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Optimization of Video Streaming Quality in 802.16

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Abstract- 802.16 also called WiMax is the latest wireless technology that is being used. Video streaming which is one among the multimedia applications is being tremendously used with the increased use of internet. In centralized networks such as WiMax the base station takes care of allocating the appropriate time or frequency slot for all the subscriber stations that are competing for the medium depending on the type of traffic that each subscriber station holds thereby providing quality of service support by differentiating among the traffic types. The traffic in WiMax is prioritized by the scheduling service that is employed at the MAC layer. However encoding video with efficient encoding codec's is necessary for optimizing the video quality thereby minimizing the distortion at the receiver end. There may be several reasons for distortion in the video quality at the receiving end like transmission errors, loss of important video frames etc, However we concentrate only on the second one i.e. distortion due to loss of any important frames which would obviously result in deteriorating of the video quality at the receiver end. The system uses MPEG-4 for encoding the video.

Index Terms*- WiMax, QOS, IEEE 802.16 and MPEG-4*

I. Introduction

IEEE 802.16 is a set of standards for building up a wireless network similar to that of Wireless Local area networks. WiMax stands for worldwide interoperability for micro wave access technology. The name "WiMax" was created by **WiMax Forum**, which formed in June 2001.The basic intention was to promote conformity and interoperability of the standard. However what differs WiMax from WLANs is that WiMax supports high data rates and more coverage range (i.e. more number of subscriber stations can be served by a base station). It is said that Wi-Fi is capable to cover up to a range of just 30 meters where as WiMax can cover a radius of 50 km. WiMax supports TDMA or FDMA duplexing techniques at the mac layer. These techniques are responsible for performing the Quality of service differentiation i.e., base station may serve the subscriber stations either by employing TDMA or FDMA.With TDMA each subscriber station is given a different time slot (slots are of variable lengths depending on the type of traffic the station has to transmit) with same frequency. Incase of FDMA all subscriber stations can transmit at same time but at different frequencies. Unlike 802.11, 802.16 are centralized, which means any subscriber station if wants to transmit needs to get its appropriate slot from the base station. However when considering TDMA mechanism the wireless medium is divided into frames of time slots. Each frame is then divided into two sub frames. One is the uplink sub frame and the second downlink sub frame. Uplink sub frame is used for transmission from subscriber stations to the base station and the downlink sub frames are used by the base station for transmitting to the subscriber stations [8]. However what is required is that at the MAC layer both base station and subscriber stations be synchronized at all times there by transmitting in their appropriate time slots.

Different types of traffic can be streamed in network. When streaming video in any network what is important is that it needs to be sent fast and in order. In order to reach this constraint we opt for differentiated services, which means differentiating the service depending on the type of traffic. In general, WiMax at the mac layer supports five scheduling services. Through these WiMax is allowed to allocate different bandwidths to different stations depending on the priority of the application that each station holds. The five scheduling services are 1)Unsolicited Grants Service(UGS) 2)Real-Time Polling Service (rtps) 3)Extended Real-Time Polling Service(ertps) 4)Non-Real-Time Polling Service (nrtps) 5)Best Effort Service. Both UGS and rtPS scheduling services are designed for applications that do have any specific delay requirement. We concentrate only on rtps which supports real time multimedia applications that generates variable size data packets like for example MPEG (moving pictures expert group) and VOIP (voice over IP).

II. IDENTIFY, RESEARCH AND COLLECT IDEA

In this section, we describe how video is encoded in an efficient manner. With this efficient encoding the distortion at the destination is reduced to a maximum. When a video needs to be streamed in the network it first has to be encoded. There are several encoding techniques available which include H.264, MPEG2, etc., we have chosen MPEG-4 standard for compressing the video. However what is important before streaming video is to exploit video data characteristics by considering the most significant information from the application layer. This will lead to improve the quality of the streamed video at the receiver end.

The MPEG-4 standard defines three types of video frames for the compressed video stream, which include I (Intra-coded) frame, P (Predictive-coded) frame, and B (Bi-directionally

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predictive-coded) frame. The MPEG I frame is an independent frame which means both its encoding and decoding is said to be independent which means that it can be coded as a still image, with no relationship to any of its previous or successive frames. The P frame is however said to be dependent on its preceding I or P frame. Encoding P frame uses prediction from the preceding I or P frames in the video sequence. Therefore the P frame requires the information of the most recent I frame or P frame for encoding and decoding. The B frame is dependent on the preceding and succeeding I or P frames. It is encoded using predictions from the preceding and succeeding I or P frames. According to the coding relation, in MPEG-4 video stream the most significant video type is the I frame, with the P frame being more important than B frame [3].

Figure 1: Prediction encoding of MPEG-4, GOP (N=9, M=3) (from CH-Lin et al [3])

In general video sequence can be decomposed into smaller units, which we call GOP (Group of Picture), similar to a deterministic periodic sequence of frames (shown in figure 1). A GOP pattern is characterized by two parameters, G (N, M): N is the I-to-I frame distance and M is the I-to-P frame distance. For example, G (9, 3) means that the GOP includes one I frame, two P frames, and six B frames (as shown in figure 1). The next I frame successive to the 9 frames marks the beginning of the next GOP. B frames and P frames decoded are dependent on the preceding or succeeding I or P frames and are indicated by the arrows in the above figure. For MPEG-4 video stream, loss of any important frames would obviously result in deteriorating of the video quality at the receiver end. For example, the loss of one I frame would make all frames in the same GOP to be undecodable; while at the same time, one B frame loss just affects itself. However based on the importance of video frames at the MAC layer they are prioritized with the I frame as the highest; the P frame below I but above B's priority, and the B frame given the lowest priority.

Generally compressed video data is said to be highly sensitive to errors. There may be several reasons that cause the decoding failed at the receiver end and shall result in distortion of the pictures. Two of such reasons include transmission errors or dropping of some important video frames due to network congestion etc. What is important is to take care of the encoded video by properly setting up the priority to the encoded video frames in such a way that the most important frames say like I frames are not lost. Loss of such frames will result in the entire GOP to be undecodable by the receiving station. This results in minimizing/avoiding loss of the most important frames thereby avoiding the distortion at the receiver end. Therefore before directly streaming the encoded video we set priorities to the frames depending on their type to inform the network about their prioritization. WiMax takes care of setting up the priorities to the different traffic types that the different subscriber station holds. The Mac layer takes care of QOS management in WiMax. However multiple traffic streams by a single subscriber station is a total different concept. Also a PRD model is used to relate the available power with the rate of encoding so as to minimize the distortion [Cheng et al. 2006]:

$$D(P_s, r_s, \sigma_s, \mu_s) = \sigma_s^2 2^{-\mu_s r_s g(p_s)}$$

Where σ^2_s is the video sequence variance, μ_s is the encoder efficiency, and the function $g(p_s)$ represents the mapping between the video coder complexity and the microprocessor power consumption which is given by: $g(p_s) = p_s^{1/\gamma}$ where $1 \leq \gamma \leq 3$ is a system parameter [8].

III. Simulation Setup

To evaluate the performance of the proposed solution we have used network simulator NS-2. The video used in the simulation is YUV QCIF (176 x 144), Foreman. Each video frame was fragmented into packets before transmission, while the maximum transmission packet size over the simulated network is 1000 bytes. A total of 15 nodes were used in the simulation setup. Apart from video there exist other flows like voice, constant bit rate and TCP traffic flows, for which voice is given the highest priority, video the second, CBR the third and finally TCP. Among these nodes one is a video sender and the other a video receiver. Where in a sender sends to the receiver via the base station. Higher priority video frames (I frames) are given higher priority during the transmission. However different traffics are started at different simulation times. Incase if the nodes move far away from the coverage range of the base station then the packets are said to be dropped.

IV. Performance Evaluation

In this section, we report our evaluation results that illustrate the effectiveness of our encoding based quality optimization in 802.16 against the optimization in 802.11e Wireless LAN. Fig.1 shows a resultant graph that reads the X and Y axis values from two trace files resulting in a comparison for distortion with 802.11e versus 802.16.

The x-axis indicates the time and Y-axis the distortion. The distortion in case of 802.11e for optimizing the video quality is said to slowly increase as time passes when compared to that of the 802.16 optimization of video quality with an efficient encoding technique. Also it is clear that after certain time in the simulation the distortion is said to be stable in case of the proposed solution but it is not so in case with 802.11e WLANs.

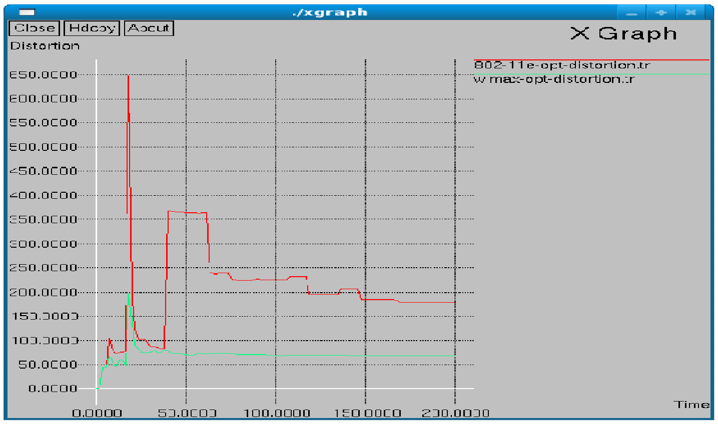
Fig. 1. X-Graph comparison for distortion in case of 802.11e with 802.16

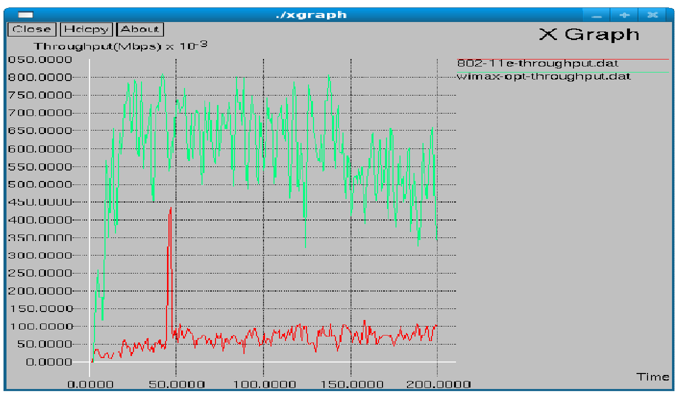
Fig. 2. X-Graph comparison for distortion in case of 802.11e with 802.16

Fig .2 shows a resultant graph that reads both the X and Y axis values from two trace files resulting in a comparison for throughput with 802.11e versus 802.16.

The x-axis indicates the time and Y-axis the throughput. The throughput in case of 802.16 optimization of the video quality is very much high when compared to the throughput of 802.11e optimization of video quality.

V. CONCLUSION

We present the optimization of video streaming quality in WiMax networks with an efficient encoding mechanism that actually prevents the loss of the most important encoded video frames during their transmission in the network. Thereby reducing the distortion of the streamed video at the receiving end. Our investigation showed that this solution achieved good performance when compared against the cross-layer framework for optimization of video quality in WLANs.

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