

Quality of Service Provisioning for Multimedia Transmission over UWB Networks

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Abstract—In this paper, the Quality of Service (QoS) for multimedia traffic of the Medium Access Control (MAC) protocol for Ultra Wide-Band (UWB) networks is investigated. A protocol is proposed to enhance the network performance and increase its capacity. This enhancement comes from using Wise Algorithm for Link Admission Control (WALAC). The QoS of multimedia transmission is determined in terms of average delay, loss probability, utilization, and the network capacity. In addition, a new parameter is aroused for the network performance.

Index Terms—Ultra Wide Band, Medium Access Control, resource allocation, and Quality of Service.

I. INTRODUCTION

UWB is a technology for transmitting information spread over a large bandwidth (>500 MHz) under the right circumstances. A February 2002 Report and Order by the Federal Communication Commission (FCC) [1] authorizes the unlicensed use of UWB in 3.1–10.6 GHz. This is intended to efficiently usage of exceptional radio bandwidth while enabling both low and high data rates. The FCC defines UWB signal as the emitted signal bandwidth exceeds the lesser of 500 MHz or 20% of the center frequency. Over there, pulse-based systems can access the UWB spectrum under these rules. Each pulse in a pulse-based UWB system occupies the entire UWB bandwidth, thus reaping the benefits of relative immunity to multipath fading (but not to intersymbol interference) [2].

Multiple Band Orthogonal Frequency Division Multiplexing (MB-OFDM) and Direct Sequence- UWB (DS-UWB) were proposed for the physical layer in IEEE 802.15.3a Task Group [3], [4]. The 802.15.3 MAC mainly works within a piconet which is a small network [5], [6]. It consists of data devices and one of them is taken as the piconet coordinator (PNC). The PNC is responsible for devices association/disassociation and the basic timing of the network by sending the beacon to all devices [5]. One major challenge in UWB MAC design is the QoS investigation with efficient resource scheme. Very limited work takes into account the characteristics of UWB for real time traffic.

In this paper, the proposed protocol in [7] has been modified to achieve QoS requirements for multimedia traffic. Further-

more, additional proposed algorithm had been realized for real time traffic.

The paper is organized as follows; Section II gives an overview of UWB physical model and resource allocation. Section III introduces the detail description of the proposed protocol for QoS provisioning. Simulation results and comparison discussions between data, voice, video, and multimedia

traffic using the proposed protocol are made in Section IV. Finally, conclusion will be shown in Section V.

II. RESOURCE ALLOCATION

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 [8], 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:

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

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 [9]. 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 [8], [2], [10]. In [10], [11], 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.

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

One major challenge in UWB MAC design is the QoS provisioning with an efficient resource allocation scheme [11], [12], [13]. Although there have been large researches on real time traffic (voice and video) [13], [14], very restricted work takes into account the unique characteristics of UWB. The proposed protocol in [7] is modified to achieve the QoS requirements for multimedia traffic.

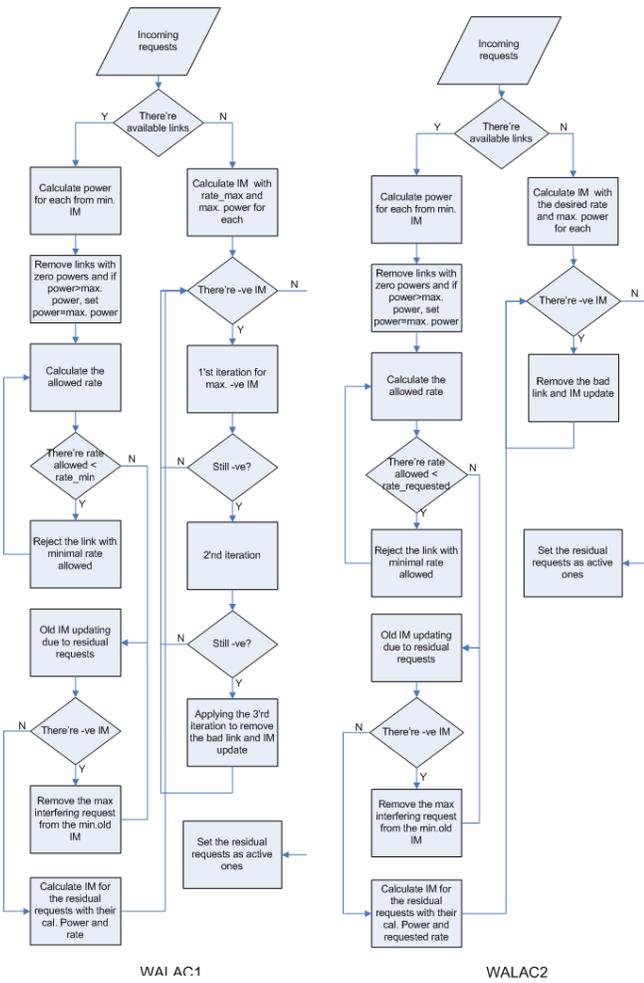


Fig. 1. Flowchart of WALAC1 and WALAC2

III. THE PROPOSED PROTOCOL OVERVIEW

The proposed protocol in [7] shows superior performance for UWB network when compared with Slotted Aloha and Packet Reservation Multiple Access systems in data traffic. It is based on two channels (data and control channels). The data channel is used for traffic transmission while the control channel is existed for requesting a link. The piconet coordinator (PNC) applies Wise Algorithm for Link Admission Control (WALAC) for the link requesting. 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.

This proposed protocol is modified to cope with multimedia challenges. Actually, the modified protocol has two proposed algorithms. One is for non real time traffic (data) which is the same as used in [7], and it is named WALAC1. The other one is made for real time traffic (voice and video) and it is named WALAC2.

In WALAC1, if a data request is valid, there are two cases. Firstly, 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 other case is there are available links in the system. In this case, the PNC calculates the allowed power for each request from the minimal IM of active links from (3). 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 (to achieve fairness) 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 to achieve fairness. 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.

As shown from Fig. 1, there are no great differences between them 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. Furthermore, the PNC has three queues for the incoming requests. The highest priority is placed for the voice queue, then the video one followed by the data one will be served respectively. That is to achieve QoS requirements.

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

$$R_{allow} = \frac{P_0 g_{0j_0}}{\gamma \left(\eta_i + T_f \sigma^2 \sum_{k=1, k \neq i}^N P_k g_{k j_0} \right)} \quad (4)$$

The modified 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 correct queue. Subsequently, it applies WALAC2 for voice and video requests respectively then WALAC1 for non real time one. Over there, The PNC informs the requesting terminals about its state, i.e., admitted or rejected.
- 3) The admitted links transmit in the next slot in the data channel while the rejected ones request again in the next slot in the control channel.
- 4) For the link termination, the PNC is informed through the control channel.

IV. SIMULATION RESULTS AND DISCUSSIONS

In this section, we study the behavior of the proposed centralized protocol through simulations. The simulation area is taken as 50m×50m 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. Durations of talkspurts and silentspurts are exponential distributions with mean values of 1 and 1.35 seconds respectively [15], [16].

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 as 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 call/sec per user. Furthermore, the buffering rate is 9600 b/s [17]. The rest of the default parameters used are shown in TABLE I.

TABLE I
SIMULATION PARAMETERS.

Parameter	Value
T_f	10 ns
σ^2	1.99×10^{-3}
η	2.56×10^{-17}
P_{max}	7 dBm
λ	30
α	4
γ	6 dB
superframe duration	10 msec
slot duration	128 μ sec
packet length	32 bytes
voice life time	20 msec
video life time	50 msec
data life time	6 sec
voice channel coding rate	8.33 Mb/s
video channel coding rate	33.3-100 Mb/s
minimum rate (R_{min})	2 Mb/s
maximum rate (R_{max})	100 Mb/s

The performance of the proposed protocol is measured according to 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 new parameter for the network performance is perused.

Figs. 2 to 4 show the average system delay (the average delay per successfully packets) for data, voice, and video traffic respectively. 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. Furthermore, from Fig. 2, the average delay for data traffic is nearly saturated. That is because the channel can not be dominated by certain

users (because there is no streaming traffic here). While from Fig. 3, the average delay for voice traffic is directly proportional to the number of users then will be saturated. That is because the channel interference is increased with the number of active users and hence less admission ratio which leads to more delay. While from Fig. 4, the average delay is too low compared with the above due to the lowest transmission time. Although the streaming nature of the traffic, the high channel coding rate prohibits users to dominate the channel (and hence, there is fairness among users). More delay can be noticed for large number of users because of the channel congestion.

The average delay for multimedia traffic can be shown from Fig. 5. A slightly decrease for the average delay of both data and voice traffic can be noticed (on the contrary of video traffic). That is because the highest priority of the voice traffic. While for data traffic, although it has the lowest priority, it has non QoS nature. Despite of the users' possession of the channel, the average delay for the data users is not greatly affected like voice users because data traffic can be transmitted with the available rate. On the contrary, video traffic must achieve the QoS requirements. Larger number of active users, larger interference in the channel will be deduced and hence, less probability of admission and more delay can be noticed.

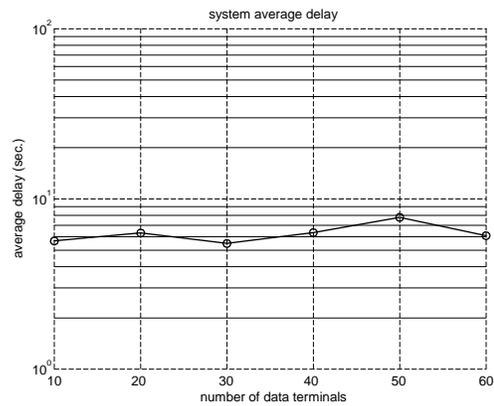


Fig. 2. System average delay for data traffic.

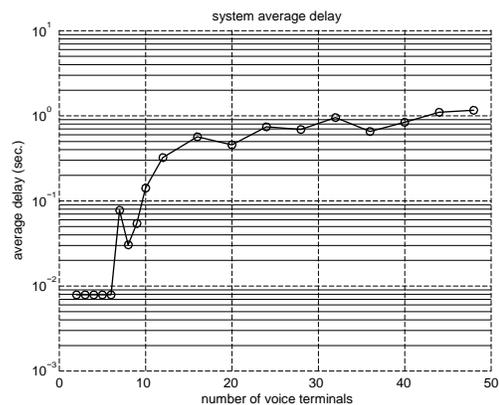


Fig. 3. System average delay for voice traffic.

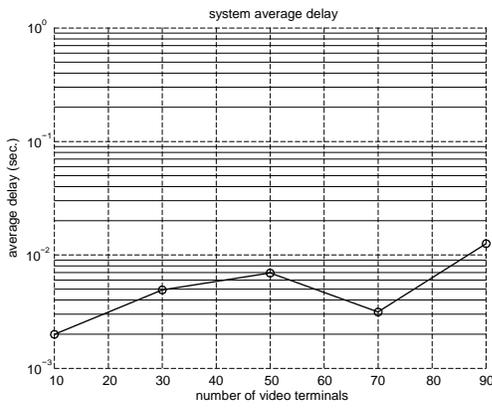


Fig. 4. System average delay for video traffic.

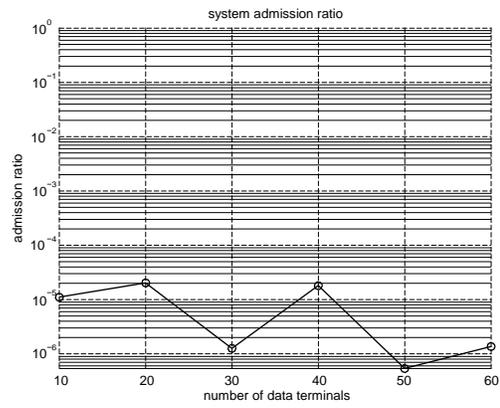


Fig. 6. Admission ratio for data traffic.

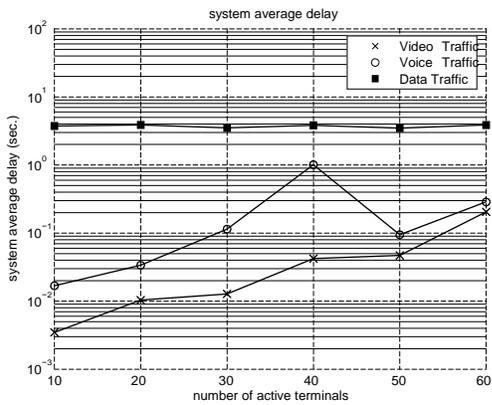


Fig. 5. System average delay for multimedia traffic.

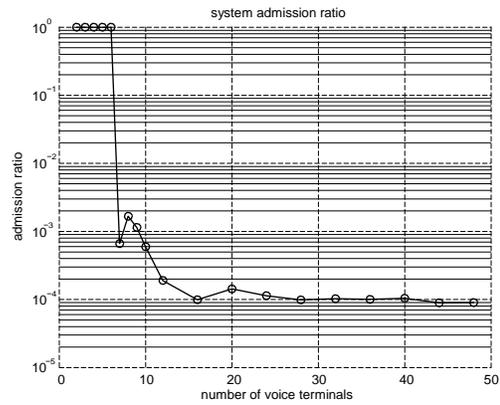


Fig. 7. Admission ratio for voice traffic.

Figs. 6 to 9 depict the system admission ratio versus the number of active terminals for data, voice, video, and multimedia traffic respectively. For data, voice, and video traffic, the admission ratio is inversely proportional to the traffic then it will be saturated between 10^{-5} to 10^{-6} for both data and video traffic and nearby 10^{-4} for voice traffic. That is because the highest priority of the voice traffic. While the low admission ratio for the data traffic is due to its very low buffering rate. Furthermore, from Fig. 8, the degradation in the admission ratio for the video traffic can be noticed for larger number of users due to the channel congestion and hence the delay will be increases as shown in Fig. 4.

For multimedia traffic as shown from Fig. 9, a slightly decrease in the admission ratio for voice traffic can be noticed; because the presence of the other traffic admitted to the channel (video and data) lowers the probability of the admission. While there are no effective changes in the admission ratio for the video traffic. That is because its streaming nature besides its high transmission rate. While for the data traffic, the admission ratio is increased and hence slightly less delay can be noticed from Fig. 5. That is because the presence of other traffic in the system which prohibits the data users to take possession of the channel as happened when it stood alone. However, the admission ratio for the video traffic still the minimum one then for the data and voice traffic respectively.

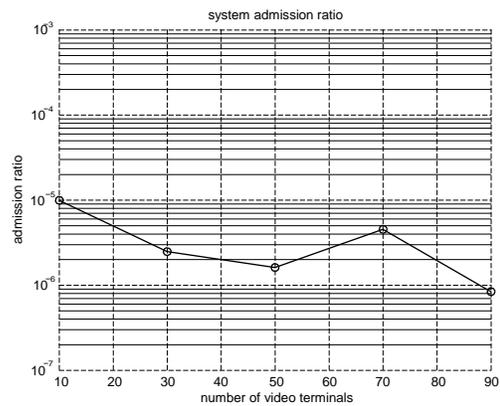


Fig. 8. Admission ratio for video traffic.

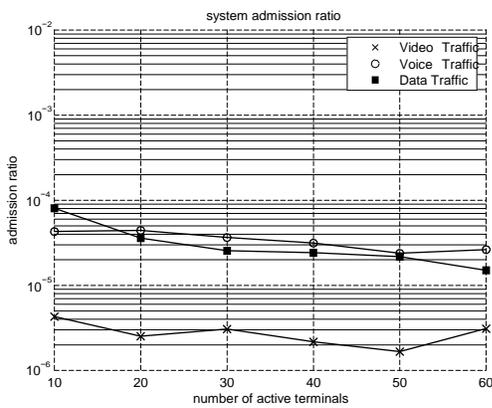


Fig. 9. Admission ratio for multimedia traffic.

The system loss probability (the ratio between the rejected transmitted packets and all transmitted packets) for data, voice, and multimedia traffic can be shown from Figs. 10 to 13 respectively. For data traffic, a very large number of data terminals can be supported. 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. While for voice traffic, the lowest threshold value of the maximum delay (to achieve real time requirements) plays a great role in the probability of loss increase. Fig. 11 shows that the system can support up to 37 voice users taking 10^{-2} as the threshold value of the loss probability. For video traffic, because of the lowest transmission time delay in the buffer, in addition to the large maximum delay threshold value, the system can support more than 90 users taking 10^{-4} as the threshold value of the loss probability as shown in Fig. 12. For multimedia traffic shown from Fig. 13, the system can support nearly 43 voice users due to its highest priority. This increase in the number of users due to the slightly decrease in the admission ratio beside the delay decrease which prohibits users to dominate the channel. While a large degradation in the video traffic is noticed as nearly 45 video users can be supported. That is because of the long time channel usage for the voice users (for the low channel coding). Therefore more delay can be noticed which leads to more losses. Despite of its non changeable admission ratio, the delay is increased, then the admitted links will be terminated due to the threshold value of the delay for video traffic. While for data users, there is no degradation noticed due to its non real time nature. Furthermore, users enhancement is predicted due to its admission ratio increase besides its average delay decrease.

The system utilization for data, voice, and video traffic can be shown from Figs. 14 to 16 respectively, while for multimedia traffic is shown from Fig. 17. The system utilization for data traffic is nearly saturated around 10^4 b/s. The low utilization because of the low traffic rate for data users. While for voice users, the system utilization is directly proportional to the number of active users and saturated around 10^6 b/s. That is because the saturation of the admission ratio and hence more users admitted for more traffic, therefore more successful transmission packets over the time. For video traffic, the utilization will be saturated around 10^6 b/s. The

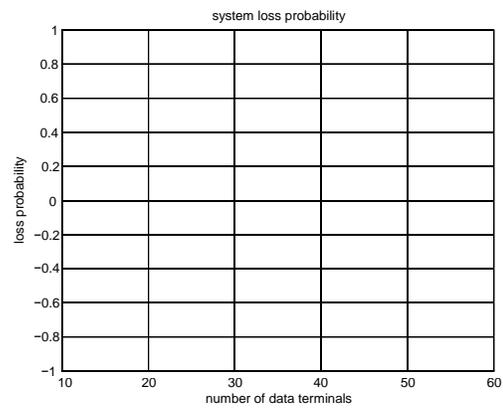


Fig. 10. System loss probability for data traffic.

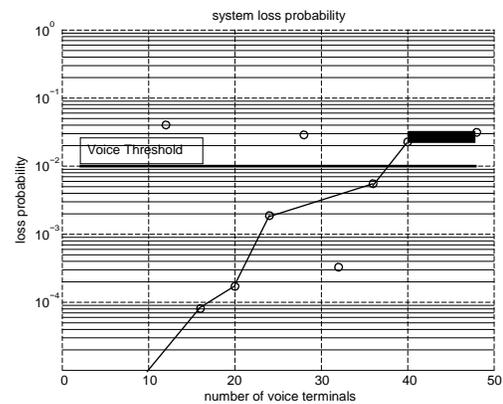


Fig. 11. System loss probability for voice traffic.

more utilization for lower users is due to the streaming nature for video traffic. For multimedia traffic as shown from Fig. 17, the saturated utilization and the better utilization for video traffic over voice and data can be noticed due to the streaming nature of the video traffic. The data traffic has the lowest utilization because of its low buffering rate.

From these discussions, the system performance is controlled by both the admission ratio and the system average delay. It can be ordered from the best to the worst as follows;

- High admission ratio with low average delay. It is like the case of the data traffic.
- Low admission ratio with low average delay. It is like the case of the voice traffic when it is alone and with the multimedia traffic. The voice enhancement can be noticed.
- High admission ratio with high average delay.
- Low admission ratio with high average delay. It is like the case of the video traffic when it is alone and with the multimedia traffic. The video degradation can be noticed.

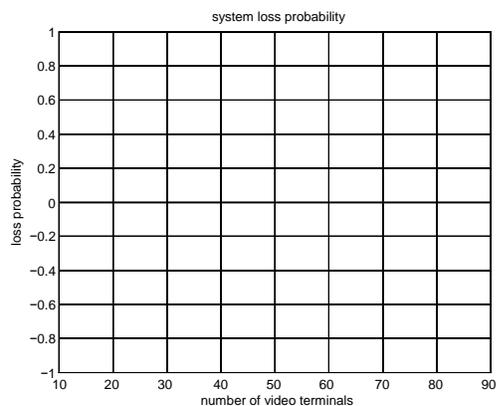


Fig. 12. System loss probability for video traffic.

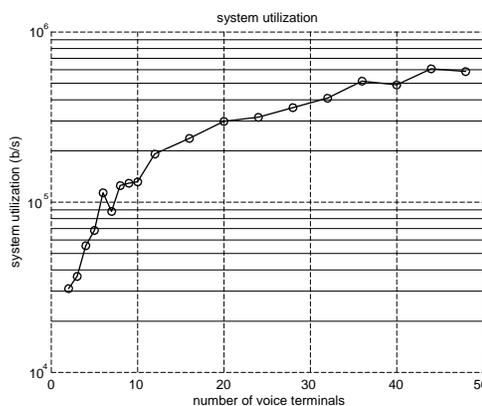


Fig. 15. System utilization for voice traffic.

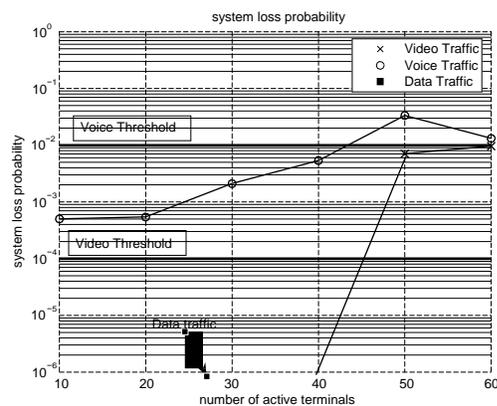


Fig. 13. System loss probability for multimedia traffic.

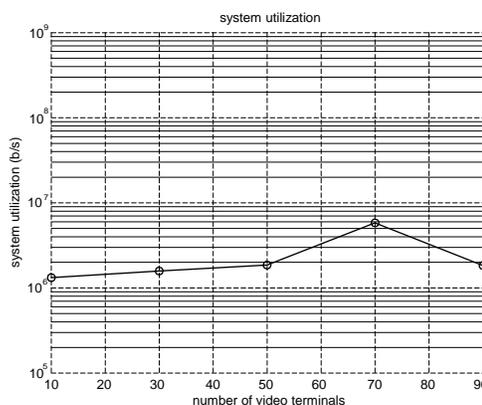


Fig. 16. System utilization for video traffic.

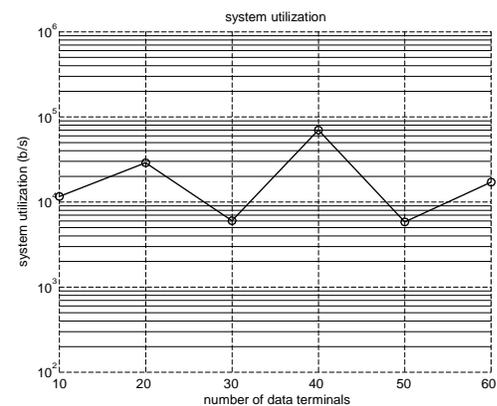


Fig. 14. System utilization for data traffic.

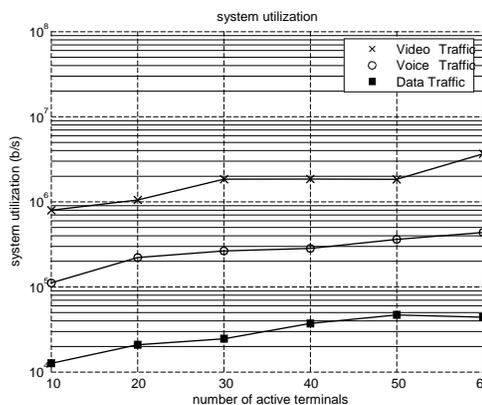


Fig. 17. System utilization for multimedia traffic.

V. CONCLUSION

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. 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 system performance can be managed by both the admission ratio and the average delay. The best performance is for the highest admission ratio with the lowest average delay, while the worth performance is for the lowest admission ratio with the highest average delay.

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