# Performance Analysis of Multimedia Traffic

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## INTRODUCTION

Videoconferencing and multimedia communication technologies in general are gaining momentum with the development of networked organizations. Efforts are currently underway to further adapt multimedia technologies to virtual organizations where multimedia communications find applications in training and education, customer relationships management, enterprise resource planning and many other areas (Camarinha-Matos, 2002).

Technologies that support voice over IP (VoIP) and video and voice over IP networks (VVoIP) are gaining wide acceptance in today's end-user communities. Internet service providers, equipment vendors and research consortiums have developed large-scale conferencing systems in support of user communities that have integrated VVoIP infrastructures into their data networks. These technologies are increasingly important and common for supporting daily work within virtual organizations.

Thus, it is becoming increasingly necessary to develop VVoIP-friendly network protocols and devices such as firewalls, network address translators (NATs), packet shapers and call-admission control systems that augment existing VVoIP system functionalities. These protocols and devices should not, however, degrade the end-user experience of audio-visual quality. In order to fulfil these requirements, the development of VVoIP-friendly network protocols and system devices requires a good understanding of the VVoIP traffic characteristics (Schulzrinne, 2005).

This article deals with metrics and models for signalling and transport protocols in multimedia communications systems. Overall characteristics of multimedia transport and application layer protocols are described. Special emphasis is made on quality-of-service requirements and constraints imposed on network performance by multimedia technologies and codecs in common use today. Specific methodologies and tools for measuring and analyzing performance of multimedia conferencing are also covered. Performance analysis, monitoring and troubleshooting techniques and tools are overviewed.

## BACKGROUND

Multimedia technologies are key components for collaborative applications and environments (Dewan, 2005). Throughout the last years, networked multimedia applications and services are becoming more and more important in Internet-based enterprise and organizational architectures. For an in depth discussion of the use of multimedia technologies in this context, the interested reader can refer to Vernadat (2005). Furthermore, concepts and technologies developed for networked games (Young, 2005) are being integrated into general multimedia collaborative applications.

Usability of multimedia communication technologies depends on optimum network performance due to time constraints as well as high and preferentially constant bandwidth requirements (Watson, 2001; Dewan, 2005). Most of these technologies share specific properties and underlying communication protocols. Multimedia traffic modelling approaches involve specific metrics and performance requirements on interconnection networks. These requirements usually focus on real-time constraints to a varying degree.

At present, the Internet lacks the quality-of-service (QoS) provisioning mechanisms that are required to support delay sensitive traffic such as real-time voice. The original design and development of Internet Protocol version 4 (IPv4) did not consider that this best-effort packet delivery network might be one day used do accommodate real-time constant bit rate (CBR) traffic such as voice and video flows. Even today, the Internet does not provide QoS. The load generated by new types of communications services related to multimedia and video transmission is becoming one of the major sources of traffic in WAN networks. Modelling this type of load is a prerequisite for any performance study (Manzoni, Cremonesi, & Serazzi, 1999).

## MULTIMEDIA TRAFFIC MODELS

IP multimedia traffic characterization can answer a variety of questions. A first class of questions addresses how users (clients) make use of multimedia material through a network. These questions include what mixes of media types and distributions of file sizes are accessed by users, what file size distributions arise for each media type, how many servers are accessed by clients and with what distribution. A second class of questions addresses the requests that a particular server receives: questions such as what is the time between successive accesses to a file in a server (of interest to cache designers), and what is the distribution of file sizes and media types requested. A third class of questions addresses how to best use mechanisms to improve network performance, such as caching and pre-fetching in a network (Watson, 2001).

To answer these questions, specific metrics, procedures, techniques and tools for measurement and analysis of network performance besides generic (application layer details unaware) are required. Some of these metrics are also specially relevant and have an specific meaning in the context of multimedia traffic (Schulzrinne, 2005).

It has been shown that audio and video traffic exhibits long-range dependence (LRD). Various processes have been proposed for modelling traffic with LRD and analyzing its effects on network performance (Beran, Sherman, Taqqu, & Willinger, 1995; Crovella & Bestavros, 1997; Ahn & Kim, 2000). Traffic and workload measurement studies have been performed for both streaming and videoconferencing systems. Quality of voice communications over Internet backbones has been studied and the results have been compiled into models for assessing quality (Markopoulou, Tobagi, & Karam, 2003).

In this context, specific models and studies of quality-of-service guarantee as well as supporting technologies and infrastructure for multimedia traffic are required (Shin, Lee, & Jay-Kuo, 2003). Measurement studies of videoconferencing traffic have revealed the impact of end-point technologies that use popular audio and video codecs as well as generic network status described by delay, jitter, loss and reordering on the end-user perception of audio-visual quality (Lee & Calyam, 2005). Metrics of user performance perception, further analyzed in the next section, can be both objective and subjective quality measures.

Streaming applications work best when there is little variation in their transmission time delay. Although such applications can tolerate occasional packet losses, variation in delays can cause noticeable degradation in the user-perceived quality of their service. A number of research results have been reported on the characterization of live streaming workloads in content delivery networks, providing insight on parameters such as popularity, arrival process, session duration, and transport protocol use. It has been found that popularity obeys Zipf's law, session arrival processes follow exponential distributions and session durations show long-range dependencies (Sripanidkulchai, Maggs, & Zhang, 2004). Audio flows exhibit significant consistency in data rates and are considerably more persistent than HTTP connections.

As for videoconferencing, H.323- (Kumar, Korpi, & Sengodan, 2001) based voice and video conferencing solutions are established popular technologies both in industry and academia. Session initiation protocol (SIP) (Johnston, 2004) applications have also been widely deployed in the Internet during the last years. However, these bandwidth intensive applications are often plagued by various performance problems (Calyam, Mandrawa, Sridharan, Khan, & Schopis, 2004). In particular, the impact on provisioning of backbones for latency sensitive traffic is significant with current technologies (Fraileigh, Tobagi, & Diot, 2003).

Performance of protocols that are employed for multimedia session signalling has been also analyzed through measurement studies and the high dependence on delay and jitter has been characterized (Fu, Atiquzzaman, & Ivancic, 2005). However, there is still a lack

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