Analysis of BER Improvement of Adaptive Equalizer using CE and STB code

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ABSTRACT

In a wireless communication channel, when the modulation bandwidth is exceeding the coherence bandwidth ISI takes place as the transmitted pulses are spread into the adjacent symbols. To combat the effects of ISI and noise and to reconstruct the signal and minimize Bit Error Rate (BER), the adaptive channel equalizer is used at the receiver end. Various kinds of interferences in a communication channel exists like Inter symbol Interference, Multipath Interference and Additive Interference which deals with the design of an Adaptive Equalizer. The idea of the equalizer is to build filter in the receiver that counter acts the effect of the channel. In essence, the equalizer must “unscatter” the impulse response and can be stated as the goal of designing the equalizer E so that the impulse response of the combined channel and equalizer CE has a single spike. We investigate channel equalization and introduce hierarchical and adaptive nonlinear channel equalization algorithms that are highly efficient and provide significantly enhanced bit error rate (BER) performance. Due to the high complexity of nonlinear equalizers and poor performance of linear ones, to equalize highly difficult channels, we employ piecewise linear equalizers.

1. INTRODUCTION

Adaptive equalizers are capable of correcting for ISI due to multipath in the same way as ISI from linear distortion in telephone channels. The rapidly increasing need for computer communications has been met primarily by higher speed data transmission over the widespread network of voice-bandwidth channels developed for voice communications. A modulator-demodulator (modem) is required to carry digital signals over these analog pass-band channels by translating binary data to
voice-frequency signals and back. The thrust toward common carrier digital transmission facilities has also resulted in application of modem technology to line-of-sight terrestrial radio and satellite transmission. Analog channels deliver corrupted and transformed versions of their input waveforms. Corruption of the waveform-usually statistical-may be additive and/or multiplicative, because of possible background thermal noise, impulse noise, and fades. Transformations performed by the channel are frequency translation, nonlinear or harmonic distortion, and time dispersion.

1.1 Adaptive FIR Equalization

Wireless channels vary over time and so we can’t simply have a static impulse or frequency response for the equalizer. Rather the equalizer must adapt to the ever changing channel called Adaptive Equalization. This adaptation to the changing channel is achieved by training the equalizer based on the frequency response of the channel and tracking the frequency response of the channel as it varies in time. Obviously it becomes increasingly difficult to track when the channel is changing rapidly. A general form of the adaptive filter is illustrated in Figure 1.1.

A general transfer filter's architecture is illustrated in Figure 1.2. Broadband wireless systems for future mobile multimedia applications need to support high data rates in the range above tens or even hundreds of megabits per second. Radio transmission techniques for these systems have to cope with frequency selective and time-‐varying radio channels. In particular, a multipath propagation scenario such as broadband wireless access gives a large delay spread.

Figure 1.1 Adaptive algorithm units

Figure 1.2 Architecture of transfer filter

\[ w(n) = [w_0, w_1, w_2... w_l] \] is the vector of the transfer filter's coefficients having the ; \( d(n) \) is the desired output of the transfer filter; \( y(n) \) is the output of the transfer filter; \( e(n) \) is the error value, and it can be written as:
The Adaptive algorithm unit represents some algorithm to update the coefficients of the transfer filter. For LMS algorithm:

\[ W(n+1) = w(n) + \mu * x(n) * e(n) \]

\( \mu \) is the step of LMS algorithm. For the LMS algorithm \( \mu \) is a constant. We also can have variable \( \mu \). For this comparison in this paper we choose constant \( \mu \) LMS algorithm.

### 1.2 Space Time Block Code Technique

Equalization technique is a type of partial-response signaling that can help to reduce the required maximum frequency because it allows for a controlled amount of ISI to be removed afterward. In space time block code scheme, channel loss and Tx/Rx equalization combine to produce ISI, which may be expressed as the z function \( 1 + z^{-1} \) [1]. Tx/Rx equalization results in binary input data being output as a space time block signal according to the function \( 1 + z^{-1} \) i.e. \( 1 + (1) = 2, 1 + (0) \) and \( 0 + (1) = 1, 0 + (0) = 0 \), where the value inside each parenthesis denotes a preceding data bit. Although space time block is not Nyquist, but a partial-response signaling, we have defined the Nyquist frequency of space time block signaling to compare its characteristics with other methods. The Nyquist frequency value was determined by the sampling frequency at which you can recover original waveforms of optimum space time block single bit response with 1.5Tsymbol transition time.

### 2. LITERATURE REVIEW

[4] The performance evaluation of the optical access network is an important factor and is measured in terms of Bit Error Rate (BER), packet delay, packet loss probability, interference etc. Scheduling algorithm, DBA algorithm, network topology and crosstalk interference are the various issues that affects the performance of an optical network.

[6] In this paper, we have focused to adjust the maximum number of retransmission packets using a wireless network technology and developed algorithm for energy efficient and reliable WSN system.

[10] In this paper, the ML algorithm can be directly utilized to estimate the target location since the probability density function of the observation data is known with our assumption. Finally, the extensive simulation studies have demonstrated the effectiveness of our proposed algorithm.

[12] Image denoising is the basic problem in digital image processing. Removing Noise from the image is the main task to denoise the image. Salt & pepper (Impulse) noise and the additive white Gaussian noise and blurredness are the types of noise that occur during transmission and capturing. To remove these types of noise we have many filters like mean filter, median filter, inverse filter, wiener filter. No single one filter can remove both types of noise. So I design a hybrid filter which can be used to denoise these both types of noises from the image.

[16] Wireless communication technology is increasing daily; with such growth sooner or later it would not be practical or simply physically possible to have a
fixed architecture for this kind of network. With the progression of computer networks extending boundaries and joining distant locations, wireless sensor networks (WSNs) emerge as the new frontier in developing opportunities to collect and process data from remote locations.

3. RESULT ANALYSIS
The objective of the purposed work is to compare linear adaptive FIR equalization and space time block code scheme, for that measurement of BER at the input and also at output of each equalization scheme.

3.1 Space Time Block Code Scheme

<table>
<thead>
<tr>
<th>No. OF SAMPLES (N)</th>
<th>PLANT</th>
<th>SNR</th>
<th>MU</th>
<th>ORDER OF EQUALIZER (N)</th>
</tr>
</thead>
<tbody>
<tr>
<td>2000</td>
<td>Ch</td>
<td>20</td>
<td>0.07</td>
<td>9</td>
</tr>
</tbody>
</table>

Fig. 3.1 BER and SNR

\[ eq = -0.0168 \ -0.0749 \ 0.2988 \ -0.7336 \ 1.5681 \ -0.6889 \ 0.3302 \ -0.1864 \ 0.0684 \]

3.2 Linear Adaptive Equalization Scheme

\( n = 4000; \% \text{BER} \), Plant \( = \text{ch} \); SNRmin = 20; SNRmax = 100.
4. CONCLUSION

They can easily be designed to be linear phase. Put simply, linear-phase filters delay the input signal, but don’t distort its phase. They are simple to implement. On
most DSP microprocessors, the FIR calculation can be done by looping a single instruction. They are suited to multi-rate applications. By multi-rate, we mean either “decimation” or "interpolation" (increasing the sampling rate), or both. Whether decimating or interpolating, the use of FIR filters allows some of the calculations to be omitted, thus providing an important computational efficiency. In contrast, if IIR filters are used, each output must be individually calculated, even if it that output will discard. They have desirable numeric properties.

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