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Optimization of the size of packets used for the transmission in wireless ARQ protocols

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Abstract. ***The article analyzes the dynamic sizing of the Medium access control (MAC) layer frame, which is transported by radio. In addition, the tolerance between the reduction of the pulse power is presented, taking into account the need to reduce the percentage of transmission errors and the signal-to-noise ratio. It is shown that the optimal transmission rate depends on the operating conditions of the data transmission channel. Analytical analysis has shown the adaptive sizing of the MAC layer in the frame in the presence of varying noise in the channel, which has a significant impact on the user-reaching bandwidth of the data transmission channel. The use of adaptive frame length control to improve energy efficiency at the desired level of productivity and to expand the usable radio range by gradually reducing throughput is analyzed. The implementation of the adaptive MAC mechanism for frame length control in combination with adaptive hybrid Forward Error Correction and Automatic Repeat Request (FEC/ARQ) error control for a reconfigurable architecture for processing packet wireless layer packets for low transmission power is described. with adaptive wireless multimedia node.***

Keywords: ***Cross-layer design optimization, Adaptive transmission techniques, Performance schemes, Automatic repeat request***

I. Introduction

Wireless network connections are characterized by speed, variability in channel weather conditions, and energy constraints on wireless mobile consumer nodes. Therefore, static connection control techniques that make sense in relatively well-functioning cable connections do not necessarily apply to wireless connections. The new adaptive connection channel is a necessary control technique that ensures stability and energy efficiency [1].

These techniques work even in the presence of variations in the order of frequency errors in the bits and other conditions of the radio channel. For example, recent research supports adaptive link layer techniques such as adaptive channel state error control and variable signal propagation protocols [2]. Although wireless networks are growing in number and popularity, many of these new networks continue to use cable line techniques to maximize bandwidth.

While in practice wireless networks usually have a single transition, without cable infrastructure extensions, they represent not so much a separate operating point, but a new set of engineering constraints. For example, endpoints in a wireless network need to be energy efficient, as they are usually not tied to a constant source power, while the cable node can usually be powered by a wall outlet [3].

As another example, since wireless nodes are mobile, they must also be tied to the physical range over which they can transmit or receive data, as this value can be a constant stream and determines whether or not the user stays in touch with the rest of the network [4]. Therefore, maximizing range is important in the wireless, while not for the wired, fixed node. The problem with this parameter is simply dictated by the range of the connecting cable. For these and similar reasons, new techniques need to be considered for the wireless network.

The wireless node environment is generally highly variable, with potentially varying levels of bit error above orders of magnitude even within a single user session as they move or when environmental conditions change. Because this is actually extremely important to consider for a system that is adaptable to environmental conditions as opposed to one that is statically defined for an operating point. In contemporary research, the adaptation of the error control link of the packet planning layer with direct sequence and increasing the propagation of the radio frequency spectrum to the channel conditions has been studied. The Stop and Wait (S&W) protocol is the simplest variant of ARQ protocol. In this protocol, the transmitter sends a packet and waits for confirmation (ARQ). If the ARQ does not arrive within a predetermined period of time, called waiting or arriving for a negative acknowledgment, the packet is retransmitted [5].

When ARQ arrives, the transmitter switches to a new packet. The S&W protocol is very suitable for half-duplex operation, which is the mode that is usually supported by current and modem technology. However, it has low connection efficiency, where the propagation delay is longer than the packet size. The efficiency of the ARQ protocol is measured by the time spent waiting and can be improved if used to transmit new packets. This is the idea behind the continuous transmission of protocols - Back N and the selective iteration protocol. However, as acknowledgments arrive while new packets are being transmitted, these protocols require full duplex operation. Therefore, despite its low efficiency, the S&W protocol seems to be the appropriate method of choice for current conversions. There are available schemes to meet the half-duplex requirements. It can also increase the efficiency of the S&W. These protocols focus on transmitting block packets rather than a single packet, thus making better use of time spent waiting for confirmations.

The article aims to illustrate an analysis that visualizes that much available information can be obtained from the variable length of the frame in terms of bandwidth reaching the user, with an effective transmission range and transmitter power for the wireless connection. In addition to these results, an analysis will be presented that uses a variable frame length to provide improved service to the wireless user, with minimal additional costs and long-term interoperability with the existing Internet down to the IP layer. The report will outline the effective ARQ variables to improve the throughput of SW, GBN and SR ARQ. The optimized ARQ protocols will offer higher bandwidth than conventional ARQ protocols. The modems operating in half-duplex will be analized and their limitation to the choice of ARQ scheme to S&W class protocols. In addition, it will be illustrated that as bandwidth efficiency increases, the modified S&W protocols offer lower sensitivity in both packet size selection and error probability.

II. Description of the optimization algorithm

Given the bandwidth, which is called goodput, as a function of frame length when varying the percentage of household errors. Goodput refers to the bandwidth that the user actually receives after accounting for all losses, including those from MAC and PHY. Beyond these costs, however, good transmission will be reduced by the appearance of lost frames and errors [6].

Even a bit error in the scope of the framework will result in the loss of this framework, as the CRC will not pass. Each frame lost directly results in a loss of bandwidth, as well as the time spent sending that time until the frame makes proportional progress. This loss can also lead to additional ARQ signaling. To consider the behavior of the system, let us first point out the various quantities of interest. Let's denote: L - length of user data; H - length of the MAC header and I_P=40 bytes; F - length of P_{HY} = 52.5 bytes (assuming n = 0); M_{TU} - L+20 bytes; R_C - radio transmission speed = 2 Mbps; BER_C - probability of error in the bits of the channel; G_{WL} - goodput.

With these values, the normalized goodput can be indicated as:

$$\frac{G_{WL}}{R_C} = \frac{L}{L+H+F}(1-BER_C)^{L+H} = \frac{1}{1+\frac{H+F}{L}}(1-BER_C)^{L+H} \quad (1)$$

The above equation gives the value of the normalized performance relative to the raw data rate in terms of frame length and BER. By making PHY and MAC overhead constant, we can change the L size of user data. Bit errors are a function of the path loss and transmitter power that occur after a white Gaussian distribution.

The result of the solution of equation (1) for different bit error rates can be visualized as the M_{TU} differing from L only by a constant. This strengthens the IP sizing relationship. The main conclusion to be drawn from it is that with the deterioration of the channel, it would be useful to use a lower M_{TU} instead of the maximum allowed M_{TU} of 1500. Properly chosen M_{TU} can even "revive" obviously non-functional connection which gives a zero result.

Another indicator that needs to be taken into account, as the length of the frame is different, is the transmission of the range. There are at least two ways to do this. First, good distance-to-distance performance can be considered for selected M_{TU} values. To do this, a coherent QPSK must be taken to correct the bit error rate as another indicator to consider, as the length of the frame is different from the range interval. Second, good performance with respect to packet transmission distance can be taken into account at selected M_{TU} values. To do this, it should be noted that for a coherent QPSK we can denote the degree of bit errors as:

$$BER_C = \frac{1}{4} erfc(\sqrt{\frac{E_b}{N_0}}) \tag{2}$$

If a model of loss of free space path with a path of the degree of loss equal to two is further adopted, this is simplified to:

$$BER_C = \frac{1}{4} erfc(\frac{1}{d}) \tag{3}$$

where d is the division of the transmitter-receiver into some appropriately normalized distance units. With this relationship between distance, BER and substitution in the equation. (1), It can be find out goodput as a function of distance at a given MTU.

The communication link introduces a propagation delay with Td = l/c, where l is the distance between the transmitter and receiver, and c=1500 m/s is the nominal speed of sound. Thus, the total time required to transmit the m packet and receive the corresponding group is:

$$T(m) = m(T_p + T_{ack}) + T_w \tag{4}$$

ARQ bandwidth efficiency is defined as the ratio of packet useful time to total time. If p denotes the probability of error in the packet, the average time required to transmit a packet using S&W-1 is given by:

$$T_1 = \frac{1}{1-p} T \tag{5}$$

The S&W-2 protocol can be considered as MS&W-1 protocols running in parallel. Each S&W-1 has an equal wait time up to T(M), and the packet error rate is still equal to p. Therefore, the average time required to successfully transmit a packet on one of the M connections is T(M)/(1-p). Because the M connections work in parallel, a total of M packets are transmitted successfully during this time. The obtained throughput is:

$$\delta_2 = (1-p)\frac{MN_dT}{T(M)} \tag{6}$$

The S&W-3 protocol begins with the transmission of a group of M packets. The waiting time is set to allow a two-way trip T(M).

At the end of the timeout, one of the following situations will occur: no packet was received successfully and the transmitter remains in the “M” state with M packets to be transmitted; one packet is received and the transmitter states in "M-1" and M-1 transmission packets left. Now the waiting time is set to T(m) and upon receipt of the confirmation the recipient goes to the state "m - k", if there are k of the m packets will be positively recognized. The probability of this event is m-k, p^m−k(1−p)k. If we denote by Tm the average time spent in the state "m" then it must have the following relation:

$$T_m = \sum_{k=0}^{m} \binom{m}{k} p^{m-k}(1-p)^k \left[T_{m-k} + T(m)\right] \tag{7}$$

This connection can be used to find the required average time TM for successful transmission of a group of M packets. The setting of the initial value T_0=0, recursive evaluation of the above expression gives:

$$T_M = \frac{1}{1-p^M}\left[\sum_{m=1}^{M-1} \binom{M}{m} p^{M-m}(1-p)^m T_{M-m} + T(M)\right] \tag{8}$$

The bandwidth efficiency depends on the normalized waiting time T_w/N_dT and the packet error with probability p. The probability of a packet error is given in conditions of probability of error in bit (symbol) Pe such as:

$$p = 1-(1-P_e)^N \approx NP_e \tag{9}$$

where the approximation is valid for $P_e \ll 1$ and it is assumed that the bit errors occur independently. By increasing the packet size, better utilization of latency is achieved, but the chances of a bit error in the packet increase.

Therefore, there is an optimal packet size for which the throughput and efficiency are maximum. While the bit error rate is determined by the channel conditions and the modulation/detection method used for the physical layer, the packet size can be varied to maximize efficiency.

The optimal package size can be estimated in closed form for S&W-1 and S&W-2 (for S&W-3 can be found numerically). For this purpose, it is enough to focus on δ_2 (δ_1 is Its special case). The efficiency δ_2 is expressed by the packet size N_d for given parameters of the physical layer P_e, R, and the bond distance l, such as:

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$$\delta_2 = (1 - P_e)^{N_d + N_{oh}} \frac{N_d}{N_d + \varepsilon} \tag{10}$$

The treatment of the packet size as a continuous variable is optimal value is obtained as a solution of $d\delta_2/dN_d=0$, given by:

$$N_{d,opt} = \frac{\mu}{2}\left[\sqrt{1 + \frac{4}{\mu\rho}} - 1\right] \tag{11}$$

with the approximation valid for Pe<<1. This packet size achieves maximum throughput efficiency:

$$\delta_{2,max} = (1 - P_e)^{N_{d,opt} + N_{oh}} \frac{N_{d,opt}}{N_{d,opt} + \mu} \tag{12}$$

Note that the efficiency δ_2 increases with the size of the group M. The practical limit of M will be determined by system constraints such as storage capacity. However, it is interesting to note that the upper limit of bandwidth efficiency is:

$$\lim_{M\to\infty} \delta_{2,max} \approx 1 - \sqrt{N_{oh}P_e} \tag{13}$$

for $N_{oh}P_e$<<1. Therefore, by increasing M, the problem of long propagation delay can be overcome and efficiency remains limited by the quality of the connection, ie. from BER.

III. Algorithm implementation and discussion

Three factors that can potentially increase bandwidth in the AWGN channel must be considered. These factors are: MRC combination; Packet size adaptation; Complex ARQ protocols such as GBN and SR.

A method for quantifying the improvement in throughput has been established, taking into account each of these factors, and including their combinations. First it is necessary to adjust the simulation equipment. It is necessary to simulate an AWGN channel from point A to point as the spectral density of the noise power is σ2. A binary phase or modulation scheme with Shift Keying (BPSK) is considered. We assume that the MAC layer packet contains 240 redundant bits, ie. h=240 bits, as in IEEE 802.11. The window size for GBN and SR ARQ is assumed to be 8 packets and 3 bits used to represent a sequence number [7]. The effect of combining MRC is achievable, as it is quantifiable how much bandwidth is obtained by stand-alone MRC combining. For this it is necessary to simulate all three ARQ protocols with and without MRC combination for different payload sizes. The values of equation 1 vary from 10 bits to 22x10^3 bits and BER=10^{-3}. Same results are shown in [8].

It should be taken into account that for all values of the packet sizes the system with MRC combination and significant increase of the throughput compared to the system without MRC is provided for where T_{MRC}(a, b) and T(a, b) are the maximum throughputs. BER values b for an ARQ protocol, not under systems with and without MRC combination, respectively.

Bandwidth is maximized for all packet sizes and the maximum value is obtained by performing simulations for different packet sizes and then selected at the maximum observed bandwidth value. Note that bandwidth and margin increase sharply for BER greater than 10^{-3}. Thus, MRC combination is much more efficient in a low SNR region, and packet size adaptation is the main reason for bandwidth and BER improvements of less than 10-3 in systems with and without MRC combination. It is first necessary to correct the packet size to 2347 bytes (maximum size for IEEE 802.11 system) to obtain the bandwidth for this packet size as a function of BER in systems with and without MRC.

The throughput values obtained above T_MR_C(b) and T(b) obtained for different BER b are then compared. It is necessary to take into account that the system with packet adaptation has a significantly higher throughput than that in the system with a fixed packet size. The simulation results for the other fixed packet sizes also give similar results. Here, using simulations, the effect of ARQ protocols on bandwidth of systems with and without MRC combination can be quantified.

It is possible to achieve the improved throughput of SR and GBN ARQ over that of SW ARQ with and without MRC combination for different payload sizes. Here we note that in the system without MRC, combining the increase in bandwidth of SR and GBN ARQ over that of SW ARQ decreases rapidly and the packet size increases. However, with the combination of MRC, the stock bandwidth of SR and GBN ARQ compared to that of SW ARQ is significant for the whole range of considered packet sizes.

This bandwidth enhancement provides a strong argument for replacing SW, ARQ with SR or GBN and ARQ. For specific results, the packet length is chosen to maximize system throughput efficiency.

The bandwidth obtained by using the packet size calculated by the analysis is close to that of the maximum bandwidth achieved in the system (ARQ protocol is SR) Thus, the analytical approach we proposed works well in practice. We assume that the MAC layer packet contains 240 header bits and a variable payload from 100 bits to 18,000 bits. We believe the Rayleigh attenuation channel. The analysis must first be done for the SW protocol. It assumes that the confirmation is received from the sender after a delay of 8 packets. After application of the method, the improvement in its bandwidth is shown.

It should be borne in mind that the throughput with MRC is higher than the throughput with optimal packet size below SNR=29 dB, while the throughput with optimal packet size is higher than the throughput with MRC above SNR from 29 dB. bandwidth with optimal packet length. SR ARQ provides more bandwidth than GBN and SW ARQ. SR ARQ throughput can be improved with optimal packet size along with combining MRC. Here, as in GBN, there are two MRC-SR crossover points with an optimal packet circuit at SNRs of 22 dB and 29 dB.

Between these points, the bandwidth of the MRC circuit is higher than that of the optimal packet length. It should be borne in mind that the problem with measuring the BER of the canal is complicated by ensuring that noise losses (eg from increased road loss) will be measured, as opposed to measuring losses due to multiple fading.

The frame length sizing described here is intended to combat the first, but will offer little help in dealing with the last channel, unless the channel fades relatively short compared to the smallest available frame sizes, or at least not much shorter. -longer than the shorter dimensions of the frame. When the attenuation is too long compared to the time required to transmit with larger packet sizes, this technique would be ineffective. The information sent by the radio channel can be useful in improving the quality of the channel estimation. By applying the analysis, such feedback on the state of the channel to the connection layer can be provided.

The power of the transmitter is increased after the packet has been fragmented based on the current best estimates of the state of the channel that is fed to the corresponding flow line according to the source that produced it. For example, a wireless node may produce compressed speech, compressed video, and FTP data simultaneously, each with different latency and bandwidth requirements, and therefore with different error control and coding needs. In this way, each is placed in its own line from where it will be fed to the respective FEC block and then to a transmission line from where it can be sent to the receiver. It should be noted that the issues of packet length and error control coding are intertwined, as the amount and type of coding required will depend on such factors as packet sizing, in fact on its own sizing, and the adapter for frame length and FEC adapter.

Therefore, the transmitter must work together to optimize the overall structure of the packet. Although there is a cost for reduced flexibility to change the channel conditions, once the packet of the IP layer is fragmented and protected from errors, it will remain unchanged when transmitting the line. Similar results are presented in [2].

If the channel conditions change, the new outgoing packets will be fragmented accordingly, but the packets existing in the line transmission, whether waiting to be transmitted for the first time or waiting to be retransmitted by some ARQ scheme, are not refracted or recoded, in interest of reduced complexity.

The transmitter packet scheduler is responsible for determining when a fragment should be sent to the receiver. This is done based on the ACK received from the counterparty at the other end of the relationship, as well as knowledge of the various constraints associated with each flow. The scheduler therefore allows packets from the corresponding line by controlling a packet.

The packets are demultiplexed according to the information in the transmitter and fed to the corresponding FEC block. The results of FEC decoding are used, as mentioned above, to assess the conditions of the channel, as well as to determine whether a particular fragment should be recognized or not. Finally, again according to the information in the transmitter, when the complete IP packet has arrived, its fragments are reassembled and fed back into the network layer. Such an analysis would provide control of the MAC layer to be implemented in the packet scheduler. With the transmitter acting as a "bitpipe", this method takes care of everything under the IP and can therefore do so in a way that is suitable for a wireless connection. It thus enables adaptive sizing of MAC frames, as well as error control and coding in a way that is transparent to networking and a higher layer of software, and yet provides significantly improved wireless network performance above what would be accessible through simple cable network extensions.

IV. Conclusion

This article provides an analysis that shows that much available information can be obtained from the variable length of the frame in terms of bandwidth reaching the user, with an effective transmission range and transmitter power for the wireless connection. While for specific systems using WaveLAN transmitters in Linux machines, the results are general, as they would be reserved for other transmitters. In addition to these results, an analysis is presented that uses a variable length framework to provide improved service to the wireless user, with minimal additional costs and long-term interoperability with the existing Internet down to the IP layer. In this article, we have proposed the effective ARQ variables to improve the throughput of SW, GBN and SR-ARQ.

The investigated ARQ protocols offer higher bandwidth than conventional ARQ protocols. A new method for packaging combined size optimization (which is a network layer technique) with MRC applied to each bit of the package has been proposed. This is observed for modified ARQ protocols, the improvement in throughput is significant at a higher bit error rate, where throughput decreases due to transmission errors.

Because modems typically operate in half-duplex, the choice of ARQ scheme is limited to S&W class protocols. High latency of the channel makes the basic S&W protocol extremely inefficient and of limited usefulness for systems that transmit at low transmission speeds and very short distances. For the multi-channel search system of interest in communication projects, this standard protocol is not a good choice. Instead, S&W schemes based on the transmission of grouped packets for which there are selective acknowledgments and the ones generated by them should be used. The bandwidth efficiency of these protocols can be maximized by selecting the optimal packet size as a function of the connection parameters and group size (M).

In addition to increasing bandwidth efficiency, the modified S&W protocols offer lower sensitivity when choosing the package size of both the product for the range and the probability of error.

In order to make full use of the limited resources of the channel, future designs of the systems that use them must focus on the application of an adaptive ARQ scheme.

Two aspects can be considered when making: Adaptive adjustment of the waiting time according to the measured instantaneous travel time; Adaptive adjustment of the packet size according to the measured instantaneous probabilities of error and delay of the connection.

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