Dynamic Redundancy Bit Allocation And Packet Size To Increase Throughput In Noisy Real Time Video Wireless Transmission

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Abstract

The typical real time wireless video-audio digital transmission process consists of capturing the signal, digitizing it, compressing it, adding cryptography to it (crypto it), adding redundancy to enable the receiver to detect and correct a number of bit errors, packetizing it and then transmitting it. Transmitting the signal via the Transmission Control Protocol TCP-IP provides a fixed number of redundancy bits, and very rigid transmission process that could result in a large number of automatic repeat requests and denial of services. In this research we develop a dynamic transmission algorithm, where by, the degree of redundancy is a function of the noise and the probability p for a bit to be corrupted. We also provide a variable number of protection depending on the importance of certain bits. In addition we provide a variable packet size depending on the noise, in order to decrease the probability of automatic repeat request. The preferred protocol to be used with our algorithm is the User Datagram Protocol (UDP) fortified with our dynamic redundancy check algorithm, a packet sequence number, number of redundancy bits, signal group size as part of the packet header. Our algorithm has two parts. The first one is noise detection and noise quantization. The second part is redundancy bit adjustment and packet size adjustment to maximize the transmission throughput. In this paper we present the analytics of keeping the correctable groups of bits in each transmission until the whole packet is received.

Key Words; Bit Redundancy. Forward Error Correction, Digital

Transmission, UDP, Wireless Transmission, Noisy Channels.

1 Introduction

Wireless channels suffer from noise interference. Therefore in wireless digital transmission to assure signal integrity it is necessary to encode the bits and enable the receiver to detect and correct errors using Forward Error Correction (FEC), where we use the code at the receiver site to correct errors, and Automatic Repeat request (ARQ) in cases where the received packet has more errors than the FEC correction algorithm can correct, in this case the receiver sends a message to the transmitter to retransmit the packet. The traditional ARQ protocols are: Selective Repeat (SR), Stop and Wait (SW), an GO-Back-N (GBN). For channels with error-rates relatively small SR is very efficient, although require relatively large buffers to store all the packets to be retransmitted. A number of modifications of the SR algorithm have been proposed by many researchers [1-7]. Our algorithm is an extension of this research. The size of the packet in our case as well as the degree of redundancy is a function of the importance of each group of encoded bits and the channel noise as is estimated by the probability of a transmitted bit to be in error. The typical real time wireless digital transmission process consists of capturing the signal, digitizing it, compressing it, adding cryptography to it (crypto it), adding redundancy to enable the receiver to detect and correct a number of bit errors, packetizing it and then transmitting it. Transmitting the signal via the Transmission Control Protocol TCP-IP provides a fixed number of redundancy bits, and very rigid transmission process that could result in a large number of automatic repeat requests and denial of services. In this research we develop a dynamic transmission algorithm, where by, the degree of redundancy is a function of the noise and the probability p for a bit to be corrupted. We also provide a variable number of protection depending on the importance of certain bits. In addition we provide a variable packet size depending on the noise, in order to decrease the probability of automatic repeat request. The preferred protocol to be used with our algorithm is the User Datagram Protocol (UDP) fortified with our dynamic redundancy check algorithm, a packet sequence number, number of redundancy bits, signal group size as part of the packet header. Our algorithm has two parts. The first one is noise detection and noise quantization. The second part is redundancy bit adjustment and packet size adjustment to maximize the transmission throughput. Here we present the analytics related to the number of trials needed until the packet is received by the receiver. In our first approach we assume that the probability for an arbitrary bit to get corrupted during the transmission is p has been estimated , and we estimate the optimal number of (n,k), where n is bit size of a group of bits and k is the redundancy needed to detect e errors and correct t of them. Subsequently we compute the optimal packet size in order to increase the probability of the computer receiving the packet to be able to correct all errors in the packet and to not issue an automatic repeat request. So our method increases the expected number of packets that can be corrected. Another measure we are concerned about is consistency which is a function of the variance associated with packet loss. This paper is organized as follows: Abstract, introduction, background information, analysis, conclusions, and references.

2 Background Information

Wireless communications is very widely used. From the infrared wireless transmission used as a remote control to change the television channels, to television, radio, wireless telephony, wireless networks, robot communication, satellite communication, and so many other applications, military and civilian. To accommodate all these digital wireless communication application there is a large number of modulation methods and wireless communication protocols. The IEEE 802.11 wireless transmission protocol includes several standards. Some of these standards are 802.11n for WLAN access point, 802.11a 802.11b to 802.11g WiFi standards. Cellular networks in the past 10 years have grown in a very fast pace. With the introduction of the iPad, iPhone, iPod touch, android, Kindle, social networking, the demand for wireless networks has grown significantly. At the present time there are over 6 million base stations (BSs) serving mobile users. The energy consumption for a BS is on the average 25MWh per year. The energy cost and carbon footprint for operating these BS stations is rising. The European Commission has recently started such new projects as "Energy Aware Radio and NeTwork TecHnologies" (EARTH) , "Towards Real Energy -efficient Network Design" (TREND). And "Cognitive Radio and Cooperative strategies in multi-standard wireless devices" (C2POWER). Civil aeronautical communication uses analogue systems at the present time, using a relatively large bandwidth for low quality noisy audio. A new digital protocol based on IPV6 is on the way that will increase the quality and decrease the bandwidth. A newly formed working group by the International Civil Aviation Organization (ICAO) has defined recently a standard defining the IP-based Aeronautical Telecommunication Network (ATN/IP). This communication standard will be used for ground to ground, ground to air and airborne network. The IPv6 will be used as the basis of this digital communication, because is a mature protocol and is actively maintained by the Internet Engineering Task Force (IETF). This new system will provide cost effective, high quality, low bandwidth communications.

3 Analysis

Let p_b be the probability that an arbitrary transmitted bit is corrupted. Then if N are the number of bits in the packet, if in is a code-vector of n-bits that we can correct k errors, and if $m = \frac{N}{n}$ is the number of code-vectors in the packet, then the theorem bellow gives the probability that an arbitrary code-vector of n-bits either will be received with no errors, or when received, will have k or fewer errors and therefore all the errors will be corrected.

Theorem 1. If the probability for an arbitrary bit received by the receiver to be corrupted is p_b then the probability that a code-vector, of n-bits with enough redundancy to correct k-bit errors, will have no error, or be able to correct all the errors in the n-bits is:

$$p_g = P(Y = 1) = (1 - p_b)^{n-k} \sum_{i=0}^{k} \binom{n}{i} (1 - p_b)^{k-i} p_b^i \qquad (1)$$

Where Y is a random variable with the value 1 if the code-vector of n-bits is correctable and 0 if it is not. If $m = \frac{N}{n}$ is the number of code-vectors of n-bits transmitted in the packet then the expected number of code-vectors of n-bits successfully received is mp_g and the variance is: $mp_g(1 - p_g)$.

Proof Let Y be a random variable having the value 1, if the received code-vector of n-bits is correctable and 0 if it is not. Then

$$p(Y = 1) = (1 - p_b)^n + \binom{n}{1} (1 - p_b)^{(n-1)} p_b + \ldots + \binom{n}{k} (1 - p_b)^{(n-k)} p_b^k \qquad (2)$$

Or if we factor out the $(1 - p_b)^{(n-k)}$ then equation (2) becomes:

$$p(Y = 1) = (1 - p_b)^{(n-k)} [(1 - p_b)^k + \binom{n}{1} (1 - p_b)^{(k-1)} p_b + \ldots + \binom{n}{k} p_b^k] \qquad (3)$$

Or

$$P(Y = 1) = (1 - p_b)^{(n-k)} \sum_{i=0}^{k} \binom{n}{i} (1 - p_b)^{(k-i)} p_b^i \qquad (4)$$

If $p_g = P(Y = 1)$, denotes the probability for a code-vector of n-bits with the redundancy to correct k-bits, to be received successfully. Then the probability that x-code-vectors out of the m-code-vectors in the packet, of n-bits for each code-vector, will be correctable has the binomial distribution, and:

$$P(X=x) = \binom{m}{x}(1-p_g)^{(m-x)}p_g^x \tag{5}$$

Thus the expected number of received n-bit code-vectors that are correctable is:

$$\mu_g = E(X) = mp_g \tag{6}$$

and the variance is:

$$\sigma_g^2 = mp_g(1-p_g) \tag{7}$$

Theorem 2.

Let the packet size be N-bits, a code-vector of n-bits, with t-bits redundancy that enables us to correct k-errors. Let m=N/n, and let Y be a random variable denoting the numbers of times needed to transmit, so that all the m-code-vectors in the packet are received by the receiver. Then the probability of Y to be r is:

$$\begin{aligned} P(Y=r) = [p_g(1-p_g)^{r-1}]^m \textstyle\sum_{k_1}\sum_{k_2}\sum_{k_{r-1}} \binom{m}{k_1}\binom{m-k_1}{k_2} \\ \binom{m-k_1-\ldots-k_{r-2}}{k_{r-1}}(1-p_g)^{-[(r-1)k_1+(r-2)k_2+\ldots+k_{r-1}]} \end{aligned} \tag{8}$$

Where k_1, code-vectors of n-bits are correct or being able to be corrected in the first trial, k_{r-1} in the r-1 trial, and $0 \leq k_1+k_2+\ldots+k_{r-1} \leq (m-1)$ are the n-bit code-vectors transmitted successfully during the first r-1 trials. The remaining n-bit code-vectors are transmitted in the rth transmission.

Proof Let r be the number of transmissions needed in order to transmit all the n-bit-code-vectors in the packet. If k_1 are the n-bit code-vectors transmitted in the first time the packet was transmitted, k_2 are the n-bit-code-vectors transmitted in the second time and k_{r-1} the the number of n-bit-code vectors transmitted successfully the (r-1) time, where $0 \leq k_1+k_2+\ldots+k_{r-1} \leq (m-1)$ then in the rth transmission $m-(k_1+k_2+\ldots+k_{r-1})$ n-bit-code-vectors are transmitted successfully and the whole packet is successfully transmitted. Now let m=N/n, and let Y be a random variable denoting the number of times needed to transmit a packet with m-code-vectors so that all the m-code-vectors in the packet are received by the receiver. Then the probability for transmitting k_1 n-bit-code-vectors in the first transmission k_2 in the second transmission k_{r-1} in the (r-1) transmission, and $m-(k_1+k_2+...+k_{r-1})$ in the rth transmission, is:

$$\begin{pmatrix} m \\ k_1 \end{pmatrix} p_g^{k_1}(1-p_g)^{(m-k_1)} * \begin{pmatrix} m-k_1 \\ k_2 \end{pmatrix} p_g^{k_2}(1-p_g)^{(m-k_1-k_2)} * ... \begin{pmatrix} m-k_1-...k_{r-2} \\ k_{r-1} \end{pmatrix} p_g^{k_{r-1}}(1-p_g)^{(m-k_1-...-k_{r-1})} * p_g^{(m-k_1-...-k_{r-1})} \quad (9)$$

The probability of the packet being received at the rth trial is:

$$P(Y=r) = p_g^m(1-p_g)^{(m(r-1))} \sum_{k_1} \sum_{k_2} \cdots \sum_{k_{r-1}} \begin{pmatrix} m \\ k_1 \end{pmatrix} \begin{pmatrix} m-k_1 \\ k_2 \end{pmatrix} \begin{pmatrix} m-k_1-...k_{r-2} \\ k_{r-1} \end{pmatrix} (1-p_g)^{-((r-1)k_1+(r-2)k_2+...+k_{r-1})} \quad (10)$$

where $0 \leq k_1+k_2+...+k_{r-1} \leq (m-1)$

The expected number of bits to be transmitted in the first transmission is:

$$\mu_g = E(X) = mp_g \quad (11)$$

Definition: In a code-vector of n-bits if b-bits are part of the message and n-b bits are the redundant then we define the b-bits as the message-load-bits in the code-vector.

The message load in a packet with m n-bit-code-vectors is m*b If we increase the number of bits to be corrected By 1 then the message-load-bits in the group of n is decreasing to d ¡ b and the number of n-groups Is increasing from m= N/n to $s = \lceil \frac{Nb}{nr} \rceil$ if this change increases the probability of transmitting the packet in less than or equal to r-transmissions then increasing the number of bits to be corrected is a better strategy. Thus there is an optimal code-vector size and message load depending on the noise during transmission.

4 Conclusion

Wireless Communications is subject to noise. Unlike analogue wireless transmission that has limited ways of dealing with noise, digital transmission uses forward error correction and automatic repeat request to provide pristine signal quality. In this research paper we provide a closed form approach to deal with noisy environments. Transmitting the signal via the Transmission Control Protocol TCP-IP provides a fixed number of redundancy bits, and very rigid transmission process that could result in a large number of automatic repeat requests and denial of services. In this research we develop a dynamic transmission algorithm, where by, the degree of redundancy is a function of the noise and the probability p for a bit to be corrupted. We also provide a variable number of protection depending on the importance of certain bits. In addition we provide a variable packet size depending on the noise, in order to decrease the probability of automatic repeat request. The preferred protocol to be used with our algorithm is the User Datagram Protocol (UDP) fortified with our dynamic redundancy check algorithm, a packet sequence number, number of redundancy bits, signal group size as part of the packet header. Our algorithm has two parts. The first one is noise detection and noise quantization. The second part is redundancy bit adjustment and packet size adjustment to maximize the transmission throughput.

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