User to Network QoS Parameter Transformation in Networked Multimedia Systems

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1 Introduction

Multimedia is currently being recognized as one of the major driving applications accompanying the "internet" rush, along with real-time systems, networks, tools, and methodologies. For example, services are emerging in both the commercial and research worlds: teleconferencing, audio/video data, and news-on-demand services. However, most systems have been custom-made without the user interface to facilitate usage for the non-expert. Typical multimedia users simply want to use the service without needing to know the technical details of the underlying systems. Since the resource allocation layer must take into consideration the underlying scheme in order to be able to effectively assign tasks to the processing units and network components, there is need for a tranlation layer that will take user-level information and produce network-level parameters.

Even though some effort has been dedicated to the API and to some forms of Graphic User Interface (GUI), the correlation of these with the underlying computing or network subsystems has not been clear. For example, at the network level, schemes have been proposed to grant resources with a channel admission algorithm, but without concern for jitter [CSZ92, ZSS94]. To solve the problem of high delays under higher loads and uncontrolled jitter, resource reservation schemes are used [BFM+96]. A versatile architecture, called $\mathcal{V}-$NET[FZM95], was introduced to address these concerns and is flexible enough to accept different scheduling disciplines in different sites, different admission control policies, and different types of traffic sources. The architecture is based on an end-to-end communication channel ($\mathcal{V}$-channel) which represents an association between the application's QoS specifications and the network resources.

Beyond real-time data transfer, the Lancaster Quality of Service Architecture (QoS-A) [CCH94], the QoS Broker [NS95], and the NetWorld [CM95] are some architectures that take into consideration both the user-level specification of the QoS and the reservation schemes for resources at the network level. The QoS Broker functions as an intermediate between the application and both the operating system and the network protocol subsystem (Tenet suite of protocols in this case [BFM+96]). QoS-A and NetWorld provide a framework to specify and implement QoS guarantees as service contracts. QoS-A incorporates QoS interfaces, management, and mechanisms across all network layers.

The NetWorld was further enhanced [CCG+96] to allow the users to select different modes of operation: *expert* and *non-expert*. In the non-expert mode, the user only needs to specify Quality of Service (QoS) such as "low", "fair", "good", and "excellent" instead of entering the specific QoS parameters, such as resolution, color, frame rates, etc. On the other hand, the expert mode requires the users to enter their own QoS parameters. For instance, for audio, the expert user must specify bit length, sampling rate and number of channels.

Even for these experts, the underlying network reservation protocol needs different parameters in order to accommodate the channels' requests for forwarding real-time traffic. Therefore, at the system level, the parameters must be expressed[1] in terms of: number of packets (n), periodicity of packets (r), and burst size (m) which are defined as network parameters in Berkeley's Dash system [ATW+90].

The purpose of this paper is to show how to convert the user-defined parameters to the network-level parameters. This is an essential part of the resource reservation mechanisms for any real-time operating system which intends to guarantee delivery of multimedia data from source to destination.

Jitter

The presence of jitter in multimedia systems can cause audio glitches, image discontinuity and errors in lip synchronization in video and audio. The amount of jitter necessary to be perceived by human subjects depends upon the media type. Experiments have been done to determine the amount of acceptable jitter and error in synchronization [Ste96]. The results of these experiments can be used to assign different level of quality of service as shown in Table 1.

Jitter is introduced when using packet networks to transport multimedia. Therefore, synchronization at the receiver is necessary. One way to handle jitter is to buffer complete data streams prior to presentations (see Concord synchronization algorithm [SNAS96])

2 Network System Parameters Conversion

The user-defined parameters are converted to the network parameters depending on the mode selected. The conversion is performed in two steps for the non-expert user, while the expert skips the first step. First, the translation depends on the selected media type: movie, data, video or audio conference, the user-defined parameters (low, fair, good, excellent). These are converted to resolution, frame rate, color, or channels, bit length, sampling rate, and bandwidth. Thereafter, a conversion into the network-level parameters take place. The value of delay, jitter, and burst for the different applications are taken or derived from [Roy94, MT93, CCH93].

[1] Since our work is based on the $\mathcal{V}-$net paradigm, we show here the system-level parameters for that architecture. Works such as the Tenet suite [BFM+96] would work analogously.

	Media	Mode, Application	QoS
video	animation	correlated	+/-120 ms
	audio	lip synchronization	+/- 80 ms
	image	overlay	+/-240 ms
		non overlay	+/-500 ms
	text	overlay	+/-240 ms
		non overlay	+/-500 ms
audio	amination	event correlation(e.g. dancing)	+/-80 ms
	audio	tightly coupled (stereo)	+/-11 ms
		loosely coupled (dialog mode)	+/-120 ms
		loosely coupled (background music)	+/-500 ms
	image	tightly coupled (e.g. music with nodes)	+/- 5 ms
		loosely coupled (e.g. slide show)	+/-500 ms
	text	text annotation	+/-240 ms
	pointer	audio relates to showed item	-500 ms/+750 ms

Table 1: Media Mapping Table

The bandwidth is defined as a product of resolution, frame rate, color and a *action factor* which is dependent upon the media type. For instance, if the media type is a normal movie, then *action factor = 1*. For an action movie, however, the *action factor = 2*, due to the greater demand on the resources to maintain the image quality with most compression algorithms. Note that the action factor does not apply to audio since the variation in bandwidth between the different types of audio sources is not of large scale.

In the second step, the network topology has to be taken into consideration, since the number of hops the data traverses is relevant to the delay that can be applied to the data. The network-level parameters (n, m, r) are computed by the following algorithm:

```
Computing r:
   if ( action factor != audio) and (quality >= good)
     r = delay / total nodes in segment * 5
   else if ( action factor == audio )
     r = jitter / 2
   else
     r = delay / total nodes in segment
```

```
Computing n:
     n = bandwidth * r / data_format * packet_size

Computing m:
   if ( r > 1 )
     m = burst / r
   else
     m = burst
```

As mentioned above, the maximum delay and the delay jitter are application dependent and can be derived based on the QoS chosen by the users.

3 Additional Work

The GUI introduced in [CCG+96] allows users to either (a) select the amount that they are willing to pay or (b) let the system compute the cost of the service for them. In either case, the system first computes the cost and correlated quality, then informs the user. For option (a), the system compares the computed cost and the user-defined cost. If the result cost is exceeded by the specified one, then the system prompts the user to pay more or select a lower quality if appropriate. On the other hand (option b), if the computed value is smaller than the selected one, then the system will automatically offer the user better quality.

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