Wireless Network project final report

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***Abstract*—Based on PHY and MAC layer, this paper proposes an adaptive cross layer design for downlink OFDM system. The purpose of this paper is to realize maximum weighted sum capacity (MWC) that balances the quality of service and capacity. In MAC layer, data scheduling is realized. The order of packet transmission depends on the weight, which is determined by delay, tolerance, packet size, quality of service (QoS) priority and other factors. The weight of the queue is calculated in the modified largest weight delay first(M-LWDF) scheme, and the weight of the packet is calculated in the packet dependent (PD) scheme. When the weight is obtained, it will be transferred to the physical layer, combined with the subcarrier and power allocation scheme to achieve flexible traffic control. The results show that the total complexity of PD scheduling is much lower than that of queue-based M-LWDF, and the system performance is better in spectrum and energy efficiency, delay and packet loss rate, especially in multi-user multi service network.**

***Index Terms*—Cross-layer design, OFDM, resource allocation, QoS, multiuser diversity, M-LWDF**

I. Introduction

Wireless communication has developed rapidly in the past decades, but the needs of users have never been met. In addition to the substantial increase in the number of users, people also put forward higher requirements for the speed of network transmission. The cross layer design realizes the adaptive resource management and active feedback control for the downlink service of multi-user multi service OFDM system. The principle of cross layer design is to eliminate the boundary between each layer, to share information, to realize dynamic feedback and compensation. This project focuses on the physical layer power and subcarrier distribution and MAC layer data scheduling optimization. The simulation results show that the optimized design can reduce the average packet delay and packet loss rate, and improve the spectrum utilization and energy utilization. In addition, the design can also achieve adjustable resource allocation to meet the needs of different businesses. However, in practical applications, the cost of architecture and unexpected development and maintenance problems can not be ignored.

In the traditional open system interconnection (OSI) model, the communication system is divided into seven independent layers to standardize the services and layer interface functions of different systems [8]. This object-oriented design ensures clear structure, simple maintenance and flexible modification, but the cost is low efficiency of information exchange between layers. Therefore, a cross layer model is proposed to provide dynamic feedback and compensation.

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As the first layer and the lowest layer, PHY layer focuses on transmitting the original bits through the channel. In this layer, channel response, noise level and available capacity can be obtained. The adjacent MAC layer is a sub layer of data link layer, which is used to realize multiplexing and traffic control. It determines which packets or queues to serve at a certain time, and thus the order of data transmission. MAC layer is very important for wireless communication, because the allocation of broadcast channel directly affects the user experience [9]. Although different networks can be very different in higher layers (such as transport layer and application layer), they conform to the similar resource allocation and scheduling implementation of phy and MAC layer [9]. In this project, the two layers are optimized to provide general support for different OFDM based communication networks. Although OSI model is more like a theoretical reference, the network has been controlled by TCP / IP protocol for decades, but it is still conducive to the analysis and design of typical communication systems [9]. Therefore, the design, simulation and discussion of this project are based on OSI model.

Based on the previous work, this paper designs an adaptive cross layer communication model between physical layer and MAC layer for downlink multi-user multi task heterogeneous OFDM system. By breaking the barrier between layers, CSI and available capacity obtained from PHY layer can be transmitted to MAC layer for data transmission. On the other hand, delay and weight information can be returned to the physical layer to achieve adaptive subcarrier and power allocation. Compared with the traditional hierarchical architecture, this method is more flexible in meeting different business requirements, more efficient in power and spectrum utilization, and more advanced in network performance such as delay and packet loss rate.

II. Reference paper

Moustafa M. Nasralla and Ikram Ur Rehman have co-authored this research paper titled QCI and QoS Aware Downlink Packet Scheduling Algorithms for Multi-Traffic Classes over 4G and Beyond Wireless Networks. It was published in the 2018 International Conference on Innovation and Intelligence for Informatics, Computing, and Technologies (3ICT). It is 7 pages long and concerns with providing fair scheduling and balanced Quality of Service for the two different traffic classes viz. Real-Time and Non-Real Time traffic. The abstract for the paper is mentioned below.

The recent advancements in wireless technologies and applications make downlink scheduling and resource allocation a hot topic of research. Hence, fair scheduling and balanced Quality of Service (QoS) delivery for different types of traffic (e.g., VoIP, video, and best-effort) are vital for next-generation wireless networks. In this paper, we analyze various downlink scheduling algorithms in terms of network-oriented performance parameters such as average throughput, system fairness, average packet loss ratio, and system spectral efficiency. Also, we show the effect of the QoS Class Identifier (QCI) parameters on different delay-aware scheduling algorithms. Furthermore, we propose a group of algorithms to improve the existing Log-rule, Linear-rule, and Modified Largest Weighted Delay First (M-LWDF) scheduling strategies. This is achieved by including in the algorithms the QCI parameters to balance the QoS delivery for different traffic-classes with improvement to the overall system performance. Through simulation, we show that the proposed scheduling algorithms utilizing the QCI and QoS parameters introduce improved QoS performance for different traffic classes (i.e., real-time (RT) and non-real-time (NRT)).

The authors in their paper have simulated an environment where the user traffic has been segmented into three categories. 4/5ths of the total users are equally divided into two classes - VoIP users and video users. The remaining fifth are constant bit rate users. The author has used LTE-Sim system-level simulator to simulate the above scenario. Moreover author has tuned his network to the parameters described in Table I.

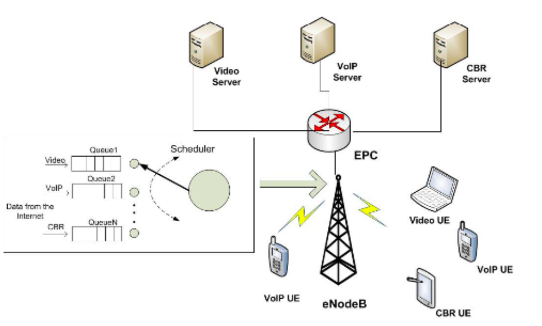

Fig. 1. Network Architecture

The paper evaluated the performance of several variants of the M-LWDF algorithm such as Packet Loss Rate(PLR) aware M-LWDF downlink packet scheduling algorithm. α_i ensures user's delay, $D_{HoL,i}(t)$ is the Head of Line packet delay.$\bar{R}_i(t)$ signifies average transmission data rate and r_i, j(t) is the available data rate/sub-channel. A Physical Resource Block (PRB) is piece of bandwidth in both frequency and time domain. Largest Weighted Delay First translates to allocating

Name	Value
Bandwidth	10MHz
Number of PRBs	50
Macrocell radius	1 Km
E-UTRAN frequency band	1(2.1 GHz)
Frame Structure	FDD
Maximum Delay (video and VoIP)	100ms
Maximum Delay (CBR)	300ms
Video Bitrate	242kbps
CBR Bitrate	100kbps
VoIP Bitrate	8.4kbps
Flow Duration	20 sec
Simulation Time	30 sec
UE Speed	3 km/hr
Path Loss and Channel Model	Pedestrain-A propagation Model

TABLE I
SIMULATION PARAMETERS FOR LTE DOWNLINK SYSTEM

the PRB to the user, whose weight is the highest among the traffic. It is a down-link scheduling scheme.

$$W_{i,j}(t) = \begin{cases} \alpha_i(t) * D_{HoL,i}(t) * \dfrac{r_{i,j}(t)}{\bar{R}_i(t)}, & for\ i\ \epsilon \text{ real-time UE} \\ \dfrac{r_{i,j}(t)}{\bar{R}_i(t)}, & for\ i\ \epsilon\ non \text{ real-time UE} \end{cases}$$

Fig. 2. M-LWDF Algorithm

The author then goes on evaluating the variants of the proposed algorithm, tweaking the algorithm by introducing the Packet Loss Ratio (PLR) which is the ratio of lost packets to transmitted packets. The author uses this algorithm to calculate weights for real time traffic. The proposed algorithms are QCI PLR aware Log Rule, QCI PLR aware M-LWDF Linear Rule. These are described in Figure 3 and 4. The author then evaluates these algorithms on four characteristics namely Packet Loss Rate, Average Throughput, Fairness Index and Spectral Efficiency.

$$W_{i,j}(t) = \alpha_i(t) * D_{HoL,i}(t) * \frac{r_{i,j}(t)}{\bar{R}_i(t)} * \log PLR \ , for\ i\ \epsilon \text{ real-time UE}$$

Fig. 3. QCI PLR Aware M-LWDF Log Rule

$$W_{i,j}(t) = \alpha_i(t) * D_{HoL,i}(t) * \frac{r_{i,j}(t)}{\bar{R}_i(t)} * PLR \ , for\ i\ \epsilon \text{ real-time UE}$$

Fig. 4. QCI PLR Aware M-LWDF Linear Rule

As part of our implementation for this paper, we have used Matlab simulator to simulate the above scenario. Comparison between M-LWDF and Packet Dependent scheduling is performed for the characteristics system throughput, average packet delay, packet drop rate and delay outage probability.

III. SIMULATION DESCRIPTION

In this section, the reasons for choosing MATLAB as project simulator will be given. Then a description of the simulation that project designed, modifications made in project and the reasons for them will be discussed. Finally presentation of project's results in graph and comparison of these results with the original paper's results. The entire simulation process is as follows.

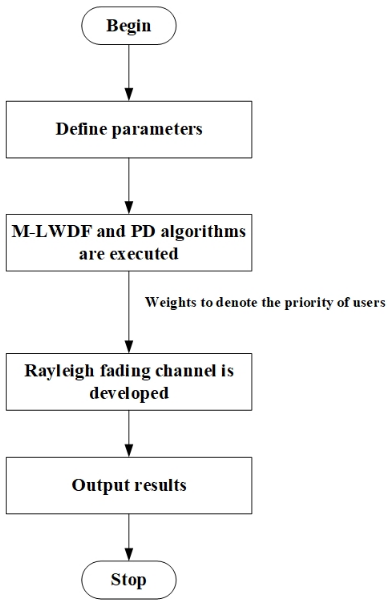

Fig. 5. simulation process

A. A description of MATLAB and reasons for choosing it

MATLAB is a commercial mathematical software produced by MathWorks in the United States. It is used in advanced technical computing languages and interactive environments for algorithm development, data visualization, data analysis, and numerical calculation, mainly including MATLAB and Simulink.

The reasons for choosing it as project simulator are following:

- It has complete graphics processing functions to realize the visualization of calculation results and programming.
- Friendly user interface and natural language close to mathematical expressions make it easy for scholars to learn and master.
- The function-rich application toolbox (such as signal processing toolbox, communication toolbox, etc.) provides users with a large number of convenient and practical processing tools.
- It is easier to implement the scheduling algorithm on MATLAB than on NS3 or other simulators.

B. Parameters of simulation

In this part, the parameters of simulation will be given as following table. Most of parameters' setting is based on the reference paper. The BE in this table means best effort.

C. Introduction to two scheduling algorithms

The first algorithm is Modified Largest Weighted Delay First (M-LWDF) Scheduling algorithm. It is a benchmark algorithm in reference paper and our project. This is a queue-based scheduling scheme which tends to serve the one with largest weighted delay with priority, thus the delay as the main consideration is kept in a bound. At the beginning of each time slot, all queues' weights are calculated. Then these weights are compared to determine which queue is to be chose in this slot. The entire process is shown by Figure 5.

Name	Value
Number of users	512
Number of queues per user	3
Number of total slots	2000
SNR	20db
Slot duration	1ms
Voice tolerance	100ms
Video tolerance	100ms
BE tolerance	300ms
Packet arriving rate of voice	8.4kbps
Packet arriving rate of video	242kbps
Packet arriving rate of BE	100kbps

TABLE II
PARAMETERS OF SIMULATION

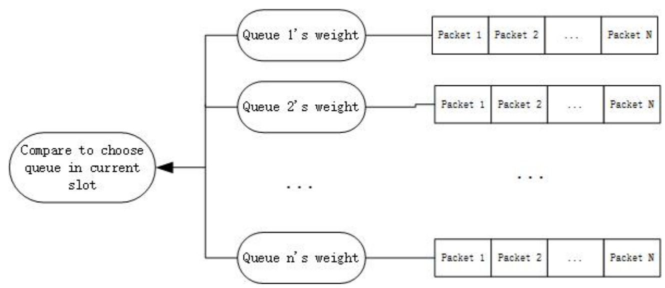

Fig. 6. M-LWDF process

From this figure, it is easy to find out that even though M-LWDF can adjust the traffic to guarantee throughput and keep the delay below a level, the mechanism determines that the transmission order for each queue is still first in first out (FIFO). For example, if a video packet arriving later than voice packets, it can be served earlier. However it should always stay behind of last video packet. So, a more flexible algorithm is required when urgency of packets in the same queue are different in large scale.

The second algorithm is Packet Dependent (PD) Scheduling algorithm. This algorithm is the algorithm for this project to improve the M-LWDF algorithm and compare its results with the results of M-LWDF. Rather than relying on queues, PD algorithms focus on signal packet. It will assign different weights to different packets based on characteristics of them for example it will choose different weight based on packet's delay, delay tolerance, packet size and QoS priority. The entire process is shown by Figure.

In a slot, weight of the packets are calculated. Then, different packets' weights will be compared and the packet with the largest weight will be chose to serve. By using this PD algorithm, the more urgent packets can be chosen to server first than normal packets. So in some cases, PD algorithm will have better performance than M-LWDF.

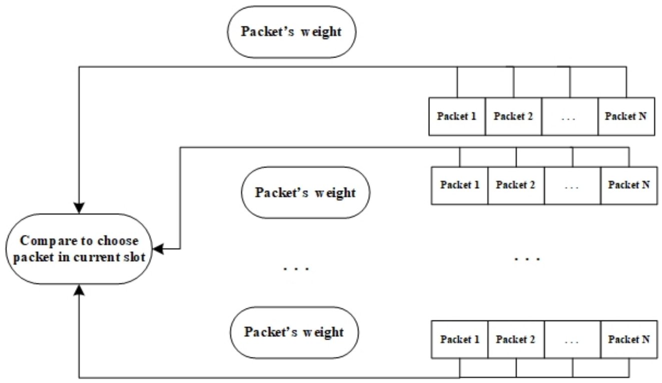

Fig. 7. PD process

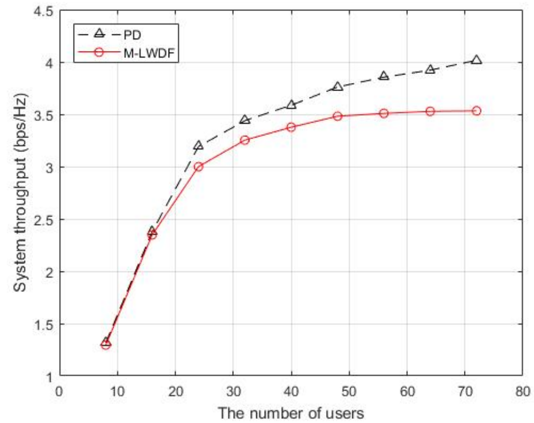

Fig. 8. Influence of user number on throughput with SNR = 20 dB

D. Rayleigh fading channel model

Rayleigh fading channel is a statistical model of radio signal propagation environment. This model assumes that after a signal passes through a wireless channel, its signal amplitude is random, that is, "fading", and its envelope follows Rayleigh distribution. This channel model can describe short-wave channels reflected by the ionosphere and troposphere, as well as urban environments with dense buildings. The fundamental parameters of the channel are shown in Table.

Name	Value
Range of SNR	0-10db
Number of subcarriers	64
Point of FFT	64
Number of paths	6
Number of blocks	1000
Number of bits per block	10000

TABLE III
PARAMETERS OF CHANNEL

E. Run simulation and output results

After setting all relative parameters, run the main function to get M-LWDF user basic.mat which contains all data from simulation. Then run the drawing function to draw all results as following figures.

Figure 4 shows the influence of the user number on the throughput with different weight design. In this case, the SNR is set to 20 dB and the number of users varies from 8 to 72. From the figure, it is easy to find out that when the number of user is small, throughput is nearly the same for M-LWDF and PD scheduling algorithms. However, with number of users increasing, the diversity of packet is increased. So, PD algorithms have better performance than M-LWDF algorithm.

Figure 5 shows the performances of PD and M-LWDF in another dimension. In this figure, it shows influence of SNR on throughput with user number = 32. From the figure, it indicates PD has better performance with low SNR. When SNR is high, these two algorithms have almost same performance because scheduling algorithms can not work in high SNR situation.

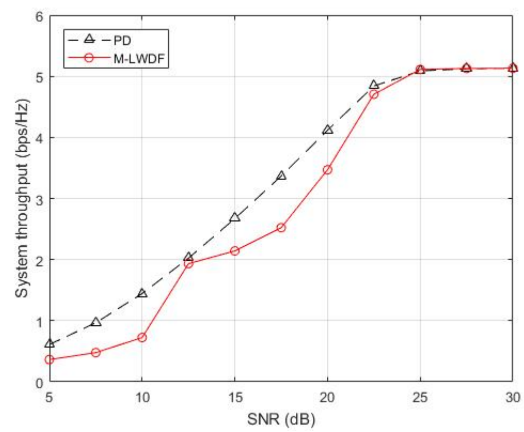

Fig. 9. Influence of SNR on throughput with user number = 32

For different user number, the average voice packet delay with SNR= 20 db is shown by Figure 6. When the number of use is from 8 to 32, the performance of M-LWDF is slightly better than PD. The reason is that due to sufficient transmission resources and zero packet loss rate, all data packets tend to have relatively low latency. As a result, no matter which scheme is adopted, voice packets should be served first. So, M-LWDF has better performance under this situation. When the number of users increases, PD has better performance. For a system with 72 users, the average voice delay is nearly 42 ms for M-LWDF, which is more than 3 times compared with PD algorithm.

Figure 7 shows the influence of SNR on average voice packet delay for 32 users. When SNR is low, M-LWDF has lower delay because M-LWDF always assigns the maximum weight to voice queue. So, the voice delay is relatively low for M-LWDF. With SNR increasing, PD has lower delay.

Figure 8 indicates influence of user amount on average delay of video packets with SNR=20 dB. When the amount of

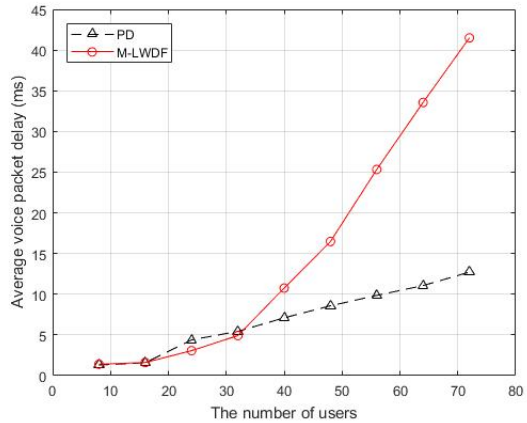

Fig. 10. Influence of SNR on average voice delay with SNR = 20 dB

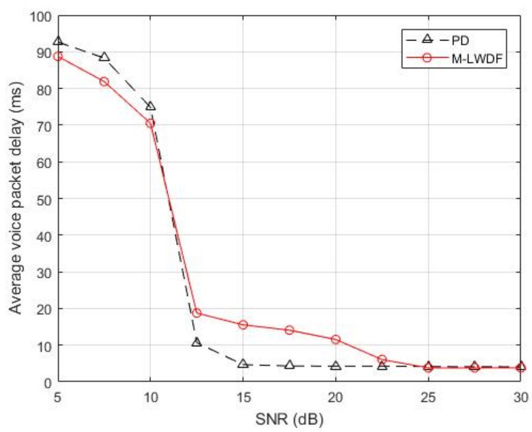

Fig. 11. Influence of SNR on average voice packet delay for 32 users

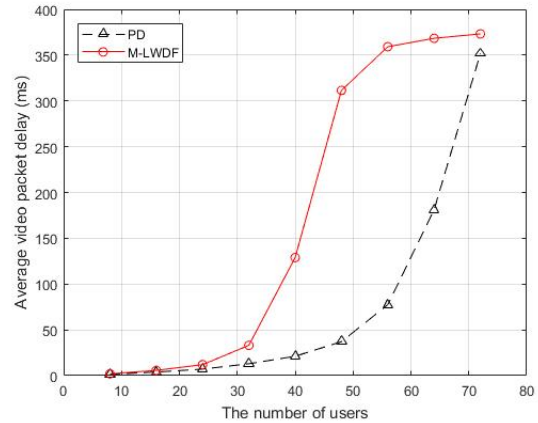

Fig. 12. Influence of user amount on delay of video packets with SNR=20 dB

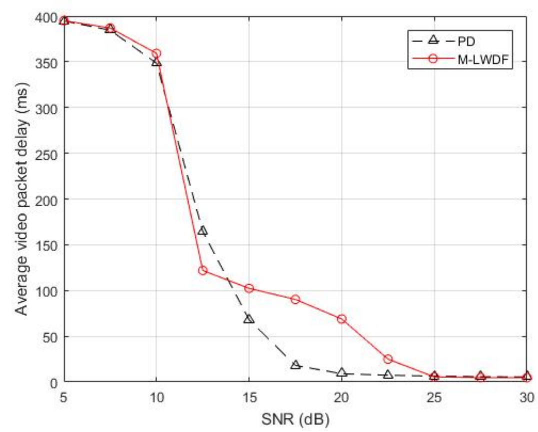

Fig. 13. Influence of SNR on video delay with user number = 32

user is small, two algorithms have almost same performance. When the number of users from 30 to 70, PD has much better performance than M-LWDF.

Figure 9 shows the result when the number of user is fixed to 32 and the SNR is varied from 5 to 30 is presented. In most cases the delay of PD is shorter than the delay of M-LWDF.

Figure 10 shows the video packet drop rate for a 32-user system with different SNR. When the SNR is low, PD has better performance because M-LWDF will allocate more resource to guarantee the performance of voice service but PD will keep a balance between voice and video packets in weight design.

Figure 11 shows influence of user number on BE packet drop rate with SNR = 20 dB. When the number of user is below 25, these two algorithms all can keep packet drop rate at 0. With the number of user increasing, M-LWDF has higher packet loss rate than PD.

According to Figure 12, when SNR is from 12db to 22db, PD has lower packet drop rate than M-LWDF. When SNR is low, these two algorithms need to guarantee voice and video service in the first place. So the packet drop rate is very high with low SNR.

Outage probability reflects the packet delays by showing the probability that the waiting time is longer than the the required threshold time. As shown by figure 13 and 14, PD has lower delay outage pro ability than M-LWDF in voice and video service. That is to say that PD algorithm can guarantee more packets with lower delay than M-LWDF.

F. Comparison of project's results with the original paper's results

This part will compare project's results with the original paper's results in parameter setting,channel chosen and overall conclusion.

For parameter setting, our project keeps the same parameter setting with the original paper to do the best to simulate model in the reference paper.

For the channel chosen, in reference paper, it chooses pedestrian-A propagation model as channel model to simulate the propagation in urban environment. Because there is insufficient description of this channel in the article and there are few simulation examples of this channel on the Internet, our

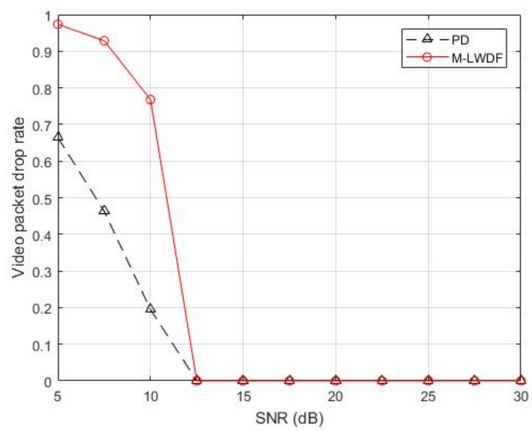

Fig. 14. Influence of SNR on video packet drop rate when user number = 32

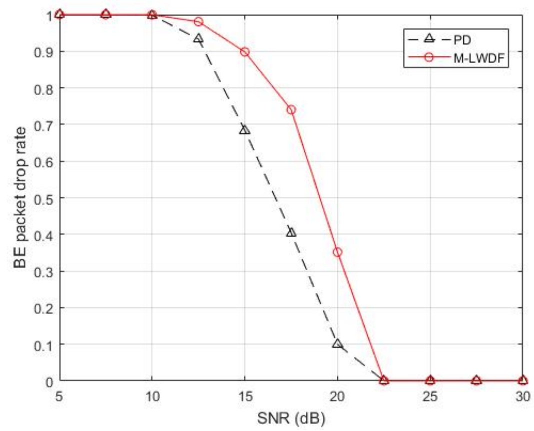

Fig. 16. Influence of SNR on BE packet drop rate when user number = 32

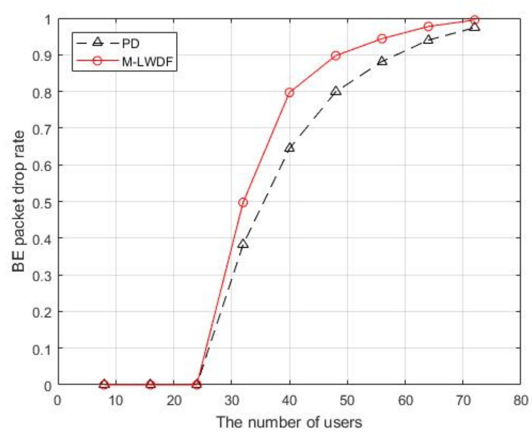

Fig. 15. Influence of user number on BE packet drop rate with SNR = 20 dB

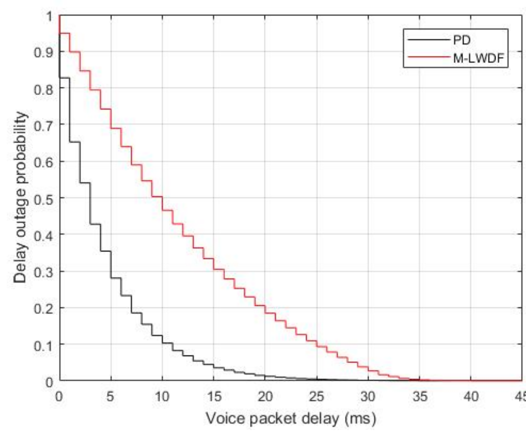

Fig. 17. Voice traffic delay probability with 32 users when SNR = 20 dB

project chooses Rayleigh fading channel model as project's channel model which is suitable to simulate the propagation in urban environment too. So, it may have little influence on results of our project.

For overall conclusion, our project's results show PD algorithm has better performance on video than M-LWDF with the same SNR when user amount is high. It is the same result with original paper. Our project results show M-LWDF may have better performance on voice service with small number of user. This conclusion is the same with reference paper too.

IV. CONCLUSION

A. Discussion on M-LWDF and PD algorithms

The simulation results suggest that by choosing proper packets for weight calculation, PD demands a lower complexity than queue-based M-LWDF, but in most cases the performance in terms of throughput, delay and packet loss rate is better, especially for multiuser multiservice system when the resource is insufficient for lossless transmission. Overall, the objectives of learning wireless channel models, understanding basic network structure, studying OFDM system, and investigating packet transmission mechanism have been achieved.

B. Difficulties

This project aims at the cross layer design of multi-user multi service orthogonal frequency division multiplexing system with heterogeneous downlink traffic. We choose MATLAB as the simulation environment of the project, and compare the M-LWDF and PD algorithms in the aspects of throughput, packet delay, packet drop rate and output probability. This process is very difficult. Although we find a demo that fits the theme better from the Internet, unfortunately, some of the errors in it can not be run, and some temporary data generated based on random numbers can not fully meet our needs, so we spend a lot of time in the process of debugging.

C. Final division of labor

Wenxuan Bao is responsible for the part of throughput, Rui Guo is responsible for the part of packet delay and packet drop rate, and JHA Abhishek is responsible for the part of output probability. Finally, we finished the PowerPoint and the report together.

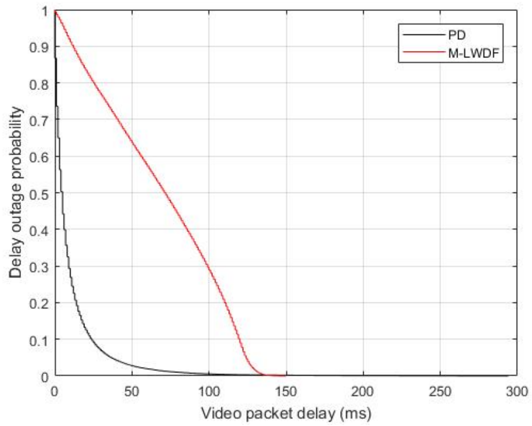

Fig. 18. Video traffic delay probability with 32 users when SNR = 20 dB

D. *Future outlook*

Although the simulation result can reflect the real situation to some extent, there are also some places that can be improved for more accurate outcomes. The most important one is the packet size distribution. With the fast advancement of communication industry, services as high-resolution audio, 4K resolution video have updated the demands of public to relevant traffics. Therefore, the packet size design and weight balancing may need reconsidering when the cross-layer design is applied in specific cases.

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