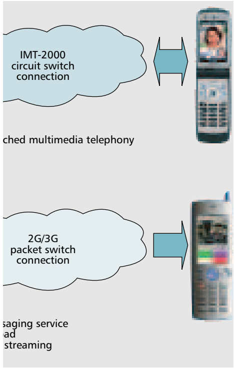
WIRELESS VIDEO APPLICATIONS IN 3G AND BEYOND

MINORU ETOH, DOCOMO COMMUNICATION LABORATORIES USA
TAKESHI YOSHIMURA, NTT DOCOMO, INC.



The authors survey wireless video applications that have been commercialized recently or are expected to go to market in 3G (and beyond) mobile networks, mainly covering error control technologies in view of "wireless video."

ABSTRACT

This article surveys wireless video applications that have been commercialized recently or are expected to go to market in 3G (and beyond) mobile networks, mainly covering error control technologies in view of "wireless video." We introduce several related 3GPP standards including circuit-switched multimedia telephony, end-to-end packet-switched streaming, multimedia messaging service, and multimedia broadcast /multimedia service. We also review the supporting technologies for those four applications. The article concludes with a discussion of error control and rate control adaptability to network QoS variation, which are distinct from wired networks and critical to wireless networks. With respect to MBMS, we point out that required cell transmission power is crucial when realizing meaningful multicast coverage, and suggest a system that integrates different unicast and multicast networks, application-layer data repair, and transmission scheduling.

INTRODUCTION

In the early 1990s, most of us could not have imagined the current mass popularity of multimedia communication over mobile networks. At the beginning of this century, we have experienced two mobile digital-network generations: the second generation (2G), which brought us digital mobile communication, and the third generation (3G), which is characterized by its ability to carry data at much higher rates. 2G-3G radio access networks include:

- General Packet Radio Service (GPRS)
- Enhanced data Global System for Mobile Communications (GSM) environment (EDGE)
- Wideband code-division multiple access (W-CDMA), also known as Universal Mobile Telecommunications System (UMTS)
- High Speed Downlink Packet Access (HSDPA)
- CDMA2000 1X Evolution, Data-Only (1X EV-DO)

For further information on these networks, please refer to the authors' article [1].

3G cellular network diffusion seems to be advancing steadily. Mobile multimedia communication has taken off with 3G networks. 3G is still being enhanced with its higher data rates. Let us take Japan's mobile network evolution path to explain current trends and emerging services in mobile communication, because Japan is one of the most advanced wireless broadband markets in the world, and its path reveals the kinds of future demands that are expected from mobile service customers. Figure 1 illustrates the recent data speed enhancement in Japan, for which market research figures show that users are now replacing their 2G cell phones with 3G terminals.

The evolution path indicates that 300 kb/s–2 Mb/s bandwidth cellular connection will be commonly available in Japan soon, and throughout the world eventually. Moreover, flat-rate pricing for unlimited access to data on 3G mobile networks is now becoming a common practice of operators. Although mobile network operators are still profiting from current transaction- or packet-based pricing, where voice mail and messaging are usually charged by air time or transaction, the pricing trend is progressing toward the unmetered Internet model. The combination of WiFi and 3G cellular networks will bring a realistic and comfortable solution beyond 3G, where 54 Mb/s in hotspots and several hundred kilobits per second with wide coverage are available.

In the last decade video compression technologies have evolved in the series of MPEG-1, MPEG-2, MPEG-4, and H.264. Given a bandwidth of several hundred kilobits per second, the recent codecs, such as MPEG-4, can transmit VCR quality video. The coding evolution together with the rapid growth of wireless communications are bringing video into our lives any time, anywhere, on any device. In this article we review wireless video applications and supporting technologies. After introducing four wireless applications, we discuss supporting technologies, focusing on wireless-related error control.

WIRELESS APPLICATIONS AND RELATED STANDARDS

This section introduces major wireless video applications specified as Third Generation partnership Project (3GPP) standards. The reason

 1536-1284/05/$20.00 © 2005 IEEE

we discuss 3GPP standards here is that those standard descriptions give unbiased technical depth about the applications. 3GPP has standardized four types of visual content delivery services and technologies:
- Circuit-switched multimedia telephony [2]
- End-to-end packet-switched streaming (PSS) [3]
- Multimedia messaging service (MMS) [4]
- Multimedia broadcast/multicast service (MBMS) [5]

Figure 2 depicts these applications. 3GPP transport technology can be viewed as a mixture of "wireless narrowband integrated services digital network (N-ISDN)" and an extended GPRS network. Multimedia telephony is realized by a circuit switch connection that provides a 64 kb/s ISDN-compatible bearer connection. This service has been widely adopted for 3G phones. MMS has already been widely commercialized over 2G and 3G packet networks. Packet-switched streaming service has already been demonstrated and is ready to use. MBMS is an emerging standard that aims to offer an efficient way to transmit data from single source to multiple destinations over radio networks. Unlike the other standards, MBMS defines transport systems that require adding new functional entities to the 3GPP architecture.

Circuit-Switched Multimedia Telephony

The 3G-324M standard was developed for circuit-switched multimedia telephony service, and it is applied to speech (AMR) and video codecs (H.263, MPEG-4), the communication control unit, and the multimedia multiplexing unit (H.223).

While various media coding schemes can be used in 3G-324M by exchanging the set of available functions through the use of communication control procedures (H.245) and by changing the codec setting by establishing logical channels, 3G-324M defines a minimum essential codec to ensure interconnection between different terminals. For the video codec, 3G-324M specifies the H.263 baseline (excluding the optional capabilities) as the essential video codec (as is the case for H.324). It also recommends the use of MPEG-4 video to deal with transmission line errors unique to mobile communications. For error resilience, multimedia multiplexing functionality is enhanced in this standard. The multiplexer also provides transmission service according to the type of information (e.g., quality of service [QoS] and framing). H.223, the multimedia multiplexing scheme for H.324, satisfies the requirements by adopting a two-layer structure consisting of an adaptation layer and a multiplexing layer.

Three adaptation layers are defined according to the type of the higher layers:
- ALl: For user data and control information. Error control is performed in the higher layer. Iterative retransmission is applied for reliable transmission.
- AL2: For voice. Error detection and sequence numbers can be added.
- AL3: For video. Error detection and sequence numbers can be added. Automatic repeat request (ARQ) is applicable.

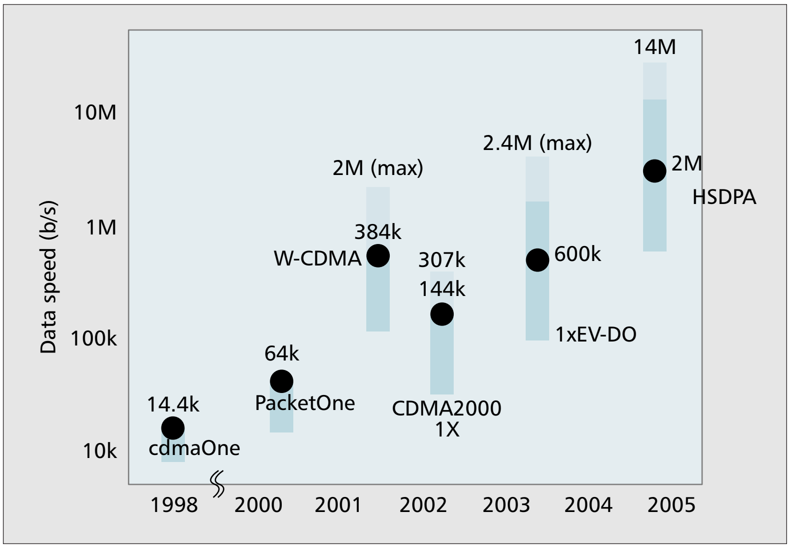


Figure 1. *Recent data speed enhancement in Japan.*

The H.223 specification teaches us technology fundamentals of error controls when transmitting multimedia data. The essential features are synchronization, error detection, forward error correction (FEC), ARQ, and data duplication. These features are combined for better error resilience depending on the type of payload: data, voice, or video.

Packet-Switched Streaming Service

When considering most typical applications over the Internet and IP-transparent seamless connectivity, media streaming will certainly be based on a packet-switched connection, whereas a combination of Internet Engineering Task Force (IETF) standards, such as Real-Time Transport Protocol (RTP), will be used over mobile networks.

Packet-based streaming services are required whenever instant access to multimedia information must be integrated into an interactive media application, allowing a Web server to respond to requests for information, deliver the requested information as fast as possible, complete the transaction, disconnect, and go on to other requests. A client connects to a Web server only when it needs information. A 3GPP standard, transparent, end-to-end packet-switched streaming service (PSS), has been specified to fulfill these requirements.

Figure 3 shows packet-based streaming protocol stacks standardized in 3GPP for a W-CDMA network. The media types specified in this standard are very similar to 3G-324M. For session control, however, RTSP and SDP are used. For the media transport protocol, RTP is used. RTP Control Protocol (RTCP) reports to the sender the reception status of images and audio transmitted with RTP to control the service quality.

Media transport technologies should differ for each type of wireless link. For real-time videoconferencing over the circuit-switched network, we use 3G-324M for error control. H.223 is a very sophisticated multiplexing scheme for bit errors, but it cannot be used for packet streaming, in which packet loss (packet erasure)

All necessary protocols underlying media adaptation are included in 3GPP PSS and the emerging TETF standards. Retransmission control and media-specific mechanisms are left intact as they are not standardized.

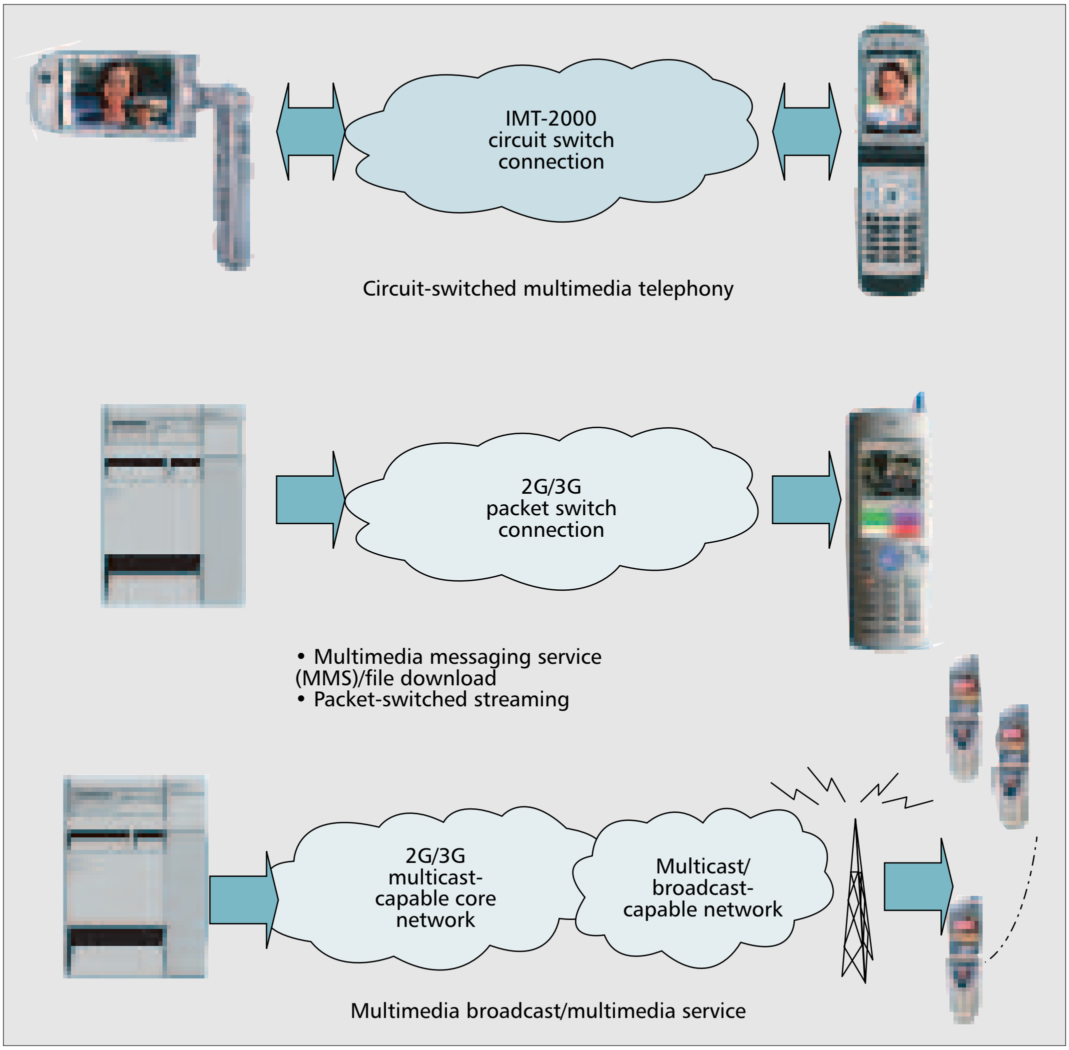
is a problem. For robustness with respect to packet loss, we must abandon the idea of "joint source-channel decoding" (the H.324 approach), where the receiver tackles bit-erroneous packets using source coding knowledge. Moreover, streaming services are allowed to have somewhat more latency, while real-time videoconferencing has a strict latency constraint (e.g., 200 ms). Among the error control fundamentals, synchronization and error detection are provided by an underlying transport layer protocol stack, such as RTP/UDP. FEC, ARQ, data duplication, and other techniques, such as data interleaving and unequal error-resilient packetization, are relevant in some ways to packet loss resilience.

All necessary protocols underlying media adaptation are included in 3GPP PSS and the emerging IETF standards. Retransmission control and media-specific mechanisms are left intact as they are not standardized. We examine specific mechanisms in the following sections.

Multimedia Messaging Service

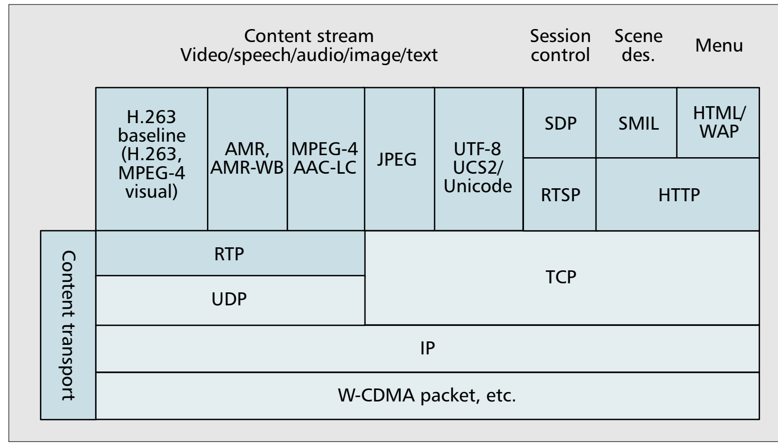
In MMS, multimedia content is delivered to the user asynchronously by means of a message. MMS allows users to transmit still images from compatible mobile phones with built-in digital cameras to virtually any device capable of receiving email. As one of the 3G services, the mail service is extended in Japan to enable users to email video clips as large as several hundred kilobytes either taken by the mobile device's camera or downloaded from sites. For this application, ISO/IEC and 3GPP standards have been adopted. The major technical issues in MMS are compression efficiency and multimedia content wrapper (file format). The file format specified by the MMS standard is the 3GGP file format.

Apart from asynchronous messaging, there are two methods of distribution between the multimedia information distribution server and a mobile device: streaming and download. The 3GPP standard-based file format is now also being used for content download service. The download method can use the file format for the distribution of media clips. To minimize initial playback latency, the file format also supports progressive download, which allows a client to start playback before the file is fully downloaded. To do this, the media tracks must be interleaved properly within the file so that the client receives short portions of each media type in turn. Here, the download method requires a reliable communication protocol between the multimedia information distribution server and a client, even though some transmission delay may

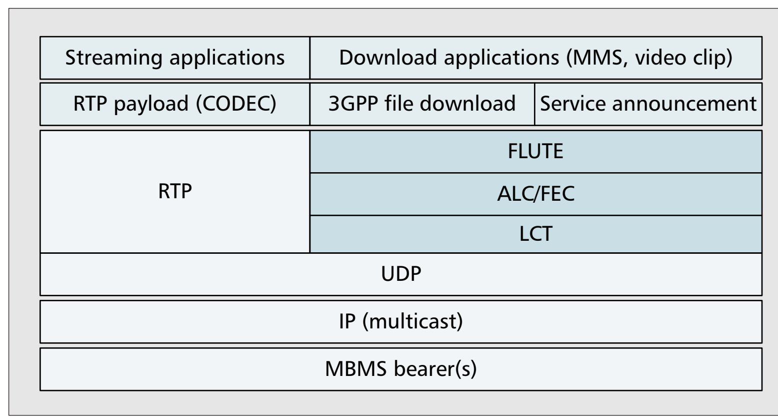


■ **Figure 2**. *Multimedia services in 3GPP standards.*

be tolerable. Communication procedures that meet this requirement include HTTP on wireless TCP/IP, which is tuned for high-latency networks.



■ **Figure 3**. *Protocol stack view of PSS.*



■ **Figure 4**. *Protocol stack view of MBMS.*

MULTIMEDIA BROADCAST/MULTICAST SERVICE

The previous three mobile applications assume the point-to-point model, where two single endpoints (e.g., client-server) are communicating one another. Because the mobile communication bottleneck exists at radio access, we expect that broadcast and multicast techniques will decrease the amount of data within the network and use the resources more efficiently. In this scenario the same information (e.g., news) can be delivered to many users simultaneously in a very efficient way.

3GPP extends the IP multicast paradigm to mobile networks. Multicast is a protocol for transmitting IP datagrams from one source to many destinations in a local or wide area network of hosts that run the TCP/IP suite of protocols. MBMS uses the IP multicast framework for users, where two logical underlying connections exist:

- Interactive bearer for service discovery, digital right management, subscription to groups, point-to-point data repair, and so on
- MBMS bearer for content delivery from a content server to clients

To enable MBMS, the core network should have a broadcast/multicast service center, and support nodes that are capable of broadcast area configuration, provisioning control, and multicast. The radio access network should also be extended to support link layer FEC, and so on.

Similar to PSS and MMS, two types of applications are anticipated:

- MBMS download (messaging): To push a multimedia message to clients
- MBMS streaming: Continuous media stream transmission and immediate playout

The protocol stack is designed to accommodate the above applications as illustrated in Fig. 4. The streaming stack is very similar to that of PSS; the download stack is unique in terms of its adoption of IETF reliable multicast standard, which ensures complete, reliable multicast/broadcast delivery in error-prone environments, even though reception periods and durations are hard to predict. Developed over several years by the Reliable Multicast Transport group of the IETF, the FLUTE standard (ftp://ftp.ietf.org/rfc/rfc3926.txt) aims to overcome the challenges of broadcast applications transmitting data concurrently to a large population of receivers.

As a protocol, FLUTE is fully specified and built on top of the Asynchronous Layered Coding (ALC) protocol of the layered coding transport (LCT) building block. File transfer is administrated by special-purpose objects, file description table (FDT) instances, which provide a running index of files and their essential reception parameters in-band of a FLUTE session. ALC is the adaptation protocol to extend LCT for multicast. ALC combines the LCT andFEC building blocks. LCT is designed as a layered multicast transport for massively scalable, reliable, and asynchronous content delivery. An LCT session comprises multiple channels originating at a single sender that are used for some period of time to carry packets pertaining to the transmission of one or more objects that can be of interest to receivers. The FEC building block is optionally used together with the LCT building block to provide reliability. The FEC building block allows the choice of an appropriate FEC (e.g., Reed-Solomon) code to be used within ALC, including using the no-code FEC scheme that simply sends the original data using no FEC coding.

MBMS standardization is still in process. It seems that its pure commercialization will need at least three more years.

CONTENT DELIVERY TECHNOLOGIES

In the previous sections we discuss standardized wireless applications. Here, we discuss supporting technologies, in light of wireless communication, from physical layers to application layers that are currently in use or planned in the near future.

LAYER 1/2 TECHNOLOGIES

Note that the trend in the emerging access networks is for use of adaptive modulation and coding (AMC). AMC adaptively changes the level of modulation — binary phase shift keying (BPSK), quaternary PSK (QPSK), 8-PSK, 16-quadrature amplitude modulation (QAM), and so on — and amount of redundancy for a error

When applying taxonomy to wireless video technologies in 3G and beyond, it is hard to provide a neat classification, because the combination of real-time/non-real-time, streaming/file download, and bi-/uni-directional communication complicates the abstraction of technical elements.

correction code. A higher level of modulation (e.g., 16-QAM) with no error correction code can be used by users with good signal quality (close to the base station) to achieve higher bandwidth. A lower level of modulation (e.g., BPSK) with more redundancy for error correction is used by users with bad signal quality (in the cell edge) to keep the channel condition, but results in lower bandwidth. The idea is to limit the number of link errors by adjusting the dedicated bandwidth through AMC in general; W-CDMA also adjusts the spreading factor and number of multiplexing spreading codes.

Regarding adaptive modulation, 1xEV-DO and wireless LAN (WLAN) have a similar strategy. Typically, the average bit error rate (BER) requirement is set beforehand, depending on the class of application. The adaptive modulation is applied to ensure the QoS that is evaluated directly by the signal-to-interference ratio (SIR) or indirectly measured by BER. Layer 1/2 transport control generally provides two distinct states, quasi-error-free and burst errors, during fading periods, while there is a large variation of bandwidth and delay.

Among the above mentioned services, video-related services require higher channel quality than the values presented in a number of research papers. In Table 1, the bottom row shows the practical operational requirements recommended by 3GPP (values with which we agree), where PLR indicates the packet loss rate at the transport (IP) layer.

For MBMS, the error requirement is defined in terms of the block error rate (BLER) and should be le-2, where block is defined as a data block passed by the physical layer to the MAC layer for a given transport channel (i.e., physical layer error rate). The block size varies with bit rate and transmission time interval (TTI), and is defined as *bit rate* × *TTI*. A 3GPP simulation reported the following results [6]. The assumptions used in the simulation are:

- MBMS data rate: 64 kb/s
- Block error rate: le-2
- TTI length: 80 ms (i.e., time length for turbo FEC coding)
- Channel model: A pedestrian model at 3 km/h
- Coverage: 90 percent (the percent of terminals that can receive the data)
- Selective combining: Over three radio links (RLs)

Selective combining (SC) is an enhancement in the recent 3GPP standard by which the network simulcasts the MBMS contents, and the terminal simultaneously receives and decodes the MBMS data from multiple radio links. Selection of the radio link is performed on a layer 1/2 transport block basis at the RLC, based on cyclic redundancy check (CRC) results and sequence numbers. The above conditions are optimistic, where three RLs are used in a pedestrian condition. To provide 90 percent coverage and 1 percent BLER in a cell, it is calculated that the fraction of cell transmission power must be about 10.7 percent. This means an MBMS service will occupy more than 10 percent of a cell radio resource even in very optimistic conditions. The report says that where no selective combining with 20 ms TTI is used, we cannot guarantee 90 percent coverage and 1 percent BLER with any realistic power consumption. We again realize the benefit of transmission power control and AMC over point-to-point radio links. Multicast communication over cellular networks is a very challenging technical issue.

Technologies above Layer 2

Major technical challenges exist when pursuing optimal media delivery in conjunction with application layer characteristics. When applying taxonomy to wireless video technologies in 3G and beyond, it is hard to provide a neat classification, because the combination of real-time/non-real-time, streaming/file download, and bi-/unidirectional communication complicates the abstraction of technical elements.

Table 1 provides a summary of our practical review of recent wireless video technologies.[1] As for source coders, we assume that MPEG-4 will be selected for video telephony and H.264 for packet streaming. MBMS is not limited to video application delivery, although we assume that MBMS is used for MMS broadcast and video clip distribution applications.

In **MPEG-4 video**, application layer error resilience tools were developed on the assumption that there is a benefit in having damaged data delivered from H.223. At the source coder layer, these tools provide synchronization and error recovery functionalities. Efficient tools are Resynchronization Marker and Adaptive Intra-Refresh (AIR). The marker localizes transmission errors by inserting code to mitigate errors. AIR prevents error propagation by frequently performing intraframe coding on motion domains. These application layer technologies are comprehensively described in [7].

As for **H.264/AVC**, there is considerable literature on the error resiliency future. H.264 is based on hybrid video coding, and is similar in spirit to other standards, such as MPEG-4, but with new sophisticated coding technologies, such as its prediction scheme. It is said that H.264 attains substantial bit rate savings (up to 50 percent) relative to other standards, such as MPEG-4, at the same subjective visual quality. The decoding complexity at least doubles that of MPEG-4, while the encoding complexity may triple (or more) that of MPEG-4, depending on the degree of rate-distortion (R-D) optimization and complexity of motion estimation. If complexity is not a crucial issue, compression efficiency is most important in the messaging and progressive download applications.

In contrast to the MPEG-4 error resilience structure, the H.264 error resilience structure is based on a flexible network adaptation structure called Network Abstraction Layer (NAL).[2] Unlike MPEG-4, the H.264 error resilience structure is based on the assumption that bit-erroneous packets have been discarded by the receiver. The error resilience structure has been designed mainly for packet loss environments, and this design concept is valid, since Internet protocols are currently widely used for content delivery. The NAL unit syntax structure allows greater customization of the method of carrying the video content to the transport

[1] *Cross-layer technologies must exist aside.*

[2] *Network adaptation is called "network abstraction" in H.264.*

Layer	Video telephony	Packet streaming	Messaging + progressive download	MBMS data
Source coder (application layer)	Error concealment, feedback-based error control (Adaptive Intra Refresh), reversible VLC	Error concealment Feedback-based error control	Compression as essential	Point-to-point data repair
Error resilience tools Network adaptation	Resync marker data partitioning Selective ARQ (AL3)	Slice interleaving, data partitioning, redundant pictures, packet scheduling, selective ARQ	Interleaving for progressive download	Interleaving for FEC above layer 2
End-to-end transport	H.223	RTP/UDP+RTCP	Wireless-TCP/IP	FLUTE
Layer-1/2 transport	FEC (BER: le-4)	FEC+ARQ (PLR: le-4)	FEC+ARQ (PLR: < le-4)	FEC

Table 1. *Possible wireless video delivery technologies.*

layer. A typical example is packetization to RTP payload. For example, the flexible syntax enables robust **packet scheduling**, where we send important packets earlier and retransmit lost packets so that we can improve the transmission reliability for important pictures. For further details, see another article in this special issue [8].

Let us move to end-to-end transport technologies. **Error concealment** has a long history; it has been available since H.261 and MPEG-2. The easiest and most practical approach is to hold the last frame that was successfully decoded. The best known approach is to use motion vectors that can adjust the image more naturally when holding the previous frame. More sophisticated error concealment methods consist of a combination of spatial/spectral and temporal interpolations with motion vector estimation.

One can also add optimization techniques to **feedback-based error control**, especially for streaming services. A feedback channel indicates which parts of the bitstream were received intact, and which parts of the video signal could not be decoded and had to be concealed. Having this feedback, the source coder typically uses the INTRA mode for some macroblocks (MBs) to stop interframe error propagation. That is also called a selective ARQ method. Originally, selective recovery of video packets in AIR works well. Reference picture selection is another well-known feedback-based control technique performed at the source coder layer.

FUTURE DIRECTIONS

Radio access networks (RANs) are going to be heterogeneous. In addition, we note that current RAN implementations offer a very low error rate QoS (approximately le-4 PLR), because layer 1/2 error control keeps error QoS as low as possible, thanks to the feedback channel interaction. Note that MBMS' multicast environment is the exception here.

We believe that emerging mobile networks will have the following characteristics:

- Large bandwidth and delay variation: Layer 1/2 transport results in large variations of round-trip time (RTT), jitter, and bandwidth, while trying to keep error rate constant.
- Two distinct error states: Error conditions are bisected into a good stable state and burst error state.

A typical RTT in recent mobile networks varies between a few hundred milliseconds and 1 s due to effects from other users and mobility. Arriving and departing users can reduce or increase the available bandwidth in a cell. Increasing the distance from the base station decreases the link bandwidth due to reduced link quality, and by simply moving into another cell the user can experience a sudden change in available bandwidth.

From these characteristics, we can conclude that the essential error control features are:

- Error control adaptability to a burst error state
- Rate adaptability to the bandwidth variation identified above

The issues are how to efficiently detect the state transition from "quasi no error" to "severe error" and how to adapt the delivery system to the state change over heterogeneous RANs. For error recovery and concealment in the burst error state, the essential element is rate adaptation to introduce data redundancy.

Concerning rate control over mobile networks, we have already discussed that rate control mechanisms utilize RTCP. Most of the rate control mechanisms in the legacy wired Internet assume that packet loss, delay, and jitter are caused by network congestion, and the available bandwidth should be measured in an end-to-end manner. In mobile networks, however; packet loss and jitter may also be caused by radio link errors. When conventional rate control mechanisms are applied to mobile networks, a sender cannot identify the network congestion condition correctly, and this leads to inappropriate rate control. A typical symptom is when a sender reduces its transmission rate even if the network is not congested. Network support to indicate the QoS change is more necessary in wireless networks. A good example is explicit loss notification (ELN) [9], which is a cross-layer approach that signals link layer information to the transport layer and then the application layer. A more generic cross-layer framework is desirable, where such signaling can contribute to more effi-

We identify rate control as the essential technology that provides extraordinary adaptability to varying bandwidth. To do that, signaling link layer information to the transport layer or upper layer should be developed. This is also important for progressive download applications.

cient congestion control and QoS adaptation by avoiding duplicated effort across layers. Although a framework has been discussed, as far as we know cross-layer video rate control, such as layer 2/3 interaction with RTP retransmission and layer 2 ARQ, has not been reported. We can expect further contributions in this area.

Broadcast or multicast video content delivery is a very challenging topic in terms of suitable layer 1 technology development and total optimization of reliable transport over heterogeneous transport. We may focus more on an application layer point-to-point data repair scheme to provide a more plausible scenario. The engineering direction should be toward more systematic total delivery system development that aims to optimize network utilization by combining point-to-point networks and point-to-multipoint networks. Content delivery scheduling of delivery timing, channel selection of unicast or multicast, and application layer selective retransmission (i.e., point-to-point data repair) are key technical issues to be explored.

Conclusion

We review four types of wireless video applications in 3G and beyond. To avoid having a biased view, we introduce related 3GPP standards of circuit-switched multimedia telephony, end-to-end packet-switched streaming, MMS, and MBMS. Then we review supporting technologies of those four applications. To highlight wireless video, we mainly discuss error control technologies and their future direction. We pointed out that layer 1/2 transport tends to provide two distinct conditions: quasi error free and burst errors during fading periods. In the former condition, upper layer error control technologies have a limited role. When considering this role, extraordinary adaptability of error control to the latter condition is essential. We also saw that the emerging mobile network QoS is going to have larger variations of bandwidth and delay. Thus, we identify rate control as the essential technology that provides extraordinary adaptability to varying bandwidth. To do that, signaling link layer information to the transport layer or upper layer should be developed. This is also important for progressive download applications.

So far, content file download typically with MMS seems a promising application. One of the greatest challenges in forecasting the usage of a new technology is predicting when and how a new killer application will emerge. No one can predict it; however, someone will create the innovation sometime in the future.

References

[1] M. Etoh and T. Yoshimura, "Advances in Wireless Video Delivery," *Proc. IEEE*, vol. 93, no. 1, 2005, pp. 111–22.

[2] 3GPP, "Codec for Circuit Switched Multimedia Telephony Service; General Description," Tech. spec., Group Services and System Aspects, June 2002, TS 26.110.

[3] 3GPP, "Transparent End-to-end Packet-switched Streaming Service Stage 1," Tech. spec., Group Services and System Aspects, Mar. 2002, TS 22.233.

[4] 3GPP, "Multimedia Messaging Service (MMS); Stage 1," 3rd Generation Partnership Project; Tech. spec., Group Services and System Aspects, Dec. 2002, TS 22. 140.

[5] 3GPP, "Multimedia Broadcast/Multicast Service (MBMS)," Tech. spec., Group Services and System Aspects, Nov. 2004, TS 26.346.

[6] 3GPP, "S-CCPCH Performance for MBMS," Tech. spec., Group Radio Access Network, Mar. 2004, TR 25.803 v. 1.3.0.

[7] R. Talluri, "Error-Resilient Video Coding in the ISO MEPG-4 Standard," *IEEE Commun. Mag.*, June 1998, pp. 112–19.

[8] T. Stockhammer and M. M. Hannuksela, "H.264/AVC Video for Wireless Transmission," *IEEE Wireless Commun.*, this issue.

[9] H. Balakrishnan and R. H. Katz, "Explicit Loss Notification and Wireless Web Performance," *Proc. GLOBECOM '98*, Nov. 1998.

Biographies

MINORU ETOH [M'92, SM'05] received his Ph.D. degree from Osaka University in 1983, 1985, and 1993, respectively. He started his career as a research engineer at the Central Research Laboratories of Matsushita Electric in 1985. In the 1990s he led an image communication research team and participated in MPEG-4 standardization. In May 2000 he joined NTT DoCoMo Laboratory, where he was involved in multimedia communication research and development. In 2002 he was appointed president and CEO of DoCoMo Communications Laboratories USA, Inc., where he has been conducting several research groups in charge of mobile network architecture, security, terminal software, and mobile media technologies. He is now at NTT DoCoMo as managing director of multimedia laboratories. He is also a visiting professor at Nara Institute of Science and Technology.

TAKESHI YOSHIMURA [M'99] received B.E. and M.E. degrees from the Department of Information and Communication Engineering, University of Tokyo, Japan, in 1997 and 1999, respectively. He joined NTT Mobile Communication Network, Inc. in 1999. Since then, he was engaged in research on mobile streaming media technology, and participated in IETF and 3GPP standardization. He is currently a research engineer of NTT DoCoMo Multimedia Laboratories, Yukusuka, Japan. His current research interests include audio signal processing and ultrasound communication.