Broadband Data, Video, Voice and Mobile Convergence – Extending the Triple Play

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Introduction

The word “convergence” has been used for 20 years to describe unified communications services. Yet today, it is far more likely that consumers have different service providers and different equipment for their landline phone, mobile phone, Internet and video services. For those with a great deal of initiative, it is possible to combine one or more communications services. The good news is we are in the early stages of convergence to a single platform that can dramatically simplify our lives.

Today, converged data, landline voice, mobile and video on a single platform from a single provider—and with a single bill—is becoming a reality. The technology exists and is available now at the right cost points. Broadband is playing a key role.

Recognizing the full potential of broadband is understanding that it extends beyond simple Internet access. It is the mechanism responsible for cost-effective delivery of compelling current and future new services and applications to the consumer. An example of an existing service that benefits immediately from broadband is the home telephone. Looking forward, the combination of broadband access technology and Wireless Local Area Networks (WLANs), such as 802.11, makes it possible to blur the lines between the telephone in the kitchen and the cellular telephone on your hip. Cellular phone manufacturers, for example, recently introduced dual-mode phones capable of communication over both cellular networks and WLANs. With such a phone, a consumer can seamlessly roam from cellular networks to a lower cost home network, transfer video from a phone to his home network, e-mail pictures to friends and family, and make video calls. And this is just the tip of the iceberg.

When the user is within range of his home, the cell phone converts into a WLAN voice handset that can also transfer data, pictures or video at broadband speeds. Ultimately, this enables video conferencing with the phone’s built-in camera. The handover will be as transparent to the consumer as today’s cell-to-cell handoff. All this functionality will be built into the personal communication device to create a mobile IP platform. The services on this platform provide one phone number for voice, video and high-speed data whether at home or away.

Service providers have a tremendous opportunity. For the “triple play” broadband provider, the benefit is a differentiated voice product. For the cellular provider, the broadband provider represents an excellent Mobile Virtual Network Operator (MVNO) opportunity. The key is building a symbiotic relationship between the cellular and broadband service providers, as well as the integration of infrastructures for a seamless handover between the cellular and home networks.

While there are some emerging activities within CableLabs®, this article will not describe them. Instead, it will provide a broad, end-to-end understanding of the technology that is critical to the cable-mobile convergence. Key aspects such as provisioning, authorization, management and transport are each sufficiently important and complex that they warrant dedicated discussion. This article will focus on voice transport issues, specifically those issues related to successfully connecting and maintaining a call.

Broadband and Convergence

Broadband technology makes convergence possible; however, care must be taken to ensure that end-to-end requirements are met for each service. For voice communications, these requirements include a converged signaling protocol, as well as controlled jitter and latency on all links in the end-to-end system. We generically refer to the jitter and latency requirements as Quality of Service (QoS). In the following sections, we will discuss the signaling and QoS requirements.

Signaling Convergence

Figure 1 shows an end-to-end system that illustrates mobile and broadband convergence.

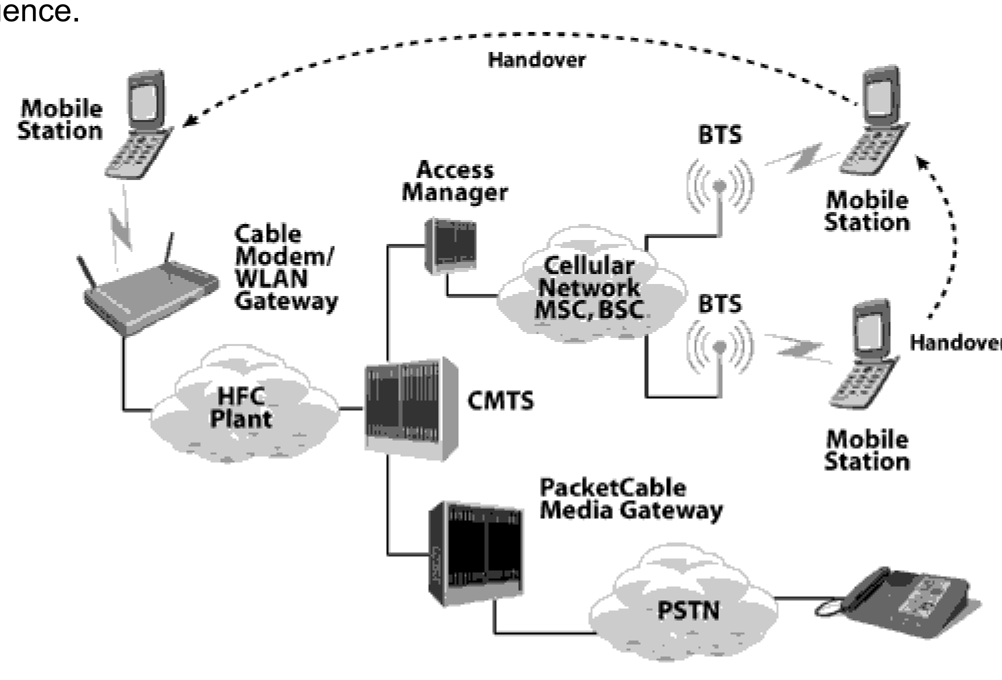

Figure 1: Broadband convergence network

In a typical mobile call handover (shown on the right side of the figure), a mobile station (MS) communicates over the radio channel with a base terminal station (BTS). The BTS connects to the cellular network “cloud,” which includes a base station controller (BSC), a mobile switching center (MSC) and other elements of the mobile infrastructure. Each BTS covers an area of a few miles.

When a call comes in for a specific mobile station, the MSC transmits a paging message to all the BSCs, which in turn transmit it over the air to the mobile stations. Each mobile station responds via the BTS/BSC associated with the relevant cell. Once the location of the right MS is identified, the MSC sends a ringing message. A dedicated voice channel is established and utilized until the end of the conversation. The protocol for an outgoing call is similar except for the paging process, since the MS initiates the call.

When the MS leaves the BTS coverage area, it is handed over to another BTS. The handover process includes radio link measurements to select the best available BTS, registration with the new BTS, and termination of the connection with the previous BTS.

The same signaling protocol can be used in a converged mobile-cable system. The radio link layer is replaced with a broadband pipe, as illustrated on the left side of Figure 1. The BSC, BTS and the cellular radio link are replaced by the Access Manager gateway (emulating the BSC), the cable modem termination system (CMTS), the cable modem (CM) with a WLAN Access Point (AP) integrated, and the WLAN radio link.

With the proper signaling, a mobile call can be handed off to a broadband pipe. Another important aspect of routing the call over broadband is the special treatment required for voice traffic in the broadband pipe. The next sections will discuss how to establish QoS in the DOCSIS®/WLAN/IP pipe.

QoS Considerations in an End-to-end Converged System

QoS is based on the idea that transmission rates, error rates and other characteristics can be measured, improved, and to some extent, guaranteed in advance. QoS is a particular concern for the continuous transmission of voice, high-bandwidth video and multimedia information. Transmitting this kind of content dependably is difficult in public networks using ordinary "best effort" protocols.

The human ear is very sensitive to latency during a two-way conversation. Latency exceeding 200 milliseconds (msec) is annoying. It causes two problems: echo and conversation overlap. Most echo issues can be addressed with echo cancellers. Conversation overlap (or the problem of one person stepping on the other person’s speech) becomes significant if the one-way delay becomes greater than 200 msec.

The end-to-end delay budget is the driving requirement for reducing delay through a packet network. Each element in the end-to-end system must minimize its contribution to total latency. Latencies arise either from direct delay of communications elements or from de-jitter buffers needed to compensate for the variable arrival time of packets in certain links. Table 1 shows a typical latency budget between two mobile stations connected through two cable-WLAN networks.

Segment	Latency [msec]	Variable [msec]
Speech coder	10	
WLAN	10	5
DOCSIS	8	4
Routing	40	25
DOCSIS	8	4
WLAN	10	5
De-jitter buffer	50	
Speech decoder	10	
Total	**146**	

Table 1: Typical latency budget

Each part of the network should use mechanisms to control latency and guarantee the variance of the packet arrival time (jitter). Next, we will examine QoS on each part of the network from a mobile station, through a WLAN access point and DOCSIS, into a Media Gateway.

QoS Considerations in Voice over Internet Protocol (VoIP) Codecs

VoIP codecs encode voice packets in mobile stations and other VoIP endpoints on the network. High-performance digital signal processors (DSPs) and codec algorithms in VoIP systems minimize the voice coding latency. Different coders work in different ways, so delay varies with the selected voice coder.

For example, algebraic code excited linear prediction (ACELP) algorithms analyze 10 ms blocks of PCM samples and compress them. In addition, the compression algorithm must have knowledge of what is in the next sample block to accurately reproduce sample block N. This “look ahead” generates an additional delay, called algorithmic delay, and increases the length of the compression block.

Packetization delay is the time needed to fill a packet payload with encoded/compressed speech. It is a function of the sample block size required by the codec and the number of blocks placed in a single frame. Packetization delay can also be called accumulation delay because the voice samples accumulate in a buffer before being released. Clearly it is important to synchronize the packetization with packet timing to eliminate unnecessary wait time. This means that the codec output should be ready just prior to being sent to the output interface.

QoS on the WLAN Link

802.11e (or derivatives of 802.11e, such as WME and WSM) introduced options for controlling QoS. Legacy 802.11 designs implemented only the best-effort media access controller (MAC) protocol, as defined in the original standard.

802.11e provides two mechanisms to support applications with QoS requirements.

- Enhanced Distributed Channel Access (EDCA) delivers traffic based on differentiating user priorities. This differentiation is achieved by (1) varying the amount of time a station senses the channel to be idle before back-off or transmission and (2) adapting the length of the contention window to be used for the back-off (duration a station may transmit after it acquires channel access). This is done using the Arbitration Inter Frame Spacing (AIFS) that is variable based on the QoS requirements.

- Hybrid-coordinated Controlled Channel Access (HCCA) allows for the reservation of transmission opportunities (TXOPs) with the access point (AP). The station requests TXOPs at the beginning of the session. The AP schedules TXOPs for both the AP and the station in a Contention Free Period (CFP) which occurs repetitively on a regular basis. During the CFP, the AP initiates transactions. For transmissions from the station, the AP polls the station based on the parameters supplied by the station at the time of its request. For transmissions to the station, the AP delivers the queued frames to the station directly.

The HCCA mechanism is illustrated in Figure 2.

The HCCA mechanism may be better suited than EDCA for real-time applications such as voice and video, which may need periodic service from the AP. This is because EDCA is more stochastic in nature and thus increases the average latency and jitter. Nevertheless, the EDCA scheme is simpler to implement because it does not involve complex reservation and scheduling mechanisms. It is expected to be more popular initially.

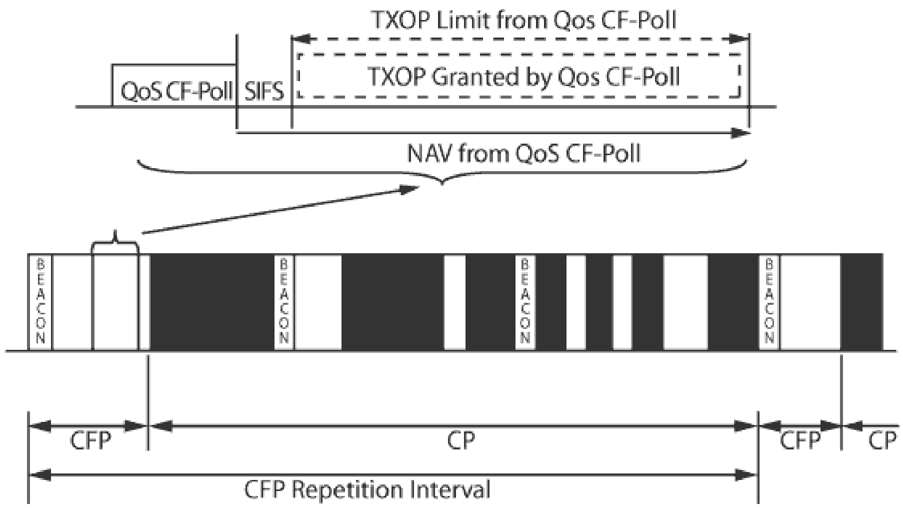

Figure 2: Contention Free Period and HCCA operation

QoS on the DOCSIS Link

DOCSIS 1.1 supports QoS mechanisms by using service flows and reservation schemes. The Unsolicited Grant Service (UGS) and its derivatives (such as UGS with activity detection [UGS-AD]) provide the technological backbone for minimizing latency and jitter and assuring the required bandwidth for the application. PacketCable™ utilizes these mechanisms to ensure QoS for voice traffic.

The principal DOCSIS 1.1 QoS mechanism classifies packets traversing the Radio Frequency (RF) MAC interface into a Service Flow, a unidirectional flow of packets that receives a particular QoS. The cable modem and CMTS provide this QoS by shaping, policing and prioritizing traffic according to the QoS Parameter Set defined for the Service Flow.

The primary purpose of the QoS features on the DOCSIS link is to define transmission ordering and scheduling on the DOCSIS RF interface. However, these features often need to work in conjunction with mechanisms beyond the RF interface in order to provide end-to-end QoS or to police the behavior of cable modems.

To address this issue, scheduling services have been designed to improve the efficiency of the poll/grant process.

By specifying a scheduling service and its associated QoS parameters, the CMTS knows the throughput and latency of the upstream traffic and provides polls and/or grants at the appropriate times. The UGS is designed to support real-time service flows that generate fixed-size data packets on a periodic basis, such as VoIP.

The service offers fixed-size grants on a real-time periodic basis to eliminate the overhead and latency of cable modem requests and assure that grants will be available to meet the flow's real-time needs.

QoS on the IP Routing Protocols Link

Differentiated services (diffserv) and Multiprotocol Label Switching (MPLS) are two standards that attempt to solve the IP quality problem. Diffserv takes the IP type of service (TOS) field, renames it in the differentiated services field (DS Field) and uses it to carry information about IP packet service requirements. It operates at Layer 3 only and does not deal with lower layers. On the other hand, MPLS specifies how Layer 3 traffic can be mapped to connection-oriented Layer 2 transports. MPLS adds a label containing specific routing information to each IP packet and allows routers to assign explicit paths to various classes of traffic. It also offers traffic engineering and techniques that can boost IP routing efficiency.

MPLS involves setting up a specific path for a given sequence of packets, identified by a label put in each packet, thus saving the time needed for a router to look up the address to the next node and to forward the packet. MPLS allows most packets to be forwarded at the Layer 2 (switching) level rather than at the Layer 3 (routing) level. In addition to moving traffic faster overall, MPLS makes it easy to manage a network for QoS.

Connecting It All Together

Each segment of the communication network has its QoS mechanism. The challenge is to make them work together seamlessly to provide end-to-end QoS.

Figure 3 illustrates an end-to-end protocol stack representing the network of Figure 1. The stack represents a call between a mobile station connected through a cable modem access point, to either a cellular or mobile station endpoint or a telephone endpoint through the public switch telephone network (PSTN). To keep it simple, only the voice bearer protocols are shown (for the most likely scenario), but it is still a quite complex system.

- Each protocol allows many QoS mechanisms and an even higher combination of selecting operating parameters. Careful coordination between the protocols is essential to an optimal solution. For example, both DOCSIS and WLAN provide scheduled QoS mechanisms (UGS in DOCSIS and HCCA in WLAN). Combining them is likely to provide better overall end-to-end QoS than combining UGS with EDCA.

- Synchronization between all segments reduces total latency and jitter, especially when using scheduled services. Coordination between the Mobile Station's DSP, CMTS and AP schedulers is not a trivial task but will help to reduce total latency and jitter.

- System integration, validation and interoperability testing require an industry-wide effort and are essential to create a good end-to-end solution.

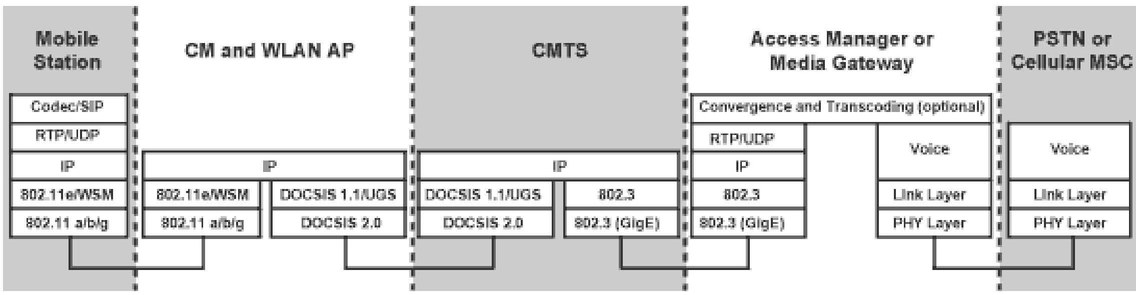

Figure 3: End-to-end voice bearer protocol stack

Conclusion

Broadband technologies are becoming mature enough to make the vision of converged data, voice and mobile communications a reality. The industry is more than halfway to a solution:

- High-speed always-on features are already built in broadband technologies.
- Each element of the end-to-end converged mobile-broadband network has QoS hooks. In some cases, multiple methods of QoS methods are available.

To achieve full seamless convergence, work is needed to specify the interaction of these individual QoS mechanisms. This task will be addressed in two ways: Through relationships formed by cellular and broadband service providers to provide consumers with an extended triple play that includes a total voice service package; and by standards bodies providing a framework for end-to-end convergence.

About the Author

Yigal Bitran is the Chief Technology Officer for the Broadband Communications Group at Texas Instruments Israel. He is responsible for driving TI's technical innovation and leadership in the broadband industry. Bitran earned a bachelor's degree and master's degree in Electronic Engineering from Tel-Aviv University and an MBA degree from Kellog-Recanati executive program. He may be contacted at: bitran@ti.com

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