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The Technology Guide Series

The Converged Network Infrastructure: An Introductory Guide

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An Introduction to the Converged Network

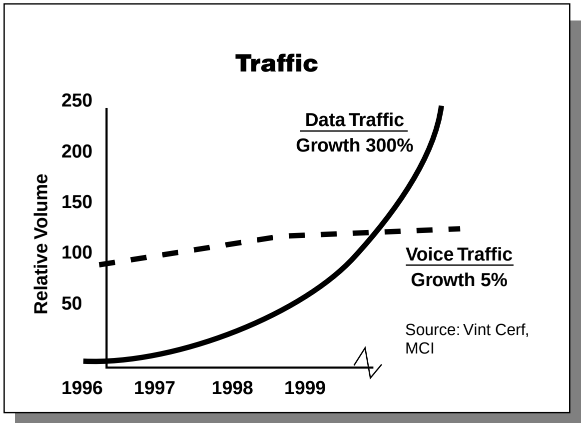
It's a whole new world of communications services, and that world is packet based.

The legacy model, in which voice, video, and data are delivered separately over dedicated, single purpose networks, no longer meets the needs of 21st century businesses. Today's businesses require more dynamic services; and they need versatile computing platforms and real-time, interactive applications to support their new, business-critical needs. As time goes on, it is increasingly clear that the legacy model is just not up to it. Instead business needs are being increasingly supported by a next-generation network that manages diverse traffic types within the context of a converged service in which voice, data, and video coexist on a single, shared facility.

This converged network supports all of the user's traffic types with packet-based protocols such as ATM, Frame Relay, or IP. Of these, the dominant protocol in the access, edge, and core of the network is the Internet Protocol (IP). A managed IP infrastructure for both voice and data transmission is a concept that has begun to revolutionize the industry. It leverages the Internet so that enhanced services can be offered at lower costs, and it is motivating the development of a family of dynamic, next generation, real time applications.

However, it's not clear sailing! As most people recognize, voice and data traffic have significantly different characteristics and are difficult to reconcile within a single network. Data traffic tends to be bursty, consuming large volumes of bandwidth for occasional, short intervals whereas voice traffic is predictable and requires a steady, low-delay, transmission path from end to end.

Figure 1: Comparative Traffic Growth Figures for Voice and Data



Existing Time Division Multiplexing (TDM) circuit-based networks were originally designed to carry these predictable streams of voice traffic but do not efficiently support bursty data traffic. With data growth now outpacing voice, (Figure 1) service providers face the challenge of shedding inefficient TDM infrastructures while preserving the integrity and quality of private-line and voice traffic. This change is leading to a gradual replacement of the existing circuit switched infrastructure with new packet-based equipment.

Two additional factors provide an even more compelling need for service providers to speed up their infrastructure conversion:

- Deregulation and competition between incumbent and new service providers; and
- The worldwide explosion in mobile communications.

Deregulation, Competition and Survival

The worldwide waves of deregulation that have opened up new opportunities for both new and old players in the marketplace have introduced competition at every level of networking. As the race becomes increasingly fierce among communication services companies to acquire and maintain market-share, the importance of rapidly introducing new services while responding to their customers' changing needs becomes a key competitive advantage and a critical aspect of survival.

From a network equipment provider's perspective, this has two key elements: supporting service providers' need for improved time-to-market and enabling them to rapidly implement product innovation. These are essential in order for them to keep up with the pace of change in customer requirements. This is where the converged network, based on Internet protocols and featuring distributed Internet-based applications is superior to traditional methods. Unlike the older communications service model, the converged IP-based model is one in which product innovation and development allows access to Internet application developer skills, with a resulting time-to-market measured in Internet time—months rather than years.

In this new time-constrained environment, network equipment suppliers generally don't have the resources or time to build all of these solutions "from scratch" and so need to outsource much of this activity. For outsourcing to work in this environment, solutions must be based on open standards, so that diverse equipment can work together and networks can properly interoperate. These solutions require standards to be developed that allow equipment vendors to compete for various parts of the overall network solution, thus accelerating the overall pace of change. This move towards outsourced open standard solutions enables the industry to cope with the demand for continuous technological innovation.

Wireless Communications

If the demand for new services in the land based communications space is growing, then the demand for new voice and data services is even more significant for wireless communications. Existing second generation wireless networks (such as GSM) have only limited data capability (typically 9.6 kbps) but new technologies are quickly emerging. The advent of third-generation (3G) wireless communications promises a truly mobile data network with real-time Internet connectivity.

- ***Third Generation Wireless (3G)*** The concept of 3G wireless technologies is a major shift from circuit-switched, voice-only services, to services that are packet-based and multimedia-oriented (voice, data, video, fax). While the rollout of 3G wireless networks across the world will be gradual, there are existing technologies that enhance the current 2G network and which allow better data services to be implemented.

- ***General Packet Radio Service (GPRS)*** offers a technology that supplements today's Circuit Switched Data and Short Message Service on 2G networks. GPRS involves overlaying a packet based wireless interface onto the existing circuit switched network, thus improving data rates up to 115 kbps and enabling more data intensive services.

- ***Enhanced Data for GSM Evolution (EDGE)*** represents the final evolution of data communications within the GSM standard. EDGE uses a modulation scheme that enables data throughput speeds of up to 384 kbps using an existing GSM infrastructure. EDGE offers incumbent GSM operators that do not have third-generation licenses an alternative route to providing data services.

The common thread in all these improvements is the use of packetized voice and IP throughout the network. Thus, wireless communication becomes another major driver for the converged network.

The IP Based Converged Network Model

Voice over IP (VoIP) and other packetized voice/data services can be employed to provide a more cost effective, efficient, and flexible way of building networks. These technologies are based on open standards and provide for the separation of functions such as call control and switching. The distributed nature of VoIP allows innovation and enables service providers to compete for different parts of the network continuum, while at the same time, interoperable standards ensure that the overall network model remains consistent.

The story of the converged network is really built on three key elements:

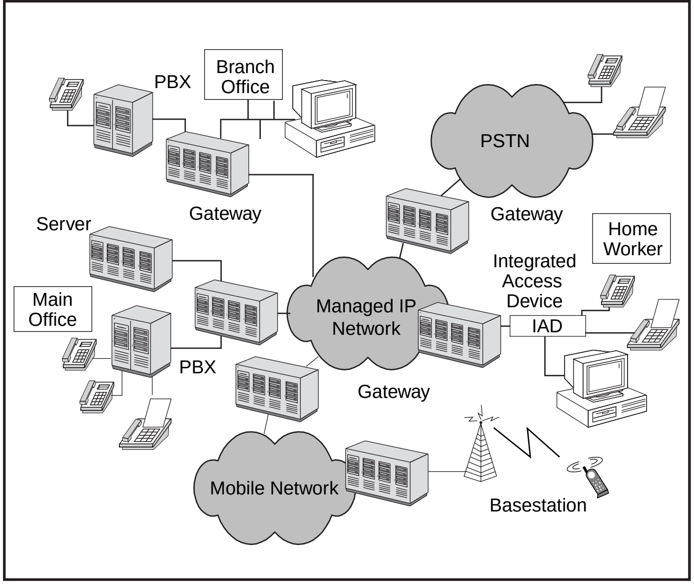
- ***VoIP technology***—this enables voice services to be offered on a data network.
- ***A multi-purpose converged network***—built on a new, functionally distributed, IP-based network architecture.
- ***Open Systems***—a mature set of internationally defined protocols and standards for interoperability and performance assurance.

The converged network allows many different communications systems to interoperate so that users are able to share data and voice services (as shown in Figure 2). So for example, a field worker is able to use his or her mobile phone to communicate through both the mobile network and the IP network in order to access data that is held on the corporate server. These same techniques would also be used for calling the office, customers, or to check voice mail.

Each section of the converged network may use different techniques for handling data, voice, and fax. So at each stage of this process, the network traffic may use a different communications standard. These media streams need to be seamlessly reformatted at each stage—a task that may require a lot of signal processing to achieve. This is the task of a crucial element called a media gateway, and the advent of improved signal processing technology and open standard interconnects has made the deployment of high-density media gateways a reality.

Voice over IP (VoIP): An Enabling Technology

Figure 2 - Voice over IP (VoIP): An Enabling Technology



VoIP is a term that describes the delivery of voice traffic using the Internet Protocol (IP). In general, this means sending it in digital form in discrete packets rather than in the traditional circuit switched protocols of the PSTN.

Voice Quality in VoIP

IP-based data networks are not inherently designed for the real-time bandwidth requirements of speech. Fundamentally, IP packets don't need to arrive at destinations within the narrow time windows needed for acceptable voice quality. And although real-time speech has a reasonably low bandwidth requirement, it cannot tolerate jitter or uneven service levels. While many service providers have adequate capacity to handle real-time voice traffic on their data networks without inhibiting the flow of other non-voice traffic, linear and nonlinear voice compression techniques are still used, particularly when voice is transmitted to the desktop. Nonlinear compression can be a major cause of reduced voice quality.

VoIP networks rely on network processes (often built into gateways) to address some voice-quality problems. For example, silence suppression is used to prevent packets from being created and transmitted during the quiet periods between spoken phrases. Also, Echo Cancellers are needed to eliminate echo that becomes perceptible when delay is introduced. If these kinds of processes do not work properly, then voice quality suffers.

Quality of Service (QoS)

Quality of Service (QoS) is the idea that transmission rates, error rates and other traffic characteristics can be measured, improved, and to some extent guaranteed in advance—an essential function needed in multiservice, "packetized" networks. QoS is of particular concern for the continuous transmission of high-bandwidth video and multimedia information. Transmitting delay sensitive communications such as voice and video in packetized networks, utilizing "best efforts" protocols is often not sufficient and requires more predictive transmission speeds, which are defined by QoS protocols.

Resource Reservation Protocol (RSVP)

RSVP is a QoS enabling protocol. Essentially, it attempts to assure QoS by reserving resources for designated traffic types. Using RSVP, a packet passing through a gateway host can be expedited based on policy and reservation criteria arranged in advance.

VoIP has emerged from the work of the VoIP Forum, an effort by major equipment providers to promote the use of the International Telecommunications Union's (ITU) H.323 standard for supporting voice and video using IP on the public Internet and within intranets.

Benefits of VoIP

Cost Reduction. Most cost reductions using VoIP accrue when the voice traffic travels as IP traffic over an existing IP network such as the Internet.

Simplification. An integrated and converged infrastructure that supports all forms of communication allows more standardization and reduces the total equipment complement. Using IP protocols for all applications reduces complexity and increases services flexibility.

Consolidation. Combining operations, eliminating possible points of failure, and consolidating directory, accounting and security services are just some of the benefits of consolidation.

Advanced Applications. Long term VoIP benefits will be derived from new multimedia and multi-service applications.

The IP-Based Converged Network Elements

The key to the next-generation IP-based converged network is in the way that important functions are broken out into logical components, each of which can be supported by special purpose equipment. In this model, scalable and interoperable solutions can be constructed to meet the different needs of the various service providers at a reasonable cost, while enabling services to be supported uniformly throughout the network. The major advantages of this approach are:

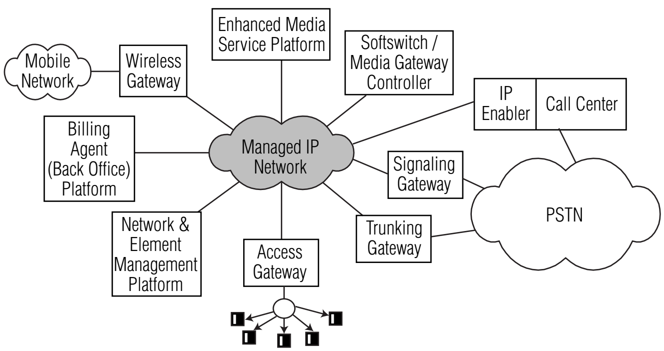
First, providers can improve speed of deployment of their solution through the use of off-the-shelf network elements; with each of which adhering to published and agreed upon international standards that can be custom fit into a unique, yet interoperable and scalable solution.

Secondly, competition among equipment suppliers is promoted by the demand for agreed upon open standards. This keeps costs low and drives innovation so the prohibitive cost barrier to introducing network access is significantly reduced.

Thirdly, separating control and the media elements enables the rapid development of a variety of applications; such as extensive telephony services, vertical market applications and integrated voice/data services, including new contact center capabilities.

Fourth, this approach enables the evolution of a fully managed IP network that supports all of the separate applications and enables integration with the Internet, which further improves the range of available services.

Figure 3



As can be seen in Figure 3, the managed IP network provides gateways into all of the various new functions that are part of the modern network, including wireless and contact centers, as well as traditional PSTN access.

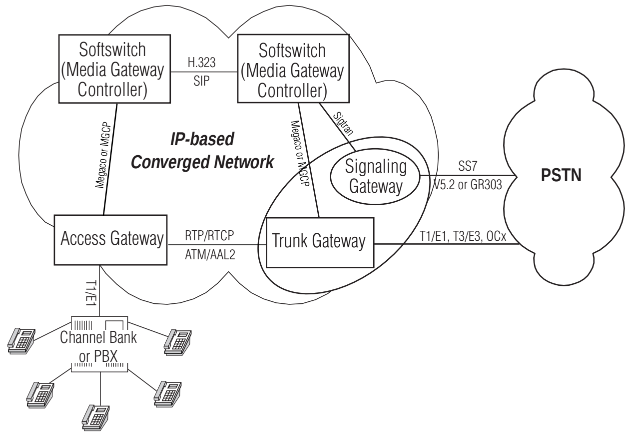
The principal new components of this platform relating to voice services and interworking with existing networks such as the traditional PSTN are:

- Media Gateways that provide interconnection between the IP network and network services, such as those provided by the PSTN and wireless networks.

- Signaling Gateways that translate the call signaling protocols between the various networks.
- Media Gateway Controllers or Softswitches, which provide coordination between gateways according to signaling information they receive from signaling gateways.

Additionally, there are all the normal back-office elements for billing, network management, and customer support that would be required in any service provider network. The following sections describe the typical functions performed by each element, and the open protocols between them.

Figure 4: Elements and Protocols in the Converged Network



Media Gateways

A media gateway is the critical interworking element that translates between networks of differing standards. It provides conversion of streamed media formats such as voice or video, and manages the transfer of information between the different networks. Commonly, a media gateway is required to offer some or all of the following features:

- Multiple network termination types (for example T1/E1 digital trunks and Ethernet or ATM)
- Voice codecs (for example G.729A, G.723.1, PCM, ADPCM, GSM AMR)
- Echo cancellation
- Tone detection and generation (for example DTMF dial tones, call progress tones)

This element requires a high degree of Digital Signal Processing (DSP) capability together with significant packet handling capacity, and as such is a prime candidate for an outsourcing component model to take advantage of the continuous improvements in DSP and network processor technology.

A number of specific media gateway types are defined according to studies performed by ETSI (Tiphon) and the IETF. The most commonly encountered are:

- ***The Access Gateway:*** Access Gateways connect user network interfaces (such as ISDN or traditional analog services) to a VoIP or voice over ATM network. As such, it will typically terminate TDM call signaling and pass this information to a Media Gateway Controller or Softswitch for call control decisions to be made.
- ***The Trunking Gateway:*** Trunking gateways interface between the PSTN telephone network and the IP-based network (or ATM network). Such gateways typically manage a large number of digital virtual circuits and TDM bearer circuits with signaling carried on a separate path (through a signaling gateway).
- ***The Network Access Server:*** The NAS is a specialized form of the Access Gateway designed to terminate modem calls or HDLC connections and provide data access to the IP network.

The Signaling Gateway

The Signaling Gateway (SG) is responsible for termination of Switched Circuit Network (SCN) signaling (typically SS7) and transport of signaling messages to MGC or Softswitch elements across the managed IP network. It also allows remote devices on the IP network to exchange messages with the PSTN network for call setup or for querying SCN database servers which support IN services such as local number portability, toll free numbers, etc. The SG implements the functions and interfaces defined by the SIGTRAN standard.

The Media Gateway Controller or Softswitch

Another key building block is the Media Gateway Controller (MGC)—commonly also referred to as a Softswitch—which defines signaling mediation, call processing and control of media gateways. It consists of software-based functions that reside on a high-availability computing platform and is used to control the flow of voice and data traffic in a converged telecommunications network; mediating signaling between IP, wireless and PSTN sub domains. The Softswitch itself does not perform an actual switching function. Rather, its function is call control and mediation. In this sense, a Softswitch performs many of the common tasks done by its technical predecessor, the Gatekeeper. Call control functions typically include call routing, user authentication, connection control (setup and tear down) and signaling.

A Softswitch is equivalent to the unbundling of call control functionality inherent in a central office switch. It allows interworking between different networks types (i.e. PSTN and packet based) with different signaling systems and media transports.

In regard to the PSTN domain, it must understand call control protocols like SS7, V5, GR-303, and R2. On packet networks it implements H.323 and SIP for call establishment.

Each signaling system has its own unique set of requirements, which make interworking between them complex. The Softswitch can interface to the various signaling networks directly, or via Signaling Gateways. The connection between Softswitch and Signaling Gateways uses the SIGTRAN protocol.

The MGC/Softswitch hides the incompatibilities of different signaling systems and controls the Media Gateways to perform interworking bearer functions. Connections between the Softswitch and Media Gateways employ the MGCP or H.248 (MEGACO) protocols.

Element Control Protocols

The other key element that is used to support VoIP is the set of standards that have evolved over time to address the various problems associated with voice traffic.

H.323

The ITU-T H.323 standard was developed to address call setup and transmission of multimedia voice and video over packet-switched networks that do not guarantee Quality of Service (QoS), such as the Internet and intranets. H.323 utilizes the Real-Time Protocol (RTP/RTCP) from the IETF, along with internationally standardized codecs (the ITU-T G.xxx series such as G.729).

H.323 was the first protocol used in Voice over IP implementations, but pressure is being put on the H.323 community by the IETF's Session Initiation Protocol (SIP), which has gained momentum in the VoIP marketplace due to a simpler call set-up mechanism and better scalability.

To respond to this, the ITU has enhanced the H.323 protocol suite, including improvements for

large-scale deployment and several new annexes that speed up call establishment and deal better with the needs of simple terminals.

H.323 is maintained and improved by the ITU. However, there are many Web sites with excellent tutorials and papers on H.323. To find out more about H.323, a good place to start is on the IMTC Web site (which supports the main interest groups).

ITU-T: www.itu.int/itudoc/itu-t/rec/h/h323.html

IMTC: www.imtc.org

IEC Web Tutorial: www.iec.org/tutorials/h323

SIP

The Session Initiation Protocol (SIP) is an application-layer protocol that can establish, modify and terminate multimedia sessions or calls across an IP network. SIP performs a similar function to H.323 within the new converged network. SIP sessions can include multimedia conferences, distance learning, Internet telephony and similar applications. SIP can invite both persons and "robots," such as a media storage service to both unicast and multicast sessions.

SIP became a proposed standard and was published as RFC 2543 in March 1999. For more information, see the IETF Web site or visit the SIP Forum Web site:

SIP: www.ietf.org/rfc/rfc2543.txt

SIP Forum: www.sipforum.org

MGCP

Media Gateway Control Protocol (MGCP) is used for controlling media gateways from external call control elements (media gateway controllers or call agents). MGCP assumes a call control architecture where the call control intelligence is outside the gateways and handled by external call control elements (usually on Media Gateway Controller elements). A gateway assumes that these call control elements will synchronize with each other to send coherent commands to the gateways under their control.

The MGCP call model views media gateways as a collection of endpoints that may be joined together in a connection. Endpoints can either be physical (such as an analog telephone line or a digital trunk connection) or virtual (such as an RTP stream across a UDP/IP connection). Endpoints can also be internal (for example, a voice announcement service). Communication can be established by connecting endpoints together with appropriate echo cancellation or voice coding.

MGCP is described in the IETF document RFC-2705. See the IETF Web site for more details.

MGCP: www.ietf.org/rfc/rfc2705.txt

MEGACO/H.248

The Media Gateway Control Protocol, (MEGACO) is intended to supersede MGCP as the standard for media gateway control - used between elements of a physically decomposed multimedia gateway. The protocol creates a general framework suitable for gateways, multipoint control units, and interactive voice response units (IVRs).

The connection model used by MEGACO is similar in concept to that employed by MGCP. MEGACO views media gateways as a collection of terminations that can be associated with each other within a particular context. A termination is the source or sink for media streams (which can be multimedia). Like MGCP, terminations can be physical (such as a bearer channel) or virtual (such as an RTP stream). The media stream parameters, including bearer and modem parameters, are part of the termination properties. When a termination is placed in a context with other terminations, communication takes place—the context specifies media translation properties. So for example, a call forwarding application would be implemented by moving a termination from one context to another; a conference would be initiated by joining many terminations together in the same context.

Different Media Gateways implement different terminations and there are many options. So to achieve interoperability, the properties of common termination types are grouped together into packages. MEGACO supports an audit mechanism whereby an MGC can identify supported packages to ensure that connections between multiple MGs will work satisfactorily before starting data flow.

MEGACO was developed within the Media Gateway Control Working Group of the IETF (RFC-3015) but it is a standard that has also been reviewed and approved by the ITU (as H.248) so it has wider overall support than MGCP. See the IETF Web site for more details.

MEGACO: www.ietf.org/rfc/rfc3015.txt

SIGTRAN

SIGTRAN is a suite of protocols and adaptation layers that are used to transport signaling information over IP based networks. It is the key transport component in a distributed VoIP architecture and is used in elements such as the Signaling Gateway, Media Gateway Controller, Gatekeeper or IP-based Service Control Point. SIGTRAN functions can be divided into SCTP and Adaptation Layers:

- SCTP is responsible for reliable signaling transport, streaming, congestion avoidance and control, bundling and un-bundling, multi-homing and association management, plus security and user-transparent fault management.
- The Adaptation Layers transport the signaling information from the corresponding signaling layers using SCTP services. It provides user-data segmentation and bundling as well as protection against masquerade, fraud and repudiation, as well as Congestion Control and generic interfaces to support non-signaling applications.

Sigtran is defined by IETF document RFC 2719—A Framework Architecture for Signaling Transport. See the IETF Web site for more details.

SIGTRAN: www.ietf.org/rfc/rfc2719.txt

Forums and Standards Organizations

A number of standard bodies and global forums are playing key roles in defining and promoting the standards and solutions for next-generation networking, based on packetized multiservice communications. They include:

European Telecommunications Standards Institute (ETSI)

ETSI (the European Telecommunications Standards Institute) is a non-profit organization whose mission is to produce the telecommunications standards that will be used for decades to come throughout Europe and beyond. In particular, the Tiphon body within ETSI is instrumental in defining how the new packet based network should interwork with the existing circuit switched network. Also, the 3GPP body is closely involved with the definition of the 3rd generation wireless network.

Tiphon: www.etsi.org/tiphon

3GPP: www.etsi.org/3gpp

Internet Engineering Task Force (IETF)

The Internet Engineering Task Force (IETF) is a large open international community of network designers, operators, vendors, and researchers concerned with the evolution of the Internet architecture and the smooth operation of the Internet.

The actual technical work of the IETF is done in its working groups, which are organized by topic into several areas (e.g., routing, transport, security, etc.). There are currently more than 80 working groups responsible for developing Internet standards. Since the converged network is based on Internet protocols, most of the working groups have some relevance.

See www.ietf.org

International Softswitch Consortium (ISC)

The International Softswitch Consortium exists to promote worldwide compatibility and seamless interoperability of Softswitch operation. The International Softswitch Consortium was formed in May of 1999 to help advance the formation of open standards and interoperability of Internet-based real-time multimedia applications by establishing a forum for the exchange of ideas among like-minded companies and individuals. The Softswitch Consortium currently numbers more than 140 members, including communications service providers, equipment vendors, and software developers.

See www.softswitch.org

International Telecommunication Union (ITU)

The ITU, headquartered in Geneva, Switzerland is an international organization within which governments and the private sector coordinate global telecom networks and services.

The purpose of the ITU-T is the elaboration, adoption, distribution, and follow-up of recommendations (non-binding standards) in order to standardize telecommunications on a worldwide basis. The main study group that is involved in the converged network protocols is Study Group 16.

See www.itu.org

International Multimedia Telecommunications Consortium (IMTC)

The International Multimedia Telecommunications Consortium, Inc. (IMTC) is a non-profit corporation comprising more than 150 companies from around the globe. The IMTC's mission is to promote, encourage, and facilitate the development and implementation of interoperable multimedia teleconferencing solutions based on open international standards (predominately H.323).

The IMTC's fundamental goal is to bring all organizations involved in the development of multimedia communications products and services together to help create and promote the adoption of the required standards.

See www.imtc.org

The Multiservice Switching Forum (MSF)

The charter of the Multiservice Switching Forum is to develop and promote implementation agreements for protocols and interfaces that enable an open architecture for multiservice switching systems. A Multiservice Switching System is a distributed switching (frame, cell or packet based) system designed to support voice, video, private line, and data such as Asynchronous Transfer Mode (ATM), Frame Relay, and Internet Protocol (IP) services. Multiservice switching systems may use a broad range of access technologies, including traditional Time Division Multiplexing (TDM), Digital Subscriber Line (xDSL), wireless data, and cable modems.

MSF Implementation Agreements define the requirements of the interfaces between components of a MSS and build on the work of the IETF, ETSI, ITU etc. to define specific architectures using the defined elements and protocols discussed in this Guide.

See www.msforum.org

The PCI Industrial Manufacturers Group (PICMG)

The PICMG (PCI Industrial Computer Manufacturers Group) is a consortium of over 500 companies that collaboratively develops specifications that adapt PCI technology for use in industrial and telecommunications computing applications. PICMG Specifications include CompactPCI for Eurocard, rackmount applications and PCI/ISA for passive backplane, standard format cards.

The PICMG-developed standard CompactPCI is a key open standards technology that is enabling equipment suppliers to use off-the-shelf components to accelerate time-to-market.

See www.picmg.org

The Equipment Vendor Challenge

Internet Time

The requirements of next-generation multiservice networks are pushing equipment makers to drive down the cost of ownership of communications equipment, while providing an ever-increasing number of value-added functions and features, while at the same time, driving towards simplicity in function with open service creation environments.

All this is now happening in "Internet Time"—a rapid development cycle more aligned to the creation of software applications than to the long development cycles of communications equipment. Most equipment manufacturers, however, are finding that this can no longer be effectively achieved through isolated in-house efforts. These manufacturers see that by outsourcing specific components and features development to component suppliers they have a cost-effective way of ramping up production quickly with a more rapid time to market. In fact, companies that are careful and selective about what they outsource can actually "future proof" their products by adhering to open standards, both inside and outside their products and equipment.

Take, for example, the notion of developing a standard component such as voice-gateway circuit boards for VoIP network equipment. A network equipment manufacturer that designs VoIP gear might choose to develop these components on its own, using a proprietary approach. But as time goes on, new innovations may demand significantly increased port densities. These innovations may be difficult, or even impossible to incorporate into a proprietary board, but if the component had been built to open standards, it would provide a clear insertion point to accommodate this enhanced feature.

The outsourcing component model encourages greater competition in the OEM supplier space, since an equipment supplier using open standards no longer has to develop everything internally and can instead leverage technology from multiple suppliers. Greater competition, in turn, encourages more rapid innovation, which becomes the key to success. A wide uptake of open standards underpins the drive to greater competition, since the approach encourages new entrants. Even for incumbent network equipment makers, open standards solutions offer a compelling model for overall product evolution. A network equipment company may give up some of its ownership of a proprietary solution, but in return stands to gain much more rapid time to market advantage in terms of product enhancements and a better overall product roadmap for lower cost.

Using Open Standards to Help

Hardware and Equipment Practice—CompactPCI

CompactPCI is an established standard for computer backplane architecture and peripheral integration, defined and developed by the peripheral component interconnect (PCI) industrial computers manufacturers group (PICMG). CompactPCI and PCI technologies have emerged from the search for a standard interface between peripherals on PC—compatible CPUs and backplanes. In keeping with the requirements for true interoperability, the PICMG—an independent and cooperative consortium of vendors and manufacturers—has overseen the development of these specifications.

CompactPCI uses a vertically mounted backplane consisting of five connectors, as is defined in the PICMG's CompactPCI specification 2.0. This method of mounting is significantly more robust that that available from a standard PC and also provides better access for cooling because air can flow past more easily.

The major benefit of CompactPCI over its predecessors is the capability for hot plugging boards to allow maintenance and upgrades without losing service.

Software Protocol Stacks

The major benefit of agreed-upon functional elements, and open protocols between the elements, is that there then exists a market in which software and algorithm developers can create protocol stack implementations. Such implementations can be developed and tested independently, reducing the risk of interoperability problems. Additionally, each customer of a stack provider gains the benefit of the extensive testing the implementation receives from all the other customers.

Integrated Hardware and Software Subsystems

While open systems definitions provide a basic level of interworking between products from different manufacturers, there is a growing trend for outsourcing that relies on the provision of hardware and software elements that are guaranteed to work together as a higher-level subsystem. One example of this might be an MGCP Media Gateway subsystem consisting of DSP boards and control software—designed for integration into a CompactPCI chassis. Subsystems further reduce the risk for system integrators and allow them to concentrate on developing the service creation and other management aspects of their product.

Summary

The communications world is in massive transition. The traditional domains of voice and data are converging and the situation is creating a critical challenge and opportunity for both services companies and equipment makers.

Communication services drive the current model and pace of business, enabling a whole new way of transacting commerce and interacting with customer and employees. The emergence of global competition and the continued domination of the Internet has driven a need for much more sophisticated multifunctional communication services in order for businesses to reach customers. Carriers are responding to the challenge through the building of much more versatile platforms, both through new network deployments and the transformation of their traditional singular communication networks to multiservice architectures.

For traditional carriers, the importance of co-existence of existing and newer technologies and equipment is an important reality. Next-generation multiservice networks, based on packetized technologies will not entirely push out existent circuit-switching equipment and solutions for many years. This is where a network element called a “media gateway” is needed—it performs both signaling and media format conversion between different network encoding and transmission standards. For new communications service providers that don’t have a legacy of traditional networks, co-existence is moot. Flexible and versatile multiservice network architectures can provide an immediate competitive advantage, although since these are relatively new technologies, there are still issues of proven capability to be achieved. However, these new service providers must still work with existing networks to maximize customer coverage and appeal.

Key to success in this new communications market is the ability to rapidly create new communications solutions that offer carriers and other communication services providers a competitive edge in attracting new and retaining existing customers. The adoption of open standards in equipment development generates a virtuous circle where a market for outsourced components fosters competition which in turn drives innovation, so visionary network equipment makers are looking to feature and function development partners who adhere to open standards for leading-edge technology development.

For new equipment makers with no legacy of existing products, looking for quick entry into this lucrative and important market, there’s no need to reinvent the wheel. These companies can rapidly ramp up to a generic open standards-based platform provided by external equipment suppliers, then focus on adding on unique value-added features and functions.

Symmetry Communications Takes Advantage of Blue Wave's ComStruct™ Open Standards for Next Generation Wireless Infrastructure

This example of how the converged network is being accelerated by the increasing acceptance and use of commercial-off-the-shelf (COTS) products draws on innovation for the wireless infrastructure. GPRS is a high bandwidth wireless data service that connects legacy 2G infrastructure with the converged IP network. Symmetry Communications has developed a GPRS support node using industry standard hardware with a future-proof architecture, including ComStruct™ platforms from Blue Wave Systems. This approach has not only reduced the time to market for its original implementation, but also enables Symmetry to migrate rapidly to the latest platforms through a simple technology insertion.

General Packet Radio Service or GPRS is a new non-voice value added service that allows packet data to be sent and received across a second generation (2G) mobile telephone network. Sometimes referred to as 2.5G, it is the most advanced wireless data communications technology available today and is a recognized stepping stone in the evolution path to third generation systems.

Many operators are now evaluating GPRS as a long-term solution to respond to the predicted explosion in demand for wireless data at speeds comparable to those offered by fixed wire. For those operators who go on to implement 3G systems, the network hardware additions required to support GPRS may also provide the backbone for handling data traffic in a 3G environment.

A technically superior solution

Using up to eight timeslots, GPRS is theoretically capable of maximum speeds of up to 171.2 kbps, although the most quoted figure is 115 kbps. The speed is dependent on the realities of constraints in the network and terminals.

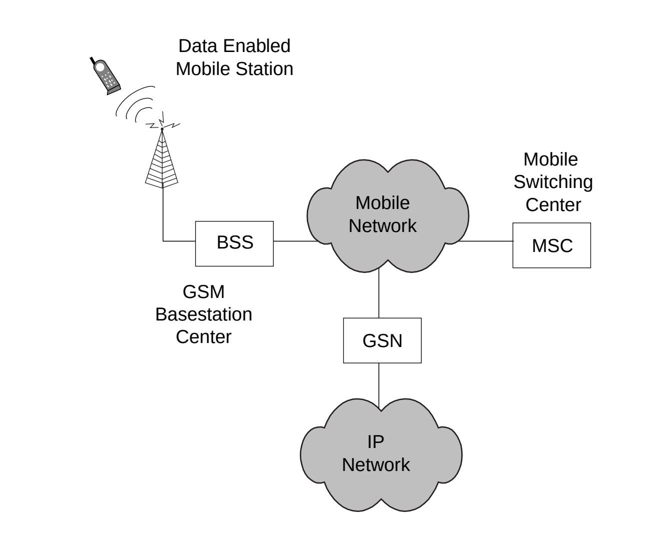
This is ten times as fast as current circuit switched data services on GSM networks. By allowing information to be transmitted more quickly, immediately and efficiently across the mobile network, GPRS enables many more applications and services compared to circuit switched data.

GPRS also enables instant connections such that information can be sent or received immediately as the need arises, subject to radio coverage. No dial-up modem connection is necessary. This is why GPRS users are sometimes referred to be as being 'always connected'. This immediacy is another advantage of GPRS when compared to circuit switched data and is a very important feature for time critical applications such as remote credit card authorization where it would be unacceptable to keep the customer waiting for even thirty extra seconds.

Unique and innovative solution for the 'Information Super Skyway'

Symmetry Communications is a leading provider of wireless packet data solutions for Internet and next-generation telecommunications services. The Symmetry S2000 GPRS Support Node (GSN) offers a very powerful yet highly flexible solution for operators of wireless networks and enables third generation wireless networks for fax, e-mail, the Internet, teleconferencing, and corporate intranets,

Figure 5: GSN and the Converged Network



all with continuous connectivity. Figure 5 shows the position of the GSN in the converged network and how it offers a gateway between 2G mobile and the converged world.

Enabling GPRS on a GSM network requires the addition of two core modules: the Gateway GPRS Support Node (GGSN) and the Serving GPRS Support Node (SGSN). The GGSN acts as a gateway between the GPRS network and IP-oriented Public Data Networks as well as bridging other GPRS networks for user roaming. The Serving GPRS Support Node provides packet routing to and from a particular service area. A single Symmetry S2000 GSN can be configured as an SGSN, a GGSN or both.

Symmetry's solution is very scalable thanks to its use of standard-based technology. It allows for an initial low-cost deployment for several hundred active users per GSN, expandable to hundreds of thousands of users. It can also be co-located at the base station controller, and so avoids costly E1/T1 connection to the Mobile Switching Center (MSC). Alternatively, for large operators seeking a centralized solution, high capacity S2000 GSN systems can be installed at the MSC.

Keeping ahead of the game

In developing its GPRS system, Symmetry recognized that producing a system that is at the leading edge of today's wireless technology was not enough. To maintain its market leadership, the company needed to ensure that its GSN could easily evolve to meet changing customer needs. Symmetry identified the benefits of open standards technology and commercial-off-the-shelf (COTS) products.

New functionality requires increased processing power and using open standards technology, such as the CompactPCI® form factor, can significantly ease the continuing development of telecom infrastructure equipment. Flexibility was not the only factor in the decision to use open standards COTS technology as Walter Tijiboy, Symmetry's Vice President of Operations explains:

"In the fast-moving wireless infrastructure industry, we could not compromise on time to market. We have created products that are unique in the industry, built on an open standard and modular hardware platform that results in low-cost starting points for service delivery, yet scale massively as the offered services become more popular."

Open standards the key

To provide the encryption, compression and decompression of data within its GSN, Symmetry selected the ComStruct™ communications processing platform from Blue Wave Systems. Based on high performance Digital Signal Processors, ComStruct's flexible and scalable open standards architecture was in tune with Symmetry's own product strategy. Walter Tijiboy continues:

"Blue Wave's roadmap for ComStruct will enable future expansion to higher capacities which is in line with our needs to offer low-cost deployment and highly scalable, powerful solutions."

Symmetry has integrated Blue Wave's ComStruct CompactPCI-based CPCI/C6400 DSP resource board into its GSN system architecture to provide rapid connection and fast data transmission rates for packet transfer with data compression, security features and encryption.

Symmetry's GSN uses the ComStruct solution to perform the necessary compression, decompression and encryption in conjunction with a Sun Microsystems host processor. It is this ability to combine best in class products that underlines the power of using open standards—the customer is not tied to a particular proprietary technology and can select the best solutions for the task.

A clear path to the future

Symmetry's strategy paid off when Blue Wave Systems launched a new communications platform, the CPCI/C6402. This CompactPCI-based resource board offers more than double the performance of its predecessor and—most importantly—the two ComStruct solutions use the same architecture.

For Symmetry, this means that to more than double the performance of the DSP-based processing in its GSN simply requires a simple software change to support a hardware technology insertion. Blue Wave Systems has ensured that the two hardware platforms are compatible, so integration and testing are minimized.

This development further strengthens Symmetry's decision to use open standards and off-the-shelf technology. ComStruct is ideally positioned for evolving applications such as GPRS with its modular and upgradable architecture for hardware and software providing a future-proof deployment platform.

2G – 2G wireless systems use digital encoding and include GSM, TDMA and CDMA. Except for GSM's SMS text message service, 2G systems have been used mostly for voice.

3G – 3G is a specification for the third generation (analog cellular was the first generation, digital PCS the second) of mobile communications technology. 3G will provide increased bandwidth up to 384 kbps when a device is stationery or moving at pedestrian speed, 128 kbps in a car, and 2 Mbps in fixed applications. 3G will work over wireless air interfaces such as GSM, TDMA, and CDMA.

3rd Generation Partnership Project (3GPP) – A cooperation of standards organizations (ARIB, CWTS, ETSI, T1, TTA and TTC) throughout the world that is developing the technical specifications for IMT-2000, which increase data rates for 3G wireless communications.

Adaptive Differential Pulse Code Modulation (ADPCM) – ADPCM is a form of pulse code modulation that produces a digital signal with a lower bit rate than standard PCM. ADPCM produces a lower bit rate by recording only the difference between samples and adjusting the coding scale dynamically to accommodate large and small differences.

Asynchronous Transfer Mode (ATM) – A broadband communications technology standard, defined by the ITU-T, for the transport of consistent-sized "cells" across a network. ATM is a multiservice transport where information for multiple communications service types (voice, video and data) is conveyed in small, fixed-sized packets called cells.

CompactPCI – CompactPCI is a very high performance industrial bus based on the standard PCI electrical specification. The theoretical top speed of CompactPCI is 133 Mbps, using 32-bit wide data or 266 Mbps using 64-bit wide data. CompactPCI is designed for applications that require very high data transfer rates, including data communication interfaces such as ATM, and broadband ISDN.

Digital Signal Processing/Processor (DSP) – DSP implies the use of a data compression technique. A digital signal processor is a special type of coprocessor designed for performing the mathematics involved in DSP. Most DSPs are programmable, which means they can be used for manipulating different types of information, including sound, images, and video.

Digital Subscriber Line (DSL) – DSL technologies use modulation schemes to transmit data through copper wires. These are technologies used only for connections from a telephone switching station to a home or office.

E1 – E1 is the European format for digital transmission and a similar alternative to North America's T-1 service. E1, however, is a slightly higher bandwidth, carrying signals at 2.048 Mbps (32 channels at 64 kbps). E1 and T1 lines may be interconnected for international use.

Enhanced Data rates for Global Evolution (EDGE) – An enhancement to the GSM and TDMA wireless communications systems that increases data throughput to 384 kbps.

European Telecommunications Standards Institute (ETSI) – Based in France, this engineering group helps define and promote worldwide standards for building next-generation communications networks. ETSI claims to comprised of approximately 789 members from 52 countries from both inside and outside of Europe, and represents administrations, network operators, manufacturers, service providers, research bodies and users.

Frame Relay – A packet-switching protocol for connecting devices on a Wide Area Network (WAN). Frame Relay networks typically support data transfer rates at T-1 (1.5 Mbps) and T-3 (45 Mbps) speeds. In Europe, Frame Relay speeds vary from 64 kbps to 2 Mbps. Frame Relay is considered a more efficient communications service than X.25 – the telecommunications offering that it has generally replaced.

General Packet Radio Service (GPRS) – General Packet Radio Service (GPRS) supports digital cellular networks (GSM, DCS, PCS). It utilizes a packet radio principle and can be used for carrying end user's packet data protocol (such as IP and X.25) information from/to a GPRS terminals to/from other GPRS terminals and/or external packet data networks. GPRS is standardized in ETSI (European Telecommunications Standards Institute).

Global System for Mobile Communications (GSM) – A digital cellular system that uses narrowband TDMA, which allows eight simultaneous calls on the same radio frequency.

GR-303 – GR-303 defines a set of requirements for Integrated Access Systems (IAS) that include open interfaces for the mix and match of Local Digital Switches (LDS) with Remote Digital Terminals (RDTs) as well as Element Management Systems (EMS).

GSM-AMR – GSM-AMR is an implementation of the Global System for Mobile (GSM) Adaptive Multi-Rate (AMR) Algebraic Code Excited Linear Prediction (ACELP) voice coder, and is fully compliant with the ETSI GSM 06.90 specification.

H.323 – H. 323 is a set of standards that specifies the components, protocols and procedures that provide multimedia communication services– real- time audio, video, and data communications– over packet networks. Approved by the ITU, H.323 defines how audiovisual conferencing data are transmitted across networks. In theory, H.323 should enable users to participate in the same conference even though they are using different videoconferencing applications.

High-level Data Link Control (HDLC) – HDLC is a transmission protocol that embeds information in a data frame and allows devices to control data flow and correct errors.

Integrated Services Digital Network (ISDN) – A communications standard for sending voice, video, and data over digital telephone lines or normal telephone wires. ISDN supports data transfer rates of 64 kbps (64,000 bits per second).

Interactive Voice Response (IVR) – IVR describe a type of application used to deliver products or services without human phone operator intervention. IVR solutions can, for example, capture credit card number and expiry data and immediately obtain a bank authorization for a transaction.

International Telecommunications Union (ITU) – A global, inter-governmental organizations through which public and private organizations develop telecommunications standards and technologies. The ITU is responsible for adopting international treaties, regulations and standards governing telecommunications.

Internet Engineering Task Force (IETF) – A standards organization for advancing the Internet. The IETF is a large open international community of network designers, operators, vendors, and researchers concerned with the evolution of the Internet architecture and the smooth operation of the Internet.

International Multimedia Telecommunications Consortium Inc. (IMTC) – Based in San Ramon, Calif., this 150-company member group focuses on promoting a uniform base of standards for the adoption and interoperability of multimedia communications products and services.

Internet Protocol (IP) – IP specifies the format of communications packets or “datagrams” and the addressing scheme. Most networks combine IP with a higher-level protocol called Transport Control Protocol (TCP), which establishes a virtual connection between a destination and a source. IP by itself is often described as something like a postal system, which addresses a communications packet and drops it in the network system. However, there’s no direct link between sender and recipient. TCP/IP, on the other hand, establishes a connection between two hosts, allowing each to send messages back and forth for a period of time.

Kbps – Thousands of bits per second.

Media Gateway – A media gateway translates traffic from some type of access link to a time-division multiplexer or IP-based backbone carrier network.

Media Gateway Controller – A Media Gateway Controller allows inter-working between different networks (PSTN and Packet based) with different signaling systems and media transports. Each signaling system has its own unique set of requirements, which make inter-working between them complex. A Media Gateway Controller hides the incompatibilities of different signaling systems and controls the Media Gateways to perform bearer interworking functions.

Media Gateway Control Protocol (MGCP) – MGCP is a proposed control and signal standards for the conversion of audio signals carried on telephone circuits (PSTN) to data packets carried over the Internet or other packet networks.

MEGACO/H.248 – An emerging set of standards designed to enable voice, fax and multimedia calls to be switched between the public switched telephone network and emerging IP networks by separating the signaling and service capabilities, SS7 termination capability and media transmission capability. MEGACO stands for Media Gateway Control.

Network Access Server (NAS) – A server in a network dedicated to authenticating users that log on. It may refer to a dedicated server or to the software service within a server.

Original Equipment Manufacturer (OEM) – A manufacturer that sells equipment to a reseller for rebranding or repackaging.

Public Switched Telephone Network (PSTN) – The long-established, traditional international telephone system based on copper wire technology and designed to carry analog voice data.

Pulse Code Modulation (PCM) – PCM is a sampling technique for digitizing analog signals, especially audio signals. PCM samples the signal 8000 times a second – each sample represented by 8 bits for a total of 64 kbps.

Quality of Service (QoS) – QoS is a specification for providing a guaranteed throughput level for communications. It defines a measure of performance for a transmission system that reflects its transmission quality and service availability.

Real-Time Transport Protocol (RTP) – An Internet protocol for transmitting real-time data. RTP itself does not guarantee real-time delivery of data, but it does provide mechanisms for sending and receiving applications to support streaming data.

Request for Comment (RFC) – An RFC is a series of notes about the Internet. An RFC can be submitted by anyone and, if it gains enough interest, may evolve into an Internet standard. Every RFC submitted is assigned an RFC number and never changes even when published. Modifications to an original RFC are assigned a new RFC number.

Resource ReSerVation Setup Protocol (RSVP) – RSVP is an Internet protocol that is designed to enable specified support for Qualities-of-Services (QoSs) on the Internet. Using RSVP, an application is able to reserve resources along a route from source to destination. RSVP-enabled routers are designed to schedule and prioritize packets to fulfill the QoS requirements.

Session Initiation Protocol (SIP) – SIP is a signaling protocol for creating, modifying and terminating sessions with one or more participants. Sessions may include Internet multimedia conferences, Internet telephone calls and multimedia distribution. Members in a session can communicate via multicast or a mesh of “unicast” relations, or a combination of these.

Short Message Service (SMS) – SMS is the transmission of short text messages to and from a mobile phone, fax machine and/or IP address. Messages must be no longer than 160 alpha-numeric characters and contain no images or graphics. Once a message is sent, it is received by a Short Message Service Center (SMSC), which must then get it to the appropriate mobile device.

Signaling Gateway – A Signaling Gateway allows remote devices on an IP network (such as IP based Signaling End Points, Media Gateway Controllers, Call Agents, PBXs) to exchange messages with the PSTN network via an interface over IP.

SIGTRAN – SIGTRAN defines a suite of protocols and adaptation layers for transporting signaling information over IP based networks. It is the key transport component in a distributed VoIP architecture and can be integrated into products such as Signaling Gateways, Media Gateway Controllers, Gatekeepers and IP-based Service Control Point to develop convergent voice and data solution.

Simple Computer Telephony Protocol (SCTP) – SCTP is a protocol to exchange telephony signaling over IP networks and serial communication lines.It can be used to build client/server telephony applications, and to create a common interface for intelligent digital telephone devices. SCTP is designed to work over TCP/IP networks (using TCP sockets), and over RS232 serial links.

Softswitch – A softswitch is software that resides in either a server or another network element and is designed to separate the call control functions of a phone call from the media gateways that carry it. Some vendors include the media gateway or signaling gateway, too, as part of the softswitch itself. Essentially, a softswitch is designed to let carriers provide traditional and introduce new services at a lower cost.

Switched-Circuit Network (SCN) – Also referred to as the public switched telephone network (PSTN), a carrier network that provides circuit switching among public users. Switched circuit or circuit switching is a method of routing communications traffic through a switching center, from local users or from other switching centers. A “circuit” connection session is established between the calling and called stations until the called or calling station releases the connection.

Signaling Systems 7 (SS7) – Signaling System 7 is an ITU-defined telecommunications protocol that specifies a way to offload PSTN data traffic congestion onto a wireless or wireline digital broadband network. SS7 architectures are set up in a way so that any node can exchange signaling with any other SS7-capable node, not just signaling between switches that are directly connected.

T-1 – T-1 lines are a popular leased-line communication services for businesses. A T-1 line supports data rates of 1.544 Mbps, but actually consists of 24 individual channels, each of which supports 64 kbps. Each 64 kbps channel can be configured to carry voice or data traffic.

Time Division Multiplexing (TDM) – A type of multiplexing (placing signals on a single channel) that combines data streams by assigning each a different time slot in a set. TDM repeatedly transmits a fixed sequence of time slots over a single transmission channel.

Third-Generation (3G) – 3G is an ITU specification for the third generation (analog cellular was the first generation, digital PCS the second) of mobile communications technology. Among other things, 3G promises increased bandwidth, up to 384 kbps when a device is stationery or moving at pedestrian speed, 128 kbps in a car, and 2 Mbps in fixed applications. 3G will operate on a range of wireless air interfaces, including GSM, TDMA, and CDMA. A new specification, called the EDGE (Enhanced Data rates for Global Evolution) air interface, has been developed specifically to meet the bandwidth needs of 3G.

User Datagram Protocol (UDP) – UDP is a connectionless protocol that runs on top of IP networks and provides few error recovery services. Instead it is a direct way to send and receive datagrams over an IP network and used primarily for broadcasting messages over a network.

V5 – V5 identifies a full range of procedural and protocol interface requirements that allows vendors to develop non-proprietary, standards-based local access and local exchange equipment. One of the main goals of defining this interface is to ensure connectivity and compatibility between competing vendors, giving operators greater choice when purchasing equipment.

VoIP – A private intranet or WAN used for the two-way transmission of audio over an IP network.

Wireless Application Protocol (WAP) – Is a specification that provides a means of communications that allows users to access information from handheld wireless devices such as mobile phones, pagers, two-way radios, “smartphones” and “communicators.” WAP supports most wireless networks and operating systems, including those specifically engineered for handheld devices including PalmOS, EPOC, Windows CE, FLEXOS, OS/9, and JavaOS. WAP is specifically devised for small screens and one-hand navigation devices that don’t use a keyboard. WAP is a fairly new standard put forward by companies including Motorola, Nokia and Ericsson, but is not yet formally approved.

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