Video Delivery over Broadband Mobile Networks

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ABSTRACT

Video streaming over wireless networks has undergone enormous development recently due to the continuing growth in wireless communication, especially since the emergence of 3G wireless networks. The new generations of wireless networks pose many challenges, including supporting quality of service over wireless communication links. This is due to the time-varying characteristics of wireless channel. Therefore, a more flexible and efficient bandwidth allocation scheme is needed. This paper is a part of ongoing work to come up with a more robust   scheme that is capable of rapidly adapting to changes in network conditions. The proposed scheme focuses on the wireless part of the network, providing high quality video service and better network resource utilization.

Keywords: Video streaming, 3G mobile Networks, Bandwidth Allocation.

I.   Introduction

The increasing number of mobile users and with the rapid progress of both wireless communication networks and computer networks, different kinds of mobile wireless environments have been presented. Also, the growing demand for new multimedia streaming services have spurred a great deal of research on video streaming over wireless networks. 2G, 2.5G and 3G of cellular mobile telephone networks have already been well known all around the world. 4G networks supporting IP are expected to be widely deployed in the near future. On the other hand, Wireless LAN (WLAN) standard IEEE802.11a,b&g, and   Personal Area Network (PAN) standard IEEE802.15 and Bluetooth are increasingly applied for wireless access to public internet. As a result, video communications in wireless IP networks is the major research area in recent years [1].

The third generation (3G) wireless systems, e.g. CDMA2000 and Universal Mobile Telecommunications System (UMTS), are becoming available, which, together with the advance of low bit-rate video compression technology, will lead to wireless video streaming era in the near future. Therefore, future broadband mobile networks are expected to support applications with diverse traffic characteristics and performance requirements such as multimedia applications.

In general, compressed video sequences are highly bursty in nature, i.e. various amounts of data might be generated during varying time intervals. In this case, the amount of aggregate incoming traffic is greater than the outgoing wireless link speed and packets have to be buffered. If this situation persists, packets will be dropped due to buffer overflows, which will in turn cause a degradation of an application's QoS. Thus, to guarantee the transmission of compressed video with consistent perceptual quality, the maximum bandwidth for encoded data has to be allocated. Also, the bandwidth is limited in wireless networks and the capacity of a wireless channel is fluctuating in response to the changing distance between the base station and the mobile host.

Due to the above problems, a more flexible and efficient bandwidth allocation scheme is needed. Thus, from the system point of view, it is important to allocate the available wireless network resources dynamically. On the other hand, with regard to traffic source, it is also necessary to employ a scalable compressed video in wireless environments. It has been shown that scalable multilayer video is suited to handle the variability of network conditions gracefully[9][25]. Therefore, this paper is a part of ongoing work to come up with a more robust scheme that is capable of rapidly adapting to changes in mobile network conditions.

The proposed scheme in this paper   focuses on the wireless part of the network, providing high quality video service and better network resource utilization. Thus the scheme is applicable to wireless and mobile networking environments due to the existence of large scale mobility requirements, limited radio resources, and fluctuating network conditions. The proposal is an attempt to develop a QoS-aware video delivery scheme that is required for the new generations of mobile and wireless networks, such as 3G and 4G networks. The multilayer video delivery scheme allows applications to delegate responsibility for augmenting or reducing the perceptual quality of video flows to the transport system. The augmentation or reduction is performed when mobile network resource availability increases or decreases respectively. This strategy is basically a QoS-controlled handoff that uses the notation of an adaptive network service. Different Differentiated services connections are provided for the scalable multilayer video streams, to the base layers and the enhancement layers, as the mobile devices roam. The active transport system uses the notation of mobile transport objects which can be dispatched on demand to strategic points in the network like Radio Network Control (RNC) in UMTS networks to provide value-added QoS support when and where needed.

II.   Multilayer Video Encoding

When one compresses a video sequence the following parameters must be determined: frame size, frame rate, data rate, and de-compressed quality. One of the problems with

many video compression methods is that these parameters are fixed at the encoding time and cannot be easily changed. There are several ways of adjusting the quality of compressed video stream. These may include adaptive encoding, switching among multiple pre-encoded versions, and multilayer (hierarchical) encoding. Considering adaptive encoding, the encoder re-quantizes data on-the-fly based on network feedback [18][19][20][21]. Since encoding is CPU-intensive, sources are unlikely to perform that for a large number of receivers. An alternative option is for the source to retain several versions of each video stream with a different resolution. Thus depending on the network's available bandwidth, the source would switch between the lower and higher resolution quality streams as required. This option requires large buffers at traffic source.

Multilayer encoding is also another option that is part of a family of signal representation techniques. It is a scalable compression technique that allows compressing the video data once and then decompressing it at multiple data rates, frames rates, spatial resolutions, and/or video quality (SNR). Such a encoding technique would be very desirable from a networking viewpoint as it allows differentiated quality and bit rates depending on the availability of network resources. Generally, it refers to an approach in which the video compression source scales its output compression rate by partitioning the video stream into sub-streams or layers, each layer representing a portion of the signal. The aggregation of these layers reconstructs the original data, but subsets of the data can also provide various degrees of approximation to the original signal as illustrated in Figure 1 and Figure 2. The greater the number of layers received at the end stations, the better is the quality of the reconstructed signal [8][9][16]. In this approach as more bandwidth becomes available, more layers of the encoded stream are delivered. If the available bandwidth in the wireless part of the network decreases, the network access points (RNC) would then drop some of the layers. The multilayer encoding usually have the decoding constrain that a particular enhancement layer can only be decoded if all the lower quality layers have been received. There are several advantages of multilayer encoding, including that of less storage requirements at the source, and provision of an opportunity for selective prioritisation of the important information. The design of an effective multilayered video transmission system basically entails the design of an efficient drop and adds layered mechanism that can maximize the perceptual quality of the received video stream.
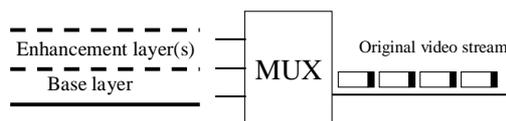
Figure 1: Multiplexing of layered data onto a single stream

Three scalability models could be used for multilayer encoding. On these models is SNR scalability, in which the base layer consists of a coarsely quantized version of the video, and the enhancement layers contain the refinement information. The SNR scalability relies on the DCT approach. However spatial scalability employs spatial pyramid encoding. In the temporal scalability approach frames are distributed between base and enhancement layers **[16]**. Temporal scaling can be accomplished easily if each frame is compressed independently without motion compensation. In this case, the frames can be freely distributed over different layers. However, MPEG compression schemes take motion compensation into account between subsequent encoded frames. So, if we distribute them among a number of layers without taking the frame dependency into account, a receiver, which eventually receives some of these layers, will not be able to decode the video. There are two possible ways to scale motion compensated video. The first approach is to encode the video independently on each layer. This approach results in transmitting groups of pictures (GOPs) on each layer between such groups dependencies are avoided. The second approach takes the structure of the GOPs into account. This paper adopts the second scaling approach, where all independent coded frames (I-frames) have to be transmitted in the base layer. On the second layer, the predictive-code frames (P-frames) are transmitted and the highest layer transports the bidirectionally predictive-coded frames (B-frames) [17]
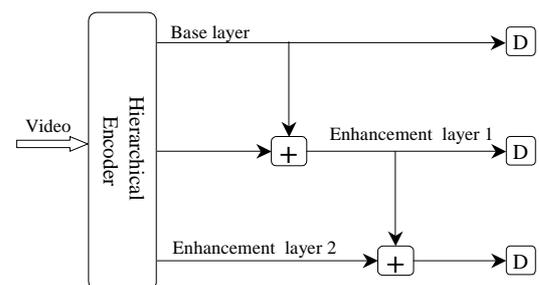
Figure 2: Multilayer video encoder.

III. Multilayer Video Delivery Scheme

The proposed scheme in this paper offers the possibility of implementing such scheme, on an end-to-end basis, both at the application and at the network levels. During the session lifetime when, a change occurs in the state of the mobile host, such as an occurrence of a handoff, in accordance with the proposed model, an indication is made by the mobile host to the bandwidth renegotiation unit in Figure 3, which in turn will ask the network to change the state of the connection. Such a dynamic mechanism offers the mobile host the guarantee of acceding to the requested level of quality, while letting flexible applications reach unused resources (higher utilisation rate).

The multilayer compressed video delivery scheme establishes several Differentiated services connections to transfer multilayer encoded video stream. Each connection has different traffic definitions. Thus the base layer, since it carries more important information than the enhancement layers, is transmitted separately on a more reliable connection. Each receiver must receive a base layer quality on a certain

connection and sufficient number of enhanced layers in other connection(s) depending on the mobile network available bandwidth [24].

The framework of the proposed scheme is illustrated on Figure 3 [24]. The framework shows the different components required for scheme implementation. The scheme's procedure is presented as a flowchart in Figure 4.
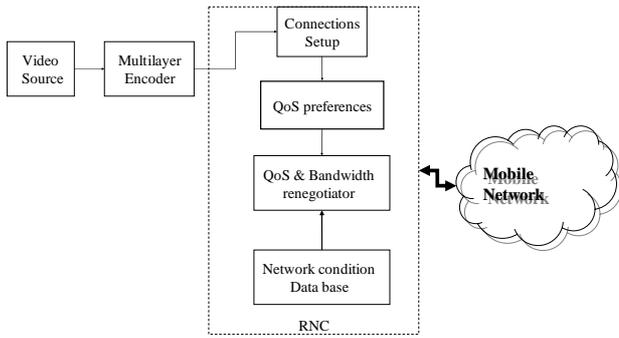
Figure 3: Multilayer compressed video scheme framework

The bandwidth renegotiation unit in Figure 3 makes its decision (based on the mobile network available bandwidth) If the mobile network has sufficient bandwidth, then the requested resources will be granted, otherwise a graceful degradation on source's QoS parameters should be considered at that time. This bandwidth constraint for adding a new layer is still not sufficiently conservative, as it may result in several layers being added and dropped with each cycle of renegotiation. Such rapid changes in quality would be disconcerting for the viewer. One way to prevent rapid changes in quality is to add buffering condition, such that adding a new enhancement layer does not endanger existing enhancement layers[36]. Thus, the RNC may add a new layer when: The instantaneous available bandwidth in the mobile network is greater than the required service rate for the base layer and the existing enhancement layers plus the new enhancement layer,

$$R > R_b + C_{new} + \sum_{i=1}^{n_a} C_i \qquad (1)$$

Or there are sufficient total buffering at the mobile host to survive an immediate backoff and continue playing all the existing enhancement layers plus the new enhancement layer.

$$B_b + \sum_{i=1}^{n_a} buf_i \geq \frac{\left(R_b + \sum_{i=1}^{n_a} C_i - \frac{R}{2}\right)^2}{2S} \qquad (2)$$

where
R : is the current transmission rate,

R_b : is the transmission rate for base layer,

n_a : is the number of currently active enhancement layers,

buf_i : is the amount of buffered data for enhancement layer i ,

B_b : is the amount of buffer space allocated to base layer connection,

S : is the rate of linear increase in service rate, and

C_i : is the required service rate for enhancement layer i .

The above constraints are the minimum requirements for adding a new enhancement layer. If these constraints are held, a new layer can be kept for a reasonable period of time during the renegotiation periods. If the available bandwidth on the mobile network is less than the required bandwidth to transport all enhancement layers a backoff occurs, the correct action is to immediately drop the highest enhancement layer. This is will reduce the required bandwidth to transport a multilayer video stream $(\sum_{i=1}^{n_a} C_i)$ and hence reduces the required bandwidth for recovery. If the available bandwidth in the mobile network is still insufficient, the RNC should iteratively drop the highest enhancement layer until the amount of bandwidth is sufficient.

$$While\left(R_b + \sum_{i=1}^{n_a} C_i > R + \sqrt{2S(B_b + \sum_{i=1}^{n_a} buf_i)}\right) \qquad (3)$$
$$Do \quad n_a = n_a - 1$$
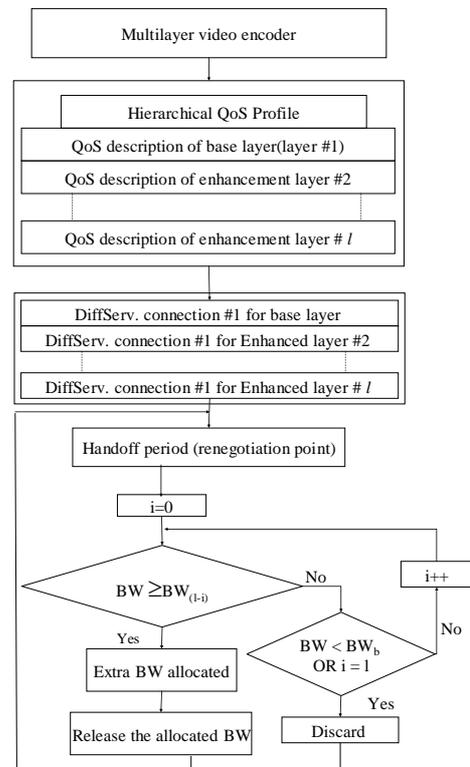
Figure 4: Scheme's proceduer

IV. Performance Evaluation

This section evaluates the performance of the multilayer compressed video delivery scheme. Several simulation experiments have been conducted using MPEG-4 traces. We start with description of MPEG-4 traces used in the simulation, then we proceed to present performance results in terms of loss rate, throughput, resources utilization, and signal to noise ratio (SNR).

The performance of the scheme has been evaluated and compared with traditional static video delivery schemes that uses a non-layered encoded video where a statistical amount of bandwidth is allocated permanently[31]. The availability of bandwidth in the mobile network is assumed to be limited and fluctuating as a saw-tooth function as shown in Figure 5 [18]. This is because our scheme is proposed to work under situations of high bandwidth contention environment. When this is not the case both traditional static and dynamic schemes will perform the same.

Traces of actual video sequences "Star Wars" are used with CIF sequences format and encoded with MPEG-4 at 10 frames per GOP [32]. We use the typical three-layer temporal-scalable MPEG 4 frame structure that was discussed in section 2. Two scenarios were studied through the simulation: firstly, the performance of traditional static bandwidth allocation scheme using non-layered encoded video, and secondly, the proposed scheme, which is basically a renegotiation-based dynamic bandwidth allocation scheme uses a multilayer compressed video.

In our proposal we assume that the bandwidth required by the base layer is always available in the mobile network. However, the service rate for the enhancement layers is adjusted dynamically according to the available bandwidth in mobile part of the network. Our model also assumes the loss is occurred only from the unavailability of bandwidth. A GOP is assumed lost if the available bandwidth in the mobile network is not sufficient to accommodate the whole GOP. The loss is calculated by the following equation:

$$Loss = \begin{cases} 0 & \mu \geq \lambda_{base} + \lambda_{en1} + \lambda_{en2} \\ \sum_{i=1}^{N} fr_i^{en2} & \lambda_{base} + \lambda_{en1} \leq \mu \leq \lambda_{en2} \\ \sum_{i=1}^{N} fr_i^{en2} + fr_i^{en1} & \lambda_{base} \leq \mu \leq \lambda_{en1} + \lambda_{en2} \end{cases}$$

$$(4)$$

where, fr_i^{en1} and fr_i^{en2} are frame sizes for enhancement 1 and enhancement 2 layers respectively in bits, N is the number of frames in GOP, where λ_{base}, λ_{en1} and λ_{en2} are the arrival rate of base, enhancement 1 and enhancement 2 layers respectively and μ is the available bandwidth.

Figure 6 shows loss ratio for non-layered video delivery scheme and the multilayer video delivery scheme. The larger loss is due to both high correlation and heavy tail distribution

of the input video stream. We have conducted the simulation under assumption of high contention and limited network bandwidth in the mobile network, to be able to compare our scheme to the traditional schemes fairly. Therefore an excessive cell loss will occur unless a large buffer size is provided. Even if one can afford this large buffer size, the problem of delay remains for delay sensitive applications. Even our proposed scheme permanently allocates a subset of bandwidth for the base layer of multilayer video stream, still it has better performance in terms of loss ratio than traditional video delivery schemes that use non-layered video compression.

Figure 7 compares the average throughput of traditional video delivery scheme uses non-layered encoded video with the proposed scheme. The performance of our scheme can be examined also from the bandwidth utilisation point of view, as shown in Figure 8. The multilayer compressed video delivery scheme achieves higher bandwidth utilization than traditional video delivery schemes.

The average utilization of bandwidth is calculated as the ratio between the instant required bandwidth to the reserved bandwidth.

$$U = \frac{1}{3}\left(\sum_i fr_i^{base}/B_{base} + \sum_i fr_i^{en1}/B_{en1} + \sum_i fr_i^{en2}/B_{en2}\right)$$

$$(5)$$

where, $B_{base}, B_{en1}, B_{en2}$ are the reserved bandwidth for base, enhancement 1 ,and enhancement 2 layer respectively.
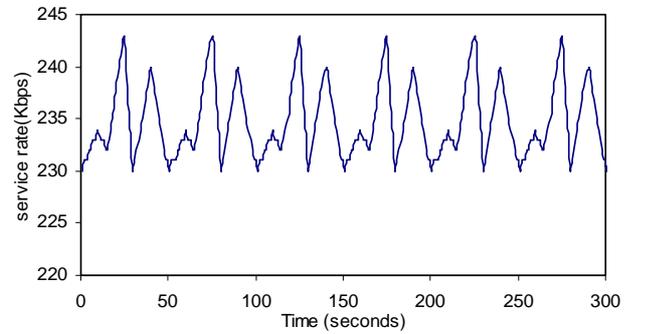
Figure 5: Service rate
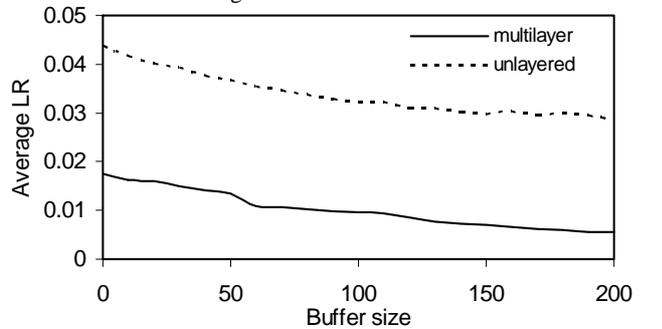
Figure 6: : Average Loss
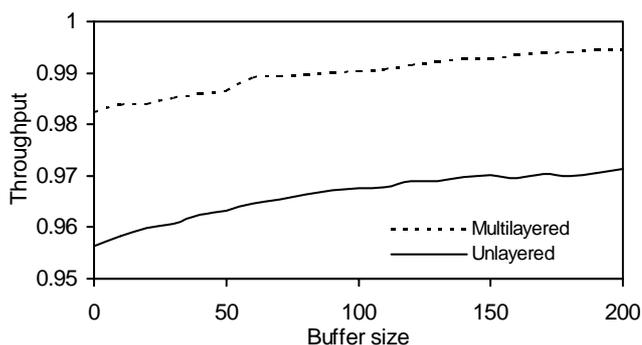
Figure 7: Average throughput
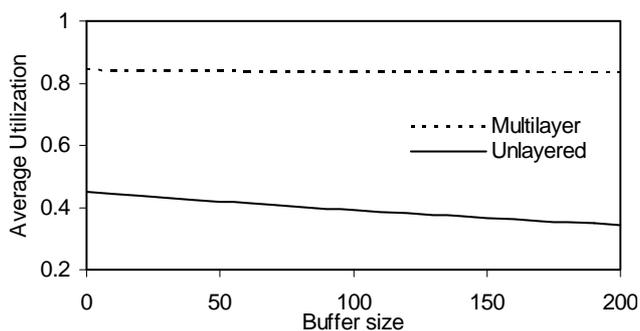
Figure 8: Bandwidth utilization

V. Conclusions

Wireless video streaming suffers from the same fundamental challenges due to congestion and the resulting best effort service. Packets still experience variable delay and loss. The proposed multilayer video delivery scheme as discussed above is important to mitigate these problems. The mobile radio channel, however, introduces specific additional constraints, and many of the resulting challenges still hold interesting research problems. Fading and shadowing in the mobile radio channel leads to additional packet losses, and hence TCP-style flow control often results in very poor channel utilization.

This paper has proposed a video delivery scheme to transport a multilayered compressed video for broadband mobile networks. The scheme aims to find a solution to the problem of varying bandwidth constraints over band-limited mobile networks. Since the compressed video is very bursty in nature, multilayer video encoders, like MPEG- 4, for example, may not alone be sufficient to achieve a high level of video quality and network utilization because of mobile network's bandwidth availability and source bit rate often vary from time to time.

To improve bandwidth utilization of such a network and to optimise the quality of the received video at the destination, the RNC must dynamically adjust the allocated bandwidth of the connection during its lifetime. Also in this model, when the mobile network cannot provide extra resources, the RNC then gracefully drop some of the enhancement layers of the compressed video based on the currently available bandwidth.

According to our simulation results, the proposed video delivery scheme has achieved better performance results than the traditional video delivery schemes based on static bandwidth allocation scheme using an non-layered video compression model. This scheme works under high contention and limited network resources in wireless part of the mobile network.

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