Internet and Multimedia Communications

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INTRODUCTION

Multimedia communications involve digital audio and video and impose new quality of service (QoS) requirements on the Internet (Lu, 2000). Different multimedia applications have different QoS requirements. For example, continuous media types such as audio and video require hard or soft bounds on the end-to-end delay, while discrete media such as text and images do not have any strict delay constrains. In addition, video applications require more bandwidth than audio applications. QoS requirements are specified by the following four closely related parameters: (1) bandwidth on demand; (2) low end-to-end delay; (3) low delay variation (or delay jitter); and (4) acceptable error or loss rate without retransmission, as the delay would be intolerable with retransmission. Multimedia applications are classified into the following three categories:

- *Two-way conversational applications,* which are characterized by their stringent requirement on end-to-end delay that includes total time taken to capture, digitize, encode/compress audio/video data, transport them from the source to the destination, and decode and display them to the user.
- *Broadcasting services* where the source is live. The main dissimilarity from the conversational applications is that it is one-way communication and it can stand more delay.
- *On-demand applications* (e.g., video on demand) where the user requests some stored items and the server delivers them to the user.

In designing and implementing multimedia applications, the characteristics of these application types should be used to provide required QoS, but using network and system resources efficiently. Even though we say that QoS should be guaranteed, the user states the degree of guarantees. Usually, there are three levels of guarantees:

- *Hard guarantee,* where user-specified QoS should be met absolutely. Reserving network and system resources based on the peak-bit rate of a stream achieves hard guarantees.
- *Soft guarantee,* where user-specified QoS is supposed to be met to a certain precise percentage. This is suitable for continuous media, as they usually do not need 100% accuracy in playback. This type of guarantee uses system resources more efficiently.
- *Best effort,* where no guarantee is given and the multimedia application is executed with whatever resources are available. More networks function in this mode.

These different types of guarantees may all be needed in a multimedia session established using proper association control protocols such as C_MACSE (Kanellopoulos & Kotsiantis, 2006). Different levels of guarantee are used for different types of traffic and the user determines which type of guarantee to use. Besides, the charging policy is related to the level of guarantee and the most expensive is the hard guarantee, while the best effort is the cheapest. At the source, multimedia data are either captured live or retrieved from storage devices. The transport module accepts these data, packetizes and passed them on to the Internet. At the destination (sink), multimedia data are reassembled and passed to the application for playback of audio/video. Packet processing time differences, network access time differences, and queuing delay difference can cause delay jitter, which has to be removed at the destination before data being played out.

BACKGROUND

Through a number of subsystems multimedia data flows. For example, an audio segment is encoded at the audio server program, sent through the underlying transport network, and decoded at the receiving application. In a multimedia multiparty call, the end-user issues diverse QoS parameters values, which are interpreted onto specific performance parameter values in the communication subsystem and the operating system frameworks. To provide end-to-end QoS guarantees, an intensive effort is necessary from all subsystems, including end-subsystems, network hardware, and communication protocols, of a multimedia system.

Bandwidth On Demand

The speed of network links and routers in the next generation Internet will be improved radically so that network congestion will be uncertain and QoS guarantees will be provided by design. This endeavor will include optical wavelength-division multiplexing (WDM) technologies, being considered by Next Generation Internet (NGI) initiative (http://www.ccic.gov/ngi/).

Multicast Support

It is a common requirement of multimedia communication to send data from one source to multiple destinations. Efficient multicasting protocols can reduce bandwidth requirements (Paul, 1998). Given the multireceiver nature of video programs, real-time video distribution has emerged as one of the most important IP multicast applications and it requires bandwidth adaptability. Real-time video multicast applications have to adapt to the dynamic network conditions, but still offer reasonable playback quality to the receivers. Liu and Zhang (2003) present a survey on adaptive video multicast solutions. Because video and shared data are essential to many distributed tasks, audio of sufficient quality is a necessary condition for almost any successful real-time interaction.

Synchronization

Continuous media are characterized by well-defined temporal relationship between subsequent presentation units to be played. A presentation unit is a logical data unit that is perceivable by the user. *Multimedia synchronization* is the process of preserving the temporal order of one or more media streams. The problem of maintaining continuity within a single stream is referred as *intra-stream synchronization*; where as the problem of maintaining continuity among the streams is called *inter-stream synchronization*. These two types of synchronization are necessary for both live streams and for stored media streams presentations. Manvi and Venkataram (2006) proposed an agent-based synchronization framework to handle three synchronization mechanisms (point, real-time, and adaptive) at application service level depending on the life/run-time presentation requirements of the multimedia applications.

Adaptive Media Coding

Multimedia data should be coded in a way such that acceptable audio/video playback quality is still achieved, when some data packets are delayed extremely or lost. Coding multimedia data into multiple layers is the basic suggestion. Some layers are assigned high priority and they contain essential data to generate basic acceptable basic play out audio/video quality. Extra layers contain data that add additional details (or quality) to the basic quality and are assigned low priority. In the case of system overloading, low priority data are dropped first, leading to little effect to play out quality. This effect is named *graceful quality degradation*, and it can be obtained by the use of error control techniques such as forward error correction (FEC).

End System Support

End systems must offer mechanisms to handle multimedia data efficiently and effectively such as to provide end-to-end QoS guarantees (Lu, 1996). The end-system support for multimedia communications is required for two reasons. Firstly, the communications protocol stack is implemented mainly in software and it has thousands of instructions executed by the end system. If, for these instructions, the end system cannot guarantee the execution time, there will be no real-time communications system regardless of how well networking support is offered. Secondly, if the media data need to be compressed and decompressed before presentation, the processing time should be predictable. If not, a meaningful presentation is not obtained.

Hardware: Multimedia applications impose the following requirements on the hardware architecture:

- Digital audio and video are very data intensive, and therefore the hardware must have high data transfer throughput and high processing power.
- Parallel hardware architectures are preferred, as a lot of multimedia applications have to access several input/output devices simultaneously. In addition, multimedia host computers have usually I/O buses, which support lower transfer rates than of those of high-speed networks. This situation leads to the problem called "*mismatch in bandwidth.*" For the solution of this problem various network interface units must be implemented.
- The hardware architecture must be scalable to accommodate new input/output devices and applications.
- To support different types of data and applications, the architecture should be versatile and programmable.

Multimedia operating systems: They should meet the following requirements (Steinmetz, 1995).

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