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PROPOSED MAC PROTOCOL VERSUS IEEE 802.15.3A FOR MULTIMEDIA TRANSMISSION OVER UWB NETWORKS

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Abstract—In this paper, a Medium Access Control (MAC) protocol is proposed to investigate Quality of Service (QoS) for multimedia traffic transmitted over Ultra Wide-Band (UWB) networks and increase the system capacity. This enhancement comes from using Wise Algorithm for Link Admission Control (WALAC) which has three suggested versions. The QoS of multimedia transmission is determined in terms of average delay, admission ratio, loss probability, utilization, and the network capacity. In addition, a new parameter is aroused for the network performance. Comparisons between the IEEE 802.15.3a protocol and the proposed one are done. The proposed protocol shows better results in both sparse and dense networks for real time traffic transmission.

1. INTRODUCTION

With the rapid evolution of wireless technologies, UWB technology is expected to play an essential role in emerging technology for future wireless communications. In addition to UWB communication applications, UWB devices can be used for imaging, measurement, and vehicular radar [1].

According to the Federal Communications Commission (FCC), the fractional bandwidth or the transmission bandwidth of UWB signal should be greater than 0.2 or 500 MHz. Additionally, the approved unlicensed spectrum which is 3.1–10.6 GHz band with power spectrum density limited to $-41.3\,\mathrm{dBm/MHz}$ [2, 3].

The standard released for UWB Wireless Personal Area Network (WPAN) is IEEE 802.15.3 [4, 5] updated by TG3a (the third task group) and unleashed IEEE 802.15.3a [6, 7]. They defined an alternative physical layer (PHY) which includes high bit rates over short ranges with low power consumption and high capacity. There are two ways to use the bandwidth available for UWB which are impulse radio approach and multiband approach [8, 9].

In this paper, a new protocol which can support QoS achievement for multimedia transmission over UWB channel in the piconet, is proposed. This proposed protocol is based on using a suggested algorithm for bandwidth allocation named WALAC3. This proposed protocol shows better performance when compared with IEEE 802.15.3a MAC protocol.

The paper is organized as follows; Section 2 gives an overview of IEEE 802.15.3 MAC protocol. Section 3 inquires into the UWB physical model and the resource allocation. Section 4 introduces the detail description of the proposed protocol. Simulation results and the comparison discussions between the IEEE 802.15.3a protocol and the proposed one are shown in Section 5. Finally, conclusions are made in Section 6.

2. THE IEEE 802.15.3 OVERVIEW

The IEEE 802.15.3 consists of several components, as shown in Fig. 1. The piconet coordinator (PNC) controls the ad hoc manner (where nodes may join or leave the network at any time) piconet. It provides the basic timing for the piconet with the beacon that marks the start of a superframe. Furthermore, it manages the QoS requirements and the access control to the piconet [4].

The superframe consists of three parts, namely the beacon, the Contention Access Period (CAP), and the Channel Time Allocation Period (CTAP) [10]. The CAP is assigned for both non real time traffic and requests to the PNC, while the CTAP is subdivided into slots for real time traffic transmission. Additionally, the CTAP may contain management CTA (MCTA) slots for real time commands or requests. After the real time node successes to gain CTA slot from the PNC (after contention for sending the request and the PNC permission if there are available slots), it will transmit in the next superframe whose beacon will contain the beginning and the duration of this slot.

Carrier Sense Multiple Access Collision Avoidance (CAMA/CA) is used for CAP transmission similar to IEEE 802.11 MAC protocol [11, 12]. The basic operation of CSMA/CA can be illustrated as follows; if the medium is free for a time period longer than Distributed coordination function Inter Frame Space (DIFS), the station can transmit immediately. Request to Send/Clear to Send

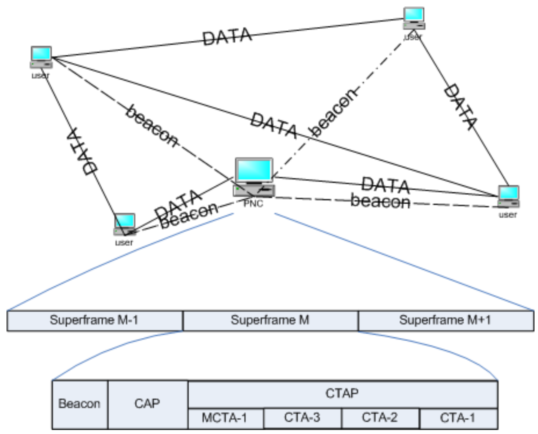

Figure 1. UWB network architecture.

(RTS/CTS) mechanism is used for hidden nodes problems. For long data periods, the data fragmentation is done using Short Inter Frame Space (SIFS) as time spacing between the fragmented data packets. If the medium is busy, the station defers its transmission and a random exponential backoff interval is then selected according to the contention window (CW) length. The backoff timer is decreased as long as the channel is sensed as idle for more than DIFS, and stopped when a transmission is detected on the channel.

Time Division Multiple Access (TDMA) is the mechanism used for real time transmission in CTA slots. While the transmission in MCTA slots (if found) uses Slotted Aloha technique.

3. UWB PHYSICAL MODEL

The physical layer of wireless networks specifies communication parameters such as bandwidth, modulation, and coding [13]. For UWB networks, to utilize the bandwidth and achieve desired QoS, an effective resource allocation scheme is needed to specify power level and transmission rate of each node to access the wireless medium.

In [14], the general approach used for resource allocation is based on a joint management of rates and powers of the nodes. Specifically, the channel capacity for UWB network is bounded by the Signal to Interference plus Noise Ratio (SINR) threshold which is given by:

$$SINR = \frac{P_i g_{ij}}{R_i \left(\eta_i + T_f \sigma^2 \sum_{k=1,k\neq i}^{N} P_k g_{kj}\right)} \geq \gamma_i \qquad (1)$$

where P_i is the average transmitted power for the link i, g_{ij} is the path gain from the transmitter i to the receiver j which can be calculated as $d_{ij}^{-\alpha}$ where α is the path gain constant usually between 2–4 and d_{ij} is the distance between the transmitter i and the receiver j , η_i is the background noise energy, T_f is the pulse repetition frequency, σ^2 is an operation parameter depending on the shape of the pulse, R_i is the rate of the link i, N is the number of active links in the network, and γ_i is the threshold value of the SINR [15]. Then powers and rates are chosen in order to match the the maximum allowed power ($0 \leq P_i \leq P_{\max}$) and the threshold value of SINR [3, 14, 16].

In [17], UWB system characteristics, compared with narrow-band wireless systems, were shown. There are no collisions in UWB Transmissions due to the use of Time Hopping Spread Spectrum (TH-SS) or Direct Sequence Spread Spectrum (DS-SS) as a multiple access techniques [8, 18] furthermore, hybrid techniques can be used [19] and hence multiple simultaneous transmissions can be occurred in the UWB network. However, the near-far problem due to the strong interference from the nearby interferer nodes still exists. This problem cannot be solved by the power control at the physical layer, but should be using a jointly radio resource allocation at the data link layer. The physical layer uses adaptive modulation [20] or adaptive coding [15] to adjust the rate to maintain SINR constant at the receiver. In [16, 17], Interference Margin (IM) approach has been assumed to avoid the frequent power reconfigure for each new admitted link. Each active link has an IM given by (2), which donated the additional interference by the new links.

$$IM_i = \frac{P_i g_{ij}}{R_i \gamma_i} - \eta_i - T_f \sigma^2 \sum_{k=1, k \neq i}^{N} P_k g_{kj} \tag{2}$$

One major challenge in UWB MAC design is the QoS provisioning with an efficient resource allocation scheme [17, 21, 22]. Although there have been large researches on real time traffic (voice and video) [22, 23], not too much work takes into account the unique characteristics of UWB.

4. THE PROPOSED PROTOCOL DESCRIPTION

The proposed protocol in [24] showed superior performance for UWB network in data traffic. This proposed protocol was modified in [25] to cope with multimedia challenges. Both protocols were based on two channels (data and control channels). The data channel was used for traffic transmission while the control channel was existed for requesting a link. The transmission is based on the superframe which consists of a number of slots in addition to the beacon in its header for both synchronization and broadcasting the piconet information. Furthermore, the control channel is divided into the same number of slots and each one is subdivided into uplink (for requesting) and downlink (for acknowledgment) subslots. The PNC applied Wise Algorithm for Link Admission Control (WALAC) for the link requesting in [24]. While in [25], there were two proposed algorithms. One is for non real time traffic (data) which is the same as used in [24], and it is named WALAC1. The other one is made for real time traffic (voice and video) and it is named WALAC2.

In WALAC1 as shown in Fig. 2, if a data request is valid, there are two cases. **The first case** is there are no available links in the system, and in this case the PNC calculates IM for all incoming requests using the maximum rate and power from (2). It checks the negative IM and applies the iteration procedure to the maximum negative IM link, if there are negative IM found. It updates IM for that link using the median then the minimum value of the rate. If it still negative, the PNC rejects that link and update the other IM and repeats this procedure till there are no negative IM links. All the residual positive IM links will be admitted. **The second case** is that there are available links in the system. In this case, the PNC calculates the allowed power for each request (P_0) from the minimal IM of active links from (3), where "0" referred to the new link. Then remove the links with zero power value and let $P_0 = P_{\max}$ (if $P_0 > P_{\max}$). Calculate the allowed rate in the system for each request from (4). If there are rates lower than the minimum allowed rate in the system ($R_{\min}$), reject the request with minimal allowed rate then repeat again till all allowed rates be greater than $R_{\min}$. Update all active links in the network. If any one be negative IM, remove the maximum interfering request from the minimal IM. Then update the IM again and repeat till no negative IM in the links. Calculate the IM for the residual requests with their calculated power and rate which will be considered as the maximum rate for that request and then apply the same procedure as if there are no links available in the network.

Figure 2 shows that there are no great differences between WALAC1 and WALAC2 except that in WALAC2, there are no iterations as in WALAC1. In addition, IM is calculated using requested rate not the maximum or allowed rates as in WALAC1.

$$P_0 = \min\left\{\frac{IM_i}{T_f \sigma^2 g_{0i}}\right\} \quad \text{where } 1 \leq i \leq N \tag{3}$$

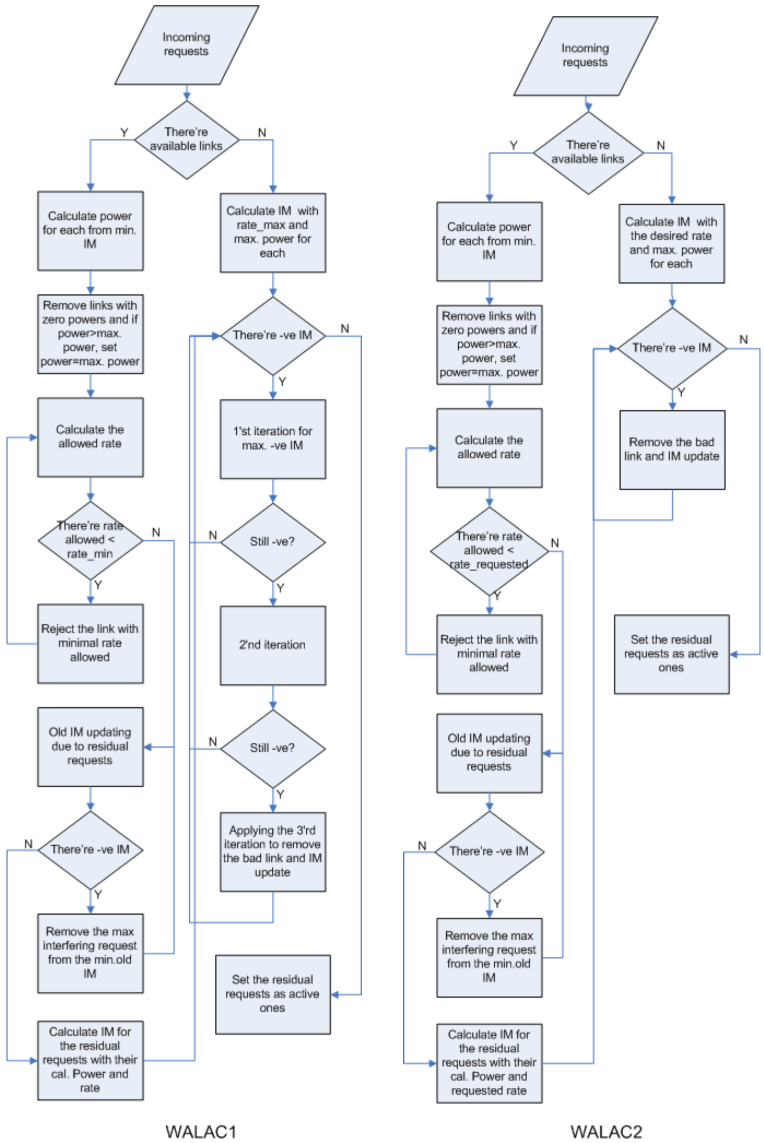

Figure 2. Flowchart of WALAC1 and WALAC2.

$$R_{allow} = \frac{P_0\, g_{i0j0}}{\gamma\left(\eta_i + T_f\sigma^2 \sum\limits_{k=1,k\neq i}^{N} P_k g_{kj0}\right)} \tag{4}$$

The proposed protocol in this paper depends on the same data and control channels. In addition, the data channel is divided into three parts; beacon, Non Real Time Period (NRTP), and Real Time Period (RTP) as shown in Fig. 3. NRTP is assigned for non real time (data) transmission. RTP is specialized to real time traffic (voice and video). RTP is divided into a number of Channel Time Allocation (CTA) slots which vary every superframe according to the real time traffic. Each CTA slot is assigned for some real time users according to IM limitation. Each real time user is permitted to transmit only in its assigned CTA slot.

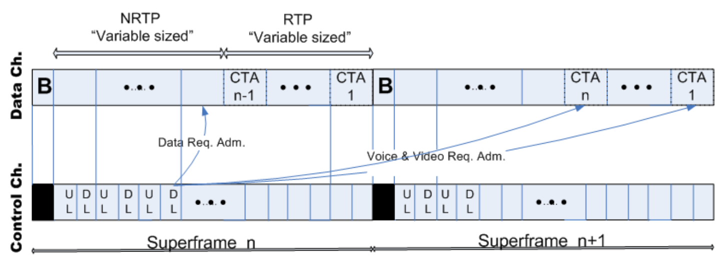

Figure 3. Piconet superframe structure.

The PNC has three queues for the incoming requests as [25]. The highest priority is placed for the voice queue, and then the video one followed by the data one will be served respectively. That is to achieve QoS requirements.

For non real requests, the PNC applies WALAC1 and the admitted links will transmit from the next slot of the data channel in the same superframe if valid (if the next slot is CTA one, then the admitted links will transmit in the beginning of the next superframe). While for real time requests, the PNC applies the proposed algorithm WALAC3 which considered as a bandwidth allocation procedure. The use of WALAC3 allows the PNC to find an appropriate CTA for the incoming real time requests without horrible influence on the existing links as well as without data channel usurpation. The data traffic are transmitted during the whole NRTP while the real time traffic are transmitted only in their assigned CTA slot.

The flowchart of WALAC3 is represented in Fig. 4. For the incoming real time requests, two cases are existed. Firstly, if there are no active real time links in the system and hence no available CTA. In this case, the PNC applies the first case of WALAC2 which results two matrices; one for rejected requests and the other for the admitted ones. If the admitted matrix is empty, the PNC sets all the requested links as rejected ones and breaks. That is because of the background

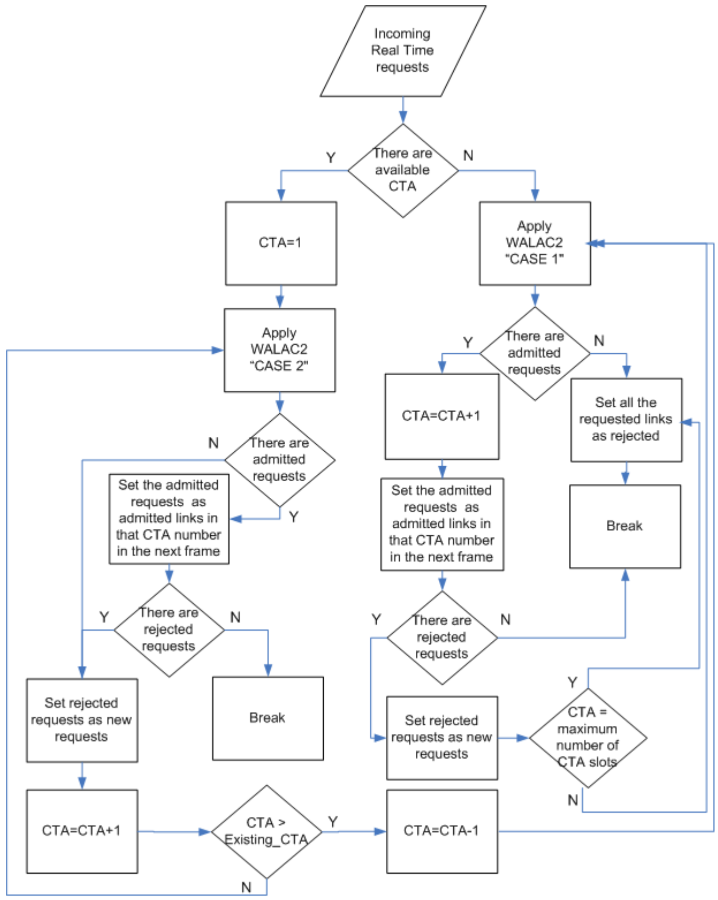

Figure 4. Flowchart of WALAC3 for bandwidth allocation.

interference is larger than the requested rate. While if the admitted matrix is not empty, the PNC increases CTA slots by one and set these links as admitted ones to transmit in the next superframe in this CTA slot. After that the PNC checks the rejected matrix. If it is empty, the PNC breaks from WALAC3 as all requests are admitted. If not, the PNC set them as new requests and repeat again the same procedure but before that it will ensure that the number of CTA slots does not reach the maximum designed number. If that happens, it will set these new requests as rejected ones and break.

In the opposite vein, if there are available real time links in the system and hence CTA slots are found. The PNC applies the second case of WALAC2 with the available links in the first CTA slot. It results the same two matrices as before. If the admitted matrix is empty, the PNC sets the rejected matrix as new requests and repeats but for the next CTA slot. If not, the PNC sets the admitted matrix as admitted links which will transmit in the next superframe in this CTA slot. Then it checks the rejected matrix. If it is empty, the PNC breaks from WALAC3, else if not, the PNC sets it as new requests and repeats the same procedure in the next valid CTA slot. If all existing CTA slots are finished and the rejected matrix is still not empty, the PNC tries to find a new CTA slots taking the same procedure as there are no CTA slots in the system.

From this discussion, WALAC3 tries first to admit new requests in already existing CTA slots then in new ones. That is because WALAC3 is considered as bandwidth allocation algorithm.

The proposed protocol can be summarized as follows:

1) Terminal with traffic desired to be sent, requests a link from PNC using the uplink subslot in the control channel. This request includes the transmitter and receiver identifications as well as the traffic type. Each terminal transmits with a certain code. Therefore there are no collisions.
2) The PNC collects all requests and places them in the appropriate queue. Afterwards, it applies WALAC3 for voice and video requests respectively followed by WALAC1 for non real time one. The PNC informs the requesting terminals about its state, i.e., admitted or rejected, through the downlink subslot in the control channel.
3) For data admitted requests, they will transmit from the next slot in the data channel if valid. While for the real time admitted requests, they will transmit in the assigned CTA in the next superframe of the data channel as shown in Fig. 3.
4) For link termination, the PNC is informed through the control channel. If all links in a certain CTA slot are terminated, the PNC remove that PNC slot and update the following ones.
5) The PNC informs all the active links about the new format of the superframe in the beacon.

5. SIMULATION RESULTS AND DISCUSSIONS

Comparisons between the IEEE 802.15.3a MAC protocol and the proposed one are done for multimedia traffic in this section through simulation programs. This comparisons are done in terms of QoS parameters such as the average delay and the loss probability. In addition, the system utilization (the ratio between the successfully transmitted bits averaged over the time) and the network capacity are considered. Furthermore, the admission ratio (the ratio between admitted requests and all incoming requests) as a good parameter for the network performance is perused. The simulation area is taken as $100\,\mathrm{m} \times 100\,\mathrm{m}$ with nodes randomly distributed.

Three types of traffic are considered. First of all, the constant bit rate source model (voice traffic) which has the highest priority according to its real time characteristics. It generates a signal of talkspurts separated by silentspurts with a rate of 32 Kb/s. A speech activity detector can be used to detect this pattern [26, 27].

The second priority traffic is the variable bit rate source model, i.e., video traffic. It generates stream traffic with a variable time rate. The source rates are generated based on truncated Gaussian distribution between 128–384 Kb/s with mean rate of 256 Kb/s. The slice time is 33 msec.

The last priority traffic is held for the data traffic which is generated based on Poisson process with λ arrival/sec per user. Furthermore, the buffering rate is 9600 b/s [6]. The rest of the default parameters used are shown in Table 1.

Figure 5 displays the average delay (the total delay per successfully transmitted packets) of the multimedia traffic for both protocols. The average delay is directly proportional with the number of users but it will be rapidly increased for the proposed protocol more than the IEEE 802.15.3a. Due to the low buffering rate for the data traffic, its transmission time is high (26.7 msec) compared with voice and video traffic (7.8 msec and 2 msec maximum respectively) and hence, its average delay is somewhat large compared with voice and video traffic for the proposed protocol while the opposite for the previous one. That is because in the proposed protocol the successfully admitted links have to wait to the next slot in the NRTP in order to have permission to send. For higher number of real time users, it will be nearly one slot for NRTP and hence the data users have to wait till the next superframe which is not found in CSMA/CA which is used by the previous protocol.

On the contrary, the average delay for real time users of the proposed protocol is lower than the previous one. That is because of the wasted contention time during CAP duration in order to send requests to the PNC. While the suggested control channel in the proposed protocol solves this problem. Additionally, the use of WALAC3 allows the best utilization of the CTA slots without affect the QoS and hence, more than one real time users can transmit in the same CTA slot which is not found in the previous protocol.

For the previous protocol, the delay is nearly the same for both

Table 1. Simulation parameters.

Parameter	Value
T_f	100 ns
σ^2	1.99×10^{-3}
η	2.568×10^{-21} W/Hz
$P_{\max}$	7 dBm
λ	30
α	4
γ	7 dB
superframe duration	30 msec
beacon	5 msec
slot duration	5 msec
minimum CAP or NRTP duration	one slot
RTS	20 bytes
CTS	14 bytes
ACK	14 bytes
SIFS	20 μsec
DIFS	60 μsec
CW length	7,15,31,63
contention slot duration	20 μsec
packet length	32 bytes
voice life time	50 msec
video life time	125 msec
data life time	15 sec
voice channel coding rate	6 Mb/s
video channel coding rate	6–18 Mb/s
minimum rate ($R_{\min}$)	2 Mb/s
maximum rate ($R_{\max}$)	18 Mb/s

voice and video traffic. While voice has a larger delay than video traffic in sparse network and the opposite in dense network for the proposed protocol. That is because the larger buffering delay for the voice users. Due to the highest priority of the voice traffic, it will be served first and hence, a larger delay can be noticed for the video traffic in dense network.

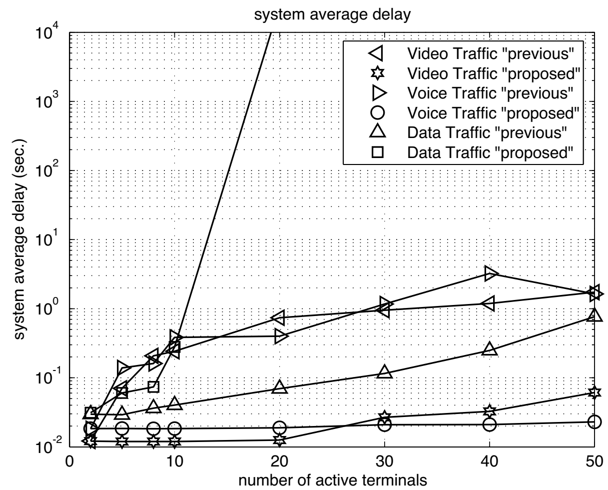

Figure 5. System average delay for multimedia traffic.

Figure 6 depicts the system admission ratio versus the number of active terminals. It is well shown that the admission ratio for real time traffic of the proposed protocol is greater than the previous one while the opposite is for non real time traffic. That is because of the lower size of NRTP and the non real time nodes have to wait to the next slot in NRTP to be admitted. According to the proposed protocol utilizes the CTA slots better than the previous one (due to the use of WALAC3) and hence, more real time users can be admitted.

For the previous protocol, the admission ratio for both voice and video is nearly the same. While the admission ratio for voice traffic is slightly greater than video traffic for the proposed protocol in dense network. That is because of the highest priority of the voice traffic. Furthermore, the admission ratio of data users is lower than real time users due to the QoS requirements.

The system loss probability (the ratio between the rejected transmitted packets and all transmitted packets) for multimedia traffic can be shown from Fig. 7. More than 50 data users can be supported by the previous protocol and the same for the proposed protocol but with less quality (with 5.5×10^{-2} dropping probability). That is because of the large threshold value of the maximum delay for data traffic, in addition to its non QoS nature. Therefore there are nearly no lost packets for the previous protocol but for the proposed one, there are small numbers of packets are lost due to its large delay as explained before.

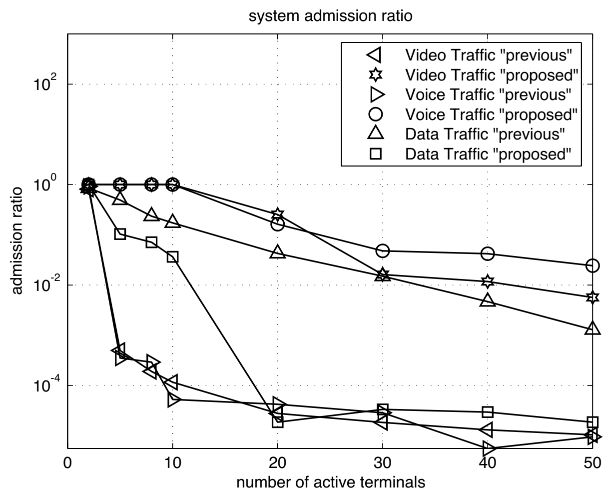

Figure 6. Admission ratio for multimedia traffic.

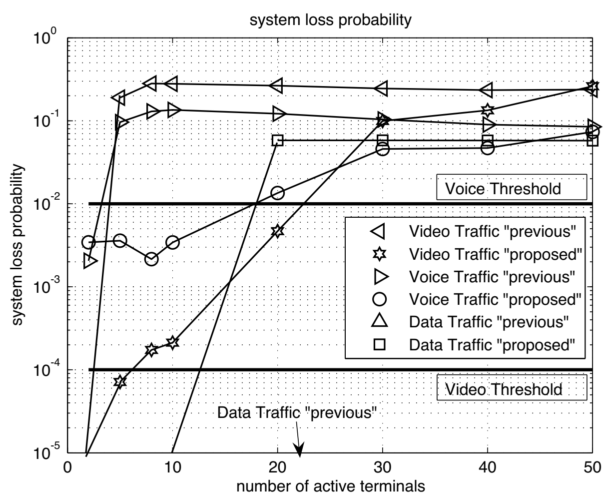

Figure 7. System loss probability for multimedia traffic.

While for voice traffic, the lowest threshold value of the maximum delay (to achieve QoS requirements) plays a great role in the probability of loss increase. The previous protocol can support up to nearly three voice users taking 10^{-2} as the threshold value of the loss probability while the proposed protocol can support up to nearly 18 voice users taking the same threshold value. For video traffic, the previous protocol can serve nearly two video users while up to nearly six video users can be served by the proposed protocol taking 10^{-4} as the threshold value of the loss probability. This enhancement comes from the best usage of CTA slots dedicated for real time traffic. The proposed system supports larger number of voice traffic whereas it has the highest priority for admission.

The system utilization for multimedia traffic can be reported from Fig. 8. The system utilization for data traffic of the previous protocol is nearly saturated around 2×10^5 b/s and around (20 b/s) for the proposed one. For voice traffic, the system utilization will be saturated around 2×10^4 b/s for the previous protocol and around 9×10^5 b/s for the proposed one. While for video traffic, it will be saturated around 10^6 b/s for the previous protocol and around 10^7 b/s for the proposed one. The proposed system shows better performance than the previous one in real time traffic. That is because the better performance of the proposed protocol in both system average delay and admission ratio.

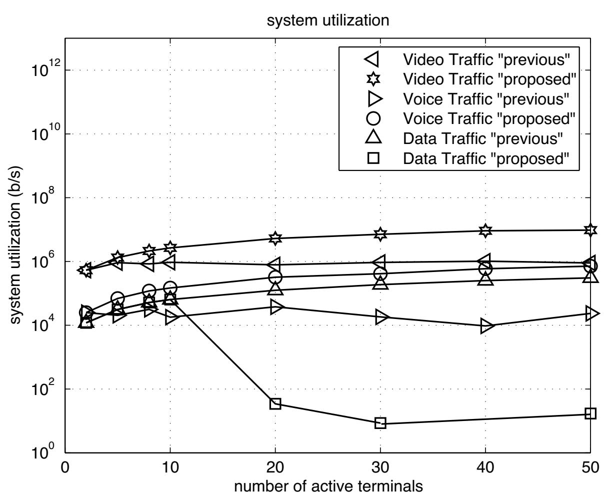

Figure 8. System utilization for multimedia traffic.

According to the lowest admission ratio and the largest average delay for the data traffic in the proposed protocol, it has the lowest system utilization. That is because of the decrease of NRTP size as mentioned above. Therefore, more delay leads to packet loss and less system utilization. While the lower delay and better admission ratio for real time traffic leads to better system utilization. In addition, the system utilization for video traffic is better than voice traffic due to the streaming nature of the video traffic.

In the following figures the effect of the superframe length will be considered in modeling both protocols. The number of users used in this simulation is three users for each traffic type. The greater superframe length is, the larger number of CTA slots in it will be and hence, more possibility for real time traffic admission can be predicted as shown in Fig. 9. While data traffic and all traffics in the proposed

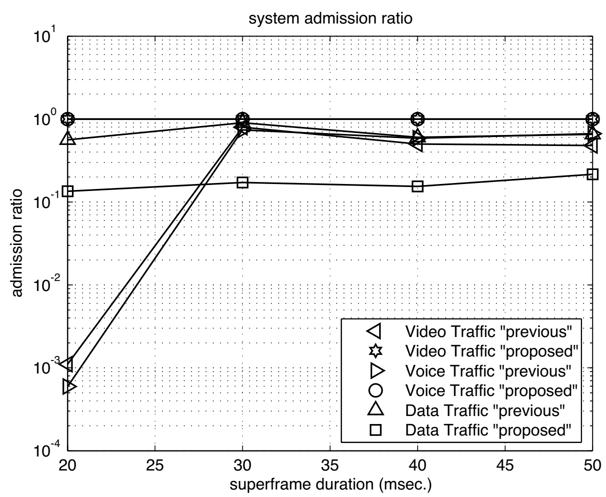

Figure 9. System admission ratio.

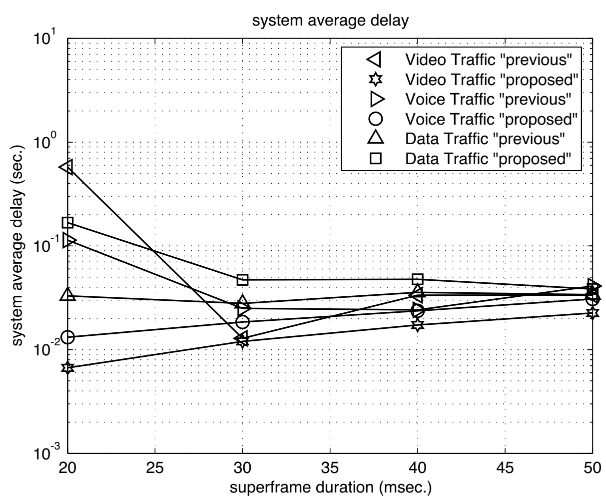

Figure 10. System average delay.

protocol are not affected too much by the superframe length. That is because the light traffic compared with the available channel. The best superframe length for both protocols is 30 msec as shown in the figure.

Figure 10 confirms the above forecast. Data traffic for the proposed protocol decreases at 30 msec superframe length and then nearly saturates with the same value as the data traffic in the previous protocol. On the same way, real time traffic for the previous protocol have the lowest delay at 30 msec superframe length. While a slightly increase of the delay of real time traffic, for the proposed protocol with the increase of the superframe length, is noticed.

From the above discussions, the best value of the superframe length is 30 msec to have nearly the best results for both protocols.

6. CONCLUSIONS

Extensive simulation programs were performed to investigate the possibility of transmitting multimedia over UWB networks. A proposed protocol was explained to achieve QoS for multimedia transmission over UWB networks. Comparisons were done between the IEEE 802.15.3a protocol and the proposed one. The extended results showed evaluation of sensitive parameters affecting real time traffic transmission such as the delay guarantee and the loss probability, as packets with a large delay should be discarded. The number of stations the network can support was determined. In addition, the admission ratio parameter and the system utilization were aroused for the system performance. Furthermore, the effect of the superframe length variation on the system performance is studied in order to determine the best superframe length.

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