A STUDY ON MULTIMEDIA COMMUNICATION

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ABSTRACT

Multimedia communication deals with the transfer, the protocols, services and mechanisms of discrete media data (such as text and graphics) and continuous media data (like audio and video) in/over digital networks Such a communication requires all involved components to be capable of handling a well-defined quality of service. The most important quality of service parameters are used to request (1) the required capacities of the involved resources, (2) compliance to end-to-end delay and jitter as timing restrictions, and (3) restriction of the loss characteristics In this paper we describe the necessary issues and we study the ability of current networks and communication systems to support distributed multimedia applications. Further, we discuss upcoming approaches and systems which promise to provide the necessary mechanisms and consider which issues are missing for a complete multimedia communication infrastructure. Multimedia communication is a part of everyday life and its appearance in computer applications is increasing in frequency and diversity. Intelligent or knowledge based computer supported communication promises a number of benefits including increased interaction efficiency and effectiveness. This article defines the area of intelligent multimedia communication, outlines fundamental research questions, summarizes the associated scientific and technical history, identifies current challenges and concludes by predicting future breakthroughs including multilinguality. We conclude describing several new research issues that systems of systems raise.

Keywords: Multimedia, Communication, Quality of Service, Reservation, Scaling

I. INTRODUCTION

Multimedia systems have attracted much attention during the last few years in the society as a whole and in the information technology field in particular. Multimedia communication comprises the techniques needed for distributed multimedia systems. To enable the access to information such as audio and video data, techniques must be developed which allow for the handling of audiovisual information in computer and communication systems. In this paper, we discuss requirements for the handling of such data in communication systems and present mechanisms which have been developed already or which are under development to fulfill the tasks necessary in distributed multimedia systems. We also discuss necessary issues which have not been studied in sufficient detail today and which must therefore be addressed in future. We first describe in a more precise way what we mean by the term “multimedia”. Unfortunately, “multimedia” has become a buzzword used to denote any kind of “new digital media” being manipulated or displayed by machines. This very imprecise (and very often employed) notion leads to a labeling of all types of media data computation, transmission, storage, manipulation and presentation with the term “multimedia”. Since the mid eighties we have proposed (and even from time to time we imposed) a much more crisp and restricted specification.

II. DEFINITION OF MULTIMEDIA COMMUNICATION

Multimedia communication is the presentation of digital information in the form of text, images, sound, motion graphics, web sites, and video.

Multimedia is the field concerned with the computer-controlled integration of text, graphics, drawings, still and moving images (Video), animation, audio, and any other media where every type of information can be represented, stored, transmitted and processed digitally.

A **Multimedia Application** is an Application which uses a collection of multiple media sources e.g. text, graphics, images, sound/audio, animation and/or video.

Timeline:

There has been interest in computer supported multimedia communication for the past three decades. We briefly characterize the major problems addressed, developments, and influence to related areas. We characterize some landmark developments using the above distinctions of analysis of input, generation of output, and interaction management.

Late 1950s

Input/output: Natural language interfaces (NLI) discussed at Dartmouth AI Conference. First integrated graphics/pointing system developed & deployed (SAGE)

1960s

Input/output: Laboratory investigations of VR, initial interest in NLI

General: First Conference on Computational Linguistics

1970s

Input: Applications of NLI

Output: Template generation systems. First speech to text systems.

Management: focus and dialogue coherence models

General: emerging commercial systems

1980s

Input: Commercialization of NLI; First integrated speech and gesture (e.g., Bolt's "Put that there").

Output: Creation of techniques for domain independent, rhetorically structured text (e.g., rhetorical schemas, communicative plans). Improved sentence/clause planning/realization. First multilingual generation systems. Automated graphics design.

Management: User and Discourse modeling. Model-based interfaces.

General: International workshops on user modeling, text generation, multimodal interaction (e.g., VENACO); government programs (e.g., DARPA intelligent user interface program), industrial visions of intelligent multimodal, multilingual interaction (e.g., Apple's "Phil", AT&T).

1990s

Input: Increasing spoken language applications. More sophisticated input analysis prototypes (e.g., partial, synchronous, and ambiguous input).

Output: Coordinated, multimodal generation prototypes. Standard reference model for presentation systems. Management: User adapted systems. Agents appear in commercial software (e.g., Microsoft™ Office Assistant). General: DARPA (e.g., Intelligent Collaboration and Visualization program (http://snad.ncsl.nist.gov/~icv-ewg/, "Communicator" spoken language architecture) and European Community Intelligent Information Interfaces (www.i3net.org) programs. First ACM international conference on intelligent user interfaces (IUI) . Readings in IUI. Emergence of media content analysis for new applications, e.g., news understanding, video mail and/or VTC indexing and retrieval.

III. REQUIREMENTS OF DISTRIBUTED MULTIMEDIA APPLICATIONS

Distributed multimedia applications have several requirements with respect to the service offered to them by the communication system.1 These requirements depend on the type of the application and on its usage scenario. For instance, a non-conversational application for the retrieval of audiovisual data has different needs than a conversational application for live audiovisual communication (e.g., a conferencing tool). The usage scenario influences the criticality of the demands. For example, a home user video-conference, say between parents and children, is not as critical as a video-conference used as part of remote diagnosis by a physician. Furthermore, the requirements of applications regarding the communication services can be divided into traffic and functional requirements. The functional requirements are multicast transmission and the ability to define coordinated sets of unidirectional streams. The traffic requirements include transmission bandwidth, delay, and reliability. They depend on the used kind, number, and quality of the data streams. For example, a bandwidth of 1.5 Mbit/s is typically required for the playback of MPEG-1 coded video. The end-to-end delay is more stringent for conferencing than for playback applications. In the former case, the delay should not be more than 150 ms. The play out of related streams must be done in tight synchronization (< 80 ms skew), hence, appropriate measures must be obeyed during data transfer. On the other hand, the reliability requirements are sometimes lower than for traditional communication applications, e.g., if a fault-tolerant data encoding scheme is used. Furthermore, retransmissions (which are used traditionally for the provisioning of reliability) increase the end-to-end delay and are often worse than lost data for multimedia applications

IV. QUALITY OF SERVICE

A system designed for the presentation of audiovisual data must take timing constraints into account which are due to the characteristics of the human perception of such information. Therefore, an overall QoS must be provided to ensure that such constraints are fulfilled. Since distributed multimedia applications need end-to-end QoS, all hardware and software components participating in this process (retrieving, transmitting, processing, and displaying the data) must handle the data accordingly – from the local resources at the sender side via the transport system, including all network components, to the local resources at the receiving side. This applies to end-systems, servers, and networks as well as to system software and applications. Most of the participating resources are shared among users and various processes. One approach would be to (over-) design them based on peak demands such that collisions between demands of multiple applications can never occur. Then it would not be necessary to provide any resource management functionality. Yet, such a scenario would result in huge costs and low resource utilization and, hence, is typically not practical. Thus, to provide a constant QoS during

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the run-time of an application, resource reservation and scheduling techniques must be applied. Another technique is to use filtering and scaling mechanisms which adapt the generated workload to the available resources by changing the characteristics of the transmitted data stream, e.g., lowering the frame rate of a video stream. Such mechanisms allow a smooth decrease in quality and are described in Section 7. Here, we discuss resource management techniques based on reservation and scheduling.

4.1 QoS Provisioning Steps and Components

In order to provide QoS by using resource reservation and scheduling, the following steps must be performed in turn at each system and component participating in the end-to-end application:

• QoS specification: the workload (i.e., the amount of traffic) and the expected QoS (e.g., the delay) must be specified to enable the system to determine whether and which QoS can be ovided.

• Capacity test and QoS calculation: When an application issues its QoS requirements, the mission control of the system must check whether these demands can be satisfied taking existing reservations into account. If so, the best-possible QoS which can be provided is calculated and the application is given a certain QoS guarantee accordingly.

• Reservation of resource capacities: According to the QoS guarantees given, appropriate source capacities, as, e.g., transmission or processing bandwidth, must be reserved

• Enforcement of QoS guarantees: The guarantees must be enforced by the appropriate scheduling of resource access. For instance, an application with a short guaranteed delay must be served prior to an application with a less strict delay bound.

4.2 QoS Classes and Layers

Several classes of QoS are typically distinguished, the extreme on one side is a hard "guaranteed QoS" where the reservation is based on peak requirements, the other is the "best-effort" approach where no reservation is made at all (and should therefore, at least with respect to QoS provisioning, better be called "no-effort"). Between these exist various forms of weaker QoS (statistical, predictive) based on average case and predicted assumptions. The hard QoS guarantee requires more resources and is, hence, more costly, than the 'weaker' approaches. Yet, in the latter cases, resources may not be available when needed for processing leading to worse quality.

V. ROLE OF RESOURCE RESERVATION PROTOCOLS

Besides the local resource management mechanisms at the participating end systems and routers, reservation protocols are needed to exchange and negotiate QoS requirements between these systems. These demands are accumulated in a Flow Spec (Flow Specification). Reservation protocols perform no reservation of required resources themselves. They are only the vehicles to transfer information about resource requirements and to negotiate QoS values between the end-systems and the intermediate network routers – they leave the reservation itself to local resource management modules. The initiator of a resource reservation is not necessarily a sender in the enforcement phase. E.g., with RSVP the reservation initiator is the data receiver. Nevertheless, the network nodes need to know always the direction of data flows for making reservations, e.g., for physical transmission lines with asymmetric capacity, and generally, for asymmetric reservations. In Section 5, we will discuss such protocols and their use for end-to-end QoS provision in some more detail.

VI. INTEROPERATION OF THE INVOLVED MODULES

The individual resource management systems need not necessarily work identically. They must be able to communicate using reservation protocols and should have a similar understanding of QoS specifications to avoid errors which might occur if specifications are translated between various forms. The QoS requirements may be mapped to resources in different ways at distinct nodes. In order to maintain QoS, subsystem builders establish a service policy. E.g., implementations might re-use the resource reservation mechanisms for, say, peak rate services until the definition of newer, say, VBR, services is finalized and a viable policy has been investigated

VII. QOS MODELS

A wide-variety of QoS models and architectures has been developed, e.g., Tenet (at UC Berkeley and ICSI) [BFMM94], HeiTS/HeiRAT (at IBM ENC, Heidelberg) [VoHN93][VWHW97], QoS-A (at Lancaster University) [CaCH94], etc. In the following we briefly describe the approach followed in the Internet community due to its foreseen influence on the future use of the Internet. The Integrated Services (IntServ [RFC1633]) activity approaches, in relation with the work on the RSVP protocol, a general solution for QoS guarantees in the future Internet. The RSVP protocol is used to transport FlowSpecs that adhere to Intserv rules. Two types of descriptions are used for the QoS specification: the traffic specification (TSpec) describes the behavior of a flow, and the service request specification (RSpec) describes the service requested under the condition that the flow adheres to the

VIII. QUALITY OF SERVICE IN COMMUNICATION SYSTEMS

8.1 Local-Area Networks

QoS can only be guaranteed in a networked environment if it is supported at the data link layer of a communication system. The widespread Ethernet networks have never been able to guarantee any kind of QoS due to the indeterminism of the CSMA/CD approach towards sharing of network capacity. The reason for this is that the collisions, which occur in the bus-based Ethernet if two stations start sending data on the shared line at the same time, lead to delayed service time. Although even the 10 Mbit/s version can sustain one single high-quality (Main Layer – Main Profile) MPEG-2 video stream, an interference of data traffic to other end-systems on the same network cannot be prohibited. Using Ethernet switches (Figure 2), instead of a bus-based topology, a star-wired network topology can be configured where each link connects a single end-system to the switch. Then, collisions can only occur inside of the switch and can be resolved by appropriate design of the switching unit. If an end-system participates in two or more concurrent communication sessions, one transmission can be delayed or aborted. Hence, the probability of collisions is reduced but guarantees cannot be given. A different approach would be to avoid collisions by controlling the beginning of transmissions from each station, e.g., in software using tightly synchronized clocks.

8.2 Network Layer – the Internet Protocol

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The provision of QoS has not been considered in IP Version 4, today's most important network layer protocol. It is designed to provide flexible, self-repairing communication. The type-of-service field in the IP header is typically unused; furthermore, the options in this field give more indications than detailed information usable for QoS support. Overall, support for continuous media was not an issue at the time of IP's design. A few details, however, have been included in the meantime. Multicast communication is provided by defining a set of multicast addresses in the IP address space. Multicast groups are maintained by adding and removing IP addresses to and from the multicast group using IGMP [RFC1112]. A multicast-capable router uses link-layer multicast support to forward a packet or, if that is not possible, forwards the data packet to multiple destinations. Version 6 of IP, the successor to the current IPv4 protocol, does not contain QoS support by itself but has been equipped with hooks which can be used by other means to set up reservations. The concept of a pseudoconnection, called Flow, is introduced which is a packet stream between sender and receivers. Flows may be established by means external to IPv6, e.g., by a reservation protocol such as RSVP. The header of each packet contains a Flow Label which can be used to identify for each packet to which flow it belongs. After a router has determined the flow a packet belongs to, it can identify the QoS to be supported and the resources allocated for its processing. In addition to the flow concept, IPv6 has a priority field which can be used by routers to process packets according to their urgency, e.g., high-priority packets containing data of interactive applications are preferred over low-priority traffic.

8.3 Real-time Transport Protocol

One of the Internet protocols that can be used in conjunction with reservation models at the network layer is the Real-time Transport Protocol (RTP) [RFC1889]. RTP is an end-to-end protocol for the transport of real-time data. An important application type supported by RTP is multi-party conferencing because of its support for synchronization, framing, encryption, timing and source identification. RTP has its companion RTP Control Protocol (RTCP), which is used to interchange QoS and failure information between the QoS monitor applications in the end-systems. RTP does not define any kind of QoS itself and does not provide re-ordering or retransmission of lost packets. However, it provides a sequence number that enables the application using RTP to initiate such steps. RTP is typically used directly on top of UDP/IP or on top of ST-2. In the former case, QoS can be guaranteed by the use of RSVP's reservation mechanisms for the UDP datagrams. In such a combination, the RTP stack provides the information necessary to make educated guesses about the behavior of the data stream based on RTP's knowledge of the data format. In addition to the base RTP specification, a number of companion documents exist that provide encapsulations for various continuous media formats such as M-JPEG or MPEG. Hence, RTP itself provides no real QoS support; it relies on other appropriate protocols and mechanisms.

8.4 Telecommunication Systems

Outside the Internet, wide-area multimedia communication is generally based on telecommunication networks. Analog telephone systems play only a minor role in multimedia communication due to their limited bandwidth. For wide-area connections, the bit failure rate in analog communications is exceptionally high, which may become fatal even for multimedia streams. Without these problems, the connection-oriented approach of telephony would be well-suited for multimedia communication. ISDN (Integrated Services Digital Network) is replacing analog telephony, at least in Europe and East Asia. As the name states, its goals are to provide for the integration of various services besides telephony. The bandwidth definitions for ISDN are made according to SDH (Synchronous Digital Hierarchy). The smallest data channel in this hierarchy is the B-channel with a fixed guaranteed data rate of 64 Kbit/s – this way, an audio sample is transmitted every 125µs. The data transmission is frame-oriented with variable-length frames. In contrast to the Internet protocols, ISDN uses out-of-band signalling as usual in telephony. The D-channel provided for that signalling offers a data rate of 16 kbit/s. The signalling traffic often does not fully occupy the D-channel. The spare capacity is used for additional data transmission services, but with lower priority than for signalling.

8.5 Open Issues

Many pieces of an overall infrastructure for distributed multimedia applications have been developed over the last years. So far, only some parts have found their way into systems of daily use. Others will be deployed in the future while some have been put aside (for varying reasons, e.g., complexity, questionable usefulness, ’political’ incorrectness, …). Various parts for the multimedia communication infrastructure are still missing and must be developed in the future to offer a complete solution. Examples for missing or incomplete parts, especially with respect to the QoS provisioning part of multimedia communication, are QoS routing and pricing mechanisms. Perhaps most important will be the verification of the suitability of the proposed mechanisms for large-scale use: for (few) large multicast sessions with many receivers as well as for many small, concurrent sessions.

8.6 Scalability

The multimedia communication methods designed for shared and distributed components must be scalable. With respect to multimedia applications, e.g., multicast video conferences, scalability has at least two aspects: (1) scalability with respect to the number of participants in one application, (2) scalability with respect to the number of concurrent applications. The first requirement states that it must be possible to transmit a flow (distributed via multicast) to a potentially very large number of participants. This is, for instance, the case in transmissions from IETF meetings or prominent lectures. To fulfill this requirement (as discussed in Section 5.2), mechanisms for resource sharing among participants and for the aggregation of reservations must be provided, furthermore, there should be no central component which has to process requests from new participants joining resp. old participants leaving such a conference. Based on their receiver-oriented reservation and flow joining concepts, RSVP and IP multicast support this requirement. ATM offers now the new leafinitiated join feature to reduce the load of any central component, yet, grouping concepts are still missing.

8.7 Routing

QoS driven routing algorithms are needed for the efficient establishment of reservations. These algorithms suggest one or multiple suitable paths towards a given target considering a given set of QoS requirements. Then one attempts to make a reservation on such a path. Without appropriate routing mechanisms which take QoS requirements into account, the setup of reservations becomes a mere trialand-error approach. A QoS driven routing algorithm has to consider the currently available capacity of a resource to avoid an immediate rejection of the reservation attempt and the QoS requirements of the reservation to find a route best-suited for this QoS. It should also consider the resource load after the routing decision to avoid using up the majority of resources on this route. Some of the problems to be solved with QoS routing are: how much state information should be exchanged among the routers; how often should this state information be updated; must there be a distinction between exterior and interior systems and if yes, how can it be made; is it possible to hide internal details of an autonomous system; can the complexity of path computation be managed?

8.8 Pricing

An important issue for the future success of distributed multimedia applications is the cost for any data transmission (i.e., with or without QoS) and the question ’why should a user ask for less then the best quality’ has always been answered with ’costs’. QoS methods have to take cost into account as an additional (possible negotiable) parameter. However, as discussed in [SCEH96], most research has focused on specific issues; architectural issues have most often been neglected. The issues to be attacked are among many others: who pays for a service, and how is this indicated, especially if the receiver benefits from the transmitted data; can the user specify a limit on its expenditures; how can fairness be provided such that each receiver within a multicast session pays its share, how can payment cross a firewall, how can a department or group be charged instead of the overall company? In addition to these aspects which apply to transmissions without QoS, further questions have to be answered in QoS provisioned systems, e.g.: how can resource consumption be “weighted” (e.g., delay vs. loss); what QoS do users accept for a specific price and which pricing schemes do they understand; how can fairness be provided such that all users – benefiting from a reservation made for a multicast transmission – share the costs in a fair manner?

IX. CONCLUSIONS

Multimedia communication has been (and certainly will be much more) used by various distributed applications: Video-conferencing, retrieval systems and video-on-demand will address all network types, LANs (e.g., in-house information systems), MANs (e.g., city information systems, campus networks) and WANs (e.g., distributed lectures). The provision of a well understood QoS is a crucial issue for the successful delivery of audiovisual and any other time sensitive data over networks and hence, for distributed multimedia applications. Within current networks this requires mechanisms like resource reservation or adaptive mechanisms such as scaling and filtering. The need for reservations was highly controversial a couple of years ago. Now, the concept of reservation-based QoS has found wide-spread acceptance, nevertheless there are still ’reservations about reservations’6 whose advocates consider reservations as too complex and propose adaptive mechanisms as overall solution. Neither reservation-based nor adaptation-based QoS support would be necessary if the available system resources would become abundant. Yet, we believe that resource demand grows at least at the same pace as available resources, hence, reservation will be necessary for quality demanding applications and users in the future

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