

Optimization of Gain Control Loops for Packet Based Data Transmissions

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Abstract

In the packet based systems for supporting burst data traffic networks, the fast synchronization is required in the short period of time upon receiving the data frame at the receiver. This important operation is generally carried out at the header part of the packet which is usually called the preamble. While detecting the existence of the valid signal frame through the carrier sensing it is also important to adjust the signal level at the input of the baseband blocks. This is generally called automatic gain control and it is essential to minimize the processing time for the gain decision for incoming received signals. In this paper, target level of the AGC is determined based on the bit-error-rate (BER) performance and SQNR for the varying power of the AGC input signal. The other parameters are optimized by utilizing statistics of the incoming signal including the degree of saturation as well as the incoming signal power. The proposed optimization method provides the improved convergence speed which in turn results in the reduced processing time.

Keywords: Gain Control Loop, ADC, Signal to Quantization Noise Ratio, OFDM

1. Introduction

As the mobile internet access is becoming a part of daily lives, smart mobile devices capable of handling high data rate traffic are now an essential part of the wireless networks. In general the high data rate system requires broadband in frequency domain which is vulnerable to frequency selective fading effects which result from multipath propagation of the radio signal due to various types of geographical obstacles in the radio transmission path. The orthogonal frequency division multiplexing (OFDM) is suitable for highly multipath channel which results in severe inter-symbol interference in time domain. In packet based WiFi system which is one of the advanced transmission standard based on OFDM, automatic gain control (AGC), synchronization and channel estimation are processed based on the preamble located in the beginning of the data packet [1]. Since the time duration is limited available for frontend operations, it is essential to minimize the processing time in the AGC which is the first block in the receiver. The effect of asynchronization which could be originated from the poor signal quality due to non-optimized AGC level in the preamble interval may cause the loss of orthogonality between subcarriers as well as inter-symbol interference which results in performance degradation [2-3].

Types of AGC are categorized into the continuous and burst mode. The continuous mode can achieve superior performance using the feedback loop which can be continuously updated for the relatively longer period of time. This scheme is more suitable for slowly changing

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amplitude variations with the continuous signal stream. The burst mode scheme controls the gain using the particular portion of the corresponding frame. This method is required for the packet based data traffic. In this mode if the AGC does not have an enough response speed, it is difficult to achieve a desired performance since it affects the various parameter estimations in the process of data retrieval. Therefore, the fast processing speed of the AGC is essential to achieve the mission within the limited time frame. The goal of AGC is to maintain the input power at the desired level to analog-to-digital convertors (ADC) while minimizing the quantization noise level at the output of ADC. In a previous research work, an AGC circuit with two different update loops is proposed for the MIMO-OFDM system [4]. In this previous work the saturation level over the ADC dynamic range is decided based on four samples. If the input signal level is relatively small, processing speed can be reduced with this method. However, if the input level is large, relatively longer processing time is required with this method.

In this paper, a modified AGC scheme and a method for the optimization of the related parameters are proposed. In this proposed method more gain loops are utilized and more observation samples are utilized for determination of the saturation statistics for faster convergence. The remainder of this paper is organized as follows. In the second section, the system model is discussed. The next section discusses the proposed AGC scheme and the optimization method is described. Performances of the proposed scheme are discussed next. Conclusions are drawn at the end.

2. System model

This section describes the system model where the proposed AGC scheme is embedded. As described in Figure 1, the IEEE 802.11a/g, one of the most popular OFDM systems is assumed for the transmission system model although the proposed scheme can be utilized in any systems which require the fast processing and convergence time of the AGC. At the transmitter the user data is first scrambled to reduce the unwanted frequency component [5]. The scrambled user data is now coded with convolutional encoder to protect the valuable data contents from the noise and the various types of interferences [6]. This encoder is often called error control code where the redundant information is added to recover the corrupted data at the receiver end. At this channel encoding stage the output of the encoder generates more data to accommodate redundancies. In the WiFi system, the basis code rate is one half. Although the robustness is increased by channel encoder, the data throughput is decreased in proportion to the code rate. In an effort to compensate the code rate loss in the throughput, the puncturing block is added [7]. In our system, rates of two thirds and three fourth of codes are implemented through two types of stolen bit pattern. The adoption of puncturing is quite effective to increase the data rate while keeping the other system parameter constant while sacrificing the error rate performance at the receiver side. To further increase the resistance to the bursty and consecutive errors the block-wise interleaver is utilized after convolutional encoder [8]. The interleaver combats errors that might occur due to the deep frequency selective fading which resides over the wide range of frequency band. This can be achieved by writing the data in the direction of the width of the interleaver block while by reading the data in the direction of depth of the block or vice versa. This operation can effectively spread the consecutive errors into the wide range of the frequency band where the frequency diversity can be achieved as well [9]. The interleaved data bits are grouped to form the symbols at the mapping stage. The mapper identifies the number of bits per symbols depending on the control signals from the upper layer (MAC: Medium Access Layer) and the bits of predefined number are sub-grouped to be a symbol for the subcarrier which eventually

becomes part of the OFDM symbol. The number of bits per symbol for each subcarrier is determined by modulation scheme selected. In IEEE 802.11a/g system, BPSK, QPSK, 16QAM and 64QAM are supported. The different modulation for subcarrier can provide the flexibility in terms of adaptability to the channel environments. This means that various types of users associated with the same access point (AP) can be served with appropriate data rate depending on the channel link quality and the distance between the AP and the user. Upon the completion of the mapping stage symbols are assigned to subcarriers in the conceptual frequency domain. The training sequences are also mapped to the predefined subcarriers. In our system there are two types of training sequences: short preamble and long preamble. The name "preamble" implies that this sequence located before the information contained data sequence. The mapping rules for subcarriers of short preamble, long preamble, Signal field and Data field are all defined in frequency domain. They define the phase and the amplitude of the corresponding subcarriers. Theoretically, sinusoidal waves with the carrier frequency corresponding to each subcarrier which are orthogonal to each other are required to transmit all the assigned symbols. However, this operation can be achieved without separate carrier waves by utilizing inverse discrete Fourier transform. Various types of fast processing algorithms have been developed to improve the computational efficiency of the discrete Fourier transform which is called fast Fourier transform (FFT) [10]. The inverse FFT (IFFT) block transforms the discrete frequency domain information into the discrete time domain. In this process an OFDM symbol is generated for each predefined number of user data bits which are determined by the data transmission mode. For each OFDM symbol excluding short preamble cyclic prefix (CP) is added to protect the symbol from the inter-symbol interference (ISI) [11]. The insertion of CP is necessary to maintain the performance of the OFDM under the harsh channel environments while the CP acts as an overhead to the actual data transmission. The trade-off between the performance and the deterioration of the throughput should be carefully taken into account at the system design stage. Although details are not shown in the block diagram, up conversion to the defined channel, transmit filter to satisfy the frequency band regulatory requirements and/or possibly pre-distortion to reduce the peak to average power ratio (PAPR) are needed before the radiation via the transmit antenna [12].

As shown in the same figure the AGC block is located at the beginning part of the receiver end of the baseband operation. The radio frequency signal which might be centered around 2.0GHz or 5.0GHz band is radiated from the transmit antenna. As discussed in the IEEE 802.11a/g standard short and long preamble part can potentially provide valuable information for the carrier sense, diversity selection [13], carrier frequency offset estimation [14], time synchronization [15] and channel impulse response estimation [16]. The quality of the estimated parameters obtained during the preamble period affects the performance of the entire system. The AGC operation should be done well before the frontend blocks begin. The receiver frontend carries out the parameter estimations which motivate the fast processing and convergence speed of the AGC. The object of the AGC is relatively simple which can be summarized as to maintain the pre-determined signal power level at the input of the ADC by adjusting the gain of the variable gain amplifier (VGA) located in the receiver circuit chain. The signal level should be determined such that the signal swings within the dynamic range of the ADC to minimize the quantization or clipping errors. The proper operation of AGC is essential to provide a basis for the improvement of the level of estimation accuracy.

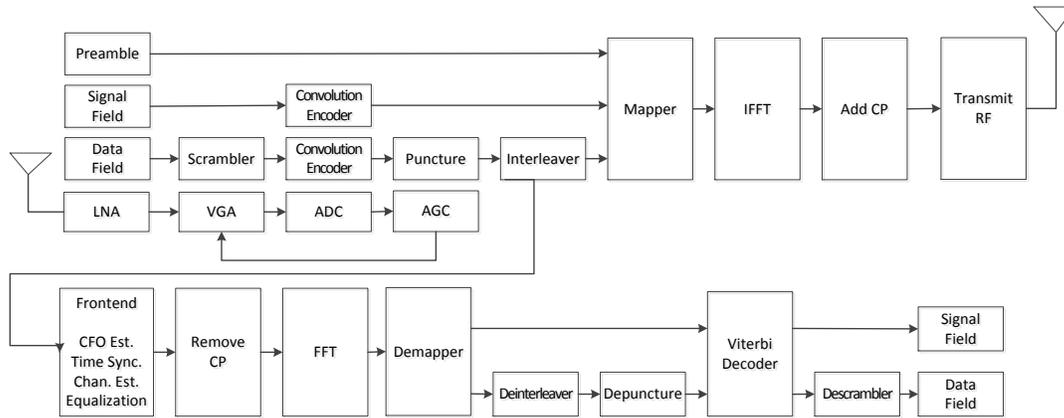


Figure 1. Block diagram of the IEEE 802.11a/g based OFDM transmission system

The remaining sub-blocks of the receiver carry out the exactly inverse function of the corresponding transmitter sub-blocks. The descrambler re-orders the scrambled data to obtain the user data transmitted. The Viterbi decoder extracts the scrambled data encoded with the convolutional channel encoder which implements the maximum likelihood sequence detection in an efficient manner [17, 18]. As the block name indicated de-interleaver also re-order the data written in the interleaver block. In the depuncture block the stolen bits at the transmitter are filled with dummy bits which are expected to be recovered in the decoding process. Removing of CP and FFT operation are performed after the frontend functions are finished. The multi-path channel which is not overflowed out of the CP range does not affect the information included in the OFDM symbol. The FFT translates the discrete time sequence into the frequency domain where the mapped symbols can be recovered. Once the valid subcarriers are identified the phase and the amplitude are estimated to recover the associated data bits whose process is called demapping. There are two types of demapping method. One of them is hard-decision type and the other soft-decision [19]. In the hard-decision scheme the bits (0 or 1 in binary data case) are decided at the output of the demapper whether they are reliable or not. However in the soft-decision case the likelihood number is assigned at the output instead of definite number. The likelihood information is utilized in the Viterbi which can further improve the detection performance [20].

3. Proposed optimization method

If AGC does not have a sufficient response rate or the operation takes long time before completion, the following signal in the same frame might be saturated or not provide enough precision for the remaining demodulation operation. This can be one of the reasons for failure of time synchronization. Even though it barely succeeds the synchronization, the poor precision of the received sample could act as a major source of BER performance degradations. It is because the estimation of the impulse response and the state information of channel could be inaccurate even though synchronization succeeds.

The first step in the AGC parameter setting is the determination of the target or steady state level. In the description of the proposed scheme a 10-bit ADC is assumed without a loss of generality. The histogram shown in Figure 2 is obtained by using the inverse ratio of the original signal to the error signal for all samples index in one packet defined in [1]. For the input signal level of 0dB the majority of the input signal causes relatively large

quantization errors which are caused by the small input signal levels compared with the ADC dynamic range. On the other hand for the input level of 60dB the samples are divided by two groups. The group in the relatively lower error region can be explained by improved signal precision due to the increased input signal power level. The other group centered in the higher error range is due to the clipping samples exceeding dynamic range limits. This phenomenon indicates that input signals with too small or large level results in high quantization noise. The remaining histogram with input level of 30dB which is centered at around -40dB shows relatively good overall ADC quality in terms of signal to quantization noise ratio (SQNR).

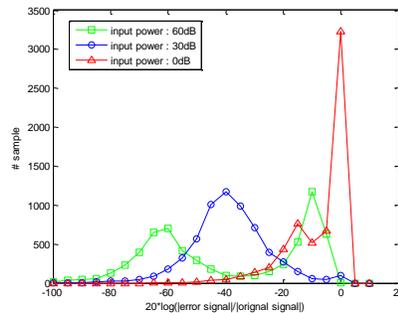


Fig. 2. Histogram of quantization noise to signal ratio for various input signal level

Another SQNR experiments are carried out to determine the optimal target signal level. The graph shown in Figure 3 describes the SQNR for the varying values of the input signal level. The BPSK modulation is assumed for the data field part of the OFDM symbol. As it can be seen in the figure the SQNR value is proportional to the signal level until the signal power reaches about 46dB. At that point we can achieve about 64dB of SQNR which can be translated into ADC quality observed at the output of the device. However the SQNR value begins to decrease dramatically after the peak point. The reason for the steep degradation of SQNR is that the saturation level reaches to a threshold which in turn degrades the signal quality abruptly. This experiment indicates that saturation counts as well as the SQNR should be considered for the determination of the AGC level.

The bit-error-rate (BER) performance as well as SQNR can also provide good baselines to find the optimum operating point of the AGC. There might be a trial to set a peak point shown in Figure 3 as the AGC target level. Although this is a good candidate in terms of SQNR performance this point is too sensitive to the minor change of the signal level. The BER simulation results are shown in Figure 4 where we can observe the signal level range for the best BER performances. The range is almost the same for the all the modulation scheme we consider in the simulation. The rightmost edge for this range is located around the signal level of 49dB which is consistent with the results obtained through the SQNR observation. Also observed is the steep degradation in BER performance at the small increased value of the input signal level. This indicates that the peak point in SQNR may not be stable target although it still provides the good BER performance. In our case input power levels between 18dB and 45dB shows the best BER performance although the range is narrowed down as the constellation size of the modulation increases. Considering all the experiment results the middle value, around 29dB is chosen as the final steady state operating point of the AGC.

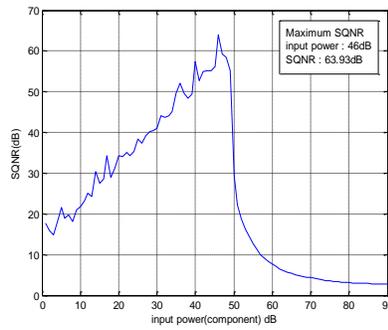


Figure 3. SQNR for the varying input signal levels (BPSK modulation assumed)

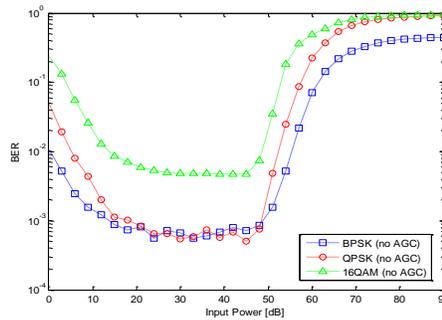


Figure 4. BER performance for varying input signal levels

Another important statistics other than input signal level which can be utilized in the proposed scheme is number of saturated samples. The graph shown in Figure 5 represents the number of samples exceeding the dynamic range of the ADC within a pre-defined period of time. Also indicated are boundaries of the region (R1, R2 or R3) where the different AGC gain is assigned depending on the saturation counts. The detailed assignment of the gain control values are summarized in Table 1 where our target is assumed to be around 29dB as discussed above. As it can be expected larger values are assigned for more saturation counts. In the proposed scheme 16 samples are being monitored for saturation statistics which is equivalent to one period of short preamble. In our example the gain is set -28dB for saturation count of 1~13, -46dB for 14~15 and -70dB for 16.

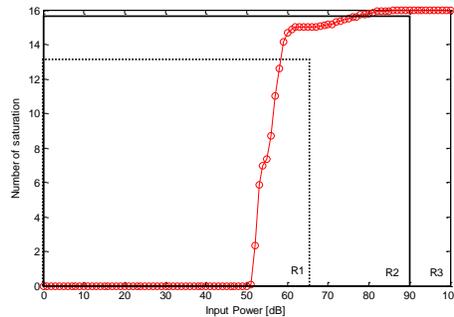


Figure 5. BER performance for varying input signal levels

When the received signal exceeds the saturation threshold, the precision level of the measured power is decreased. This means that the gain control level should be determined by saturation count rather than relatively inaccurate measured input power when the saturation is monitored. This process is repeated until quantized signal does not exceed the saturation threshold. As a final step when the saturation does not occur anymore, control gain value is set based on the difference between current signal level and the target level which is kept constant until the end of the corresponding OFDM frame at the receiver.

3. Simulation results

For the AGC performance simulations several cases of control gain sets are compared which are summarized in Table 1. For each case three different control gains are preset which are associated with the corresponding saturation count interval. The first simulation results are shown in Figure 6 which summarizes the required number of AGC loops to reach the final steady state signal levels which are related with the convergence speed. The results represent the average values based on the independent transmission of 100 packets at around BER of 10^{-3} . It can be shown that case 3 outperforms the other cases in terms of convergence speed although more analysis should be carried out for the final optimization. More important aspect of the AGC performance can be drawn from the overall BER simulation.

Table 1. Control gain sets for AGC performance simulations

set of control gain		case 1	case 2	case 3	optimized
saturation count (input power [dB])	1-13 (48~68)	-10 dB	-30 dB	-80 dB	-28 dB
	14-15 (66~89)	-20 dB	-60 dB	-100 dB	-46 dB
	16 (90~)	-30 dB	-90 dB	-150 dB	-70 dB

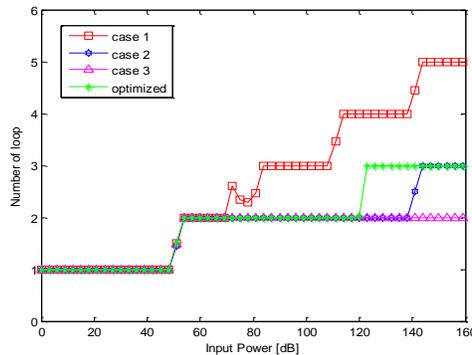


Figure 6. Required number of AGC loops for varying input signal levels

In Figure 7 BER simulation results are shown for the parameter sets described in Table 1. Here OFDM symbols are modulated with BPSK scheme which is equivalent to the data rate of 6Mbps. The uncoded BER is measured at the output of the demapper. The E_b/N_0 is assumed to be set to 6dB for the simulation where BER of 10^{-3} can be achieved with an ideal AGC. Although the required loop count is smaller for case 3 than case 1 or 2, the duration of the input power interval where the AGC is not working is quite wider than the others. Case 1

shows better BER performance than case 3 by sacrificing the loop count or equivalently convergence performance. It can be also noted that the set of control gain can be optimized to balance the convergence and BER performance. In the optimized set case the performance is maintained while trying to minimize the AGC loop counts for the input power of up to 160dB.

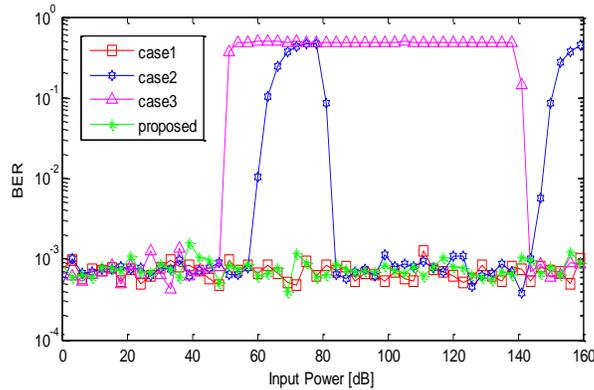


Figure 7. BER performance for various sets of control gain

In Figure 8 the comparison of the BER simulation results are shown to confirm the operation of the AGC optimized by the proposed method. The E_b/N_0 is set to 6dB for all modulation schemes and the uncoded BER is assumed for the simulation. Results show that systems with optimized AGC shows unchanged performance for varying signal levels while the non-optimized cases shows dependency of the performance on the signal levels.

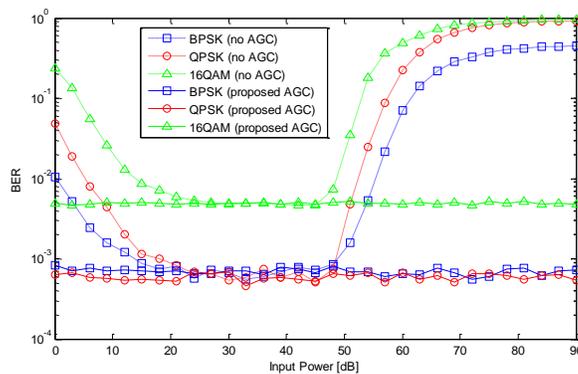


Figure 8. Comparison of BER performance before and after the proposed scheme is applied

4. Conclusion

If the AGC does not have an enough response speed in burst mode, it is difficult to achieve the desired performance since it affects the accuracy of the estimation on various parameters at the demodulation stage. Therefore the minimized processing time for gain decision is essential in the AGC block. In this paper, an optimization method for the high speed AGC to control the gain of the received signal in the OFDM system. The modified structure of the AGC is also discussed to improve the accuracy of the input power level and to expedite the

convergence. In the proposed method statistical properties of the ADC output signal is utilized to determine the set of control gain. To verify the performance of the optimized AGC, the BER performances are tested for all ranges of input signal power. The proposed AGC scheme shows optimized performance in terms of BER performance and the convergence speed.

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