3G networks. With its complete set of standards-based video networking infrastructure and development toolkits for voice, video, data and wireless communications, RADVISION is driving the unified communications evolution. This is accomplished by combining the power of video, voice, data and wireless-for high definitions video conferencing systems, innovative converged mobile services, and highly scalable video-enabled desktop platforms on IP, 3G and emerging next-generation networks. For more information about RADVISION, visit www.radvision.com

The Unified Communications Handbook | Version 3

THE UNIFIED COMMUNICATIONS HANDBOOK

A Guide To The World Of Unified Visual Communications

Version 3



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THE UNIFIED COMMUNICATIONS HANDBOOK

A Guide To The World Of
Unified Visual Communications

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RADVISION Corporate Marketing Department.

Special thanks to Pierre Hagendorf, Associate VP for Business Development and Asher Shiratzky, Chief Technology Officer for the Technology Business Unit and to all of RADVISION's technology experts for their invaluable contribution to the technology glossary.

Dedicated to all of our visionary customers and partners who are making visual communications a reality.

If you would like to receive regular updates and news on revisions to the RADVISION Visual Communications Handbook, please register at www.radvision.com/resources/handbook.

FOREWORD

Visual communications is fast becoming part of our everyday lives — and changing the way we work, learn, collaborate and play. High-speed broadband connectivity, new communications standards that enable interoperability between all types of disparate devices, and full-featured, multimedia applications are the foundation of a far-reaching paradigm shift in the way we communicate.

Vision is key to staying in the game — and we mean this in the literal sense of the word. All eyes are turned toward the technologies and services that are making unified visual communications as commonplace and easy-to-use as a telephone. Because visual communications is a delicate integration of technologies, network infrastructure, hardware, communications protocols and human behavior, sometimes you may feel that the vision isn't very clear at all.

This Handbook will help you get the big picture into focus, and understand how these elements come together to enable a unified visual communications experience that's easy to use and familiar.

We hope this revised and updated Handbook and glossary help you navigate the labyrinth of technologies, trends and applications that are enabling us to set our sights on a future filled with visual opportunities.

The background features abstract, flowing lines in shades of green and blue, creating a sense of movement and depth. The lines are semi-transparent and overlap, giving the image a layered, ethereal quality. The colors transition from deep blue on the left to bright green on the right, with some yellowish-green highlights.

The
RADVISION
Experience

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EXPERIENCE

Meeting Anywhere

WHAT IS UNIFIED VISUAL COMMUNICATIONS?

Imagine if you could communicate with anyone you wanted to - any way you wanted to - from any device you wanted.

From your desktop PC, you could open a messaging or conferencing application, initiate a multiparty video conference with your boss whose office is in another country, possibly share and collaborate on a presentation with a colleague, and invite him or her into your conference. When you're finished (or when your microphone is muted) you can use your PC to make a video call to your grandmother on her new 3G phone that you bought her so she can see her grandchildren. In the meantime, on your PC screen, you have your eye on the kids at home, sneaking a peek at the TV instead of doing their homework.

But that's not the whole story. Imagine doing all this from your WiFi PDA or smartphone - on the go. Or from your company's high definition room conferencing system.

Unified visual communications means using only one application to enjoy a full-featured voice, video and data experience: one familiar PC or mobile handset application, like Microsoft Outlook or Messenger. You can choose the application or device you are most comfortable with and that suits your needs.

The unified visual experience means no battling with network and device compatibility issues, or integrating multiple interfaces and applications, or dedicated, closed, "members-only" networks.

Unified visual communication isn't a dream. It is happening today - on mobile operator, service provider and enterprise networks across the globe. It's time to get online, see and be seen... on your terms. All you have to do is choose how.

The Impact of Video in the Communications Arena

Video is a vital part of the unified communications experience.

Visual communications means people communicating as if they were in the same room. People in many locations, standing or sitting in front of video cameras, camera-enabled 3G cell phones, smartphones or PDAs, or in standard or high-definition equipped video conferencing rooms. Visual communications enables people to simultaneously talk and view each other on PC monitors, wall screens, mobile handsets, PDAs, telephones or televisions. Unified platforms expand these capabilities even more - to enable viewing and sharing of digital documents, such as spreadsheets and images. It's about unifying all communications resources into a single visual collaborative environment - not unlike a meeting room.

The real value? Visual communications delivers richer, more intense human interaction and enhances collaboration — while saving time and resources to bridge vast geographic distances within seconds.

The History of Visual Communications

For many, it all began at the 1964 World's Fair in New York City. At the AT&T Pavilion, people waited in long lines in the grueling heat for hours to talk with - and see at the same time - a stranger

in another fair pavilion. It was the historic debut of Bell Labs' Picturephone to the public. The excitement could be felt in the air; people were convinced that the future had arrived.

But the truth is that a rudimentary version of visual communications had been around for decades before. It was first demonstrated by Bell Labs' technicians who displayed a crude link between Washington, D.C. and New York City in the 1920s.

Those in the industry at the time hoped this "visionary" medium would soon become a commercial reality. Even after the excitement of the 1964 World's Fair exhibition, classical video conferencing failed to reach a broad audience for another thirty years.

It was simply a vision before its time.

The underlying technologies that make visual communications possible have dramatically improved. Bandwidth potential continues to multiply exponentially. Advanced DSP technology enables us to make digital video devices smaller than a coin. Telepresence is no longer a concept from the future, but a viable commercial possibility on the visual communications landscape. That's why more and more consumers and businesses are demanding the visual experience across the globe.

Today, visual communications is focused on a unified paradigm that allows ad hoc, on-demand visual communications from any application, over any network, any protocol and any device. This is being made easy by new network architectures, standards and

communications protocols. It is now a key tool in a wide range of industries, such as telemedicine, security and surveillance, government and education.

We've come a long way since classic video conferencing was first introduced. The spread of affordable IP over broadband has enabled the move from ISDN-based video conferencing systems and PSTN circuit networks toward converged voice, video and data over single IP networks. The emergence of 3G technology in the mobile arena has opened a world of on-the-go video opportunities. And finally, new network architectures, like IMS, are fostering fixed mobile convergence that will allow IP networks - and all their functionality - to reach every type of device anywhere.

Across the Board Benefits

In an enterprise, organization, educational institution or service provider framework, everyone benefits from unified voice, video and data communications.

Mobile Operator and Service Provider Benefits

- Enables rapid deployment of revenue-generating value added interactive video services
- Adds the personal video element to customer relations and contact centers
- Reduces churn and boosts customer loyalty
- Enhances the subscriber's communications experience

Enterprise & Organizational Benefits

- Improves enterprise-wide communications
- Enables more personal contact between employees and colleagues at remote locations
- Improves the ability to respond to business needs, perform tasks and manage projects
- Removes the need for travel between remote sites
- Delivers more effective training for both employees and customers
- Enhances connectivity with colleagues and supply chain partners
- Aids resellers in ad hoc marketing activities, such as product distribution demonstrations
- Reduces telecommunications costs over IP networks

Employee Benefits

- Reduces stress of long-haul travel
- Increases proportional productivity and efficiency
- Enables employees to build closer relationships with colleagues despite geographical distance
- Enhances the user experience

EXPERIENCE

Collaboration Anywhere



THE EVOLUTION OF CONVERGED NETWORKS

Collaborative visual communications was not created in a vacuum. It evolved over time, led by application and networking giants like Microsoft, Cisco, IBM, Nortel and Motorola; together with video conferencing technology innovators. It developed, from separate networks that handled the new and exciting capabilities that emerging technologies offered.

Converged Voice, Video & Data Networks

First there were voice networks, followed by data. Video began on separate dedicated closed video conferencing networks.

Video conferencing was traditionally ISDN-based, while voice communication was based on PSTN circuit networks. Data networks and the Web are based on IP (Internet Protocol). Ubiquitous and affordable IP over broadband networks led the way toward converged voice, video and data on a single IP network. This momentum has continued at full force, as IP networks extend to reach desktops and all types of mobile devices.

Market research reveals that the move towards single, converged, IP-based voice-video-data networks is well underway, with the widespread deployment of new signaling protocols like SIP, and next generation network architecture like IMS.

Separate But Equal?

To deliver on the promise of the converged revolution a large variety of standalone rich media applications emerged. Each worked on converged networks, but offered services that were restricted to users of the same network. Also, they were dedicated strictly to one form of media, such as Instant Messaging, Voice over IP, interactive video, e-mail, etc.

It is important to remember that video emerged as a popular platform on PCs, PDAs, 3G mobile devices and in room systems on the foundation of its predecessors — voice and data. Real growth to truly ubiquitous video is subject to it being an inherent element in converged collaboration applications.

THE ARRIVAL OF UNIFIED COMMUNICATIONS OVER CONVERGED NETWORKS

As we said earlier, computational power on the chip level has increased so that industry-standard platforms now have more than enough power to process all types of media — including video. This brought about a fundamental shift in development — transferring the burden for heavy-duty media processing from complex hardware designs to more flexible software platforms. This shift toward software-based solutions provided the scale necessary for large enterprises and service providers to deliver not only faster implementation, but easy integration with IP-driven applications. The result is more use of converged networks.

Web conferencing, Instant Messaging (IM), and presence on the desktop or mobile handset are no longer “cool” applications for kids — they are mission-critical tools for enterprises and organizations, and important value added services in the mobile arena. These IP-driven applications are the ideal choice for converged networks, and the unique combination of IM and presence is transforming ad hoc conferencing to a new level, complete with robust support for multimedia.

Web-based conferencing has enabled organizations of all sizes to spread their wings — and communicate not only internally but also along the supply chain, and with business partners, colleagues, and associates around the globe.

In contrast to group visual communications, where participants

congregate in a specially configured conference room equipped with expensive video and voice equipment. Web-based conferencing allows participants to be at their home or office PC. Wherever they need to be and are most comfortable.

All they need is a Web browser and Web cam. IP-based local and wide area networks, including the Internet, make the connection and do all the rest.

But don't be fooled, web-based and desktop video conferencing are not mere "spin-offs" of room-based group video conferencing. The Web or desktop paradigm centers on ad hoc video calls — as easy to make as a standard telephone call.

These exciting developments in multimedia communications over converged networks are being spearheaded by two apparently divergent sides of the industry. On the one hand, application developers, such as Microsoft, IBM/Lotus, Oracle, and Adobe/Macromedia, are embracing multimedia with gusto. On the other side are the network communication vendors, such as Cisco, Avaya, Siemens, and Nortel. They are supplying innovative network infrastructure and platforms based on high definition video and telepresence that drives these trends. With advocates like these, it is no surprise that rich media conferencing and visual communications are becoming part of the mainstream. RADVISION works with these important partners on both sides, offering the technology that is driving the visual experience.

All Your Applications Working Together

Today, so many capabilities and services are available from within common, congruent interfaces, or as an embedded functionality within workflow applications.

Video, the richest modality, is the vehicle that enables us to leverage all the power of the unified communications experience. Using one network and one application to seamlessly integrate multimedia conferencing and communications is perhaps the real “killer application” of the IP age - adding real value in terms of improved productivity in business, education, healthcare and consumer communications.

The background features a large, vibrant green shape on the left that tapers towards the right. On the right side, there are several overlapping, flowing shapes in shades of light blue and dark blue, creating a sense of movement and depth. The overall aesthetic is clean and modern, with a focus on organic, fluid forms.

EXPERIENCE

Green Communications

VISUAL COMMUNICATIONS: FOR A MORE PRODUCTIVE WORKPLACE

Ask any manager and they will tell you that human resources are an enterprise's most valuable asset and a prerequisite for business success. Having said that, global competition makes the competitive landscape rough terrain, and in order to compete, global collaboration has become a key factor in the ability of a workforce to be efficient and productive.

It is a challenge to collaborate when the staff is literally deployed around the four corners of the globe. Moreover, the myriad of communications alternatives keeps employees busier than ever before by just trying to stay on top of their messages and communiqués. From the phone, IMs and emails to generating reports, creating presentations, and running to the next meeting, employees are so busy trying to communicate it may seem they never have the time to do what they were hired to do!

Unified communications, accessible directly from familiar workplace applications, is changing all this. It is empowering employees to work more efficiently by allowing them to conveniently and effectively collaborate with colleagues and clients from home, the office, or on the road.

Despite all the “productivity” technologies that have been developed over the last decade, some of these have fallen short of expectations. For instance, group video conferencing facilitates face-to-face collaboration without travel. Handheld email and Instant Messaging

enable “continuous presence”. While designed to enhance productivity, sometimes we may wonder if they make life easier for the stressed employee?

In their present state, these tools can increase complexity. While most employees do in fact master the technology they think they need, it is unreasonable to expect them to master a veritable menu of independent technology applications in order to communicate.

It is this complexity that has relegated most of these applications to the realm of scheduled meetings — often with the support of an IT team.

How can enterprises bring these impromptu casual one-on-one discussions, Instant Message exchanges and ad hoc meetings between colleagues into the realm of everyday life while leveraging them to foster productivity and creativity?

That’s where visual communications in the workplace comes in. Unified visual communications is more than just an enabler of video conferencing. It brings geographically dispersed employees “virtually together” in the same workspace — to create an atmosphere conducive to teamwork and cooperation between isolated resources.

The background features abstract, flowing shapes in shades of blue and green, creating a sense of movement and depth. The shapes are layered and semi-transparent, giving the overall composition a dynamic and modern feel.

EXPERIENCE

Enhanced Communication

MOBILITY EVERYWHERE AND ANYWHERE: THE 3G REVOLUTION

Business people are not the only ones who need to communicate while on the go: the whole world is using mobile communications. It is no surprise that as more and more video-enabled infrastructure equipment and endpoints emerged from costly room systems, the mobile arena was next in line. Mobile operators are deploying advanced networks that support video-enabled applications. Users want to connect from single video-enabled mobile devices. Converged communications platforms and methodologies are making this happen.

A Paradigm Shift to Total Mobility

Perhaps today only a small fraction of mobile phone subscribers own 3G phones, but the number of advanced mobile devices that have video capabilities is growing rapidly. Operators are just beginning to meet the potential initiated by handset developers by offering interactive video services.

As a result, subscribers are demanding the mobile visual experience — to see, share, interact, collaborate and communicate from their mobile handsets. Operators and service providers are deploying customized video services that seamlessly integrate with existing networks to meet this demand. Interactive video services that connect video-enabled PCs to the mobile world can jump-start 3G video usage and expand the 3G video community exponentially.

These compelling services leverage one-time infrastructure investments, offer ongoing revenue streams and increase chargeable bandwidth.

While not yet as ubiquitous as science fiction and action films would have us believe, the market for mobile video communications is growing at a steady pace.

Recently published research indicates that dozens of operators worldwide offer 3G services using WCDMA radio transmission technologies; some use 1xEV-DO networks. Solid commercial offerings are already in place, and even more pilots are underway — for almost any type of personal mobile video communications service you can dream of: from video dating and video blogs to video contact centers and multiparty video conferencing services.

Progress in 3G enabling technologies, such as IP and the emerging IP Multimedia Subsystem (IMS), improved audio and video codecs, miniaturization, digital signal processors, and display technologies are driving this revolution.

Mobile Challenges

Connecting carrier-class circuit switched networks, and the IP networks that are used for advanced video applications, was one of the main technological challenges in delivering visual communications over mobile networks. Gateways and bridges helped overcome the immediate challenges to a certain degree. Today though, a true high-quality interactive experience requires much more than merely linking the IP and mobile worlds.

Video telephony, or multiparty conferencing and real-time video applications, require a minimum amount of upstream bandwidth to ensure a quality video experience. Advances in codec development and implementation, such as H.264, are improving the video experience even at lower bandwidths. Today, 3G-324M networks can already offer quality video calling with only 64 kbps upstream with the quality of service needed to roll out carrier-grade services.

The key is bridging the circuit switched and IP networks with mobile gateways. These were soon deployed by most 3G operators to power video services, such as video conferencing and video mail. This though, is only the beginning. As a critical mass of endpoints is reached, more interactive video services will be deployed to increase user satisfaction and network usage.

Mobile Technology to Catch Up With Demand

Developing networks, such as HSDPA, which offers higher bandwidth for downstream, and HSUPA, which will offer high bandwidth for upstream, will take the market to new levels of quality.

Divergent networks each offer both technological advantages and challenges. With the growing use of WCDMA in Europe and Asia Pacific, and TD-SCDMA in China, the challenge remains to be able to leverage the inherent power and ease-of-use of the circuit-switch layer — while simultaneously harnessing all the video power of the IP layer. CDMA 2000 EV-DO Rev A, used in the US, Korea, and South America, widens the challenge to provide consumer and enterprise video services over pure IP networks.

The true technological breakthrough in the mobile arena though is emerging next generation networks, like IMS. The challenge today is to deploy interactive video services that will evolve with IMS.

The background features a complex, abstract pattern of flowing, translucent shapes in shades of blue, green, and yellow. These shapes overlap and blend into each other, creating a sense of movement and depth. The overall effect is reminiscent of liquid or smoke captured in motion, set against a light, neutral background.

EXPERIENCE

Unified Communications

VIDEO NETWORK ELEMENTS

The world of visual communications is a seemingly complex menagerie of network topologies, communications protocols, standards and technologies.

To make matters more complex, communications is a realm where the well-known saying — “the only thing that is constant is change” — rings true in every sense.

Keeping that in mind, we will try to lay out some of the fundamental elements of this dynamic world.

Multiparty visual communications is based on Internet Protocol (IP)-centric network endpoints, terminals and network infrastructure.

User Equipment

Voice and video-enabled end-user terminals such as 3G mobile devices, SIP terminals and desktops, and H.323 terminals. These terminals must support many video protocols, which are quite complex. For a rich multimedia conference, terminals support compliant audio and video codecs to achieve audio and video capability. Of course, the requisite peripheral equipment like microphones, speakers and video cameras are essential.

Network Infrastructure

- **Multipoint Control Units (MCUs)**

MCUs are what enable multipoint conferences. Today, MCUs are usually hardware-based, due to their processing and real-time requirements. However, software MCUs are common for desktop video conferencing applications.

- **IP Media Server**

An IP media server enables the development and deployment of many multimedia services. This type of platform handles complex video details, such as call control, media processing, video transcoding and conferencing, DTMF control, extended connectivity with any device and terminal — 3G, wireline or wireless — and integration with back-end systems.

- **Gateways**

Gateways essentially allow intercommunication between IP networks, legacy networks and 3G networks. Gateways can also perform call setup and clearing on both sides of an IP to switched-circuit connection.

- **Proxies, Gatekeepers & Network Element Management**

Proxies and gatekeepers control the flow of calls in the network, or "manage" the network. Besides handling translation of IP addresses and people-friendly aliases, gatekeepers offer an array of services, such as call routing, call transfer and forwarding, line hunting, registration services, routing rules and more.

The background features several thick, flowing, translucent ribbons in shades of blue, green, and yellow. These ribbons curve and overlap, creating a sense of motion and depth against a light, off-white background. The lighting is soft, highlighting the edges of the ribbons.

EXPERIENCE

High Definition

SECURITY AND ENCRYPTION

Encryption has become a very important factor in the multimedia collaboration environment, particularly in the government and defense sectors. Since the events of 9/11, the demand for secure video conferencing has increased significantly. Secure systems are currently being designed and deployed not only for enterprise collaborative communications, but also for real-time combat situations.

Deploying visual communications in security sensitive sectors presents complex challenges, such as bandwidth limitations. This is especially true in the defense and government sectors, where it is vital that visual communications infrastructure support all types of endpoints — from meeting rooms, desktop, command centers and field laptops.

Encryption allows enterprises and government agencies to transport sensitive and classified information across the backbone securely. There are multiple methods of “locking down” the data, depending on the level of sensitivity, using software or hardware encryption methods.

Depending on the customer environment (e.g., open or closed network), there are several ways to provide network access. In most situations accessing a closed IP network via ISDN would be accomplished by using a gateway device in order to keep the data secure over the public switched telephone network (PSTN). Type 1 encryption devices (e.g., KIV-7s) could be integrated into the solution. These Type 1 encryption devices would terminate on the serial

gateway interface which provides secure entry into the closed network. This ensures that all the data is protected from the core to the edge. Once on the internal IP network, Advanced Encryption Standard (AES) can be applied to protect the data.

The gateway serial interface enables connection to encryption devices which are commonly used in federal and military applications. In addition to serial interfaces, users also receive packet handling, advanced call functionality, and integrated network resource management, enabling feature-rich, high-quality secured video conferencing among remote sites, regardless of the device or network protocol used at each location.

Features for Secure Encryption

- **Loss of Synchronization (LOS)**

The use of encryption devices raises network link synchronization issues. Because of the “real-time” nature of video conferencing applications, they are very susceptible to network conditions such as latency and packet loss which can result in noticeably poor video quality. Therefore, many encryption devices are equipped to support re-synchronization of the attached network link. This condition is commonly known as Loss of Synchronization or (LOS). Terminal equipment such as video endpoints and gateways need to be able to support automatic and manual initiation of encryption re-synchronization. This will eliminate or reduce the need for user intervention. If the video endpoint or gateway detect that there are problems with the network or video quality, a properly-equipped and configured device will automatically trigger the encryption

mechanism to re-synchronize the network link in order to improve the level of video quality.

- **Secure IP Network Synchronization**

Secure clocking for communication devices is of vital concern, especially in the defense industry. Many legacy H.320 based gateways recommend a secure primary clock from ISDN based networks. This physical connection between classified and unclassified networks can create network vulnerability. IP centric gateways address this vulnerability by optimizing primary clocking from the internal IP network. This accurate timing and synchronization across networks is critical not only for security, but for a growing number of applications and communications protocols.

- **Support for Serial Interfaces and Signaling Protocols**

Suitable secure gateways must support a wide variety of serial protocols and cables, providing direct connectivity to the most commonly used encryption devices. These include the V.35, RS-449, EIA-530, EIA-530A and RS366 signaling protocols, and support for standard and dedicated cables for interfacing with specific encryption devices.

- **Density**

Mission-critical communications is dependent upon a distributed and scalable deployment strategy. Gateways and infrastructure that support encryption should have multiple serial interfaces, enabling several simultaneous calls at the appropriate bandwidth.

Deploying secure real-time applications on converged network architecture requires well thought out solutions that embrace common network practices and industry standards. Whether deploying a highly classified collaborative environment for the government or rolling out an enterprise collaborative network, maintaining a solid security strategy to protect data should always be a priority.

The background features several thick, flowing, semi-transparent lines in shades of green and blue. These lines originate from the left side and curve downwards and to the right, creating a sense of movement and depth. The colors transition from a deep blue on the left to a bright green on the right, with some yellowish-green highlights. The overall effect is that of a dynamic, organic structure, possibly representing a network or a path.

One Meeting Many
EXPERIENCES

THE TECHNOLOGY & STANDARDS OF UNIFIED VISUAL COMMUNICATIONS

High Definition (HD Technology)

More and more companies and organizations use visual communications and other multimedia collaboration technologies to meet their objectives. Resolution and bandwidth issues that have kept video quality at standard definition levels, are one of the challenges of achieving widespread deployment and use.

The introduction of High Definition (HD) has completely changed the face of video conferencing and overcomes this challenge. High Definition dramatically improves the viewing experience for remote video conferencing, and is a key catalyst for growth and increased usage. Endpoint companies, such as LifeSize, have pioneered new and exciting tools to make interactive video a vivid, high definition experience.

Currently, the high definition resolution commonly used for video conferencing is almost three times (2.66X) the resolution of SDTV (Standard Digital TV) and nine times (9X) the resolution of FCIF.

Video network infrastructure that supports high definition video resolutions of 720p and 1080p and AAC high fidelity audio are making widespread use and deployment of video conferencing possible.

Standards: The Key to Interoperability and Interworking

In order to deliver multiparty visual communications over IP and 3G networks, all the networking devices and terminal equipment needs to be interoperable, or to "interwork." Standards are what we use to achieve this interoperability.

Global IP networks are most often built around multiple IP communications standards, such as SIP, H.323, MEGACO and MGCP. Inter-standard or multi-protocol bridges or gateways help to achieve the interworking we need for video communications to operate correctly.

One of the video conferencing industry's most significant achievements over the last decade has been to create and widely adopt open international standards. This allows all types of video conferencing equipment to work together and interoperate smoothly with equipment from another vendor.

These standards are set and developed by various standardization bodies, such as the ITU-T, IETF, the ISO/IEC Moving Picture Experts Group (MPEG), and others. While the video conferencing industry does not directly control the creation of standards which affect its development, certain developers and manufacturers play a role and sit on steering committees and boards.

The arrival of SIP as an alternative H.323 for VoIP, and the rapid development of 3G standards for the mobile arena set the stage for an open-standard approach that is advancing the industry forward. The following section discusses the IP standards related to IP and

3G multimedia visual communications, and details the key devices needed to establish and sustain it.

IP Standards

The ITU-T and IETF, the two major standards bodies, have established the following widely-deployed standards:

- **ITU-T H.323**

This is a complete suite of protocols that facilitate multimedia communication over IP. Incorporated within this standard are video codecs and audio codecs, such as G.711, G.729 and others.

- **ITU-T T.120**

This standard contains a set of communication and application protocols and services that provide support for real-time, multipoint data communications. These multipoint facilities are important building blocks for a whole new range of collaborative applications, including desktop data conferencing, multi-user applications, and multi-player gaming.

- **IETF MGCP**

- IETF MEGACO/ ITU-T H.248**

These standards address the requirements of production IP telephony networks that use decomposed gateways.

MGCP is the protocol allowing control of Media Gateways by Media Gateway Controllers. A Media Gateway is a network element performing conversion between media signals on telephony network and media over IP. MGCP is designed for

voice only networks. MEGACO/H.248 is successor of MGCP which allows multimedia communication. It is the result of a joint effort of ITU SG16 and IETF.

- **IETF SIP**

This is a signaling protocol that focuses on session initiation, modification and termination. It is a strong competitor to H.323, and is the signaling protocol selected for use in IMS (IP Multimedia Subsystem), the multimedia subsystem of the third generation networks.

- **IETF RTSP**

This is a signaling protocol for use in streaming media systems which allows a client to remotely control a streaming media server. This protocol is used for IPTV VOD (Video on Demand), PSS (Packet Switched Streaming) and surveillance systems. It requires the use of RTP/RTCP for sending actual media.

- **IETF RTP/RTCP**

This is the predominant standard transport for real-time streaming of data, such as audio or video. Almost any IP protocol used for signaling, such as SIP and H.323, employs RTP/RTCP for sending actual media.

IP Standards Development

The H.323 protocol is best known as the original call signaling protocol that made real-time voice and video over IP possible. Today it has virtually been replaced with SIP, the most widely deployed and interoperable signaling solution.

SIP delivers an unprecedented easy mixing of video with voice and Web over IP to the desktop. The power of presence enables easier communications. SIP's use of Instant Messaging is an effective way of setting up and initiating a multimedia call, starting a chat session, upgrading to voice, data collaboration, and of course — video.

Microsoft's support for SIP in Windows[®] is the result of a significant development effort, and fostered its widespread use in conferencing.

Most major vendors, including RADVISION, were involved in the definition of special SIP extensions for media control that allow SIP-based multipoint conferencing.

MGCP and MEGACO/H.248 are complementary protocols to SIP and H.323, and are deployed "internally" within the network. These "internal protocols" are designed specifically for interfacing between "intelligent" Media Gateway Controllers (MGC) and "dumb" Media Gateways.

A key factor to the future growth of IP-centric visual communications lies in the infrastructure, which enables interworking of all the protocols described above. The infrastructure is what enables the seamless, end-to-end connectivity between all types of endpoints and network devices that creates the video conference experience.

- **3G Standards**

3GPP and 3GPP2 are the main standardization bodies dealing with 3G. In most cases, they rely and build upon standards defined by the IETF and the ITU-T in terms of signaling.

- **3GPP 3G-324M**

Based on ITU-T H.324M, the 3G-324M standard is used for mobile video telephony on 3G handsets. Virtually any new 3G handset in the market today supports 3G-324M. 3G-324M is based on a circuit switched network technology, exploiting legacy telephony infrastructure for dial plans, authentication, billing, etc.

- **3GPP PSS**

Packet Switched Streaming (PSS) is based on IETF RTSP. It handles one-way streaming of content from the network to handsets.

- **3GPP IMS**

IMS grew out of a group of standards created by the 3rd Generation Partnership Project (3GPP). IMS (IP Multimedia Subsystem) is a Media over IP network technology that defines how to set up advanced services for 3G cellular networks. It uses the Session Initiation Protocol (SIP) as its base signaling protocol.

A close-up, artistic photograph of a white lily flower. The petals are soft and white, with prominent green veins. The stamens are bright yellow and partially visible. The background is a soft, out-of-focus light green and white, creating a clean and elegant aesthetic.

EXPERIENCE

Video Everywhere

FIXED MOBILE CONVERGENCE

While it may seem natural to think about the world of IP and 3G as separate entities, our mobile lifestyle is dictating another reality. With so many mobile phones in the hands of consumers and business users, and cellular operators' competitive pricing, more and more subscribers are demanding all the functionality of their mobile devices on their fixed wireline telephones and IP devices.

Fixed to Mobile Convergence (FMC) is currently one of the most important strategic issues in the telecommunications industry. With the ability to connect mobile phones to fixed line infrastructure, operators and service providers can offer advanced services no matter where they are, how they access the network, or what type of device they are using.

The key drivers behind fixed to mobile convergence are cost and convenience. Fixed to mobile substitution offers business opportunities for mobile operators and mobile subscribers. Lifestyles, network reach, pricing, personalization and compelling applications may encourage users to opt for mobile. But long-term strategies and quality issues still need to be addressed to sustain long-term growth.

This fixed mobile convergence is coming about due to emerging next generation network architectures like IMS.

Next Generation IMS – The Next Frontier

IMS (IP Multimedia Subsystem) is a generic architecture for multimedia and voice communications over IP services, defined by the 3rd Generation Partnership Project (3GPP). IMS supports multiple access types including GSM, WCDMA, CDMA2000, WLAN, wireline broadband and other packet data applications.

IMS is based on SIP for call signaling, and originally designed for mobile networks. The IMS core network allows delivery of value added IP services to both fixed and mobile customers from calls whose destination is either an IP or circuit-switched (CS) network — or from any fixed wireline device, cellular phone or mobile device.

IMS enables delivery of Internet technologies and services that mobile users are familiar with from the cellular world — like voice, text, location, presence, pictures and video, or any combination of these. IMS also lets operators introduce new services — at the top level of packet-switched networks.

IMS-based services inherently handle this multi-mode communication in a personalized manner and securely. This is why IMS is the key to unified visual communications over both fixed and mobile networks, and the catalyst that is driving fixed to mobile convergence.

Using IMS architecture in their networks, mobile, cable and wireline operators can offer their own VoIP services, set policies and manage these value added services.

The background features abstract, flowing lines in shades of green and blue, creating a sense of movement and convergence. The lines are semi-transparent and overlap, with a bright light source from the top right creating a glow and lens flare effect.

EXPERIENCE

Converged Communications

THE FUTURE OF VISUAL COMMUNICATIONS

There is no question that we live in a video world. In some cases, such as for security and surveillance, it is hard to imagine how we ever lived without it. Economic fluctuations and increased security threats underscore just how much we have to gain from video-enabled communications.

With the proliferation of unified, easy-to-use solutions, technological advances in this market are coming from the bottom-up — fueled by demand. This can be seen in the excitement created by user-generated content, interactive TV and telepresence.

In a few years it will be commonplace to be visually in touch with family from a business trip or vacation using a handset or PDA and a home-based PC. Or collaborate with colleagues miles away while on the road on a sales presentation. Or conference in a service engineer for real-time instructions on how to fix the office printer, washing machine or elevator. It is not hard to imagine how unified visual communications will revolutionize the babysitting industry!

No matter from what vantage point you are coming from — it's a win-win situation. Fixed and mobile operators, service providers and application service providers enhance revenues by offering rich visual applications.

At the same time, enterprises and consumers benefit from cost-effective, feature-rich IP-based communications, increased

productivity and the next best thing to being there that video collaboration offers — from the desktop, mobile phone, PDA or smartphone.

Users are embracing visual paradigms more and more. Some people will never feel comfortable in front of a camera — even the tiny one on their mobile phone! Others take to it naturally. In any case, these habits are being learned and can become second nature.

In the business sphere, unified visual communications is about saving time, working more productively and reaching new levels of collaboration. That is important. But visual communications is also being used to do so much more. Like improve healthcare, further education and support top level government and military communications.

So the benefits are clear. Yes, we still have a lot to learn. But don't let visual communications "stage fright" slow you down. Visual communications has already changed so much about the way we communicate. So embrace the future and be seen. Whether at work, on the go, or at play, visual communications is here to stay.

THE VISUAL COMMUNICATION GLOSSARY



3G

Third Generation Mobile System - The generic term for the next generation of mobile wireless communications networks.

3GPP

Third Generation Partnership Project - A body comprising several organizational partners working to produce technical specifications for a third-generation mobile system based on GSM core networks and the radio access technologies they support known as WCDMA (UMTS).

3GPP2

The 3rd Generation Partnership Project 2 (3GPP2) is a collaboration agreement that was established in December 1998. A cooperation between ARIB/TTC (Japan), CCSA (China), Telecommunications Industry Association (TIA) (North America) and Telecommunications Technology Association (TTA) (South Korea) to make a globally applicable third generation (3G) mobile phone system specification within the scope of the ITU's IMT-2000 project. In practice, 3GPP2 is the standardization group for CDMA2000, the set of 3G standards based on earlier 2G CDMA technology.

3G-324M

The Third Generation Partnership Project (3GPP) has adopted the ITU-T Recommendation H.324 Annex C, also referred to as H.324/M (Mobile) or H.324M (with very few changes), on their 3G-324M as the standard for multimedia telephony over 3G circuit switched cellular networks. This standard is used for handling real-time video telephony in 3G mobile handsets.

4CIF (COMMON INTERMEDIATE FORMAT)

A standard video format used in video conferencing. CIF has 4 times the resolution of QCIF. 4CIF has four times the resolution of CIF and is equivalent to standard PAL TV resolution (704 by 576 pixels).

AAA (AUTHENTICATION, AUTHORIZATION AND ACCOUNTING)

Authentication refers to the confirmation that a user who is requesting services is a valid user of the network services requested. Authorization refers to the granting of specific types of service (including "no service") to a user, based on their authentication, what services they are requesting, and the current system state. Accounting refers to the tracking of the consumption of network resources by users. Widely used AAA protocols are Radius and Diameter.

AAC (ADVANCED AUDIO CODING)

A standardized, lossy digital audio compression scheme and part of the MPEG-4 standard. Designed as an improved-performance codec relative to MP3. AAC is the current default iTunes codec, the media player which powers iPod. AAC is also widely available in a range of portable media players and 3G mobile handsets.

ACCOUNTING

Accounting refers to the tracking of the consumption of network resources by users. This information may be used for management, planning, billing, or other purposes. Typical information that is gathered in accounting is the identity of the user, the nature of the service delivered, when the service began, and when it ended.

ADDRESS RESOLUTION

A mechanism for identifying the address of a called endpoint in terms of the network, such as an IP address.

AEC (ACOUSTIC ECHO CANCELLER)

Acoustic Echo is caused by direct and indirect feedback from speaker to microphone. An acoustic echo canceller electronically removes both direct coupling and reflected echo, enabling true full-duplexhands-free telephony for both mobile phones and desktop speakerphones.

AJB (ADAPTIVE JITTER BUFFER)

A jitter buffer is designed to remove the effects of jitter from the decoded voice or video stream, buffering each arriving packet for a short interval before playing it out. A fixed jitter buffer maintains a constant size whereas an adaptive jitter buffer has the capability of adjusting its size dynamically in order to optimize the delay/discard trade-off.

ALIAS

An alternative identification string for an IP address. An alias can be a name, a URL address, an e-mail address, a transport address in the form of "IP address:port number" or a Party Number.

ALG (APPLICATION LEVEL GATEWAY)

Serves as communicators between two networks. ALGs are protocol-aware entities that examine application protocol flows and only allow messages that conform to security policies to pass. See also Proxy Server.

ALTERNATE GATEKEEPER

Support for an Alternate Gatekeeper enables you to make Gatekeeper failures transparent to the endpoints that are registered to the Gatekeeper. In RADVISION implementations, a backup Gatekeeper (the “Secondary” Gatekeeper) runs in parallel to each online Gatekeeper (the “Primary” Gatekeeper).

AMR (ADAPTIVE MULTI-RATE)

An audio data compression scheme optimized for speech coding. AMR was adopted as the standard speech codec by 3GPP.

AMR-WB (ADAPTIVE MULTI RATE - WIDEBAND)

A speech coding standard. The codec provides excellent speech quality due to wider speech bandwidth of 50 - 7000 Hz compared to narrowband speech codecs which in general are optimized for POTS wireline quality of 300-3400Hz. AMR-WB is codified as G.722.2, an ITU-T standard speech codec.

AS (APPLICATION SERVER)

An IMS element that implements service logic, such as conferencing servers, multiparty video calling and presence. The AS abstracts the underlying network architecture and protocols, enabling application developers to easily and quickly introduce new services.

ATCA (ADVANCED TELECOMMUNICATIONS COMPUTING ARCHITECTURE)

A very large specification effort with more than 100 companies participating. Known as AdvancedTCA™, the official specification designation is PICMG 3.x. AdvancedTCA is targeted to requirements

for the next generation of "carrier grade" communications equipment.

ATM (ASYNCHRONOUS TRANSFER MODE)

A high bandwidth, high speed (up to 155 Mbps), controlled-delay packet switching and transmission system integrating multiple data types (voice, video, and data). Uses fixed size packets also known as "cells". ATM is often referred to as "cell relay".

AUTHORIZATION

Authorization refers to the granting of specific types of service (including "no service") to a user, based on their authentication, what services they are requesting, and the current system state. Authorization determines the nature of the service which is granted to a user.

AUTHENTICATION

The process of verifying the identity of a user trying to log on to a system, or of the sender of a message.

B2BUA (BACK-TO-BACK USER AGENT)

A Back-to-Back User Agent (B2BUA) is a method of implementing call control servers in SIP. A B2BUA retains control over the call during the duration of it while the proxy is typically involved only in the setup phase of the call.

B CHANNEL

Bearer channel. In ISDN communications, a B channel transmits data or voice at 64 or 56 Kbps.

BANDWIDTH

The transmission capability of a communications channel. In analog communications, bandwidth is measured in Hertz (cycles per second). In digital communications it is measured in bits per second (bps). Ethernet bandwidth is typically 10, 100 or 1000 Mbps.

BASE STATION

A two-way radio installation in a fixed location, used to communicate with one or more mobile or portable radio transceivers. They are typically used by two-way radios such as mobile phones, portable phones and wireless routers.

BEARER

Refers to a service that allows transmission of information signals between network interfaces.

BER (BIT ERROR RATIO)

The ratio of the number of bits incorrectly received to the total number of bits sent during a specified time interval. The error ratio

is usually expressed in scientific notation; for example, 2.5 erroneous bits out of 100,000 bits transmitted would be 2.5 out of 10⁵ or 2.5×10^{-5} . The bit error ratio will be affected by both the data transmission rate and the signal power margin. BER directly affects the media quality of calls.

BEST EFFORT QoS/SERVICE

Best effort delivery describes a network service in which the network does not provide any special features that ensure the quality of the service delivered.

BILLING

Billing in a telecommunications environment refers to the capacity of a service provider to capture, rate, and bill communications events. These events include voice, video, data, and electronic content.

BONDING

Bandwidth on Demand. Increasing the bandwidth of a call by linking two or more B channels of an ISDN line.

BORDER GATEWAY

A gateway at the border between two networks. Responsible for enforcing network policies, such as security, QoS, type of service, priority, authorization, etc.

BRI (BASIC RATE INTERFACE)

An ISDN subscriber interface supporting two bearer B channels at 64 Kbps and one data D channel at 16 Kbps.

BRIDGE (CONFERENCE BRIDGE)

A connection between two or more conference sites, enabling data, voice or video traffic to pass simultaneously between the sites. Video conferencing bridges are often called MCUs.

BRIDGE (INTER-NETWORK)

Gateways, interworking gateways, session border controllers (SBCs) and other entities that interconnect between networks are often referred to as bridges.

BROADBAND

A method of transmitting larger amounts of data, voice and video than telephony networks allow. Traditionally connecting to the Internet meant dialing a specific number from a PC via a fax/modem at a typical speed of 56kbps (narrowband) to then gain access the internet or the WWW (World-Wide-Web).

As speeds improved the term Broadband became popular, and is often accompanied by the speed, or bandwidth, of information or data (i.e. 512kbps, 1Mbps, 8Mbps) that travels down a fiber-optic cable or telephone line per second depending on your choice of service. In ISDN, broadband channels support rates above the primary E-1 (2.048 Mbps) and T-1 (1.544 Mbps) rate.

BROADCAST

Transmission of data to everybody on the network or network segment.

CABLE

The delivery of Internet service over cable infrastructure. The proliferation of cable modems, along with DSL technology, has ushered in the age of broadband Internet access via cable.

CAC (CONNECTION ADMISSION CONTROL)

The set of actions taken by the network during the call setup phase (or during the call re-negotiation phase) to establish whether a VC/VP connection can be accepted.

CALL CONTROL

Also called Call Processing. Refers to the signaling involved in setting up, monitoring, transferring and disconnecting (tearing down) a call.

CALL FALLBACK

Support for Call Fallback enables you to configure a Gatekeeper Forwarding policy to deal with cases, such as Gatekeeper failure to resolve a destination address in the IP network, lack of Gatekeeper bandwidth resources, or unsuccessful call setup to the destination endpoint due to network failure.

CALL SIGNALING ROUTING

Two alternative modes for routing the Call Setup (Q.931) and Control (H.245) channels. Routed Mode routes the Call Setup and Control channel through the Gatekeeper. See also Routed Mode, Direct Mode, Q.931 + H.245 Routed Mode.

CALL WAITING

A telephone network feature. If a calling party places a call to a

called party which is otherwise engaged, and the called party has the call waiting feature enabled, the called party is able to suspend the current telephone call and switch to the new incoming call. They can then negotiate with the new or the current caller an appropriate time to ring back if the message is important, or to quickly handle a separate incoming call.

CAPABILITY

Because terminals and servers are complex, and support multiple protocols, multiple codecs, and many optional features, calls need to communicate their capabilities so that negotiation and connection are successful to complete the call.

CASCADING

The MCU allows you to combine two or more conferences resulting in a larger conference with many more participants. This is called Cascading. Cascading creates a distributed environment that helps reduce drain on network resources.

CCCP (CENTRALIZED CONFERENCING CONTROL PROTOCOL)

A protocol belonging to the IETF related to SIP and IMS. Responsible for the transport of the conference control commands from the user agent to the (conferencing) application server.

CCXML (CALL CONTROL EXTENSIBLE MARKUP LANGUAGE)

An XML standard designed to provide telephony support to VoiceXML. CCXML is designed to inform the voice browser how to handle the telephony control of the voice channel.

CDMA (CODE DIVISION MULTIPLE ACCESS)

A form of multiplexing and a method of multiple access that does not divide up the channel by time or frequency. CDMA also refers to digital cellular telephony systems that make use of this multiple access scheme, such as those pioneered by Qualcomm, or WCDMA.

CDMA2000

A family of third-generation (3G) mobile telecommunications standards that use CDMA, a multiple access scheme for digital radio, to send voice, data, and signaling data (such as a dialed telephone number) between mobile phones and cell sites. It is the second generation of CDMA digital cellular.

CDR (CALL DETAIL RECORD)

Information in a simple text format that can be used as input to third party billing programs or other software for billing purposes.

CENTRALIZED MCU CONFERENCES

In a centralized topology, the Multipoint Controller and Media Processor unit components of the MCU work together to manage conference signaling and to perform media processing for all connected terminals. The MCU can handle multiple conferences simultaneously. See also MCU, Cascading, Clustered MCUs.

CIF (COMMON INTERMEDIATE FORMAT)

A standard video format used in video conferencing. CIF requires four times the bandwidth of QCIF (176 by 144 pixels).

CIPHER

An algorithm for performing encryption (and the reverse, decryption) — a series of well-defined steps that can be followed as a procedure.

CLI (CALLER LINE ID)

The calling party's number, and in some places, the calling party's name, sent to the called party's telephone equipment during the ringing signal. Or when the call is being set up, but before the call is answered.

CLIP (CALLING LINE IDENTIFICATION PRESENTATION)

The presentation of a number being identified by Caller ID (CLI). The opposite of CLIR.

CLIR (CALLING LINE IDENTIFICATION RESTRICTION)

The blocking of a number from being identified by Caller ID.

CLUSTERED MCUs

The Multipoint Controller (MC) and Media Processor (MP) unit components of the MCU operate independently. The MCU can be set up in a clustered layout to use a single MCU to control several units configured to operate only as MP units performing media processing. MCUs configured as MP-only units have their MC component disabled. The controlling MCU unit also makes use of the local MP component.

CODEC (CODER/DE CODER)

Hardware or software that converts sound, speech or video signals from analog to digital code, and vice versa.

COMPRESSION

A technique for reducing the bandwidth or bit rate required to encode a block of information so that it occupies less space on a transmission channel or storage device.

CONFERENCE HUNTING

The purpose of Conference Hunting is to maintain conferences and ignore Line Hunting where necessary.

CONFERENCING SERVICE

The conferencing service is a mechanism supported on the MCU which defines the qualities, capabilities and management policies of a conference.

CONFIDENTIALITY

Defined by the International Standards Organization (ISO) as "ensuring that information is accessible only to those authorized to have access". Confidentiality is the cornerstone of information security.

CONNECTIONLESS SERVICE

Network service that does not require a session connection between senders and receivers. Senders simply start sending packets to the destination. Useful for periodic burst transfers. A connectionless network provides minimal services.

CONNECTION-ORIENTED SERVICE

Connection-oriented service requires a session connection (analogous to a phone call) to be established before data can be sent. This method is often called a "reliable" network service. It can guarantee

that data will arrive in the same order. Connection-oriented services set up virtual links between end systems through a network.

CONTACT CENTER

A call center generally receives and handles customer calls. When the call center handles not only a voice channel but also a chat, e-mail or video channel, it is referred to as a contact center.

CONTENT PROVIDER

A company that provides services to subscribers or network operators. These services include shopping, Web surfing, chat rooms, gaming, accessing data, such as music and books through a server, and more.

CONTINUOUS PRESENCE

Allows you to view multiple participants on one screen at the same time. Incoming participant images are combined into a video image layout set according to the policies of the conferencing service. The range of video layouts available depends on the type of media processing supported.

CONVERSATIONAL SERVICE

A service that provides two-way, interactive, real-time, end-to-end information transfer.

COPS (COMMON OPEN POLICY SERVICE)

A protocol used for exchanging network policy information between a Policy Decision Point (PDP) in a network and the Policy Enforcement Points (PEPs). On WCDMA networks, the GPRS Gateway Serving Node (GGSN) is the PEP as part of the overall QoS system.

CORBA (COMMON OBJECT REQUEST BROKER ARCHITECTURE)

A standard for software components that defines APIs, communication protocols, and object/service information models to enable heterogeneous applications written in various languages running on various platforms to interoperate. Provides platform and location transparency for sharing well-defined objects across a distributed computing platform.

cPCI (COMPACT PCI)

A 3U or 6U Eurocard-based industrial computer, where all boards are connected via a passive PCI backplane. The connectors and the electrical rules allow for 8 boards in a PCI segment. Multiple segments are allowed with bridges.

CPCP (CONFERENCE POLICY CONTROL PROTOCOL)

A client-server protocol that can be used by users to manipulate the rules associated with the conference.

CRC (CYCLIC REDUNDANCY CHECK)

A type of hash function used to produce a checksum — a small, fixed number of bits — against a block of data, such as a packet of network traffic.

CS (CIRCUIT SWITCHED)

A networking technology that provides a temporary but dedicated connection between two stations regardless of the number of switching devices through which data is routed. Analog circuit switching (FDM) has been replaced by digital circuit switching (TDM).

The digital technology still maintains the connection until one speaker hangs up.

CSCF (CALL/SESSION CONTROL FUNCTION)

A key IMS element that makes session or call control and routing decisions in the IMS network; CSCF uses IMS SIP signaling to control sessions. See also P-CSCF and S-CSCF.

CSICS (CIRCUIT SWITCHED IMS COMBINATIONAL SERVICES)

Services for the transitional phase of early IMS deployment that leverage existing circuit-switched infrastructure to deliver real-time media, combined with ease of advanced and easy to deploy IMS services, such as presence, IM and more.

CUG (CLOSED USER GROUP)

Used to define a virtual private network on a public network.

D CHANNEL

The data signaling channel of an ISDN line. This channel is used to carry call control messages between the ISDN terminal and the public switch.

DEFERRED DELIVERY MESSAGING

A type of IMS Messaging service by which the sender expects the network to deliver the message as soon as the recipient becomes available.

DELAY

The time taken for a signal to pass through a video conference from the sending station to the receiving station.

DES (DATA ENCRYPTION STANDARD)

A cryptography method that makes it impossible for anyone without the decryption key to restore the data to its unscrambled form. DES is now considered to be insecure for many applications. This is chiefly due to the 56-bit key size being too small; DES keys have been broken in less than 24 hours.

DESKTOP VIDEO CONFERENCING

Video conferencing on a personal computer. Most appropriate for small groups or individuals. Many desktop video conferencing systems support document sharing or data collaboration.

DEVICE ADDRESS

The unique network address of a device, assigned by a carrier and following the format defined by an international standard such as

E.164 for MSISDN addresses, X.121 for X.25 addresses or RFC 791 for IP addresses. An address uniquely identifies the sending and/or receiving device.

DEVICE PROFILE

Information describing the capabilities of the device.

DHCP (DYNAMIC HOST CONFIGURATION PROTOCOL)

In a DHCP environment, IP policy is dynamic. This means that a terminal does not have a constant IP address. Management keys for identifying endpoints in a DHCP environment are the alias name or phone number of an endpoint.

DIAL PLAN

In traditional telephony systems, a dial plan is a front end system that allows users to call each other by dialing a number on a telephone. In voice and video conferencing over IP, a dial plan is a system that allows participants in point-to-point or multipoint conferences to call each other or join conferences.

DIAMETER

An AAA (Authentication, Authorization and Accounting) protocol for applications such as network access or IP mobility. The basic concept is to provide a base protocol that can be extended in order to provide AAA services to new access technologies. It is the AAA protocol chosen for the IMS environment.

DID (DIRECT INWARD DIALING)

A method in which incoming calls are routed directly to endpoints

on the LAN, without operator intervention.

DLNA (DIGITAL LIVING NETWORK ALLIANCE)

A global cross-industry organization of leading consumer electronics, computing industry and mobile device companies that share a vision of a wired and wireless interoperable network where digital content, such as music and videos can be seamlessly shared through personal computers (PCs), consumer electronics (CE) and mobile devices in and beyond the home.

DNS (DOMAIN NAME SERVER)

On TCP/IP networks, DNS converts the domain name (URL) of a host computer into a numeric IP address using the following format: xxx.xxx.xxx.xxx.

DOCUMENT SHARING

A video conferencing feature that enables multiple participants to view and edit the same computer document.

DOWNLINK

Unidirectional radio link for the transmission of signals from a UTRAN access point to the UE. Also the direction from a Network to the UE, in general.

DOWNSPEEDING

The ability of a Gateway to maintain an ISDN video call by reconnecting the call at a lower rate when one of the B channels is lost. Downspeeding contributes to a higher percentage of call completion on the network.

DSP (DIGITAL SIGNAL PROCESSING)

The study of signals in a digital representation and the processing methods of these signals. The goal of DSP is usually to measure or filter continuous real-world analog signals. The algorithms required for DSP are sometimes performed using specialized computers, which use specialized microprocessors called digital signal processors. These process signals in real time and are generally purpose-designed ASICs.

DTMF (DUAL TONE MULTI-FREQUENCY)

Also called touch-tone dialing. The tones that are heard when you press the buttons on a touch-tone telephone.

DVB (DIGITAL VIDEO BROADCAST)

A suite of internationally accepted, open standards for digital television maintained by the DVB Project, an industry consortium and published by a Joint Technical Committee (JTC) of European Telecommunications Standards Institute (ETSI), European Committee for Electrotechnical Standardization (CENELEC) and European Broadcasting Union (EBU).

E-1

The European interface for PRI ISDN lines. An E-1 connection has up to 30 B channels and 1 D channel, which transmit at rates of up to 2.048 Mbps.

E.164

The term “E.164 number” differentiates an “absolute” telephone number from the number you must dial to reach an endpoint from a specific location. E.164 numbers include country codes, national destination codes and subscriber numbers.

ECHO CANCELLATION

A process which attenuates or eliminates the acoustic echo effect on video conference calls. Echo cancellation is largely replacing obsolete echo suppression.

ECHO EFFECT

A time-delayed electronic reflection of a speaker's voice. This is largely eliminated by modern digital echo cancellation.

ECHO SUPPRESSION

A technique for reducing annoying echoes in the audio portion of a video conference by temporarily deadening the communication link in one direction.

EDGE

A mobile data service available to users of GSM mobile phones. Provides higher bandwidth than GPRS. Typically referred to as 2.75G.

ENDPOINT

A network element at the end of the network such as an H.323 terminal, a Gateway, an MCU, a PC terminal, IP or ISDN phone, or video camera.

ETHERNET

A LAN physical and data link protocol running over the lowest two layers of the OSI Reference Model at speeds of up to 10,100 or 1,000 Mbps.

ETSI (EUROPEAN TELECOMMUNICATIONS STANDARDS INSTITUTE)

A France-based non-profit organization that produces telecommunications standards used throughout Europe and beyond.

EV-DO and EV-DO Rev. A (EVOLUTION-DATA OPTIMIZED)

A wireless radio broadband data standard adopted by many CDMA mobile phone service providers in Japan, Korea, the Czech Republic, Russia, Latvia, Romania, Portugal, Brazil, Israel, the United States, Australia, Canada, New Zealand, Venezuela, Angola, Mexico, and Puerto Rico. It is standardized by 3GPP2, as part of the CDMA family of standards.

EVRC (ENHANCED VARIABLE RATE CODEC)

A speech codec used in CDMA networks.

EXIT ZONE

When you define a prefix for the Exit Zone Service, you need to dial

the prefix to reach an endpoint in another zone. This can be useful for restricting unauthorized users from making calls to other zones. The Exit Zone prefix affects the way in which the Gatekeeper tries to complete calls to other zones.

FALLBACK

See Call Fallback.

FAST CONNECT

Also called Fast Start and H.323 Fast Start. A procedure for shortening the time it takes to start a call by skipping the H.245 phase and transferring channel parameters in the Call Setup messages.

FAST START

See Fast Connect.

FCC (FEDERAL COMMUNICATIONS COMMISSION)

An independent US government agency charged with regulating interstate and international communications by radio, television, wire, satellite and cable.

FEC (FORWARD ERROR CORRECTION)

A system of error control for data transmission, whereby the sender adds redundant data to messages, which allows the receiver to detect and correct errors without the need to ask the sender for additional data. Retransmission of data can often be avoided, at the cost of higher bandwidth requirements on average, and is therefore applied in situations where retransmissions are relatively costly or impossible.

FECC (FAR END CAMERA CONTROL)

In conversational video applications, far-end camera control protocol is used by participants to control the remote camera. H.281/282 is the standard that defines how to implement FECC.

FIREWALL

A barrier device placed between two separate networks. A firewall can be implemented in a single router that filters out unwanted packets or it can use a variety of technologies in a combination of routers and hosts. Today many firewalls combine filtering functionality with Network Address Translation (NAT) functions and are typically located at the border of the network they protect.

FLAT CAPACITY

Consistent port capacity independent of video codecs or video quality.

FLAT CAPACITY PLUSTM

When users receive maximum value for their investment, with consistently optimized port capacity independent of video codecs or video quality. If bandwidth is not fully utilized, the number of available ports increases for reduced cost per port.

FLOW CONTROL

Comprised of the hardware, software and procedure for controlling the transfer of IP packets between two points on a network.

FMC (FIXED MOBILE CONVERGENCE)

The trend to provide a set of identical services accessible from different networks (fixed & mobile). Advanced implementations are also able to seamlessly switch between the networks while in a call.

FRAME RATE

The rate at which individual video pictures (frames) in a sequence

are displayed in a video conference. Frame rate is measured in frames per second (fps).

G.711

An ITU-T standard for speech codecs that provides toll quality audio at 64 Kbps using the PCM method and either $\mu\mu$ -law or A-law.

G.722

An ITU-T standard for speech codes that provides toll quality audio at 64 Kbps.

G.722.1

An ITU-T standard for speech codecs that compresses 50Hz -7KHz audio signals into one of two bit rates, 24 or 32 Kbps.

G.723.1

An ITU-T standard for speech codecs that provides good quality audio at 5.3 or 6.3 Kbps.

G.726

An ITU-T ADPCM speech codec standard covering the transmission of voice at rates of 16, 24, 32, and 40 Kbps. G.726 also introduced a new 16 Kbps. The four bit rates associated with G.726 are often referred to by the bit size of a sample, which are 2-bits, 3-bits, 4-bits, and 5-bits respectively.

G.728

An ITU-T standard for speech codecs that provides near-toll quality audio at 16 Kbps.

G.729 A/B

An ITU-T standard for speech codecs that provides near-toll quality audio at 8 Kbps. G.729 Annex A is a reduced complexity codec and G.729 Annex B supports silence suppression and comfort noise generation.

GAN (GENERIC ACCESS NETWORK)

Formerly known as Unlicensed Mobile Access (UMA), describes a telecommunication system allowing seamless roaming and handover between local area networks and wide area networks using the same dual-mode mobile phone.

GATEKEEPER

In H.323 IP telephony, a Gatekeeper is an optional server that is responsible for network-based services including registration, admission and status, for which it uses a special protocol called RAS. Gatekeeper functions include address translation, call authorization and bandwidth management, as well as providing accounting information.

GATEKEEPER ZONE

A group of endpoints together with their Gatekeeper constitute a zone. You can configure a zone by predefining endpoints that are entitled to register with the Gatekeeper. See also Predefined Endpoint, Registered Endpoint, Zone

GATEWAY

A Gateway is a network entity, which provides a bridge between networks. An H.323 Gateway provides real-time, two-way

communication between H.320 terminals on the packet-based network and other ITU terminals on a circuit switching network, or to other H.320 Gateways. A video Gateway provides real-time, two-way multimedia communication between video terminals on different networks. For example, between H.323 terminals on a packet-based network and H.320 terminals on an ISDN network, or between H.323 terminals on a packet-based network and 3G-324M, or terminals on a 3G mobile network.

GATEWAY SUPPORTED PREFIXES

H.323 version 2 enables a Gateway to specify prefixes that the user should dial before the WAN number in order to make a call using a certain media.

GCF (GLOBAL CERTIFICATION FORUM)

A partnership between network operators and terminal manufacturers that provides an independent program to ensure global interoperability of 2G and 3G mobile wireless terminals whose objective is to boost industry confidence in the performance and reliability of new 3G mobile devices.

GERAN (GSM/EDGE RADIO ACCESS NETWORK)

The interface between the EDGE radio network and the core network. Also defining an interface between two BSS.

GPRS (GENERAL PACKET RADIO SERVICE)

A mobile data service available to users of GSM mobile phones. It is often described as "2.5G" - or a technology that is between the

second (2G) and third (3G) generations of mobile telephony.

GROUP HUNTING

Group Hunting enables a Gatekeeper to perform load balancing for a group of H.323 endpoints defined with the same alias.

GSM (GLOBAL SYSTEM FOR MOBILE COMMUNICATIONS)

The standard digital cellular phone service of Europe, Japan, Australia and elsewhere.

H.223

Specifies a packet-oriented multiplexing protocol for low bit rate multimedia communications; Annex A & B handles light and medium error prone channels of the mobile extension used in 3G-324M.

H.225.0

H.225.0 specifies the procedures and messages applicable to Gatekeepers, including the RAS protocol for Registration, Admission and Status.

H.235 SECURITY

H.235 security ensures the authentication of each endpoint and the integrity of messages. The basis of the security process is the shared secret between the endpoint and the Gatekeeper.

H.239

Recommendation for role management and additional media channels for H.3xx-series terminals. Sets forth a way to have multiple video channels (e.g., one for conferencing, another for presentation) within a single session (call).

H.245

The ITU-T standard used for the Control Protocol for Multimedia Communication. H.245 is included in the H.225.0 Recommendation. H.245 provides signaling for the proper operation of the H.323 terminal, including capabilities exchange, opening and closing of logical channels together with a full description of these channels, mode preference requests, flow control messages, and general commands and indications.

H.245 PROXY

The H.245 Proxy enables routing H.245 channels in a point-to-point H.323 call. The H.245 Proxy is a natural extension of a Gatekeeper.

H.245 TUNNELING

H.245 tunneling decreases the time between the point at which an endpoint initiates a call and the point at which the call participants are ready to open multimedia channels. The endpoint must also support H.245 tunneling.

H.248

The ITU-T standard for signaling and control between circuit-switched PSTNs and VoIP networks.

H.249

The ITU-T standard enabling the creation of simple interactive user interfaces for devices equipped with advanced keypads and touch pad inputs. Using this standard, the user experience of handsets and terminals connecting to media servers can be greatly enhanced.

H.261

An ITU-T standard for compressing an H.320 video conferencing transmission. H.261 supports CIF and QCIF resolutions.

H.263

An ITU-T standard video codec. This codec is an enhancement to the H.261 codec. H.263 offers improved compression over H.261 and includes a set of tools aimed at optimizing codec performance.

H.264

MPEG-4 Part 10, or Advanced Video Coding, is a digital video codec standard which is noted for achieving very high data compression. Technically identical to the ISO/IEC MPEG-4 Part 10 standard (formally, ISO/IEC 14496-10).

H.281

The ITU-T standard recommendation for local and far-end camera control protocol for ISDN (H.320) video conferencing.

H.282

The ITU-T standard recommendation that defines a set of services and a protocol which may be used to perform remote device control in a multimedia conference.

H.320

The ITU-T standard for video conferencing over digital networks such as ISDN.

H.323

The ITU-T standard for video conferencing over packet-switched networks such as LANs and the Internet.

H.323 FAST START

See Fast Connect.

H.324

The ITU-T standard for video conferencing over analog telephone

lines (POTS) using modems.

H.324/M

The Third Generation Partnership Project (3GPP) has adopted the ITU-T Recommendation H.324 Annex C, also referred to as H.324/M (Mobile) or H.324M (with very few changes), on their 3G-324M as the standard for multimedia telephony over 3G circuit switched cellular networks.

H.450

See Supplementary Services.

H.460

Standard that promises to make NAT-firewall traversal between vendors' video conferencing equipment and different end-user organizations much easier.

HANDOVER

The process in which the radio access network changes the radio Transmitters, radio access mode or radio system used to provide the bearer services, while maintaining a defined bearer service QoS.

HARD HANDOVER

A category of handover procedures where all the old radio links in the UE are abandoned before the new radio links are established.

HD (HIGH DEFINITION)

A video system of higher resolution than standard-definition (SD), e.g. NTSC, SECAM and PAL.

HSDPA (HIGH-SPEED DOWNLINK PACKET ACCESS)

A mobile telephony protocol. Sometimes referred to as a 3.5G (or "3.1/2.G") technology. HSDPA provides a smooth evolutionary path for Universal Mobile Telecommunications System (UMTS) networks allowing for higher data capacity (up to 14.4 Mbit/s in the downlink) used for downlink communication to mobile devices.

HSUPA (HIGH-SPEED UPLINK PACKET ACCESS)

A data access protocol for mobile phone networks with extremely high upload speeds up to 5.76 Mbps. Similar to HSDPA (High-Speed Downlink Packet Access), HSUPA is considered 3.75G.

IAD (INTEGRATED ACCESS DEVICE)

A device that consolidates voice, data, Internet, and video services using DSL, ATM, TDM, or MGCP over T-1/E-1 lines. There are also IADs that connect to the service provider over wireless links like WiMAX. The IAD is a modern form of a CPE, i.e. a device that is installed in end-users' homes.

ICE (INTERACTIVE CONNECTIVITY ESTABLISHMENT)

A methodology for Network Address Translator (NAT) traversal for the Session Initiation Protocol (SIP). ICE makes use of existing protocols, such as Simple Traversal of UDP Through NAT (STUN), Traversal Using Relay NAT (TURN) and even Real Specific IP (RSIP). ICE works through the mutual cooperation of both endpoints in a SIP dialogue.

IETF (INTERNET ENGINEERING TASK FORCE)

Formed in 1986, the IETF sets the technical standards that run the Internet. IETF working groups seek the advice of the Internet community through RFCs (Requests For Comment), and then submit recommendations to the IETF for final approval.

IMEI (INTERNATIONAL MOBILE STATION EQUIPMENT IDENTITY)

A unique number allocated to each individual mobile station equipment in the PLMN and unconditionally implemented by the mobile station manufacturer.

INTERWORKING WLAN (I-WLAN)

A WLAN that interworks with a 3GPP system.

IMS (IP MULTIMEDIA SUBSYSTEM)

A standardized Next Generation Networking (NGN) architecture for telecom operators that want to provide mobile and fixed multimedia services. It uses a VoIP implementation based on a 3GPP standardized implementation of SIP, and runs over the standard Internet Protocol (IP). It supports both packet-switched and circuit-switched existing phone systems. The aim of IMS is not only to provide new services but all the services, current and future, that the Internet provides. IMS uses open standard IP protocols, defined by the IETF. IMS truly merges the Internet with the cellular world; it uses cellular technologies to provide ubiquitous access and Internet technologies to provide appealing services. Because it is access network independent, IMS enables converged fixed mobile network.

IMTC

An international community of companies working together to promote and facilitate the development and use of interoperable, real-time, multimedia telecommunications products and services based on open international standards.

INCOMING CALL ROUTING

See DID, MSN, Sub-Addressing, TCS4.

INSTANT MESSAGING (IM)

Instantly communicating between two or more people over a network, such as the Internet. Instant messaging requires the use of a client program that hooks up an instant messaging service and differs from e-mail in that conversations are then able to happen in real-time. Most services offer a presence feature, indicating whether

people on one's list of contacts are currently online and available to chat.

INTERACTIVE VIDEO PLATFORM

A scalable carrier-grade platform for service providers that enables easy creation of interactive multimedia (video and voice) services, based on advanced media processing capabilities. The platform complies with all voice/video IP standards, and protocols and can be part of H.323, SIP or IMS networks. Many services can be implemented on top of this platform; examples include video portals, video chat rooms, video dating services, video contact center applications, surveillance applications, mobile video conferencing and more.

INTERNET PHONE

A special Voice over IP (VoIP) terminal that is designed to be used over the Internet. Typically, it uses the Session Initiation Protocol (SIP) or Media Gateway Control Protocol (MGCP).

IP-CAN (IP-CONNECTIVITY ACCESS NETWORK)

The collection of network entities and interfaces that provides the underlying IP transport connectivity between the UE and the IMS entities. An example of an IP-Connectivity Access Network is GPRS.

IP ADDRESS

The unique address of a computer attached to a TCP/IP network. IP addresses are 32 bits long. Each octet is represented in decimal and is separated by dots.

IP MULTICAST

A means of simultaneous transmission of data from a server to a group of selected users on a TCP/IP network, (internal, intranet or Internet). IP multicast is used for streaming audio and video over the network.

IP NETWORK

A network that uses the TCP/IP protocol.

IP TELEPHONY

A set of technologies that enables voice, data and video collaboration over existing IP-based LANs, WANs, and the Internet. IP telephony uses open IETF and ITU standards to move multimedia traffic over any network that uses IP.

IPTV (INTERNET PROTOCOL TELEVISION)

A system where a digital television service is delivered to subscribing consumers using the Internet Protocol over a broadband/mobile connection. The DVB family of protocols utilize IPTV as their media transport mechanism.

ISDN (INTEGRATED SERVICES DIGITAL NETWORK)

An entirely digital telephone network that allows both data and voice communications over the same line. ISDN replaces the old analog local loop and operates at significantly faster speeds than the traditional telephone service.

ISDN ROLLOVER

ISDN Rollover feature ensures that a call is completed even when

call volume is high. ISDN Rollover requires support by the PSTN switch application and presumes the availability of a pool of stacked Gateways across the managed network.

ISMA (THE INTERNET STREAMING MEDIA ALLIANCE)

A non-profit organization whose mission is to accelerate the market adoption of open standards for streaming and progressive download of rich media over all types of Internet Protocols (IP).

ISUP (ISDN USER PART)

Part of the Signaling System #7 (SS7) which is used to set up telephone calls in Public Switched Telephone Networks (PSTN). It defines the protocol and procedures used to set up, manage, and release trunk circuits that carry voice and data calls over the PSTN.

ITU (INTERNATIONAL TELECOMMUNICATIONS UNION)

The most important telecom standards setting organization in the world. With headquarters in Geneva, the ITU is an international organization founded in 1865, now part of the United Nations System that sets communications standards for global telecom networks.

ITU-T

The Telecommunication Standardization Sector of the ITU, developing standards for interconnecting telecommunication equipment across networks.

IVR (INTERACTIVE VOICE RESPONSE)

A two-stage incoming call routing method supported by the Gateway.

It enables DID to a LAN terminal, even when the ISDN lines do not have multiple numbers allowing direct dialing to an endpoint.

IVVR (INTERACTIVE VOICE AND VIDEO RESPONSE)

Enhances the familiar (but audio-only) interactive voice response (IVR) self-service environment with video content to accompany the audio messages.

JAIN (JAVA APIs FOR INTEGRATED NETWORKS)

A set of APIs focused on emerging network protocols and architectures driven by convergence of traditional telecommunication and IP networks. Defined by the Java Community Process.

JITTER

The result of a temporary change in latency of packet delivery over a communication channel. Jitter will cause packets to arrive to their destination in a fluctuating manner.

JITTER BUFFER

A portion of memory specifically allocated to storing IP packets awaiting transmission, or to storing received IP packets. The buffer facilitates flow control by capturing IP packets and then transmitting packets as “playback” using speeds and rates of delay that the destination device can handle, without causing packet loss through overloading.

JITTER BUFFER MANAGEMENT

Jitter buffer management represents the trade-off between a larger buffer and increased rates of jitter.

LAN (LOCAL AREA NETWORK)

An organization's internal communications network.

LATENCY

A measure of accumulated waiting time or delay, representing the length of time required for information to pass through a network.

LBS/LCS (LOCATION BASED SERVICES/ LOCATION SERVICES)

A set of services (e.g. advertising, emergency) based on the customer location information retrieved from the cellular network.

LDAP (LIGHTWEIGHT DIRECTORY ACCESS PROTOCOL)

A protocol for accessing online directory services. LDAP is both an information model and a protocol for querying and manipulating the model.

LEASED LINE

A dedicated connection providing a telecommunications link for voice, data and Internet traffic. Leased lines are delivered along predetermined routers and enable control over, for example, QoS, bandwidth allocation and connection speed.

LINE HUNTING

A Gateway supports a list of prefixes (services). Gateway unavailability to receive a call means that the Gateway cannot accept calls with the particular prefix in question. The Line Hunting function of a Gatekeeper searches for a Gateway which is free to accept calls with this prefix.

LOAD BALANCING

A technique to spread work between many computers, processors, disks or other resources in order to get optimal resource utilization and decrease computing time.

MAC (MEDIA ACCESS CONTROL)

A system of rules used to move data from one physical medium to another.

MASTER

The side in communications which initiates and controls the session. The “slave” is the other side that responds to the master’s commands.

MCU (MULTIPOINT CONFERENCING UNIT)

A device that manages a multipoint conference by connecting the multiple sites and stations in the same video conference. The MCU can be used in conjunction with a Gateway to connect H.320 and 3G-324M video conference endpoints. The MCU combines video, audio and data streams from multiple conference endpoints into one multi-location, interactive session.

MEDIA GATEWAY

Acts as a translation unit between disparate telecommunications networks such as PSTN, Next Generation IP Networks, 2G, 2.5G and 3G radio access networks or PBX. Media Gateways enable multimedia communications over multiple transport protocols, such as ATM and IP.

MGCP/MEGACO (MEDIA GATEWAY CONTROL PROTOCOL / MEDIA GATEWAY CONTROLLER)

An IP telephony signaling protocol from the IETF. MGCP was the original protocol, which evolved into MEGACO. Both protocols are designed for implementation in IP phones and endpoints that are less expensive than SIP or H.323 phones.

MGW

See Media Gateway.

MGWC (MEDIA GATEWAY CONTROLLER)

Also known as MGC, this system is used in some VoIP architectures. The media gateway controller handles the registration and management of media resources at the media gateway using MGCP or MEGACO. The media gateway controller exchanges messages with central office switches via a signaling gateway. It is also sometimes called a call agent, call controller, or softswitch.

MIB (MANAGEMENT INFORMATION BASE)

An SNMP structure that describes the particular device being monitored. Networking element typically includes Public MIB (mandatory) and Private MIB (vendor specific).

MMS (MULTIMEDIA MESSAGING SERVICE)

A mobile system application enabling message-based exchange of multimedia content. MMS messages can be generated by mobile users with handsets running an MMS client. The MMS standards are part of the 3GPP standards that enable message composition of text, voice, video and graphics.

MNP (MOBILE NUMBER PORTABILITY)

The ability of a mobile subscriber to change subscription networks within the same country while retaining original MSISDN(s).

MOBILITY

The ability for the user to communicate while moving; independent

of location.

MPEG-4

A series of standards developed by the Moving Pictures Experts Group, designed to reduce the storage requirements of digital video. MPEG-4 provides the standardized technological elements for the integration of interactive graphics applications and interactive multimedia.

MPU (MEDIA PROCESSING UNIT)

A component of an Interactive Video Platform. Capable of accepting calls from video terminals, such as 3G mobile telephones, while connecting them to live or stored/streamed video content. It also has extensive media processing capabilities. The MPU can connect to multiple streams in parallel via multiple protocols and produce many video layouts.

MRCP (MEDIA RESOURCE CONTROL PROTOCOL)

A communication protocol, created by the IETF, which allows speech servers to provide various speech services, such as speech recognition, speech synthesis, etc.

MRFP (MEDIA RESOURCE FUNCTION PROCESSOR)

An IMS element that implements service media processing, such as in conferencing servers. It uses advanced DSP technologies to provide voice/video mixing, transcoding, text overlay and service announcements.

MRFC (MEDIA RESOURCE FUNCTION CONTROLLER)

An IMS element that controls one or several MRFPs. It uses SIP/IMS to interface the CSCF and various Application Servers to provide media processing services for different applications.

MSCP (MEDIA SERVER CONTROL PROTOCOL)

A protocol to control media server functions for interactive media and conferencing functions. The protocol also provides extensibility mechanisms allowing messages which are defined outside this document to be passed using the MSCP protocol.

MSISDN

The mobile equivalent of ISDN. The standard international telephone number used to identify a given mobile subscriber. The number is based on the ITU-T E.164 standard.

MSML

A proposed alternative protocol to the W3C-approved CCXML and VXML for the interface between the IMS application server and the MRFC.

MSN (MULTIPLE SUBSCRIBER NUMBER)

A method of incoming call routing in which a group of phone numbers is assigned to a particular ISDN line by the telephone company. PRI ISDN lines are usually assigned multiple numbers in the US and in Europe.

MSRP (MESSAGE SESSION RELAY PROTOCOL)

The preferred IMS protocol for instant messaging (IM). This extension enhances IM, as messages flow end-to-end and not through the server chain.

MULTIPOINT

Communication configuration in which several terminals or stations are connected. In contrast with point-to-point.

MULTICAST

See IP-Multicast.

MULTI-UNICAST

Transmission of duplicate data streams, one to each user. In multi-unicast, multiple users request the same data from the same server at the same time. Contrast with IP multicast, unicast.

MULTIPLEXER

A device that encodes information from two or more data sources into a single channel.

NAT (NETWORK ADDRESS TRANSLATION)

NAT devices translate IP addresses so that users on a private network can see the public network, but public network users cannot see the private network users.

NEIGHBOR GATEKEEPER

A mechanism by which a Gatekeeper optimizes inter-zone communication. A list of Neighbor Gatekeepers and their IP addresses allows the Gatekeeper to resolve destination IP addresses when the source endpoint is not in the same zone as the destination endpoint.

NETWORK

A group of stations (computers, telephones, or other devices) connected by communications facilities for exchanging information. Connection can be permanent, via cable, or temporary, through telephone or other communications links. The transmission medium can be physical (fiber optic cable) or wireless.

NSF (NETWORK SPECIFIC FACILITY)

The Network Specific Facility Information Element (NSF IE) feature enables system administrators to coordinate their network and service requirements with Service Providers.

NUMBER PORTABILITY

Where the provision of numbers dialed is independent of home environment and/or serving network.

PACKET

A block of data used for transmission in packet-switched systems.

PACKET LOSS

The higher the packet loss, the lower voice or video quality. Packet loss is typically measured in percentage of packets lost versus the total number of packets sent by the source. There can be various reasons for packet loss, such as noise on the communication channels, transmission errors, higher packet rate than available throughput, and more. Also some devices might not have adequate buffers or CPU power to handle incoming packets, and as a result some may be intentionally discarded by the device, the network or in between.

PACKET REORDERING

Packet reordering ensures that all packets reach their destination in the correct sequential order. An event of packet disordering can occur when two packets belonging to one stream take different routes to their destination, such that a packet sent later than its peer may arrive to the destination earlier.

PACKET SWITCHING

A networking technology that routes messages according to layer-2 headers to their required destination.

PARENT FILTERS

When a Gatekeeper fails to resolve a destination address, the Gatekeeper searches for the destination first among its Children, then among its Neighbors and then via its Parent. Parent filters enable the Gatekeeper to avoid unnecessary searches among its

Children and Neighbor Gatekeepers.

PARTY NUMBER

The dialing number of an endpoint. This number can be a telephone number or a number used by other mechanisms on various networks, such as telex and ISDN.

PBX (PRIVATE BRANCH EXCHANGE)

A small or medium company-owned telephony switching system. Used to enable direct dialing between endpoints on the corporate LAN, or for long-distance direct dialing between endpoints.

POC (PACKETS OVER CELLULAR)

See Push To Talk Over Cellular.

POINT-TO-POINT

Communication configuration in which communication is between two stations only. In contrast with multipoint.

POLICY

An ordered combination of policy rules that defines how to administer, manage, and control access to resources.

POP (POINT OF PRESENCE)

The point at which a line from a long distance carrier connects to the line of the local telephone company, or to the user if the local company is not involved. For online services and Internet providers, the POP is the local point into which users access the Internet.

PORT

A pathway into and out of a computer or network device, such as a switch or router.

PORTAL

An entity that aggregates access to services and content for easy reach to customers, including service level authorization and single Sign On (SSO) functions.

POST PAY BILLING

Billing arrangement between subscribers and service providers where the subscriber periodically receives a bill for service usage in the past period.

POTS (PLAIN OLD TELEPHONE SERVICE)

PREDEFINED ENDPOINT

An endpoint entitled to register with a specified Gatekeeper.

PREFIX

A prefix is part of the dialing sequence used to access a service or conference type. See also Gateway Supported Prefixes and Conferencing Service.

PREPAY BILLING

Billing arrangement between subscriber and service provider where the subscriber deposits an amount of money in advance, which is subsequently used to pay for service usage.

PRESENCE INFORMATION

In computer and telecommunications networks, presence information conveys status on a party's availability and willingness to communicate. A client publishes presence information to other systems' users to convey its communication state. Presence information has wide applications in Voice over IP and Instant Messaging.

PRI (PRIMARY RATE INTERFACE)

An ISDN subscriber interface supporting 23 bearer B channels at 64 Kbps and two data D channels at 16 Kbps (in North America), or 30 bearer B channels at 64 Kbps and two data D channels at 16 Kbps (in Europe). A PRI connection runs at 1.544 Mbps per second over a T-1 connection, and at 2.048 Mbps per second over an E-1 connection.

PROXY SERVER

An application that breaks the connection between sender and receiver. All input is forwarded out on a different port, closing a straight path between two networks and preventing a hacker from obtaining internal addresses and details of a private network.

PSTN (PUBLIC SWITCHED TELEPHONE NETWORK)

The worldwide voice telephone network. Once only an analog system, most telephone networks today are digital. In the US, most of the remaining analog lines are the ones from your house or office to the telephone company's central office.

PUSH TO TALK OVER CELLULAR

A walkie-talkie type service provided over a cellular phone network.

The application uses the packet network for instant one-to-one conversations

Q.931

A telephony protocol for Call Signaling, consisting of Setup, Teardown and Disengage. Q.931 is included in the H.225.0 Recommendation.

Q.931 + H.245 ROUTED MODE

The routing of the Call Setup channel (Q.931) and the Control channel (H.245) through the Gatekeeper. As opposed to direct mode, where the Gatekeeper is not involved in routing these protocols. See also Routed Mode.

QCIF (QUARTER COMMON INTERMEDIATE FORMAT)

A standard video format used in mobile video conferencing (176 x 44 pixels), where the phone's processing power and power consumption (and the display) limit the video resolution. QCIF also requires only a quarter of the transmission bandwidth required for CIF video resolution.

QOS (QUALITY OF SERVICE)

The ability to define a level of performance in a data and real-time communications system. For example, ATM networks specify modes of service that ensure optimum performance for traffic, such as real-time voice and video. QoS is needed to ensure end-to-end delivery of packets to guarantee a satisfactory user experience.

QSIF

A video resolution equivalent to QQVGA. 160*120.

RADIUS (REMOTE ACCESS DIAL-IN USER SERVICE)

A server for Authentication, Authorization and Accounting (AAA) of endpoints and endpoint aliases.

RAS (REGISTRATION, ADMISSION & STATUS)

In an H.323 audio or video system, the RAS protocol is a control channel over which H.225.0 signaling messages are sent.

REAL-TIME

The processing of information that returns a result so rapidly that the interaction appears to be instantaneous.

Video conferencing is an example of a real-time communication application.

REDUNDANCY

Duplicating equipment components to achieve high availability of telcom equipment and service continuity.

REGISTERED ENDPOINT

A registered endpoint is an endpoint that has informed the Gatekeeper that it is online, active and ready to receive calls, and has received confirmation from the Gatekeeper of its registration request.

ROAMING

The ability for a user to function in a serving network different from the home network. The serving network could be a shared network operated by two or more network operators.

ROUTED MODE

The routing of the Call Setup channel (Q.931) and the Control channel (H.245) through the Gatekeeper. See also Q.931 + H.245 Routed Mode.

ROUTER

A device or setup that finds the best route between any two networks, even if there are several networks to traverse. Like bridges, remote sites can be connected using routers over dedicated or switched lines to create WANs.

RTP/RTCP (REAL TIME TRANSPORT PROTOCOL / REAL TIME CONTROL PROTOCOL)

Real Time Transport Protocol / Real Time Control Protocol - RTP is an IP protocol that supports real-time transmission of voice and video. It is widely used for IP telephony. RTCP is a companion protocol that is used to maintain QoS.

RTP REDUNDANCY

A method of overcoming packet loss by doubling packet payload without increasing the number of packets sent. See also FEC.

RTSP (REAL TIME STREAMING PROTOCOL)

A protocol for use in streaming media systems which allows a client to remotely control a streaming media server.

SCCP (SKINNY CLIENT CONTROL PROTOCOL)

A proprietary terminal control protocol, originally developed by Selsius Corporation, now owned and defined by Cisco Systems, Inc. as a messaging set between a skinny client and the Cisco CallManager.

SCTP (STREAM CONTROL TRANSMISSION PROTOCOL)

A Transport layer protocol defined in 2000 by the IETF Signaling Transport (SIGTRAN) working group, equivalent in a sense to TCP or UDP. It provides some similar services as TCP, ensuring reliable, in-sequence transport of messages with congestion control.

SDP (SESSION DESCRIPTION PROTOCOL)

A format for describing streaming media initialization parameters intended for describing multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation.

SDU (SERVICE DATA UNIT)

A set of data that is sent by a user of services of a given layer, and is transmitted to a peer service user semantically unchanged. The SDU is the data that a certain layer will pass to the layer below.

SERIAL INTERFACE

A channel that transfers digital data in a serial fashion, one bit after the other over one wire or fiber. The serial port on a PC is a serial interface that is used to attach modems and scanners. Serial interfaces may have multiple lines, but only one is used for data. See also V.35 and X.21.

SERVICE PREFIX

The prefix identifies the service and can usually be a numeric code, an alphanumeric string, a name, or phone number.

SERVICES

A service is a function that is supported by a subset of endpoints in a zone. Access a service by dialing a prefix attached to the name or phone number. Services allow you to dynamically add more resources, such as a Gateway, into the system.

SGW (SIGNALING GATEWAY)

A network component solely responsible for translating signaling messages between one medium (usually IP) and another (PSTN). SGW typically handles SS7 protocol conversion to IP telephony messages.

SIF

A resolution equivalent to QVGA — 320*240. This resolution is common in security cameras and countries where NTSC is used.

SIGCOMP

Signaling compression between UE and PCSCF as defined in RFC 3320 and related standards.

SIGTRAN

The name given to an IETF working group that produced specifications for a family of protocols that provide reliable datagram service and user layer adaptations for SS7 and ISDN communications protocols.

SILENCE SUPPRESSION

Silence information within the audio stream can consume LAN bandwidth and burden voice processing. Using compression techniques, Silence Suppression can greatly reduce the wasted bandwidth in a multipoint conference and on congested networks.

SIMPLE (SESSION INITIATION PROTOCOL (SIP) FOR INSTANT MESSAGING AND PRESENCE LEVERAGING EXTENSIONS)

An application of the SIP protocol for server-to-server and client-to-server interoperability in Instant Messaging. SIMPLE is a step in bringing standardization to Instant Messaging.

SINGLE SIGN-ON

Ability for end users to move easily among services without having to repeatedly identify themselves with a new password.

SIP (SESSION INITIATION PROTOCOL)

An IP telephony signaling protocol developed by the IETF. SIP is a text-based protocol that is suitable for integrated voice-data applications. SIP is designed for video, voice and data transmission. It uses fewer resources and is considerably less complex than H.323.

SIP-CX (SIP CONFERENCING EXTENSION)

A Microsoft proprietary protocol used to control conferencing systems. This is being replaced by CCCP.

SIP SERVER

A SIP network element that receives SIP messages from other SIP

network elements and forwards the messages or provides SIP responses.

SKINNY

See SCCP.

SLAVE

The side in communications which responds to session commands. The “master” is the other side that initiates and controls the session.

SMARTPHONE

Any electronic handheld device that integrates the functionality of a mobile phone, personal digital assistant (PDA) or other information appliance. A key feature of a smartphone is that additional applications can be installed on the device, developed by the device manufacturer, operator or by any other third-party software developer.

SMIL (SYNCHRONIZED MULTIMEDIA INTEGRATION LANGUAGE)

SMIL is a W3C recommendation for describing multimedia presentations using XML. It defines timing markup, layout markup, animations, visual transitions, media embedding and more.

SNMP (SIMPLE NETWORK MANAGEMENT PROTOCOL)

A widely used network monitoring and control protocol.

SOAP (SIMPLE OBJECT ACCESS PROTOCOL)

A protocol for exchanging XML-based messages over a computer, network, normally using HTTP. SOAP forms the foundation layer of

the Web services stack, providing a basic messaging framework that more abstract layers can build on.

SRTP (SECURE REAL-TIME TRANSPORT PROTOCOL)

Defines a profile of RTP (Real-time Transport Protocol), intended to provide encryption, message authentication and integrity, and replay protection to the RTP data in both unicast and multicast applications.

SQCIF (SUB QUARTER COMMON INTERMEDIATE FORMAT)

A video format usually used to refer to video with dimensions of 88 x 72 pixels.

SS7 (SIGNALING SYSTEM 7)

A signaling protocol that carries telephony signaling information between central office switching systems.

STB (SET-TOP-BOX)

A device used in the TV industry to access the CableTV network.

STREAMING

A method of delivering multimedia content in one direction over IP networks. The content can be stored or live.

STUN (SIMPLE TRAVERSAL OF UNDERNEATH NATS)

A network protocol allowing clients behind NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up communication between two hosts that are both behind NAT routers.

SUB-ADDRESSING

Sub-addressing is a one-stage DID dialing mechanism in which a phone sends two numbers. One number is for routing on the circuit-switched network. The other number is forwarded to the Gateway inside a Q.931 Sub-addressing Information Element for IP address resolution by Gatekeeper/SIP Server.

SUB-CONFERENCE

Audio sub-conferences to which selected participants in an existing conference can be temporarily diverted by the MCU. When the sub-conference is over, the participants return to the main conference.

SUBNET

A subnet is a portion of an IP network defined by a subnet mask. Devices on the same subnet have the same subnet mask.

SUPPLEMENTARY SERVICES

The ITU-T H.450 Recommendation comprises a number of separate recommendations for implementing Supplementary Services of the H.450.x series.

T.120 DATA STANDARD

Data sharing protocol for multipoint data communication in a multimedia conferencing environment. T.120 enables white board collaborations, file transfers, graphic presentations and application sharing between participants in a conference.

T-1

The interface for PRI ISDN lines used in the USA and some Asia Pacific countries. A T-1 connection has up to 23 B channels and 1 D channel, and transmits at rates of up to 1.544 Mbps.

TCP

Transmission Control Protocol. A reliable transport layer on top of IP. TCP provides transport functions which ensure that the total amount of bytes sent is received correctly by the called party.

TCS4

TCS4 is a special routing method for incoming H.320 video calls. TCS4 allows direct inward dialing to an endpoint on the IP network via the Gateway when DID is not available.

TD-SCDMA (TIME DIVISION-SYNCHRONOUS CODE DIVISION MULTIPLE ACCESS)

A 3G mobile telecommunications standard, being pursued in the People's Republic of China by the Chinese Academy of Telecommunications Technology (CATT), Datang and Siemens AG.

TERMINAL EQUIPMENT

Equipment that provides the necessary functions to access networks, including operating adequate signaling protocols and media streams.

TISPAN (TELECOMS & INTERNET CONVERGED SERVICES & PROTOCOLS FOR ADVANCED NETWORKS)

Formerly Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) is a standardization body of ETSI, specializing in fixed networks and Internet convergence.

TLS (TRANSPORT LAYER SECURITY)

A security solution that provides authentication and encryption of data streams, usually over IP.

TOPOLOGY ISLANDS

IP subnets, characterized by homogeneous and fast LAN connectivity. Dividing the network into islands enables a Gatekeeper to direct calls through the most optimal routes, thus avoiding slow connections or bottlenecks as much as possible.

TPKT

A standard way of defining blocks of data in a TCP stream, since TCP does not have delimiters. During configuration you can define the maximum number of TPKT channels allowed.

TRANSCODING

Transcoding is the conversion of one transmission format into another using various algorithms to achieve wider support of formats at

different quality levels.

TTL (TIME TO LIVE)

A set maximum amount of time a packet is allowed to propagate through the network before it is discarded. TTL is a time, typically in seconds, after which the fragment can be deleted by any device on the network. In practice, on the IP network the TTL field is reduced by one on every hop.

TTS (TEXT TO SPEECH)

A method to automatically convert text to speech.

TURN (TRAVERSAL USING RELAY NAT) PROTOCOL

A far-end NAT (Network Address Translation) technique. The media traffic must pass central points on public Internet visible from all sides participating in the SIP session.

UCF (UNREGISTRATION CONFIRM MESSAGE)

AN RAS message that a Gatekeeper or an endpoint sends accepting the URQ.

UDP (USER DATAGRAM PROTOCOL)

A transport protocol within the TCP/IP protocol suite that is used in place of TCP when a reliable delivery is not required.

UE (USER EQUIPMENT)

Equipment at either end of a communications link that links to the network, otherwise known as endpoints.

UMA (UNLICENSED MOBILE ACCESS)

Former name for the Generic Access Network (GAN), describes a telecommunication system allowing seamless roaming and handover between local area networks and wide area networks using the same dual-mode mobile phone.

UMTS (UNIVERSAL MOBILE TELECOMMUNICATIONS SYSTEM)

One of the third-generation (3G) mobile phone technologies. It uses W-CDMA as the underlying standard, is standardized by the 3GPP, and represents the European/Japanese answer to the ITU IMT-2000 requirements for 3G Cellular radio systems.

UNICAST

An addressing form for sending information to a single destination, in contrast to broadcast or multicast where packets are addressed to more than one destination.

UPLINK

A unidirectional radio link for the transmission of signals from a UE to a base station, from a Mobile Station to a mobile base station or from a mobile base station to a base station.

URL (UNIFORM RESOURCE LOCATOR)

An Internet address. The address that defines the route to a file on a computer connected to the Internet.

USER AGENT

Any software or device that acts on behalf of a user, interacting with other entities and processing resources.

USIM (UNIVERSAL SUBSCRIBER IDENTITY MODULE)

An application for UMTS mobile telephony networks residing on the UICC smart card used for accessing services provided by mobile networks, which the application is able to register on with the appropriate security. Stores user subscriber information, authentication information and provides storage space for text messages.

UTRAN (UNIVERSAL TERRESTRIAL RADIO ACCESS NETWORK)

A conceptual term identifying that part of the network which consists of RNCs and Node Bs between Iu and Uu interfaces.

VAD (VOICE ACTIVITY DETECTION)

Used in speech encoding software to determine if the voice being encoded is human speech or background noise.

VAS (VALUE ADDED SERVICE)

A telecommunication/information service that is offered in addition to and/or in conjunction with a basic telecommunication/data service.

V.35

An ITU standard for group band modems that achieve higher data rates by combining the bandwidth of several telephone circuits. Typically used for modem-to-multiplexer communication. See also Serial Interface and X.21.

VIDEO BIT RATE

Bit rate is the speed at which bits are transmitted, in bits per second.

VIDEO BLOG

A vlog or video blog is a blog (short for weblog) which uses video as the primary content. The video is linked within a video blog post and usually accompanied by supporting text, image, and additional metadata to provide context.

VIDEO BRIDGE

Computerized switching system which allows multipoint video conferencing.

VIDEO CHAT

A visually-enhanced online chat session.

VIDEO CONFERENCING

Communication across long distances with video and audio contact that may also include graphics and data exchange. Digital video transmission systems typically consist of camera, codec (coder-decoder), network access equipment, network, and audio system.

VIDEOPHONE

A videophone is a telephone which is capable of both audio and video duplex transmission.

VIDEO MAIL

Used in IP endpoint or to 3G mobile calling. When a destination is busy or receives no answer, the video call service picks up the call and broadcasts a pre-recorded message stating that the owner of the destination 3G mobile is unavailable. The video mail then begins recording the incoming video conference.

VIDEO ON DEMAND STREAMING

Delivery of a Video on Demand stream to a viewer upon request at any given time. Contrast this to a real-time stream that is delivered when the conference is in progress.

VIDEO TRANSRATING

Video Transrating adjusts the bitrate of video received from an IP network to maintain video quality on mobile networks. A key requirement for the connectivity of broadband and mobile networks.

VISITED NETWORK

In roaming, the network delivering the service on behalf of the home network.

VLAN (VIRTUAL LOCAL AREA NETWORK)

A method of creating independent logical networks within a physical network. Several VLANs can coexist within such a network. Used in many cases for media and management separation.

VOICE-ACTIVATED VIDEO SWITCHING

Automatic switching of a video image viewed at each conference terminal according to the voice level of each participant.

VOIP (VOICE OVER IP)

See IP Telephony.

VPN (VIRTUAL PRIVATE NETWORK)

VPN modules create closed secure tunnels for communication between two firewalled LANs. VPN technology is one of the approaches being used today for providing secure communications over IP networks.

VXML

An XML language (Voice & Video XML) - a simple, yet powerful XML-based API, enabling the implementation of diverse video and voice services, while keeping the development task simple by relieving the developer from the need to deal with underlying complexities. It is based on HTTP protocol.

WAN (WIDE AREA NETWORK)

A communication network spread over a wide geographical area.

WAP

An open international standard for applications that use wireless communication to enable access to the Internet from a mobile phone or PDA.

W-CDMA (WIDEBAND CODE DIVISION MULTIPLE ACCESS)

A type of 3G cellular network. A wideband spread-spectrum 3G mobile telecommunication air interface that utilizes code division multiple access (or CDMA the general multiplexing scheme, not to be confused with CDMA the standard).

WHITEBOARD

A term used to describe the placement of shared documents on an on-screen "shared notebook." See also Document Sharing.

WIBRO (WIRELESS BROADBAND)

A wireless broadband internet technology being developed by the Korean telecoms industry.

WIMAX (WORLDWIDE INTEROPERABILITY FOR MICROWAVE ACCESS)

Defined to promote conformance and interoperability of the IEEE 802.16 standard. The Forum describes WiMAX as "a standards-based technology enabling the delivery of last mile wireless broadband access as an alternative to cable and DSL."

WNSRP (WINDOWS NUMBERED SIMPLE RETRANSMISSION PROTOCOL)

A standardized call setup reduction solution for real-time 3G multimedia services, supported by major operators, handset manufacturers, and suppliers.

WILDCARD DIGIT MANIPULATION

A mechanism which enables a Gatekeeper to manipulate an incoming call source number before searching for the destination endpoint.

X.21

An ITU standard protocol for a circuit switched network. See also Serial Interface and V.35.

**XCAP (EXTENSIBLE MARKUP LANGUAGE (XML)
CONFIGURATION ACCESS PROTOCOL)**

Allows a client to read, write and modify application configuration data, stored in XML format on a server. XCAP maps XML document sub-trees and element attributes to HTTP URLs, so that these components can be directly accessed by HTTP.

XML (EXTENSIBLE MARKUP LANGUAGE)

A W3C-recommended general-purpose markup language for creating special-purpose markup languages, capable of describing many different kinds of data. It is a simplified subset of Standard Generalized Markup Language (SGML). Used to facilitate the sharing of data across different systems, particularly systems connected via the Internet.

ZONE

An H.323 zone is a logical collection of terminals, Gateways and MCUs managed by a single Gatekeeper. A zone must include at least one terminal and may include several LAN segments connected by routers. See also Gatekeeper Zone.

ZONE PREFIX

Zone prefixes are similar to telephone area codes. If an endpoint in a zone dials a zone prefix before its number, and the Gatekeeper cannot resolve it in its zone, the Gatekeeper attempts to locate and route the call to the appropriate zone of the Neighbor Gatekeeper.

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