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Dynamic bandwidth management scheme for video stream on a IEEE 802.11 WLAN

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Abstract. This paper proposes and analyzes the performance of streaming video transmission over IEEE 802.11 WLAN. The simple bandwidth allocation method was proposed first to enable dynamic update of transmission schedule on AP. With a continuous trace of channel status of each video stream channel, the proposed scheme can efficiently utilize the network bandwidth, as it minimizes the useless transmission on the *bad* channel, without jeopardizing the timely transmission of other active streams. Simulation result shows that the proposed scheme can achieve as high as 91.2 % of achievable bandwidth.

1. Introduction

Video streaming services are services in which continuous video and audio data are delivered to an end user[1]. Video streaming enables users to view videos within system-dependent playout delay after the end user begins receiving the data. This is in contrast to other schemes that require the user to wait for the entire video to download before it can be viewed. Since it can often take several minutes and longer to download whole video files, video streaming offers the advantage of being able to view the video soon after it begins downloading. This service option is generally used in multimedia information and message retrieval, video-on -dem and, interactive news retrieval and search, multi media broadcasting and so on. Longer playout delay is allowed for streaming services because of its one-way transmission nature and buffering at the ter minal. Consequently, some transmission errors can be dealt with by utilizing retransmission. Terminal buffering is used to minimize the influence of delay jitter and retransmission delays, and to achieve seamless playback, as shown in Figure 1.

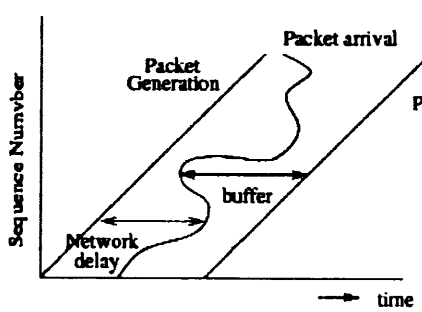

Fig. 1. Playback buffer

In the view of communication system, trans mitting video streams has proved to be a very challenging goal to attain. In most cases, trans mission of video streams requires significant resources, and can put a heavy burden on the system. Because they contain much more infor mation than voice alone, they demand much higher throughput. In order to respond to the need for reducing the throughput requirement, many video codecs are optimized for compression, so streaming service is very sensitive to any transmission errors. This service will also include system control protocols for setting up connections between parties involved with the streaming services, negotiating various codecs that the video streaming services use.

With wireless networks gaining prominence and acceptance, especially the LANs based on the IEEE 802.11 LAN standards, it is foreseeable that streaming of video will be a critical part of wireless digital infrastructure [2]. However, video streaming application has not yet been studied extensively for IEEE 802.11 based wireless networks. There are two major challenges for video streaming over WLANs, namely, fluctuation in channel quality and high bit-error rates compared with wired links. Specifically, the error resilience aspects are not explicitly considered[3]. Moreover, in case there are multiple clients, for example, in smart classroom where several mobile devices located within a small area simultaneously receive the stream service, each user will have different channel conditions, power limitations, processing capabilities, and only limited feedback channel capabilities.

This paper will propose and analyze the performance of a bandwidth management scheme for video stream traffic on a IEEE 802.11 WLAN. We focus on the video streaming scenario in the last mile between the base station, or AP (Access Point), and the mobile devices because it is likely the bottleneck of the whole video streaming system. As the WLAN is a kind of medium shared by multiple stations, its bandwidth should be deliberately maintained. AP takes the initiative of a WLAN, keeping track of all the stream connection such as connection establishment or tear-down, error state change, bandwidth requirement, and so on. According to the bandwidth allocation, AP schedules the transmission of downlink stream packets, or initiates the contention period by which a station can transmit uplink packet, for example, VCR commands that are directed to the remote stream server. AP will also continuously estimate the channel condition and on every change in the condition, bandwidth will be recalculated to avoid useless transmission and enhance the network throughput. Though it may seem to increase the complexity of AP operation, considering that the wireless channel has quite an unstable error characteristic, it does not matter if the calculation is very simple.

The rest of this paper is organized as follows: Section 2 describes background of this paper including network, error, and message models. Section 3 proposes the bandwidth management schemes and Section 4 shows the result of performance measurement. Finally, Section 5 summarizes and concludes this paper with a brief introduction of future works.

2. Background

2.1 Network model

We consider a wireless-cum-wired network scenario as shown in Figure 2[4]. A fixed node is connected with an AP through a wired link which is overprovisioned so that no packets are dropped at its end. The wireless portion is an 802.11b WLAN with N MNs (Mobile Node). Each cell is assumed to consist of a AP and multiple mobile nodes, while each flow is either uplink from MN to AP or downlink from AP to MN. In the hot spot multimedia model, most flows are downlink, and the stream flow arrives periodically from the remote server outside the cell via reliable wired link. Each MN can directly communicate only with the AP, since we focus on AP-coordinated wireless networks, disregarding ad hoc mode. The RTS/ CTS (Ready-To-Send/Clear-To-Send) mecha nism is employed, as in the general case. The AP should schedule the transmission of each flow, considering so many factors such as time constraints, accumulated service rate, current error status, and so on.

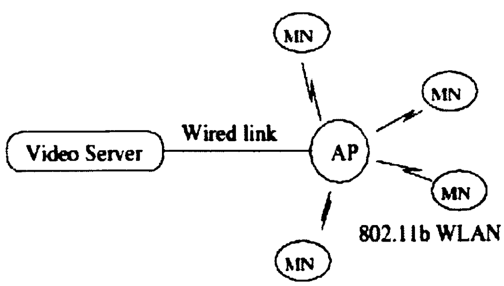

Fig. 2 Network model

The IEEE 802.11 was developed as a MAC standard for WLAN[5]. The standard consists of a basic DCF (Distributed Coordination Function) and an optional PCF (Point Coordination Function) for uplink channel access. In the contrary, for downlink transmission, neither contention nor collision will occur as there is only one traffic source to WLAN, namely, AP. AP distributes the stream traffic to the receiver via the WLAN as well as relays the other non-real-time traffic. To simplify the transmission schedule AP divides the time axis into the series of superframes, and each of them contains stream phase and general traffic phase, as shown in Figure 3. In stream phase, a portion of time amount is allocated to each stream, and the length of time interval is calculated according to the traffic requirement specified with QoS parameters. Every stream is serviced once per superframe. general phase can be thought of as the period for uplink traffics employing DCF in case the uplink and downlink time-shares the common frequency channel. This architecture is able to make the allocation procedure simple and efficiently response to the channel state change.

There has been previous works on providing QoS guarantees over wireless links using call admission and scheduling[6]. In addition, several approaches have been introduced to deal with real-time task scheduling in an overload condition, where the system cannot meet the deadlines of all tasks. The overload condition is analogous to the wireless network that experiences occasional bandwidth limitation due to network errors. The notion of (m,k)-firm deadline model, it is adequate to meet the deadline constraints of m out of k consecutive instances of a task in the overload condition. Besides, imprecise computation model is based on the minimal quality of service in the overload condition. This model is appropriate for error-prone wireless network.

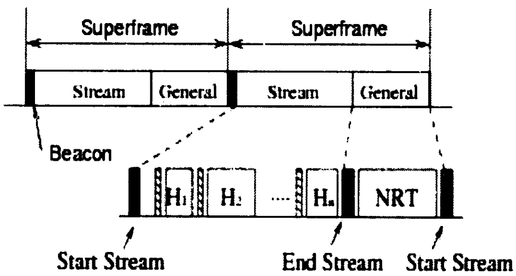

Fig. 3. Time axis of wireless LAN

2.2 Error model

WLANs may experience location-dependent channel errors, in that a MN can correctly communicate with the AP, while at the same time another one may suffer packet drops due to errors on the channel. In order to address these issues, an independent error model for each communicating pair of stations was introduced. Therefore, given N MNs, there are also N independent error models. A 802.11 radio channel is modeled as a Gilbert channel[7], where two states, good and bad, linked with a Marcov chain, represent the state of channel during an 802.11 slot time. A MAC protocol data unit is received correctly if the channel is in state good for the whole duration of transmission, otherwise, it is received in error. We denote the transition probability from state good to state bad by p and the probability from state bad to state good by q. The pair of p and q representing a range of channel conditions, has been obtained by using the trace-based channel estimation. The average error probability, denoted by ε, and the average length of a burst of errors are derived as $p/(p+q)$ and $1/q$, respectively.

In handling packet errors, one way is to simply drop the packet and another way is to reschedule the packet at some time later[6]. However, considering the bursty nature of wireless channel errors, immediate rescheduling of packet would not be either. Instead, it is desirable to delaying rescheduling the packet such that it can escape an error burst and can be delivered on time. Since errors are unpredictable, it is not clear how long rescheduling the packet should be delayed.

2.3 Message model

The video stream traffic occupies the network as a form of streaming-specific packets. For example, H.264 video encoder is configured to optimally match the networked scenario as well as to adapt to varying channel conditions[3]. In H.264, the base coding unit for transform coding is a 4 × 4 sample block and thus macroblocks are composed of 16 luminance block and 4 blocks for each chrominance component. Thereafter, consecutive macroblocks are grouped into a slice. This independently decodable slice is very useful to subdivide the coded stream into independently decodable packets, so that the loss of a packet does not affect the decoding of others. Moreover, in wireless channels, the size of the packet influences its error probability; longer packets are more likely to contain transmission errors but reduce coding efficiency.

Multimedia traffic is typically isochronous (or synchronous), consisting of message streams that are generated by their sources on a continuing basis and delivered to their respective destinations also on a continuing basis\cite{Car03}. Th QoS requirement is described and given to the network and the most important traffic characteristics of each stream are its period and message size. In case of a change in the stream set, bandwidth is reallocated. If we assume that there are N video streams in a cell, namely, S1, S2, ..., Sn, each stream can be modeled as follows: A message arrives at the beginning of its period and it must be transmitted by the end of period. The deadline is soft in that some delayed packet is permissible and transmission jitter is absorbed by a playback buffer in the video player. The period of stream, Si, is denoted as Pi, and the maximum length of a message as Ci.

3. Bandwidth management scheme

3.1 Channel estimation

AP maintains a state machine, or simply flag, associated to each video stream channel. The ACK/NAK is sent from the receiver to AP via uplink as soon as it receives a downlink packet. If the BS does not receive an ACK/NAK within predefined time-out interval, the packet will be assumed to be lost and may be, if possible, retransmitted[9]. AP sets the state to good whenever it receives from the corresponding MN, namely, a MAC-layer acknowledgment in response to a data frame, a CTS frame in response to an RTS frame, or other error-free frames. The AP sets the state to bad after a transmission failure.

Each bad channel has its own counter, and when a counter expires the AP attempts to send a single data frame to check the channel status. The duration of timer is reset to its initial value upon a transmission from bad to good, and the value is doubled whenever the probing fails in bad state. The value of timer should be set small so as to quickly recover from short channel error period.

3.2 Bandwidth allocation

The exact share of network bandwidth allocated to a flow depends on its requirements relative to the requirement of other flows. Each flow maps its minimum and maximum channel time proportion. The BM (Bandwidth Manager), co-located in AP, obtains from each flow its channel time requirement, at the start of its session, namely, in negotiating phase of video stream[10]. It uses this information to gauge proportion of channel time each flow should be allocated. The term channel time proportion is defined as the fraction of unit time for which a flow can have the channel to itself for its transmission. As only one flow can be served on the channel at a time, there is a direct correspondence between the channel time a flow uses and the share of the network bandwidth it receives.

We assume first a separate queue for each MN associated to the AP, and a channel condition estimator is also associated to each queue. Transmission is allowed only for those queues whose channel is estimated to be good. Queues attempting to access the wireless medium with a bad channel will be allowed to transmit again when their channel becomes good. By allocation, we mean the procedure of determining capacity vector, {Hi}, for the given superframe time, F, as well as message stream set, {Si (Pi, Ci)}. It is desirable that the superframe time is a hyperperiod of each stream's period and it is known that a message set can be made harmonic by reducing some periods by at most half[8]. So we assume that the superframe time is also given in priori, focusing on the determination of capacity vector. Hi limits the maximum time amount for which Si can send its message.

Let δ denote the total overhead of a superframe including polling latency, IFS (InterFrame Space) and the like, while Dmax the maximum length of a non-real-time data packet. For a minimal requirement, F should be sufficiently large enough to make the polling overhead insignificant. In addition, if Pmin is the smallest element of set {Pi}, F should be less than Pmin so that every stream can meet at least one superframe within its period. According to [11], for each superframe, at least a time amount as large as Dmax, should be reserved for a data packet to compensate the effect of deferred start of stream phase. After all, the requirement for the superframe time, F, can be summarized as follows:

$$\sum H_i + \Delta + D_{max} \le F \le P_{min} \qquad (1)$$

In addition, the minimum value of available transmission time, X_i is calculated as Eq. (2). Namely,

$$X_i = \left(\lfloor \frac{P_i}{F} \rfloor - 1 \right) \cdot H_i \quad \text{if} \left(P_i - \lfloor \frac{P_i}{F} \rfloor \cdot F_i \right) \le D_{max}$$
$$X_i = \lfloor \frac{P_i}{F} \rfloor \cdot H_i \qquad \textit{Otherwise} \qquad (2)$$

For each message stream, Xi should be greater than or equal to Ci(Xi≥Ci). By substituting Eq. (2) for this inequality, we can obtain the least bound of Hi capable of meeting the time constraint of Si.

$$H_i = \frac{C_i}{\left(\lfloor \frac{P_i}{F} \rfloor - 1 \right)} \quad \text{if} \left(P_i - \lfloor \frac{P_i}{F} \rfloor \cdot F \right) \le D_{max}$$
$$H_i = \frac{C_i}{\lfloor \frac{P_i}{F} \rfloor} \qquad \textit{Otherwise} \qquad (3)$$

The allocation vector calculated by Eq. (3) is a feasible schedule if it satisfies Ineq. (1). By this, we can determine the length of stream period Tstream and that of general Tgeneral as follows:

$$T_{stream} = \sum H_i + \delta, \quad T_{general} = F - T_{stream} \ge D_{max} \qquad (4)$$

This calculation is easily fulfilled just with simple arithmetic operations. Hence, it is possible to dynamically recalculate the bandwidth allocation with just active channels, whenever a channel status changes either from good to bad or from bad to good. If a channel triggers to bad state, the available bandwidth increases for other video streams. In that case, they can renegotiate with their servers so that more traffic will flow from the server.

4. Performance analysis

We measure the performance of the proposed scheme via simulation using ns-2[12]. To begin with, we assume that there are 10 MNs in a cell, 3-5 video streams exist simultaneously. Each video stream has the same traffic requirement for simplicity such as bit rate, error characteristic, and video packet size. Every time variable is aligned to F, for example, error duration randomly ranges from 0.00 F to 0.5 F. Error duration value 0.0 F means that there is no network error. This alignment makes it possible to abstract the network parameters which changes with the enhancement of network bandwidth, QoS requirement from the application, and error duration length. The length of packet length, Hi, is the function of F, so the larger the F, the longer Hi, for the given video stream. Within a Hi, several packets can be transmitted.

Every packet received in error can be detected by the BM. If a packet transmission duration overlaps with a time in the bad state, the packet is received error. Otherwise, the packet is received correctly. For the error duration ratio $p/p+q$, the duration Figure 4 plots the achievable bandwidth according to the error duration ratio. In addition, We are currently measuring the performance of the proposed scheme in the view of packet loss, degradation rate, delay jitter, and so on.

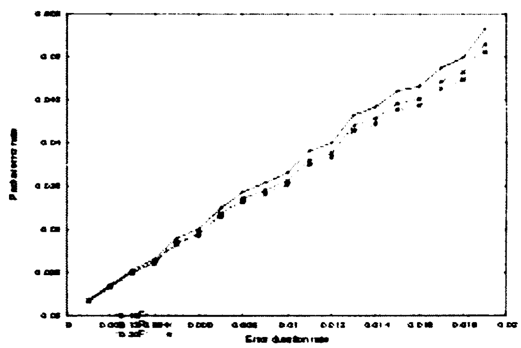

Fig. 4. Error characteristics

5. Conclusion

In this paper, we have proposed and analyzed the performance of streaming video transmission over IEEE 802.11 WLAN. The simple bandwidth allocation method was proposed first to enable dynamic update of transmission schedule on AP. With a continuous trace of channel status of each video stream channel, the proposed scheme can efficiently utilize the network bandwidth, as it minimizes the useless transmission on the bad channel, without jeopardizing the timely transmission of other active streams. Simulation result shows that the proposed scheme can achieve as high as

91.2 % of achievable bandwidth.

As a future work, we are planning to develop a retransmission scheme for the erroneously transfered video packet in WLAN to enhance the video quality. The retransmission should be completed before the packet data is played at the client, so the location of packet data in the playout buffer decides the deadline of retransmission. The dynamic scheduling in general period for the packets based on their deadline will recover the network errors.

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