Provision of QoS for Multimedia Services in IEEE 802.11 Wireless Network

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ABSTRACT

Implementation of the IEEE 802.11 technology for wireless local area networks (WLANs) has created a new opportunity to support real time and multimedia applications. The IEEE 802.11 defines a Medium Access Control (MAC), and Physical Layers (PHYs), which provides rates from 1 to 54 Mbit/s depending on specification. In IEEE 802.11 network, the PHY rate to be used for the transmission is completely determined by the source node and is closely connected with the wireless channel performance. The varying nature of WLAN channel over time can degrade the QoS level of multimedia service, which calls for elaboration of advanced QoS mechanism supporting multimedia services. The research works applied to video transmission over WLAN mostly concentrate on using a link adaptation techniques or optimisation of video stream by cross-layer signalling. It appears to be not sufficient as far as a provision of QoS for multimedia services is considered. In the paper, a new mechanism of QoS enhancement respecting also modifications in transport and network layers is discussed. A general description of this mechanism is presented with special emphasize on new approach to the multimedia rate control. The paper covers the verification of the proposal by simulation method as well as a discussion of some selected results of simulation.

1.0 INTRODUCTION

Wireless networks are nowadays one of the important elements of tactical communications. The IEEE 802.11 is one of the candidates for such communication. This standard refers to lower layers of OSI model: physical and data link layer. The available bandwidth and the nominal transfer rate defined by the standard are shown in Tab. 1.

Tab. 1 Bandwidth and transfer rate

Version	Bandwidth (GHz)	Nominal transfer rate (Mb/s)
802.11 (basic)	2,4	2
802.11b	2,4	11
802.11a	5	54
802.11g	2,4	54
802.11n	5	> 500

The wireless technology, in spite of its undoubted advantages, has also a number of shortcomings, which are mainly connected with the available bandwidth, transmission quality, interferences and user mobility. It should be notice that the basic specification of IEEE 802.11 standard does not define any QoS support mechanisms. This poses the fundamental problem in the case of providing multimedia services in military environment.

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Many projects are focused on development of effective QoS mechanism for providing multimedia services in mobile WLAN environment. In case of standard QoS mechanism used for video transmission, the coefficient of quantization or degree of compression is modified that consequently lead to the data rate adaptation [7, 8, 9]. As far as a provision of QoS for multimedia transmission is considered it seems to be not sufficient. Taking into consideration experience in providing the QoS in military narrowband networks ([1, 2, 3]) we propose use a cross-layer signalling to modify the multimedia stream across the upper layers of mobile WLAN protocol stack. The general concept of our proposal is discussed in section 2. The procedures for adaptation of video rate with using cross-layer signalling are discussed in section 3. Simulation method has been used to verify the proposed solution. The main objective of the simulation is validation of the proposed solution and assessment of its effectiveness for multimedia transmission in mobile wireless network. Section 4 describes the selected results of simulation experiments.

2.0 QOS ARCHITECTURE

The provision of QoS in mobile WLAN network is associated with dynamic reaction to the changes observed in the radio channel. In fact, the ISO/OSI protocol stack is not enough flexible, so dynamic modification of layers and protocols raise a new challenge in nowadays network design process. Applying cross-layer signalling techniques allows making better use of network resources by optimizing across boundaries of traditional network layers. The implementation of cross-layer signalling database (CLSD) as a separate part of protocol stack architecture appears very attractive one. It enables to supply the information from lower layers of ISO/OSI protocol stack to the application and TCP/UDP/IP layers. The establishment of cross-layer database that have an access to all of the protocol stack layers ensures possibility of exchanging information without destruction of protocol stack architecture [17]. The main effort is concentrated on design of interactions between layers and cross-layer signalling database.

A general concept of our proposal is shown in Fig. 1. The cross-layer signalling database collects information from physical and data link layer and transfer them into application and transport/network layers. Information on available throughput is used for multimedia rate adaptation at application layer. In contrast to the solutions given in [8, 9] we propose a new approach to the multimedia rate adaptation that is base on controlling the frame rate of video encoder and/or data buffering. The process of video frame rate adaptation is discussed in section 3 of this paper. Data buffering is realized if the available throughput is not enough for simultaneous transfer of video and data and only in case if the data priority is lower than priority of video.

The multimedia stream is composed of video and data. For data optimization the adaptation of congestion window is used at transport layer. The adaptation process includes control of congestion information. This information signalizes if the losses of the TCP packets are caused by network congestion or errors in physical layer (BER). If losses are caused by the BER than the congestion window is not modified what leading to the optimization of data transmission in transport layer. The special type of ICMP message is used at network layer for this purpose.

Available bandwidth and Frame Error Rate (FER) are used for packet size adaptation at network layer. It incorporates the headers compression, packet segmentation, packet streams multiplexing and queues handling. The proposed mechanisms have been used for managing a limited bandwidth link within the IPv6 military narrowband network. The detailed description of these mechanisms can be found in [1, 2]. We propose to use them in WLAN network between the wireless source and the wireless router (Access Point – AP). Additionally the routing information can be used as input for mechanism that selects the optimal route for multimedia transmission. An example of such mechanism is presented in [20].

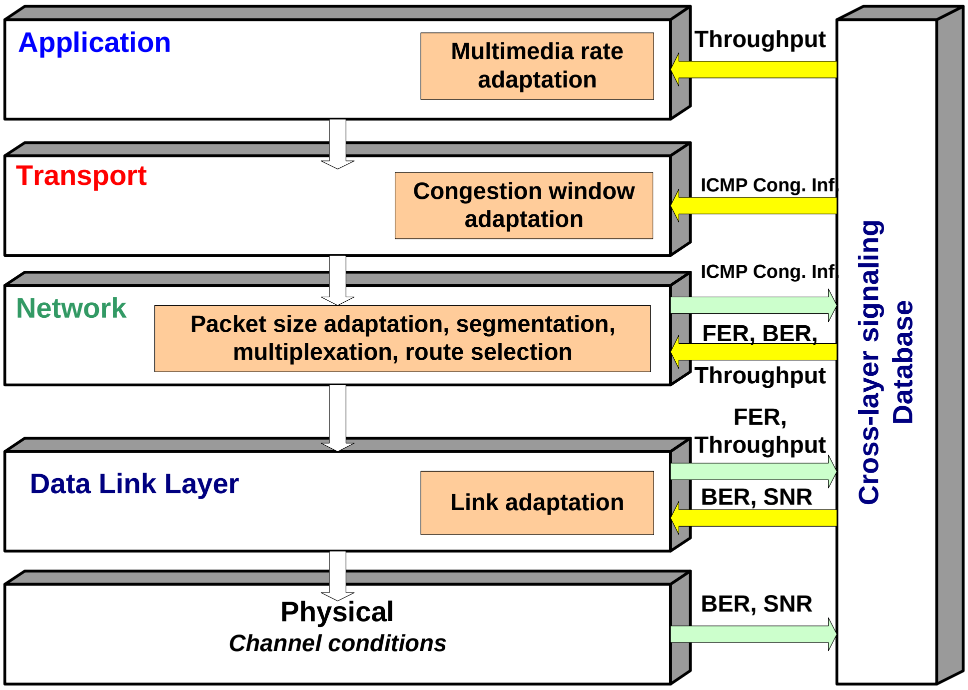

Fig. 1 General concept of QoS provisioning with cross-layer signalling

The link adaptation techniques are used at Data Link Layer for handling the changes in a radio channel. We propose to use the SNR (Signal-to-Noise) information that through the CLSD is available to the link adaptation module defined at Data Link Layer. The SNR is accessible via received signal strength measurement that is performed at the physical layer. The link adaptation module selects the modulation type satisfactory for actual value of SNR. For IEEE 802.11a implementation seven types of modulation are offered (Tab. 2).

The next section describes the proposed video frame rate adaptation mechanism that increases the efficiency of the radio channel resource utilisation.

3.0 THE CROSS-LAYER OPTIMIZATION

The following section presents our approach to implementation of cross-layer signalling for adaptation of the video frame rate to the available bandwidth of the radio channel. Digitally coded video signal symbolize the variable rate bit stream. This is because not all the video frames have the same entropy [18, 19]. Such a stream is characterized by a set of parameters that represents QoS requirements for multimedia service. The most important are the end-to-end delay, delay variation and error rate. The optimisation process of multimedia stream should guarantee adaptation of video flow to the WLAN channel condition keeping the QoS parameters in the acceptable range.

A general process of video stream optimisation is shown on Fig. 2. The stream of video bits is inserted into rate adaptation buffer with rate $R'(t)$, and processed with rate $\mu(t)$, so the output rate $R(t)$ is adequate to the available throughput.

The adaptation buffer performs regulation of bit stream when the service rate is lower than rate of video stream. Size of the buffer is evaluated on the base of expected delay. The feedback between buffer and the video codec protects the buffer against its overflow.

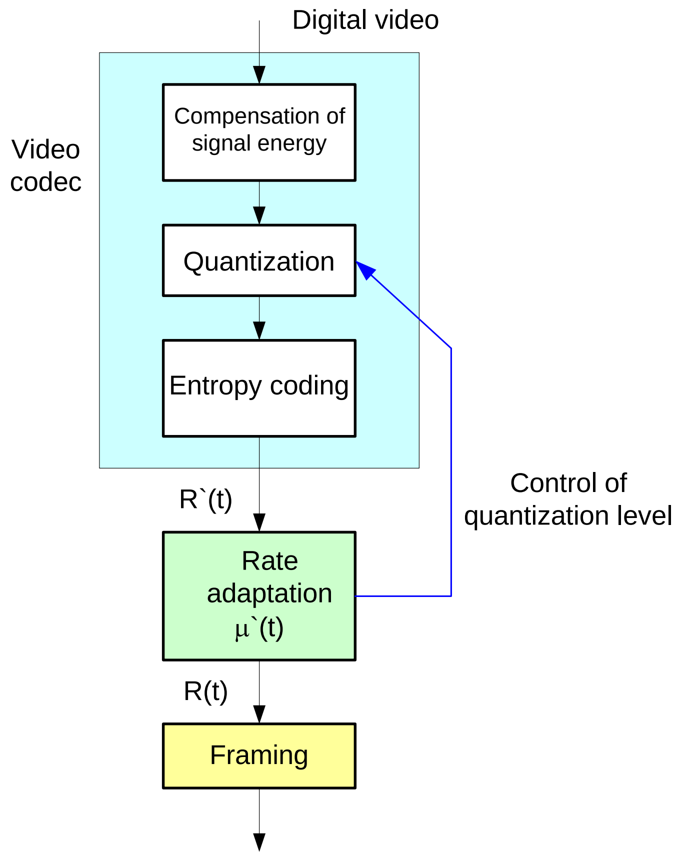

Fig. 2 General process of video rate adaptation

However, the preservation of subjective quality of received multimedia stream allows the regulation of video stream rate. When we assume the determined number of quantization levels and certain level of minimum quality we can achieve limitation of the delay and the complexity of coded video signal. Therefore, the goal of rate optimization is the limitation of changeability of $R(t)$ and at the same time minimization the influence on perceived quality [10, 11].

The simplest example of video stream adaptation is rejection of the video frames. Such an example can be found in [13]. Practical examples of video stream bit adaptation with control of quantization level in wireless environment can be found in [8, 12]. In our proposal we are going to control the video frame generation time and optimise it to the throughput available at the input to the WLAN channel.

The main requirement for the video codec is adaptation of the video stream rate to the WLAN channel throughput. This can be realized in the system that dynamically modifies the video frame rate on the basis of available throughput. Such a system is presented in Fig. 3.

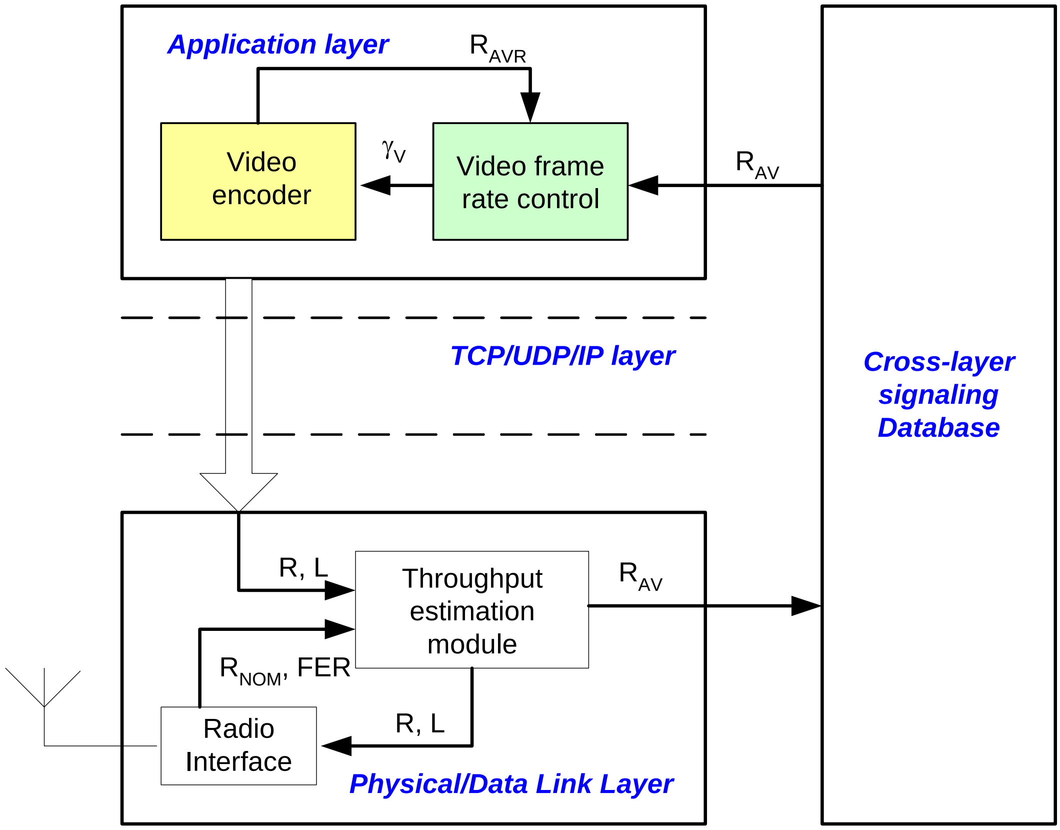

Fig. 3 Video rate adaptation

The available throughput is evaluated in the throughput estimation module (***TEM***) that is located in the Physical/Data Link Layer. Two types of information are used for estimation. The first one is obtained from Radio Interface module. This module represents an IEEE 802.11a radio interface and works in one of the seven modes of operation. The selected mode is characterized by type of modulation scheme and nominal transmission rate R_{NOM}. The relation ship between type of modulation scheme and R_{NOM} is given in Tab. 2.

Tab. 2 Mode dependent parameters (IEEE 802.11a)

Type of modulation	Coding rate	Nominal bit rate (Mb/s)
BPSK	½	6
BPSK	¾	9
QPSK	½	12
QPSK	¾	18
16 QAM	½	24
16 QAM	¾	36
64 QAM	¾	54

The Radio Interface module delivers also information about FER. This parameter is calculated based on ACK frames. The 802.11 specifications ensure that all successfully received data frames are explicitly acknowledged by sending an ACK frame. By counting the number of received ACK frames and the number of transmitted data frames one can compute the FER according to the following equation:

$$FER = \frac{Number\_of\_ACK\_frames}{Number\_of\_transmitted\_frames} \quad (1)$$

The observation time should be rather short because of changeability of propagation in radio channel. The FER and R_{NOM} are passed to the TEM where the nominal throughput is calculated as follows [14]:

$$T_{NOM} = R_{NOM}(1 - FER) \quad (2)$$

When there is no other transmission expect video than T_{NOM} is equal to T_{AV} (Available Throughput). Otherwise, T_{NOM} is evaluated with respect to the throughput of the other transmitted data streams. The necessary information is obtained from TCP/UDP/IP layer based on data stream transfer rate R and length of the packet L. Therefore the throughput of data stream can be calculated from the formula given in [8]:

$$T = \frac{8RL}{8L + bR + c}, \quad (3)$$

where:

T – throughput in Mb/s,

L – length of a packet in bytes,

R – data rate in Mb/s,

b, *c* – coefficients that depends on the 802.11 specification. For 802.11a, they assume values 161,5 and 156 respectively.

Thus, the available throughput is given as:

$$T_{AV} = T_{NOM} - T \quad (4)$$

For such evaluated value of T_{AV} and using appropriate transformation of equation (3) we can calculate the value of available rate R_{AV} that through the Cross-layer signaling Database is passed to Application Layer.

The video encoder should adapt the video rate to the value of R_{AV}. Unfortunately not all video codec's are able to operate in such a way. Therefore, for simulation experiment, we propose to use a H.263+ like video encoder [15]. A very similar characteristic has also a H.264 family of codec's [16].

The rate adaptation is realized in Video frame rate control module (VFRCM) of Application Layer every time when the R_{AV} is lower then the actual video frame rate R_{AVR}. The R_{AVR} is feedback from Video encoder module. If this condition is meet then the coefficient of rate adaptation γ_V is calculated as a ratio of R_{AV} and R_{AVR} rates. The γ_V is used for video frame rate calculation according to the following formula:

$$R_{AVR} = \gamma_V \cdot R_{AVR} \quad (5)$$

In this way the video frame rate is modified when it is required. It should be noted that video frame rate could be changed with reference to the new frame only. Therefore the time of new frame generation is evaluated in the VFRCM and after that time the new value of γ_V is calculated.

The presented solution has been evaluated by computer simulation. The results of the simulation are presented in the next section.

4.0 SIMULATION EXPERIMENT

The video rate adaptation solution has been implemented in simulation environment using the OPNETv.11.5 simulation package [21]. A basic network configuration that has been assumed for performing simulation experiments is shown in Fig. 4. It consists of two end user terminals connected together by the AP router using WLAN 802.11a and the Ethernet 100 Mb/s links.

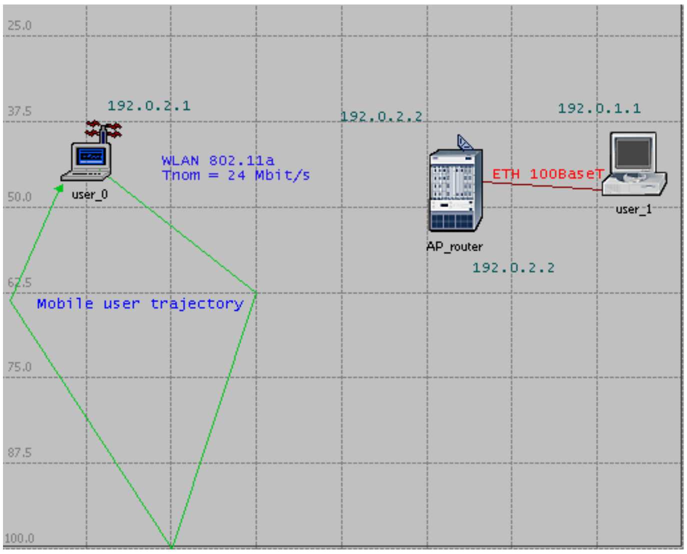

Fig. 4 Simulation scenario

The terminal *user_0* is a WLAN mobile terminal and it initiates video call and FTP transfer. The video call starts at 100 second of simulation and it last for 300 seconds, while FTP files are downloaded periodically 10 seconds after beginning of the video call. Fig. 5 shows a traffic sent by video and FTP application during the simulation time.

We can see that video data rate is about 4,147 Mb/s while rate of FTP is equal to 800 kb/s. If such traffic appeared directly in 802.11a link, especially with low data rates, it would take up nearly whole bandwidth. It will lead to the worsening of QoS parameters for video transmission. The video rate adaptation solution should protect the QoS pareameters against the degradation.

The end-to-end measures used for video quality evaluation are as follow:

- Video packets end-to-end delay
- Video packets jitter

We also analysed the WLAN media access delay to investigate how the modification of video rate influences the behaviour of MAC layer.

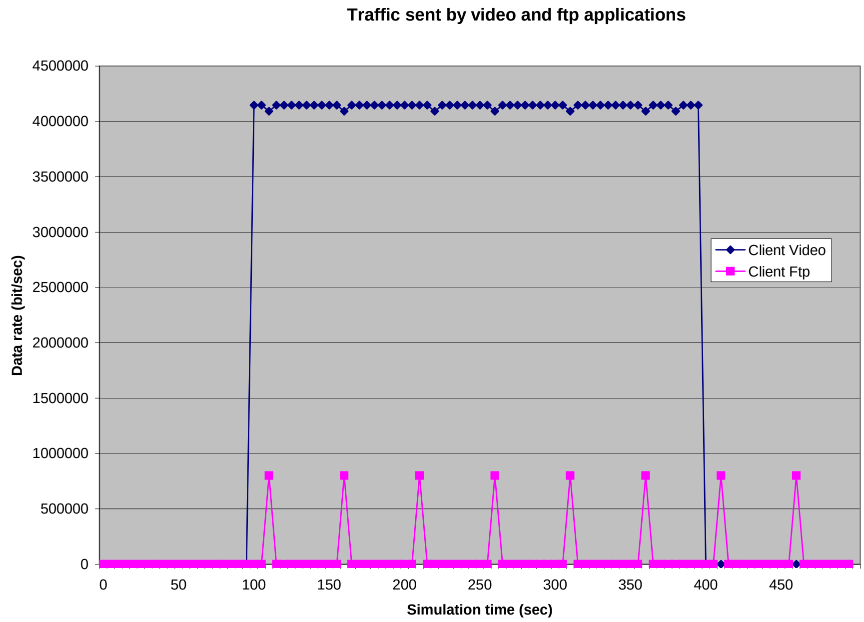

Fig. 5 Traffic sent by video and FTP applications (*user_0*) during the simulation time

Three types of experiments have been performed: video and data transmission with standard video coding mechanism, video and data transmission with video rate adaptation algorithm implemented and only video with video rate adaptation algorithm implemented. All of the experiments have been performed for different data rates in MAC layer.

The comparison of results obtained for standard video coding mechanism and new video rate adaptation solution with presence of FTP transfer is presented in the next figures. The video packet end-to-end delay registered in the configuration with MAC data rate equal to 9 Mb/s is shown at Fig. 6. We can observe that the delay for new video rate adaptation mechanism does not exceed the 70 ms while for the standard coding mechanism is about 90 ms. The typical acceptable value of video end-to-end delay is about 150 – 300 ms. The characteristic of delay is more linear in the case of modified video application and its sudden increase occurs in the moment of FTP transfer only.

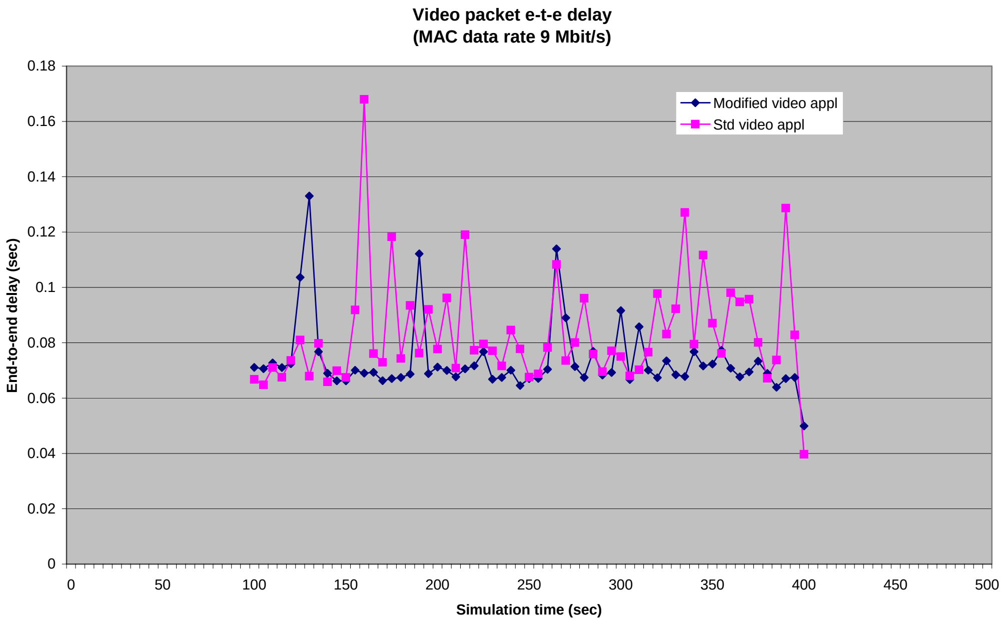

Fig. 6 Comparison of video delay for standard and modified video application

In the case of standard video application such peaks of delay occure much more frequently, thus the modification of video rate leads to the delay decreasing. The comparison of jitter that is shown at Fig. 7 confirms that implemented video rate adaptation mechanism enables improvement of qaulity of video transfer.

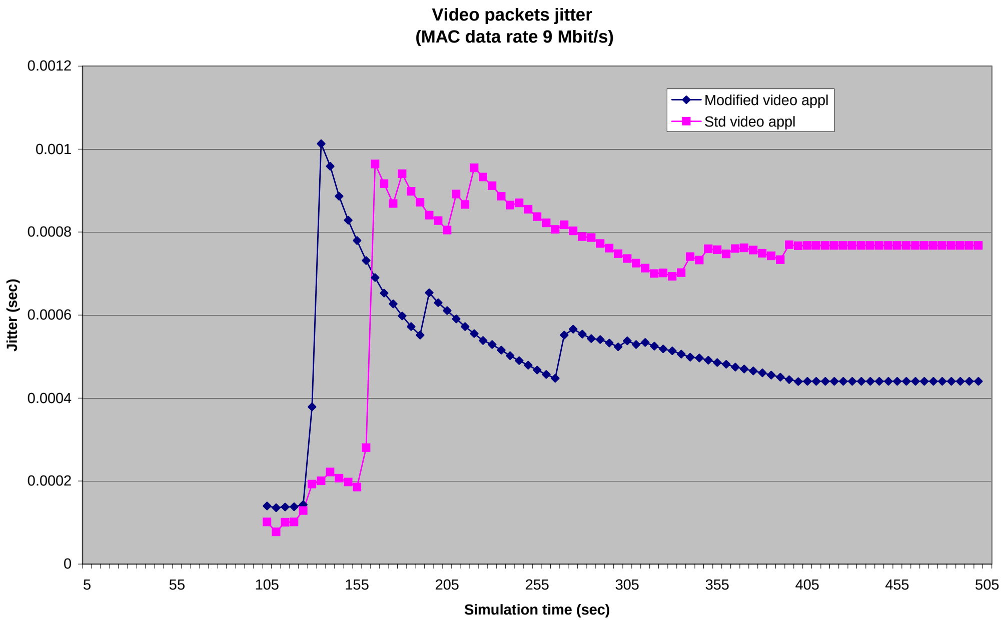

Fig. 7 Comparison of video jitters for standard and modified video application

Analysis of the results presented in Fig. 8 shows that modification of video rate at the application layer influences the method of video packet handling in MAC layer. We can see that end-to-end delay is changing in the range of 10 ms and this change occurs in the time of FTP traffic generation. It leads to the conclusion that the video packets have to wait more or less for accessing the radio channel resources (Fig. 10). It is caused by the regular broadcasting of the messages by the Access Point (AP).

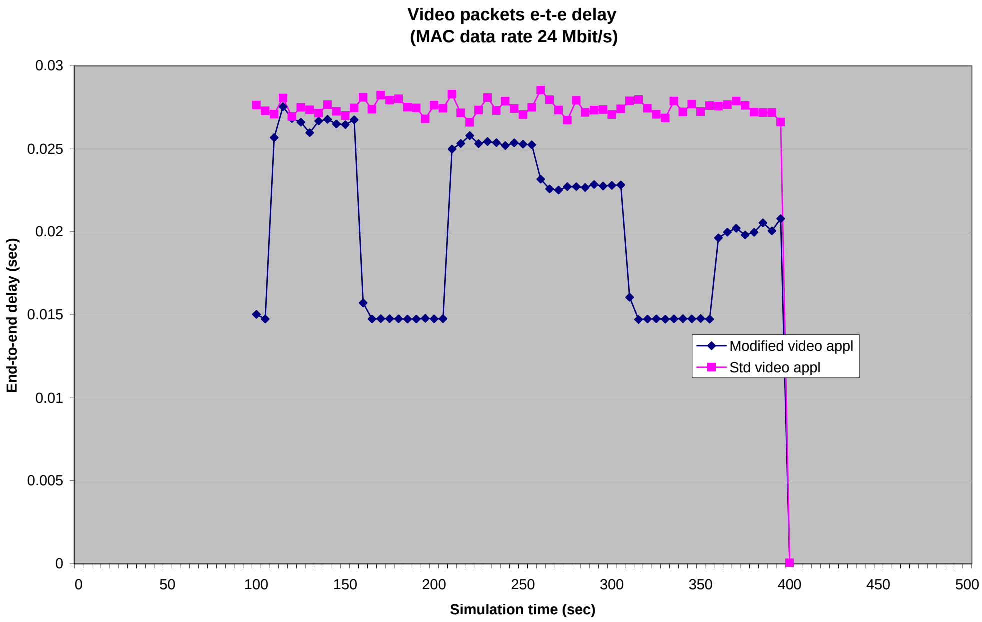

Fig. 8 Comparison of video delay for standard and modified video application

These massages are broadcasted periodically and during this action the AP can not receive the data. If the workstation has to send the packets (MAC frames) exactly in this time it than has to wait for AP accessibility. By changing the video data rate we cause that sometime packets have not to wait.

The measured values of video jitter for standard and modified video application are shown in Fig. 9. It can be observed that even though the modified video application jitter has higher value than jitter of standard application it is still in aceptable range of 80 ms.

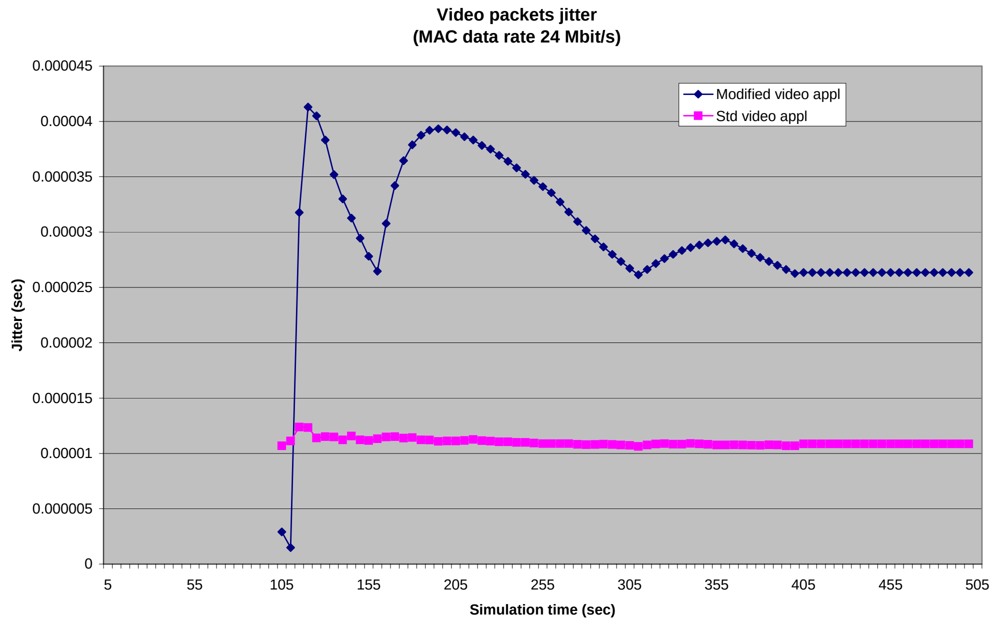

Fig. 9 Comparison of video jitters for standard and modified video application

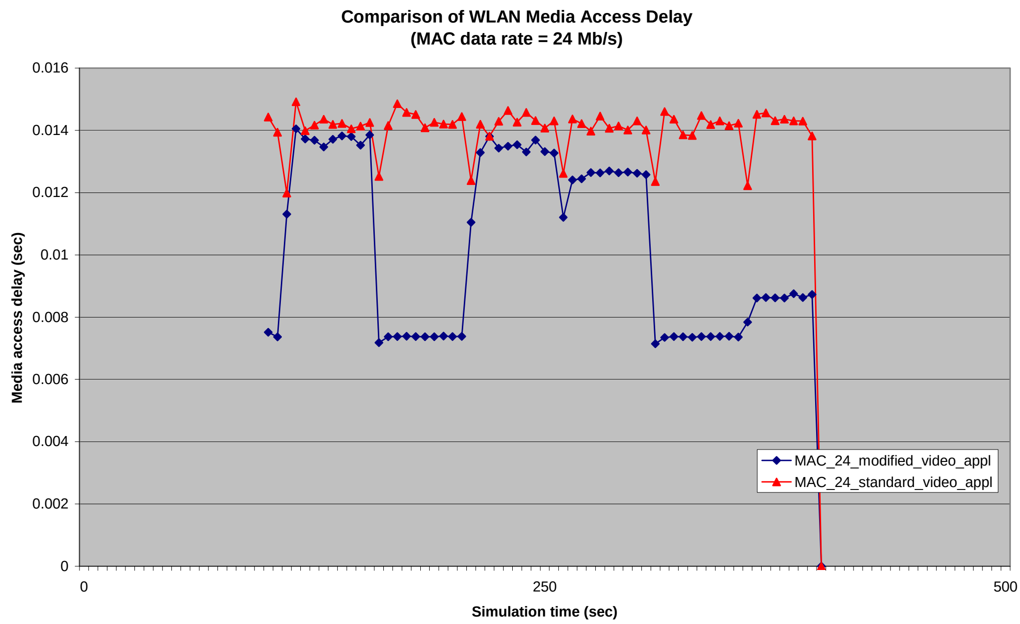

Fig. 10 Comparison of WLAN media access delay for standard and modified video application

Fig. 11 shows that video packets end-to-end delay in the presence of the FTP transfer depends on the time of FTP servicing and on the available data rate of MAC layer.

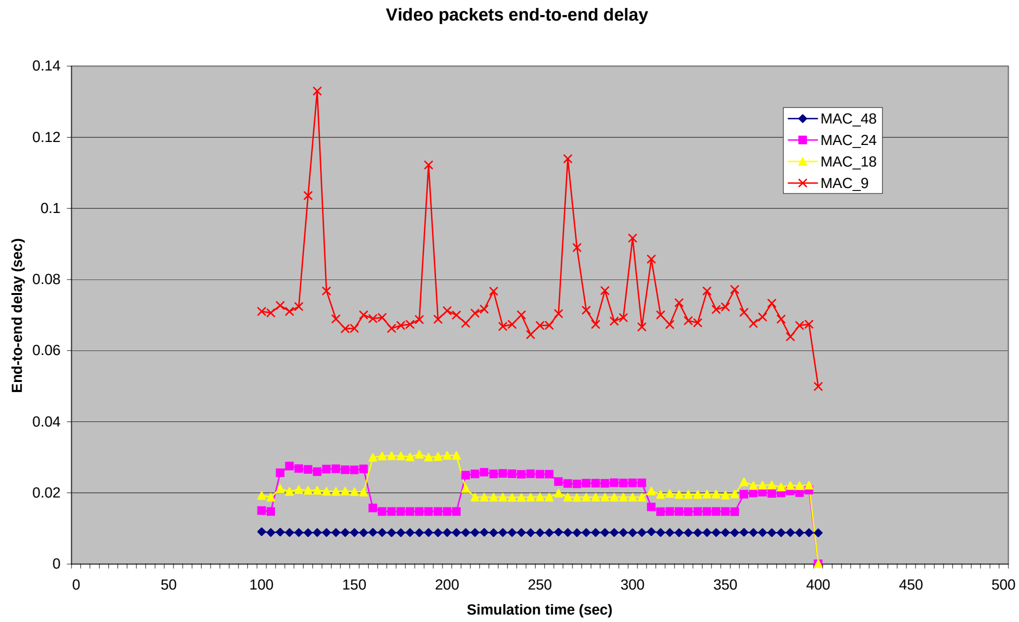

Fig. 11 Video packets end-to-end delay for different MAC's data rates

Such behaviour is caused by the nature of the WLAN that uses all of the available bandwidth for handling the data flows that come from the upper layer. This is why the video delay changes during the FTP transfer. It should be noted that the results are in the acceptable range for video transmission with QoS guaranties. In the case of 9 Mb/s data rate of MAC interface the video frames end-to-end delay does not exceed 130 ms.

A comparison of video packets end-to-end delay with and without FTP traffic is shown at Fig. 12. We can see that proposed video rate adaptation mechanism reacts on the changes in available bandwidth (FTP application starts), and than the delay increases (Video&data (MAC_24)).

The video rate is adopted also when FTP application do not sent the data (MAC data rate equal to 9 Mb/s). This is because the data rate of video application is equal to the half of WLAN channel bandwidth.

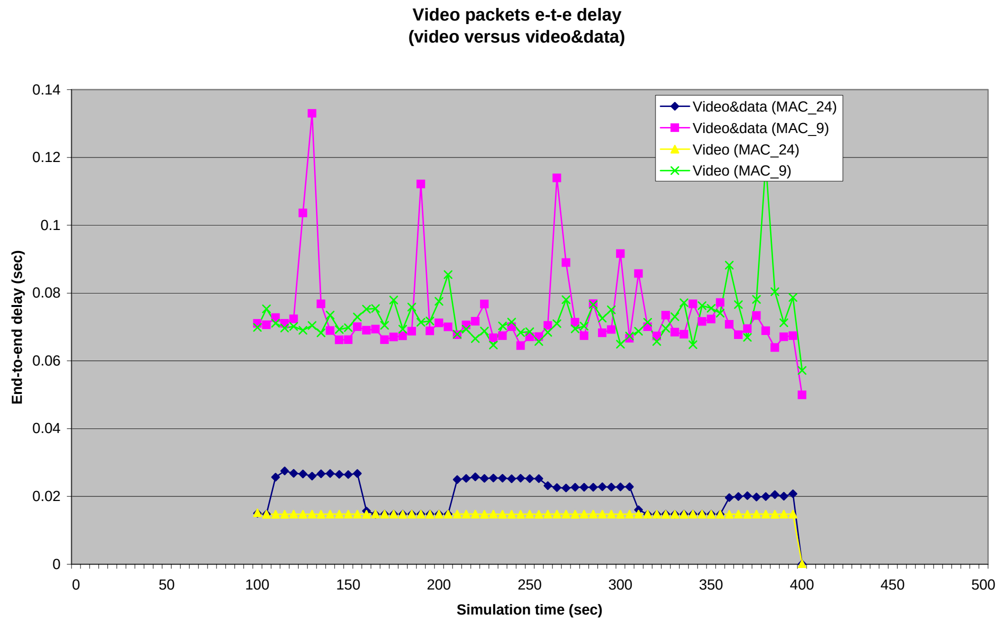

Fig. 12 Comparison of video packets end-to-end delay for video-data and video only transfer

5.0 CONCLUSIONS

In the paper, the basic architecture of QoS for multimedia services in IEEE 802.11 wireless network have been discussed. We have presented our approach to QoS support for video transfer based on the video rate adaptation with cross-layer signalling used for passing the information about available bandwidth to the video application. This mechanism has been implemented in the simulation environment. The simulation experiment shows that the new video rate adaptation mechanism has to be used in order to ensure the adequate level of video transmission quality. From the simulation we can also learn that video rate adaptation mechanism should react not only to the changes of the available bandwidth but also to the used media access algorithm.

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