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CHANNEL PREDICTION BASED ADAPTIVE PACKET LENGTH FOR WIRELESS COMMUNICATIONS

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Abstract

Wireless communication channel are subjected to different types of error which resulted in the lost of data packets. Different types of error recovery techniques where used to reduce the probability of packets being lost such as, Automatic Repeat Request ARQ and Forward Error Correction FEC. In this paper adaptive packet length in conjunction with adaptive modulation was proposed based on the prediction of the channel state. The selection of modulation type and optimal packet length was carried based on the corresponding instantaneous channel SNR. The simulation results shows that a significant improvement in throughput when adaptive packet length and modulation level compared to non adaptive technique. The channel prediction overcomes the outdated channel state condition when channel estimation used.

Keywords – Channel prediction, Packet length, Packet error

I. Introduction

Wireless communication services such as wireless networking have been greatly increased using reliable high speed communication to achieve accepted quality of service QoS over limited radio resources such as allocated bandwidth and power. The most effective technique to satisfy such requirement is to control the communication parameters such as symbol rate, modulation level, and coding rate according to the channel state condition. Wireless radio channels are typically subjected to bit errors, It is not only subjected to noise, interference, and other channel impediments, but these impediments change over time [1].

In packet radio communication the QoS is highly affected by means of Bit Error Rate BER which resulted in an increased Packet Error Rate PER and high end to end latency. In [2,3,4] investigates how frame size affects wireless network throughput. Usually Cyclic Redundancy Check CRC used to detect error occurs in packets, thus any packet will be discarded if there is one or more error bit. However longer packets are more likely to be lost due to channel errors than small packets[5], this resulted to increase the retransmission rate hence the overall performance decreased besides to inefficient channel utilization. There has been a significant amount of research on adaptive transmission schemes in order to maximize the data throughput. The main adaptation algorithms are based on varying the transmission parameters such as transmitted data rate and power where different modulation and coding schemes are adopted according to the channel state condition. Earlier research work have been carried to find an optimum packet length for certain channel conditions [2,4,6] where a tradeoff between large packets to increase throughput and small packets to reduce packet error rate.

In [7] local packet length adaptation algorithm was proposed. Each node dynamically adjust its packet length based on estimate of the probabilities of each significant type of packet loss. A constant channel BER was assumed.

In [8] a statistical analysis of protocol efficiency is carried out for a class of S&W protocols, leading to the optimal packet size which is evaluated as a function of range, Bit rate , and expected error probability for typical underwater channels.

In [3] an optimal frame size predictor based on the Kalman filter has been proposed to keep tracking variations in channel quality.

In this article packet length adaptation based on the linear prediction of the channel state in order to overcome the outdated channel state condition was investigated.
The rest of the paper is organized as follows: In section II the theoretical relationship between packet error rate and probability of error was addressed. In section III channel prediction against channel estimation was investigated. In section IV the performance evaluation of the proposed system was simulated. A concluding remarks was discussed in section V.

II. Packet Error Model

Losses can be classified into losses due to collisions which resulted when packet overlaps in time with that of another

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spatially close enough node [7], and packet loss due to channel errors. However the loss due to collision depends on the probability of a channel being busy at the transmission time and did not depend on the packet length and will not be considered here, hence the probability of instantaneous packet error rate is given by:

$$PER = 1 - \prod_N (1 - P_{se}) \qquad (1)$$

Where PER is the Packet error rate

P_{se} probability of symbol error rate

N the number of symbols within one data packet

It is clear that better utilization of the channel is by increasing packet size at the expense of increased chance of a packet to be lost due to bit errors.

Basically there are two main error recovery mechanisms, ARQ in which the retransmission of lost packet continues until it will be received correctly, and Forward Error Correction FEC where redundant data added to the original message. In ARQ the message is divided into blocks of suitable size that are transmitted after the addition of error control bits Figure 1 shows the general packet structure which consists of payload and overhead. The error detection is based on Cyclic Redundancy Check CRC consisted in the overhead of the packet.

Header	*Payload*	*CRC*

Figure 1. Packet structure

Obviously larger packet length will be more probable the packet being in error. The packet length efficiency can be defined as:

$$\boldsymbol{\eta = \frac{length\ of\ packet\ data\ field}{size\ of\ packet}} \qquad (2)$$

Shorter packets are less efficient than long packets due to the more overheads with respect to the payload data. However shorter packets are less affected by channel variations than long packets. In other words the instantaneous throughput is closely related to the dynamic channel condition due to fading. Therefore maximum throughput can be achieved if the optimum packet length was adaptively chosen by estimating the channel condition.

An ARQ scheme is widely used to achieve error free data communication. The data throughput is the only serious drawback of ARQ which is very low when the channel is bad or the receiver is far from the transmitter due to the high retransmission probability.

III.Channel Modeling and prediction

Typically wireless communication is unreliable because of the fluctuations in the received signal due to multipath variations of the received signal. Besides to the interference from other transmissions, path loss, shadowing, multipath fading and thermal noise from the receiver's electronics are affecting the recovery of the transmitted signal.

Multipath fading resulted in a dramatic variation in the received signal power because the received signal is a sum of multipath components each with independent amplitude, phase, and frequency components. Fading caused by a phenomenon known as the Doppler effect[1]. $f_d = f_c \frac{v}{c} \cdot cos\theta = f_{dmax} cos\theta$ Where f_c the carrier frequency, θ is the angle of arrival of the received signal, v is the relative velocity, and f_{dmax} is the maximum Doppler frequency. The received signal was modeled as a sum of non-resolvable multipath components each with independent amplitude, phase and frequency components. The channel is thus modeled in complex baseband as [9,10].

$$\boldsymbol{c(t) = \sum_{i=1}^{N} A_i \,.\, e^{j(2\pi f_{di} t + \theta_i)}} \qquad (3)$$

Where N is the number of scatterers, A_i is the amplitude, and f_{di} is the Doppler frequency shift of the i^{th} complex sinusoid.

Due to the correlation between channel samples, the future power level of the channel samples can be predicted using past and present channel samples. In the linear prediction model, the current sample is approximated by a linear combination of weighted past samples of the input signal [9]:

$$\boldsymbol{\grave{c}_n = \sum_{j=1}^{p} d_j \, c_{n-j}} \qquad (4)$$

Where $\grave{c}_n$ is the predicted value based on the linear combination of (p) previous values (c_{n-j}) multiplied by the prediction coefficients (d_j). And the error generated by this estimate is:

$$\boldsymbol{e_n = c_n - \grave{c}_n} \qquad (5)$$

Where c_n is the true channel value.

It is worthy to note that in order to Predict multiple future channel samples, the last predicted sample is just treated as

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an actual sample, in such a way that the channel state will be up to date at the time of receiving the acknowledgement information. Hence the effect of propagation delay of the feedback channel can be neglected.

The BER performance for different square constellation MQAM was simulated over Rayleigh fading channel as shown in Figure 2 through Figure 5. The results show that the channel prediction was much more suitable to track the channel variations at high Doppler frequencies than the frequency domain channel estimation at the same SNR.

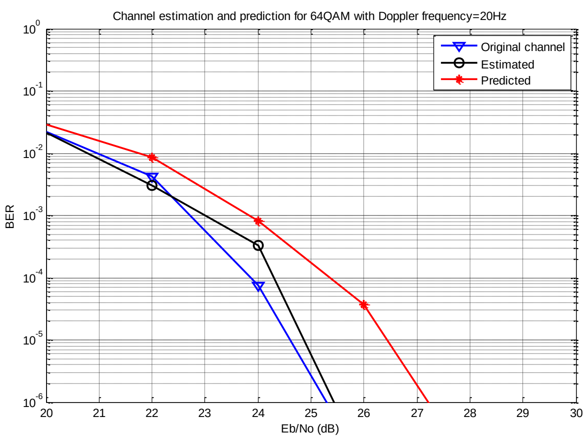

Figure 2. performance for 64QAM $f_d = 20$Hz

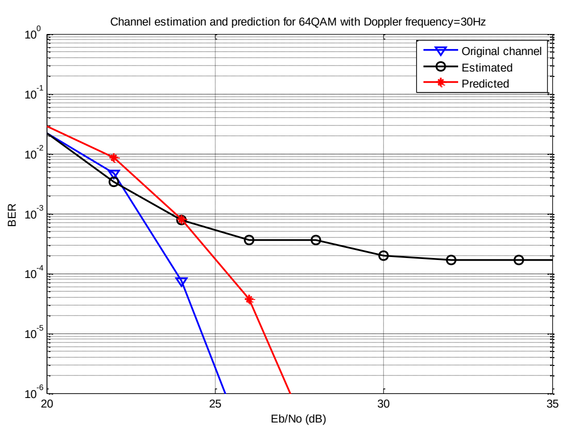

Figure 3. performance for 64QAM $f_d = 30$Hz

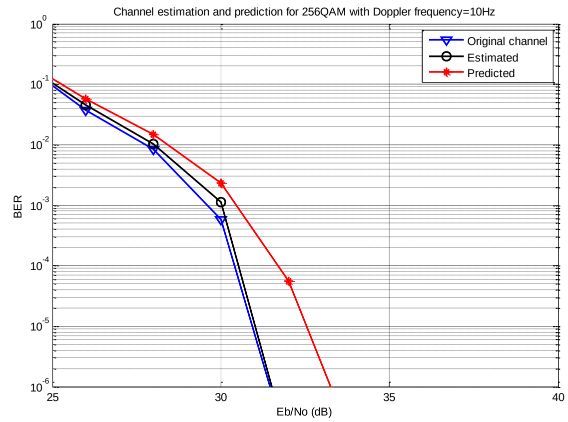

Figure 4. performance for 64QAM $f_d = 10$Hz

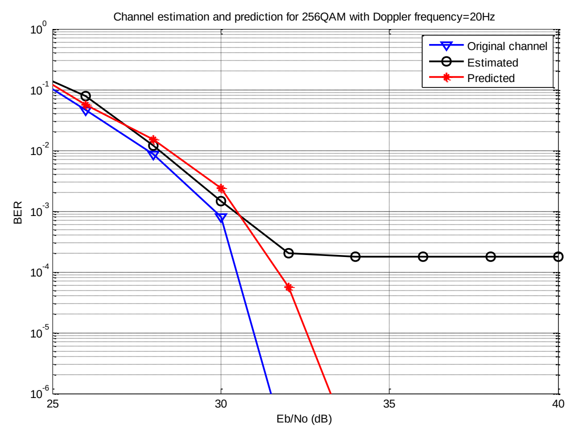

Figure 5. performance for 256QAM $f_d = 20$Hz

IV. Performance evaluation

The well known Shanon's upper bound theorem which relates the channel capacity to the allocated bandwidth and received Signal to Noise Ratio SNR is given by :

$$C = W\, log_2\left(1 + \frac{S}{N}\right) \qquad (6)$$

Where W the bandwidth of the channel, and $\frac{S}{N}$ is the received signal to noise ratio. In order to improve data throughput the restriction in both the received power and allocated bandwidth limits such improvement. The average number of correctly received data bits or throughput will be used as a performance evaluation metric with respect to frame length variation. The throughput will be reduced due to the lost of data packets, therefore retransmission will be started to overcome the packets being received in error,

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which result in waste of bandwidth CRC was used to detect packet error. For a point to point communication channel the throughput is given by:

$$T = \frac{N - N_{oh}}{N} \log_2 M (1 - P_{se})^{\frac{N}{\log_2 M}} \qquad (7)$$

Where N = $N_d + N_{oh}$; N_d is the data bits, N_{oh} is the overhead bits and M is the modulation constellation size. Principally increasing the packet size, the throughput will be improved and hence better utilization of bandwidth, at the expense of increased probability of packet being error. The optimum packet length to maximize the throughput can be found by differentiating the throughput with respect to the packet length and equating the result to zero $\frac{dT}{dN} = 0$; note that the packet is considered lost if any bit error occurs in the header or payload. Then with a certain probability of symbol error or equivalently certain signal to noise ratio there is an optimum packet size N_{opt} which results in an optimum throughput.

In spectrally efficient M-ary QAM there are a total of (M) possible states Since $M=2^m$, *m* bits per symbol can be sent. In Adaptive MQAM the transmission parameters are varied according to variation of the channel state. For every modulation mode, its error probability is directly related to the received SNR. The symbol error rate SER of MQAM is given as [1]:

$$P_{se,MQAM}(SNR) = \frac{2(\sqrt{M} - 1)}{\sqrt{M}} Q\left(\sqrt{\frac{3SNR}{M-1}}\right) \qquad (8)$$

In this paper the adaptation is performed along two dimensions, firstly the modulation type and corresponding constellation size will be chosen based on the received SNR where a maximum spectral efficiency can be obtained. Secondly the optimum packet length in accordance with the specific BER for the selected modulation is chosen. Table 1 shows the theoretical switching boundaries for optimum packet length in accordance with both SNR and P_{se}.

The following assumptions were assumed in the simulation, firstly the channel was assumed invariable over the packet duration time hence adaptation can be done over the whole packet. Secondly there is unlimited number of retransmission of lost packets, finally the feedback channel was assumed error free.

From equation (7) it is clear that the only parameter that affects the optimum packet length is the probability of error, hence once the BER or corresponding SNR from Equation (8) can be estimated, the optimum length can be selected in accordance to Table 1 with the suitable modulation constellation size.

Figure 6 shows the performance of individual fixed packet length which is arbitrary chosen to approximate the optimum length [500, 1500, 2000] and the overhead is also chosen to be 40 symbols. The performance shows that long packets are more sensitive to BER than short packets, such that the overall throughput is poor with high error rate and good throughput when BER is low.

Both the optimum and adaptive throughput was simulated as shown in Figure 7 where the optimum packet length corresponding to the specified BER was chosen. It is clearly shown that when the channel prediction is accurate then the adaptive frame length gives a very close throughput compared to optimality.

Figure 8 shows the performance throughput performance of optimum packet length with different square constellation MQAM signal. The channel symbol rate was considered fixed, then the performance throughput corresponding to the channel state as a function of SNR for both individual and adaptive modulation is as shown for each type of modulation.

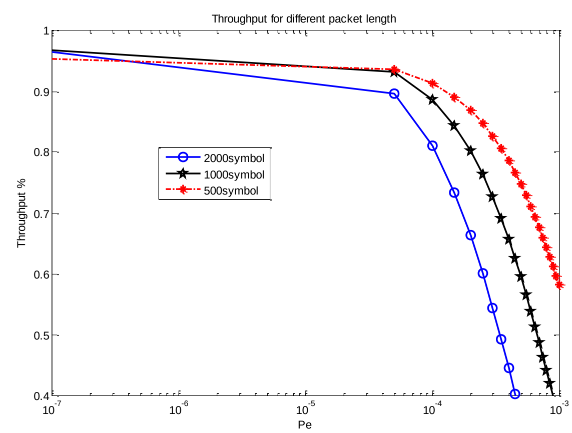

Figure 6. Throughput for different fixed packet lengths

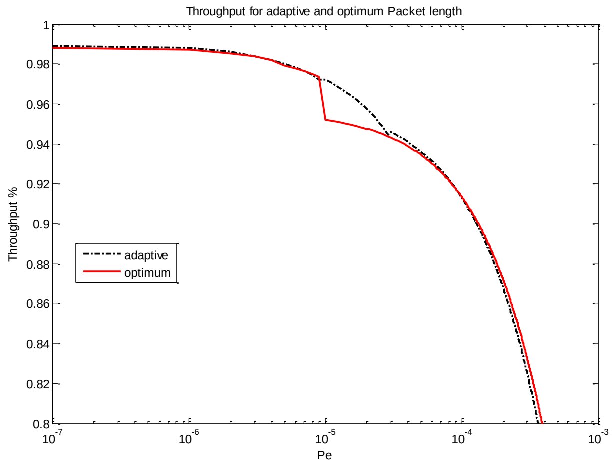

Figure 7. Throughput versus channel probability of error

V. Conclusions

A combined adaptive modulation and packet length scheme was proposed in this paper. The throughput is a function of BER which is specific for fixed modulation type, and the packet length efficiency which is a payload dependent. In this research paper the channel symbol rate was considered fixed hence the throughput at each specific channel state is dependent on the packet length and the constellation size at a given BER. The linear prediction of the future channel samples shows that at high Doppler frequency shifts or high channel variations the prediction outperforms the channel estimation besides that it overcomes the outdated feedback channel information. Results shows that significant throughput improvement can be achieved when adaptation in the selection of appropriate modulation and optimal packet length compared to fixed length.

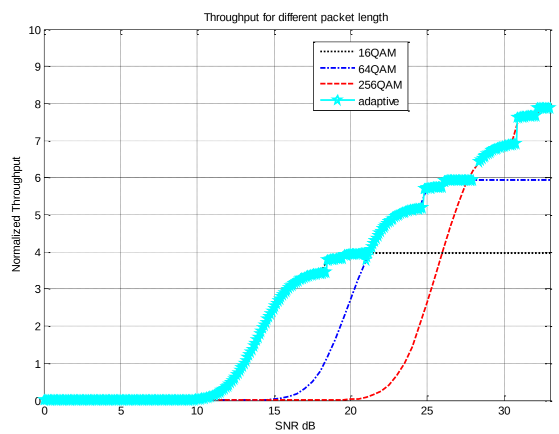

Figure 8. Tthroughput performance of optimum and adaptive system

Table 1 theoretical switching boundaries

$0 < SNR < 17.1\ dB$	No transmit	N_{opt}	p_{se}
$17.1 \leq SNR < 18.58\ dB$	16QAM	151.7	$10^{-4} \leq p_{se} < 10^{-3}$
$18.5 \leq SNR < 19.6\ dB$		457	$10^{-5} \leq p_{se} < 10^{-4}$
$19.6 \leq SNR < 20.6\ dB$		1424	$5 \times 10^{-6} \leq p_{se} < 10^{-5}$
$20.6 \leq SNR\ dB$		1835	$p_{se} \leq 6 \times 10^{-6}$
$23.3 \leq SNR < 24.8\ dB$	64QAM	151.7	$10^{-4} \leq p_{se} < 10^{-3}$
$24.8 \leq SNR < 26.1\ dB$		457	$10^{-5} \leq p_{se} < 10^{-4}$
$26.1 \leq SNR < 26.28\ dB$		1424	$5 \times 10^{-6} \leq p_{se} < 10^{-5}$
$26.28 \leq SNR\ dB$		1835	$p_{se} \leq 6 \times 10^{-6}$
$29.4 \leq SNR < 30.9\ dB$	256QAM	151.7	$10^{-4} \leq p_{se} < 10^{-3}$

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$30.9 \leq SNR < 32.16\, dB$		457	$10^{-5} \leq p_{se} < 10^{-4}$
$32.16 \leq SNR < 33.06\, dB$		1424	$5 \times 10^{-6} \leq p_{se} < 10^{-5}$
$33.06 \leq SNR\ dB$		1835	$p_{se} \leq 6 \times 10^{-6}$

References

[1] T. S. Rappaport, "Wireless Communications: Principles and Practice", 2nd edition, by Prentice Hall, 2002.

[2] P. Lettieri, M. B. Srivastara "Adaptive Frame Length Control for improving Wireless link Throughput, Range and Energy Efficiency" INFOCOM 98 Seventeenth Annual Joint conference of the IEEE computer & Communication Societies 1998.

[3] S. Ci, and H. Sharif " Improving goodput in IEEE 802.11 wireless LANs by using variable size and variable rate (VSVR) schemes" Wireless communications and Mobile computing 2004.

[4] A. Annamalia , V.K. Bhargava "Efficient ARQ Error Control Strategies With Adaptive Packet Length For Mobile Radio Networks" IEEE International conference on Universal personal communications 1998

[5] J. Korhonen, Y. Wang "Effect of Packet size on Loss Rate and Delay in Wireless Links" IEEE Wireless Communications and networking Conference 2005.

[6] Xin He, Frank Y. Li, Jiaru Lin "Link Adaptation with Combined Optimal Frame Size and Rate Selection in Error-Prone 802.11n Networks" Proceeding of the international symposium on wireless communication systems, IEEE ISWCS 2008.

[7] Michael N. Krishnan, Ehsan Haghani, and Avideh Zakhor " Packet Length Adaptation in WLANs With Hidden Nodes and Time-Varying Channels" proceedings of the Global Communication Conference GLOBECOM 2011 5-9 December USA

[8] M.Stojanovic "Optimization of a Data Link Protocol for an Underwater Acoustic Channel" Proceedings of Oceans-Europe 2005 Vol.1 pp68-73.

[9] Alexandra Duel-Hallen, Shengquan Hu, Hans Hallen, "Long Range Prediction of Fading Signals: Enabling Adaptive Transmission for Mobile Radio Channels" IEEE Signal Processing Magazine, May/2000.

Biographies

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