A Cross-Layer Architecture with Service Adaptability for Wireless Multimedia Networks

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Abstract—**In this paper, a novel cross-layer architecture based on the hierarchical protocol structure is proposed to improve the quality of service (QoS) for the video transmission over wireless multimedia networks. In addition, an average bandwidth measurement approach is used to adjust the video delivery immediately. Furthermore, the proposed cross-layer architecture uses both the bandwidth measurement and queue adaptation for the current communication protocol of wireless multimedia networks. Finally, it is shown by NS2 simulations that the methods of both average bandwidth measurement and dynamic queue adaptation do not consume any network resource, and the proposed method can yield a better performance for transmission delay time, image quality and packets throughput over wireless networks.**

Keywords- Cross-layer architecture; bandwidth measurement; queue adaptation; wireless multimedia network

I. INTRODUCTION

The development of multimedia application over wireless network is growing faster in recent years. Because of the flexibility and the low-cost of the infrastructure, different kinds of wireless network technology for mobile stations provide a platform for various real-time, multimedia and network applications over wireless networks, including video conference, emergence service, surveillance, remote medical service, and remote access technology and training, etc. These applications largely increase the convenience and entertainment of the distant communication. The best-effort service supporting current wireless network technology, however, can no longer achieve such requirements of the multimedia network applications as the sensibility to delay time, fluctuations in bandwidth, and message drop. Fortunately, some multimedia network application techniques could be used to handle the hidden error at the receiver end when the packets with continuous characteristic had been lost. Unlike the transfer of file, there is no need for real-time multimedia network application to distinguish lost packets from others. But, in order to provide the most efficient QoS (Quality of Service) performance, the cooperation between application layer and lower layers during the transmission session is necessary.

Currently, most QoS mechanisms are proposed and applied on wired network environment. It is also a fact that almost every wireless network is eventually connected to the existing infrastructure which is a wired environment. Even the performance is guaranteed on wired part by QoS, the thorough performance can always be worse than the expectation, because of the lack of QoS mechanisms on wireless part. Unfortunately, the QoS mechanism applicable to wired network does not perfectly fit in wireless environment, since a wireless connection usually has the following characteristics:

- Higher bit error rate and burst of errors,
- Wireless connection capability of limited area and time variance,
- Limited bandwidth,
- User's mobility, and,
- Signal transmission power control while moving.

Although these characteristics are unavoidable, the most common wireless technology standard 802.11x has the advantage of the same architecture as OSI reference model and the advantage of supporting the same MAC protocols.

Many researches on improving QoS for wireless network application concentrate on the exact methods of the bottleneck bandwidth measurement, which usually rely on some helpful tools of bandwidth measurement. The extra execution of the measurement tools before an application's initialization, however, might heavily affect the overall performance of the application, and the measured results could also hardly be used to adjust the transmission rate immediately, which is more likely what more application users expect.

In the original IEEE 802.11 MAC (Media Access Control) standard, the distributed coordination function (DCF) mode does not support the QoS, and the point coordination function (PCF) mode does not provide QoS parameter setting like priority, delay, and jitter, etc, though it is a polling mechanism. Since the traditional wireless protocol didn't take real-time transmission into account, the 802.11e MAC mainly enhance the wireless network with adding the feature of QoS on real-time video data transmission and define the hybrid coordination function (HCF). HCF proposes two methods: EDCA (Enhanced Distributed Channel Access) and HCCA (HCF Controlled Channel Access) methods [6][17][18]. Our focus is on the EDCA model, which is an enhanced version of the DCF model. Similarly to the DCF, the EDCA mechanism is based on the carrier sense multiple access with collision-avoidance (CSMA/CA) protocol and provides four Access Category (AC) queues with different priorities for QoS requirements.

In [19] part, which is related to present an average bandwidth measurement approach to decide the bandwidth limitation of a congested connection [12]-[14], and a cross-layer architecture [1]-[7], through which we can use an adaptive dynamic queue algorithm to adjust the cross interference among layers such as application layer and lower layers when transmitting video data over IEEE 802.11e EDCA wireless networks. However, the detailed descriptions of the proposed cross-layer architecture based on the hierarchical protocol model and features the modification of data access parameters across several layers are mentioned to improve the performance of QoS in this paper. In addition, this paper gives more detailed derivations for the predictions of dynamic bandwidth and queue adaptation, we can effectively improve the quality of multimedia video transmitted on wireless networks without the re-design of the existing protocols. Meanwhile, we also have a simulation experiment on the NS2 [11][19] to verify the video streaming transmission efficiency of cross-layer wireless network with both average bandwidth measurement and adaptive dynamic queue algorithms.

The rest of this paper is organized as follows. In section II, transmission issues on wireless networks are briefly introduced. In section III, the algorithm of average bandwidth measurement is proposed. In section IV, the cross-layer technology to dynamically configure parameters is described. In section V, some simulation results are explained and compared with existing approaches. Finally, section VI concludes this paper and the future works.

II. TRANSMISSION ISSUES ON WIRELESS NETWORKS

Transmission protocol over WLAN environment is easily interfered or affected by micro wave or obstacles, and so are the user habits. The following issues relating to wireless transmission are derived naturally: (1) Bit Error Rate: The bit error rate of the wireless channel is higher than that of wired channel, and sensible to the changing environment. (2) Network bandwidth: The bandwidth of wireless environment is not as high as that of wired network. The effective bandwidth is far less than 54 Mbps, which is the IEEE 802.11g drafted bandwidth. (3) Round Trip Time: The wireless transmission takes more delay time, since its lower bandwidth affects the total transmission rate. (4) User mobility: The wireless feature provides a more flexible connection experience to user, but the frequent hand-over among base stations easily causes unexpected connection drops. (5) Transmission power constraint: In the wireless environment, there is a limitation to the transmission capability of a mobile station: the further the transmission distance, the more power it costs. The excess of power consumption might easily get packets lost. So if a host could not judge correctly the level of congestion on network, it might underestimate the bandwidth and, in turn, cause lower application performance, decreasing transmission rate, and longer delay time.

Although the IEEE 802.11e MAC (EDCA) standard has supported the QoS feature, the management of system resource is not effective enough for multimedia network application. This is because the wireless channel has the characteristics of time-variant, limited bandwidth, and multi-path fading in an open area. The data volume of multimedia application is usually far more than that of normal application. When characteristics of transmitted multimedia data changes, it should be reflected to the underlying layers, but the lack of the cross-layer architecture makes the communication between application layer and MAC layer impossible. The communication protocols designed for wired network is not good enough to optimize the multimedia transmission performance, since they are layered model and only allow the communication between adjacent protocol layers. The wireless medium, however, is time-variant, and the latest status of wireless channel must be reflected to every protocol level, which can thereafter have the corresponding modification to improve the communication performance. Therefore, the idea of coordination among all the protocol layers is proposed and called the cross-layer concept [4][5], different from the traditional layered idea.

To get more understanding about the design concept of cross-layer architecture and make the comparison among different solutions, five approaches are suggested in the base of cross layer architecture over wireless network [4] as follows.

- Top-down approach: The protocol at the topmost level receives the optimized parameters and strategy from the lower levels. The cross layer concept must apply on the current system, that is, while the MAC layer gets the optimal configuration from PHY layer, the APP layer gets MAC's parameters and configuration as well.
- Bottom-up approach: The protocol at the lower level tries to prevent the upper levels from the damage and the interference caused by the changing bandwidth.
- Application-centric approach: Based on the requirements of APP, either top-down or bottom-up approach can help retrieve the optimized parameter values from the lower levels.
- MAC-centric approach: By forwarding information and requirements through MAC layer, APP layer can decide the parameters of packet flow in response to different bandwidths.
- Integrated approach: In this approach, the cross layer technique is thoroughly supported by all the OSI layers.

Nevertheless, the best solution for cross-layer architecture still depends on the application requirements. By the close communication on this architecture, all layers in the protocol can reflect the changing environment on time and achieve the goal of efficient system resource management.

The improvement of the current network processing capability will not be enough to avoid the network congestion. The complexity of wireless network and the required correctness of bandwidth measurement when congestion happens on video transmission make the design of the algorithm of bandwidth measurement even harder. So far, none of acceptable solution to congestion issue on wireless network is presented. This paper is to propose a cross-layer architecture for wireless transmission, and, with the techniques of both average bandwidth measurement and adaptive dynamic queue algorithm, to raise the transmission quality of video data over wireless network according to the data characteristics. The experiment shows that the proposed architecture can enhance the aimed quality as expected.

III. ALGORITHM OF AVERAGE BANDWIDTH MEASUREMENT

Transmission over wireless network has the properties of the signal strength, signal interference, and receiver's sensibility, etc. Sometimes the bandwidth utilization is too high and sometimes it is too low. Further, it results into congestion or packet lost. Most congestion control mechanism can seldom adjust the network bandwidth for transmission. Therefore, if the correct average bandwidth measures are available, the network congestion could be improved and the quality of data delivery over wireless network could be further enhanced.

Regarding a transmission path, congestion happens on the point of bottleneck [12]-[14], where, as Fig. 1 shows, is the place causing longer delay time and packet lost. The proposed algorithm with average bandwidth measures can dynamically configure the bandwidth, in order to achieve the goals of: (1) measuring the correct average bandwidth values for the basis of adjusting; (2) lowering the probability of congestion; (3) reaching the high bandwidth utilization.

In [13], a TCP protocol which can measure the end-to-end bottleneck bandwidth of a path without the network support. According to TCP protocol, TCP transmitter sends out packets to the destination end in back-to-back manner. Such back-to-back packets and their acknowledged (ACK) packets constitute packet pairs for bandwidth measurement. Therefore, when an ACK is received by the transmitter, it conveys the information that a specific packet of the TCP flow was successfully delivered to the destination end.

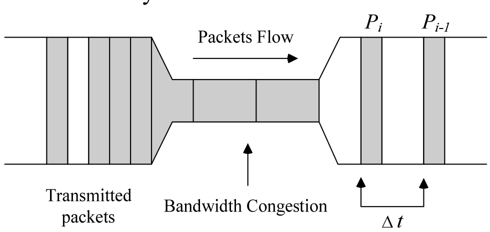

Fig.1. Network bottleneck.

As shown in Fig. 1, in the destination end, the time interval of adjacent packets results from the phenomenon of network bottleneck. In this figure, when the packets of size P_i and $P_{i\text{-}1}$ (bytes) reach the destination end, and the inter-arrival time, $\Delta t = t_i - t_{i-1}$ (ms), where t_i and $t_{i\text{-}1}$ are timing (ms) when the destination end gets the i^{th} and $(i\text{-}1)^{th}$ packets, respectively. Assume $t_i^{ack} = t_i$ and $t_{i-1}^{ack} = t_{i-1}$, where t_i^{ack} and t_{i-1}^{ack} are timing (ms) when the transmitter receives the returning ACKs of the i^{th} and $(i\text{-}1)^{th}$ packets, respectively. Then, the network bandwidth of bottleneck link can be calculated by

$$B_i = P_i / \Delta t = P_i / (t_i^{ack} - t_{i-1}^{ack}) \ . \tag{1}$$

By taking the average bandwidth measured in previous packet's arrival as a basis, it is possible to conduct the bandwidth between the transmitter and destination end. The packet on network could be lost or delayed, so the next question is how to get the average measured bandwidth. In Fig. 1, we define the weighted average of the bandwidth measured within at least 6 packets' arrival times ($\overline{B}_{n-1}$) and the current measured bandwidth (B_n) as the base of the configuration of video transmission rate, as follows.

$$B_n = P_n / (t_n^{ack} - t_{n-1}^{ack}) , \tag{2}$$

and

$$\overline{B}_{n-1} = \sum_{i=n-5}^{n-1} B_i / 5 , \tag{3}$$

where

n : number of reference packets and $n \geq 6$,

P_n : length of the n^{th} packet (bytes),

t_n^{ack}, t_{n-1}^{ack} : timing when the transmitter gets the ACKs of the n^{th} and $(n\text{-}1)^{th}$ packets (ms), respectively.

We also let S_B denote the standard deviation of bandwidth measurement defined as

$$S_B = \sqrt{\sum_{i=n-5}^{n-1} (B_i - \overline{B}_{n-1})^2 / 5} \ . \tag{4}$$

In the normal distribution, we can see that the probability content within 1, 2, and 3 standard deviations of the mean (average) are 0.68, 0.95, and 0.99, respectively. Therefore, the relation between the mean and the standard deviation can be used to adjust and improve the dynamic bandwidth measurement. In this paper, we use the philosophy of that the probability within 2 standard deviations (S_B) of the mean ($\overline{B}_{n-1}$) is around 0.95, that is, if the current measured bandwidth (B_n) is less than 2 standard deviations (S_B) of the mean ($\overline{B}_{n-1}$), then it is a point of network congestion, we need to adjust the available bandwidth for the proposed algorithm. The algorithm of proposed dynamic bandwidth measurement is presented in pseudo-code in Algorithm 1. In other words, we use the dynamic bandwidth measurement algorithm to achieve a balance between user's QoS satisfaction and bandwidth utilization.

Algorithm 1: Dynamic Bandwidth Measurement Algorithm

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1) Initialize: $i = 0$, $n = 6$.

2) Wait for next arrival ACK of packet *i*.

3) If $i < n$, then $i \leftarrow i + 1$ and transmit next packet.

4) If $i \geq n$, then $i \leftarrow i + 1$ and $n \leftarrow i$:

(i) For given *n* packets, apply (2), (3), and (4) to find the current bandwidth measurement B_n, the prevenient average bandwidth measurement $\overline{B}_{n-1}$, and the standard deviation of the bandwidth measurement S_B, respectively.

(ii) If $B_n < \overline{B}_{n-1} - 2 \times S_B$, then $i \leftarrow i-1$, wait a small time (usually a few microseconds, ex. 500ms) and transmit next packet; otherwise, transmit next packet.

5) Repeat steps 2-4 for next arrival ACKs of packets.

IV. CROSS-LAYER TECHNOLOGY TO DYNAMICALLY CONFIGURE PARAMETERS

Some methods have been suggested for efficient video transmission on wireless multimedia network, which is prone to error. These include formation of application packets, optimal scheduling of loss rate, united resource channel coding, error feedback and error hiding mechanism [4]. However, the contributions of these methods are only aiming at improving throughput or decreasing the power consumption, instead of the further characteristics of multimedia content or network flow. Regarding to the design of the lower level in the cross layer architecture, the characteristic and requirement of multimedia are getting more and more attention. The solution to optimize the video transmission by cross layer architecture has been proposed as the IEEE 802.11e MAC. Even this new MAC (EDCA) specification provides the feature of QoS, it does not guarantee the QoS on multimedia network application yet, and the resource management does not work very efficiently. This is because the time-variance property of wireless channel, the lack of sensibility to video characteristic on application layer and the lack of cooperation between application layer and lower layers.

This paper is based on the cross-layer optimization architecture [3]-[7] and enhances it even further. The algorithm and protocol in each layer usually have their own goal, so it is better to independently optimize each layer. In addition, different media streams are often treated as different kinds of information, e.g. that physical layer only recognizes the symbolic representation and depends on channel attributes, while application layer recognizes media content and characteristic. The cross-layer architecture focuses on the optimization among application (APP), network and transport (NAT), media access control (MAC), and physical (PHY) layers to take into account the parameters in each layer.

Following [3], let S_{APP}, S_{NAT}, S_{MAC} and S_{PHY} denote the set of parameters available at the APP, NAT, MAC, and PHY layers, respectively. The joint cross-layer strategy S is defined as

$$S = \{S_{APP}, S_{NAT}, S_{MAC}, S_{PHY}\}. \quad (5)$$

Then, the proposed cross-layer architecture in this paper is to find the optimal parameter set

$$S^{opt}(x) = \arg\max_{x \in S} \overline{PSNR}\,(S(x)) \quad (6)$$

such that

$$rate(S(x)) \le R_{bit} \text{ and } delay(S(x)) \le D_{\max}, \quad (7)$$

where the video quality $\overline{PSNR}$ is an average $PSNR$ (peak signal-to-noise ratio) and the value R_{bit} is the available transmission bit rate while the bound D_{max} (maximum accepted delay). Between video quality requirement and delay limitation, the dynamic bandwidth and queue adaptation are commonly considered. The average measured bandwidth is computed by Algorithm 1 mentioned in the previous section and the dynamic queue adaptation is proposed by Algorithm 2 described in this section.

In addition, consider a video sequence with M (m= 1, ⋯, M) frames (images), each of dimension $D_x \times D_y$ (x= 1, ⋯, D_x ; y= 1, ⋯, D_y) pixels, we let $F(x,y)$ and $R(x,y)$ denote the original and reconstructed video frames, respectively, then the MSE (mean square error) and $PSNR$ (dB) for the luminance (gray-level, or Y component) video frame m are defined as

$$MSE_m = \frac{1}{D_x \cdot D_y} \sum_{x=1}^{D_x} \sum_{y=1}^{D_y} [F(x,y) - R(x,y)]^2, \quad (8)$$

$$PSNR_m = 10 \cdot \log_{10} \frac{255^2}{MSE_m}, \quad (9)$$

and the average $PSNR$ quality ($\overline{PSNR}$)for a video sequence is defined as

$$\overline{PSNR} = (\sum_{m=1}^{M} PSNR_m) / M. \quad (10)$$

We use the average $PSNR$ quality of a reconstructed video frame $R(x,y)$ with respect to the uncompressed video frame $F(x,y)$ as the objective measure of received video frame. In addition, the larger the difference between $R(x,y)$ and $F(x,y)$, or equivalently, the lower the quality of $R(x,y)$, the lower the $PSNR$ value [15].

In IEEE 802.11e MAC, the EDCA uses four different Access Category queues (AC_i, for i = 0, 1, 2, 3): AC_3 for voice, AC_2 for video, AC_1 for best effort and AC_0 for background. AC_3 is assigned as the highest priority while AC_0 is the lowest. Since 802.11e EDCA maps the video data in only AC_2 neglecting the video data priority and AC congestion it may loss important video data that degrades video quality [16]. Therefore, we proposes an adaptive dynamic queue approach as Algorithm 2 to adjust the video data transmission between APP and MAC layers via NAT layer. In other words, the APP layer sends the video data to the NAT layer. Next the NAT layer passes this video data to the MAC layer and based on the instantaneous rate of increase at each ACs, i.e., the MAC layer performs the dynamic queue adaptation for appropriate ACs. The instantaneous queue length increase rate of AC_i is calculated by

$$Q_{i,r} = Q_{i,j} - Q_{i,j-1}, \quad (11)$$

where $Q_{i,j}$ and $Q_{i,j-1}$ are the instantaneous queue length of AC_i at time j and j-1, respectively. This $Q_{i,r}$ is used to predict the instantaneous queue length increase rate of AC_i during the next time increase rate according to

$$Q_{i,a} = \left\lceil (1-\mu)\cdot Q_{i,r} + \mu \cdot Q_{i,a} \right\rceil, \qquad (12)$$

where $\lceil x \rceil$ denotes the least integer greater than or equal to x, μ is the weight given to the past increase rate over the most recent increase rate and $0 \le \mu \le 1$. To simplify the computation of predicted increase rate $Q_{i,a}$, we let $\mu = 1/2$ for all AC queues. According to the adaptive dynamic queue algorithm mentioned in Algorithm 2, the choice of AC_i for the video data is based on AC_i 's priority and AC_i 's increase rate. Firstly, the video data is mapped to the highest-priority AC_3 when $|Q_{3,r}| > |Q_{3,a}|$, i.e., the AC_3 queue length is enough for sending the video data. Otherwise, the video data will be mapped to the other lower-priority AC_2, AC_1, or AC_0 depending on which one is available in sequence.

Algorithm 2: Adaptive Dynamic Queue Algorithm

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1) Initialize: $i \leftarrow 3$, $\mu \leftarrow 1/2$, $Q_{i,a} \leftarrow 0$.
2) Get the video data.
3) For given AC_i , apply (11) and (12) to find the instantaneous queue length increase rate $Q_{i,r}$ and the predicted increase rate $Q_{i,a}$, respectively.
4) If $|Q_{i,r}| > |Q_{i,a}|$ then $AC_i \leftarrow$ video data
   else $i \leftarrow i-1$.
5) If $i \ge 0$, then goto Step 3 else drop the video data.

Furthermore, the procedure in each layer of the proposed cross-layer architecture is shown in Fig. 2. While transmitting video data, APP layer could slice the original video frame with the existing video coding methods like MPEG-4 [8], H.264/AVC [9], or other scalable video coding (SVC) schemes [10], etc. The NAT layer could find the dynamic bandwidth using Algorithm 1 and establish the most stable connection of reasonable bandwidth to play the video streaming. In this way, it avoids dropping massive packets in congested condition and keeping the queue over-loaded (lowering its utilization) all the time, which are possible when adopting drop-tail scheme. The proposed MAC layer uses the adaptive dynamic queue approach for four AC queues. So when the collision happens, 802.11e EDCA can get the right to access channel very fast. When the present available bandwidth is calculated with the method of average bandwidth measurement and taken as the base to modify the transmission rate of video data, PHY layer establishes a connection between device and medium, cuts the packet from network layer into fix-sized frames, and sends them out. The procedure to transmit the multimedia video data is defined as Algorithm 3.

Algorithm 3: Cross-Layer Architecture Algorithm

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1) Select a standard or scalable video coding method, the APP layer could compress, packetize, and slice the original video frame with the existing video coding methods like MPEG-4 [8], H.264/AVC [9], or other scalable video coding (SVC) schemes [10], etc.
2) The NAT layer could find the dynamic bandwidth using Algorithm 1 and establish the most stable connection of reasonable bandwidth to play the video streaming.
3) The MAC layer provides four AC queues and uses Algorithm 2 to improve the congestion condition caused by collision and fasten the contention for the right of channel access.
4) By receiving the feedbacks of the optimal parameters $S(x)$ in (5) from each layer, it is available to get the flow scheduling based on each layer's transmission requirement, bandwidth measurement, and the dispatch of queues. It turns out that the connection can be established according to the optimal transmission environment $S^{opt}(x)$ in (6) and (7).

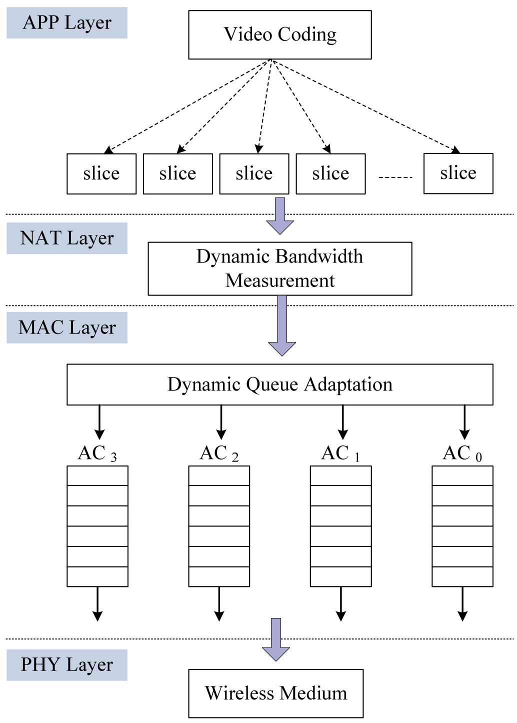

Fig.2. Procedures in each layer of the proposed cross-layer architecture.

As mentioned before, the non-cooperated OSI architecture on current wireless network environment has performance limitation and low resource utilization on the

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transmission of multimedia content. Therefore, to enhance the efficiency on wireless multimedia application, we present a change on the communication mode of current OSI architecture, and a new concept of information sharing and resource arrangement. The proposed architecture is inspired by the latest economical concept called "competition and cooperation" [4] which means that the best parameters required by each layer are beneficial in competitive environments. In wireless multimedia application, the basic competitive cooperation changes the current passive mode of fixed transmit procedure to the active mode which can adapt them. So that, the transmission strategy can match the available wireless network environment and the resource of bandwidth, share the resource and information to support the wireless transmission. The cross-layer of competitive cooperation is combined with the method of dynamic bandwidth measurement in order to allow the information sharing among the protocols in OSI layers. The idea of competitive cooperation is the fairness, which makes the chosen wireless transmission strategy more efficient and, in turn, improves the performance of multimedia application.

V. SIMULATION RESULTS

The cross-layer architecture for wireless transmission proposed in this paper is to make the adjusting of transmit rate in the transmitter end as fast as possible, and to decrease the congestion effect during transmission, which improves the overall utilization of the network resources. We select the common multimedia video coding algorithm (MPEG-4) [8] as the basis of comparison with experiment results. The observed criterions include:

- Bandwidth (BW) : By the change of bandwidth, it is clear to understand how congestion affects the transmission.
- Average throughput : It is to evaluate the overall performance of transmission.
- Average delay : Calculate the delay time of packets and check the effects to the video quality in receiver end.
- PSNR : Check PSNR to get the video quality after transmission.

The simulated network environment based on the NS2 (Network Simulator version 2) [11] is composed of a wireless transmitter and three wireless receivers. In the experiment, we establish three TCP flows and observe the transmission processes by each case of the simulated video sequences (Coastguard, News, Mother-daughter). In this paper, the cross-layer architecture for wireless transmission proposed is to make the adjusting of transmit rate in the transmitter end as fast as possible, and to decrease the congestion effect during transmission, which improves the overall utilization of the network resources. Because of the smaller video file used in this experiment, the simulated bandwidth is set to the one-tenth of the real bandwidth for simulation of the network congestion. For example, to distinguish the different effects in different levels of network congestion, we set the TCP congestion bandwidth to 0.5 Mbps and 1 Mbps, which allow us to understand the performance of MPEG-4 video delivery on the proposed cross-layer architecture in wireless environment.

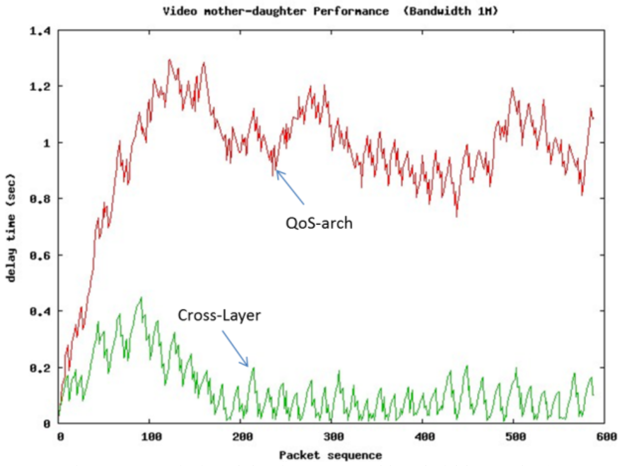

Fig.3. Transmission delay in congested bandwidth (1 Mbps).

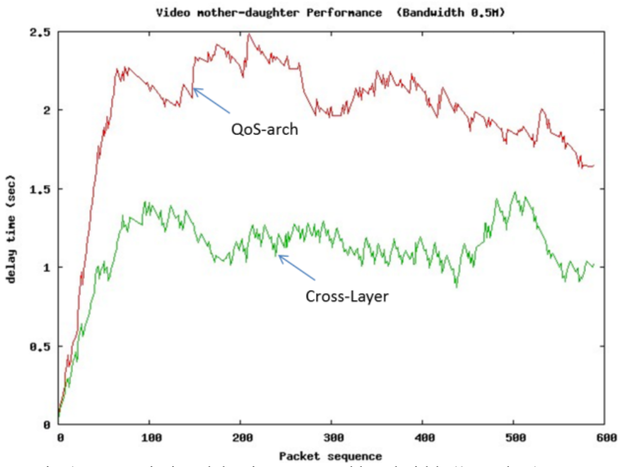

Fig.4. Transmission delay in congested bandwidth (0.5 Mbps).

Our experiments compose of three simulated video transmissions to evaluate the performance of network. In this paper, we select the common standard video coding algorithm (MPEG-4) [8] as the basis of comparison with experiment results. The observed criterions include bandwidth (BW), average throughput, average delay, and PSNR. Firstly, the source video is compressed and encoded into different MPEG-4 video streaming files, and the channel status is calculated with the method of average bandwidth measurement and taken as the basis of video transmission rate in order to establish the most stable connection of reasonable bandwidth for streaming playback. Here we set the TCP congestion bandwidth to 0.5 Mbps and 1 Mbps. Then we transmit through the cross-layer architecture with adaptive dynamic queue approach and take the records of transmit time, receive time, and packet ID, etc. In addition, we also generate the transmit log file, receiving log file and video record file for the performance evaluation later. By the analysis of those files, the compressed bit stream can be re-generated. After further decoding with the MPEG-4 decoder, the video format (CIF) the same as original video stream in transmit end can be recovered, and can be used to calculate the PSNR value for the comparison of the video quality. Besides, the receiving log file can be used to estimate some other statistics like delay time, throughput, PSNR, etc.

For the Mother-daughter sequence transmission delay analysis, we can see what happened in the proposed cross-layer architecture in normal congestion (1 Mbps), shown in Fig. 3. When congestion happened, we can see the delay time soaring and maintaining stable after a period of oscillation. The transmission on conventional QoS-arch (802.11e EDCA) wireless network makes the delay time oscillating for a longer period and weakens the transmission quality due to its unstable speed modification. Next, when the bandwidth gets lower, the transmission delay gets higher. Therefore, we will take a look on the changing in severer congestion condition (0.5 Mbps). We can see that, from Fig. 4, because of more serious congestion condition, the transmission delay of the cross-layer architecture climbed fast and immediately entered a stable status. The transmission delay of conventional QoS-arch (802.11e EDCA) will ebb and flow severely for a longer period, in the same circumstance.

For the Mother-daughter sequence transmission video quality analysis, from Fig. 5 and Fig. 6, when the congestion condition is worse, the performance of video quality with normal transmission scheme is not as good as the one with proposed architecture, because the bandwidth for transmission becomes too small and the packet easily get lost with soaring delay time. Traditionally, a conservative but easy-to-implement congestion avoidance mechanism is usually adopted on wireless environment and works perfectly fine when the bandwidth is kept stable. The proposed cross-layer architecture, however, can even dynamically adapt the congestion bandwidth, and this aggressiveness can make it have much better performance than conventional QoS-arch (802.11e EDCA) transmission scheme in a burst of congestion.

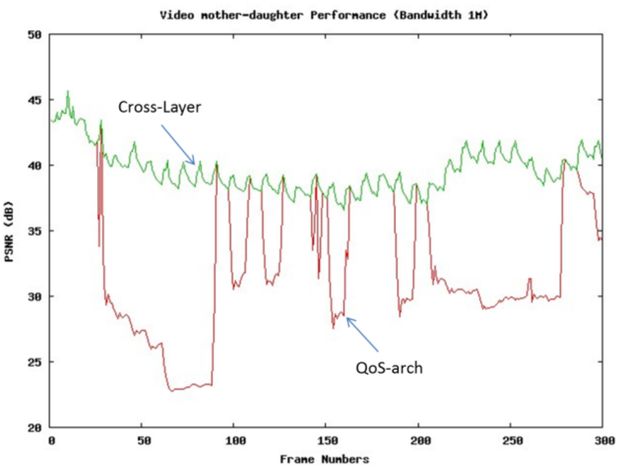

Fig.5. Transmitted video quality in congested bandwidth (1 Mbps).

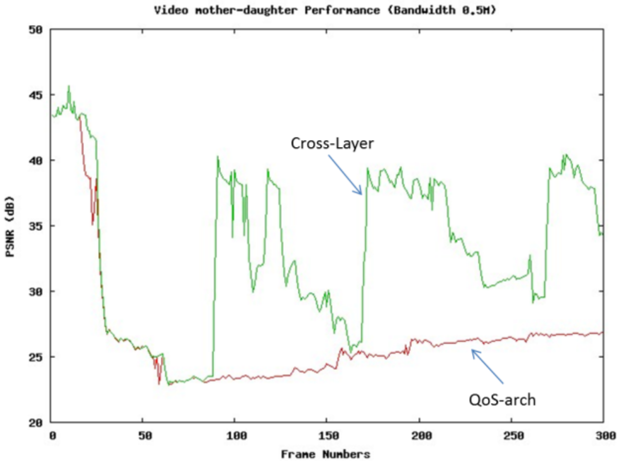

Fig.6. Transmitted video quality in congested bandwidth (0.5 Mbps).

In addition, Fig. 7 shows the 85th and 200th reconstructed frames of the News sequence in severer congestion condition (0.5 Mbps), respectively, it is easy to found that the reconstructed frames of conventional QoS-arch (802.11e EDCA) method appear many serious blocking artifacts and the proposed cross-layer method obtains a better visual quality of the reconstructed frames.

(a) The 85th reconstructed frame by conventional QoS-arch method

(b) The 85th reconstructed frame by proposed cross-layer method

(c) The 200th reconstructed frame by conventional QoS-arch method

(d) The 200th reconstructed frame by proposed cross-layer method

Fig.7. Transmitted video quality of Mother-daughter sequence in congested bandwidth (0.5 Mbps).

For the performance analysis on video transmissions, the throughput and average transmission delay time are shown in this section. Relatively, the average throughput is kept in a stable high record, and the average delay time is less than that of conventional QoS-arch (802.11e EDCA) scheme. When the delay time is large, the video quality has to be increased by more aggressive architecture. Furthermore, we study the throughput and average delay time within the three video transmissions: Costguard, News and Mother-daughter

[19]. Table I shows the average and the longest delay times when the congestion bandwidths are 1 Mbps and 0.5 Mbps, respectively. The transmission delay is improved at least 50% in the proposed cross-layer architecture. From the aspect of transmission throughput as shown in Table II, the proposed cross-layer architecture combined with the method of average bandwidth measurement has a much better performance. Table III shows the advantage of less frequency of video lag when the transmission delay is getting higher. This concept can be constructed on the 802.11e EDCA wireless standard, and the method of average bandwidth measurement does not cost any network resources. Also, the proposed cross-layer architecture can provide a more acceptable service quality by dynamically adjusting transmission rate according to congestion status and avoiding the loss of packets.

TABLE I: TRANSMISSION DELAY TIME (SEC)

Video Name	Congested Bandwidth: 1Mbps		Congested Bandwidth: 0.5Mbps	
	Conventional	Proposed	Conventional	Proposed
Coastguard	Avg:0.668 Max:1.360	Avg:0.220 Max:0.792	Avg:1.963 Max:2.673	Avg:0.942 Max:1.763
News	Avg:0.949 Max:1.318	Avg:0.201 Max:0.759	Avg:1.764 Max:2.561	Avg:1.076 Max:1.714
Mother-daughter	Avg:0.937 Max:1.297	Avg:0.120 Max:0.449	Avg:1.866 Max:2.487	Avg:1.088 Max:1.480

TABLE II: TRANSMISSION THROUGHPUT (KBPS)

Video Name	Congested Bandwidth: 1Mbps		Congested Bandwidth: 0.5Mbps	
	Conventional	Proposed	Conventional	Proposed
Coastguard	Avg:264.5 Max:325.6	Avg:294.2 Max:538.0	Avg:147.2 Max:179.7	Avg:237.6 Max:312.5
News	Avg:284.9 Max:317.7	Avg:365.3 Max:516.6	Avg:155.5 Max:179.4	Avg:254.1 Max:300.8
Mother-daughter	Avg:274.4 Max:320.4	Avg:351.0 Max:486.7	Avg:146.9 Max:174.0	Avg:246.8 Max:271.1

TABLE III: TRANSMITTED VIDEO QUALITY, PSNR (DB)

Video Name	Congested Bandwidth: 1Mbps		Congested Bandwidth: 0.5Mbps	
	Conventional	Proposed	Conventional	Proposed
Coastguard	25.6	26.5	17.9	25.5
News	30.0	34.6	22.3	27.6
Mother-daughter	33.1	39.6	26.6	33.0

VI. CONCLUSION

The IEEE 802.11e MAC standard can be used to support the QoS for wireless networks, but the bandwidth resource management is not efficient enough for the wireless multimedia network application. In this paper, based on the 802.11e EDCA wireless network standard, we construct a new architecture to provide the best video transmission quality (QoS) and to handle the issue of network congestion which affects severely the stability of wireless transmission. We allow the control messages shared among APP layer, NAT layer and MAC layer. In the NAT layer, the method of average bandwidth measurement is used to establish a stable channel with reasonable bandwidth and good enough for video streaming. Also we include the adaptive dynamic queue approach which uses four AC queues in the 802.11e MAC layer. By the simulation results, the transmission delay is improved at least 50% in the proposed cross-layer architecture, and even when the contention situation is getting more and more serious, the transmission throughput can still have a higher increase. The simulation experiment proves that the proposed method can yield a better performance for transmission delay time, image quality and packets throughput over wireless multimedia networks.

The simulation experiment proves that, the proposed architecture can be constructed on the 802.11e EDCA wireless network standard, the methods of both bandwidth measurement and queue adaptation do not consume any network resource, and the cross-layer architecture can provide a more acceptable service quality by dynamically adapting the transmission rate according to the congestion level and avoiding the loss of packets.

In the past, the related researches only focused on the methods of congested bandwidth measurement and ignored the effects of multimedia video transmission. Therefore, for the future work, we will be based on this paper and develop the transportation scheme suitable for different wireless network standard like WiMAX, Bluetooth, etc. By the network condition observed by receiver, transmitter can dynamically decide the best QoS scheme for transmission, or make another choice of path when the congestion happens. The aggressive mechanism to reflect the latest network status on the path or transmission rate can greatly enhance the total performance and the video quality in the receiving end.

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