Equalization of IEEE 802.11b Signal

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ABSTRACT
The purpose of this work is the design and implementation of an IEEE 802.11b signal in Labview. This work deals with the study of the various kinds of interferences in a communication channel like Inter Symbol Interference (ISI), and Additive White Gaussian noise(AWGN) and counteracts their effects at the receiver.

Keywords
ISI, AWGN, Equalizer, LMS, EVM, DSSS

1. INTRODUCTION
Equalization is a process in which active or passive electronic elements are used for fulfilling the purpose of altering the frequency response characteristics for any system. It is the process of adjusting the balance between frequency components within an electronic signal. Equalization compensates for the differences in signal attenuation and delay associated with different frequency components. There are various methods of signal equalization**[1].**Equalized signal can be analysed at the output using EVM **[2].**The received signal in this case is IEEE standardized by IEEE 802.11b standards **[3].**The type of modulation used is CCK **[4].** The design involves finding the channel response H(f) of the medium through which signal travelled. The received signal is afterwards passed through an adaptive filter or an inverse filter (1/H(f)) which nullifies the effect of the channel. H(f) can be estimated at the receiver by using methods like training sequence, pilot carrier etc. The main problems for inaccurate measurement of the signal by the equalizer are Inter symbol Interference (ISI) **[5]-[11]** and Adaptive white Gaussian noise **[12]** (AWGN).

2. INTER SYMBOL INTERFERENCE (ISI)
In telecommunication, inter-symbol interference (ISI) is a form of distortion of a signal in which one symbol interferes with another symbols. This is an unwanted phenomenon as the previous symbols have similar effect as noise, making the process less efficient.Mathematically the ISI can be represented by,

Let us consider the transmission of a sequence of symbols with the basic Waveform $U(t)$. To send the nth symbol b, we send $\mathrm{b} * \mathrm{u}(\mathrm{t}-\mathrm{nT})$,where T is the interval and transmitted signal given by

$$x(n) = \sum_{n=0}^{N} (\mathrm{b} * \mathrm{u}(\mathrm{t}-\mathrm{nT})$$

Based on the dispersive channel model, the received signal can be represented by,

$$r(n) = \sum_{n=0}^{N} (\mathrm{b} * \mathrm{u}(\mathrm{t}-\mathrm{nT}) + n(n)$$

Where $\mathrm{r(t)} = \mathrm{x(t)} * \mathrm{h(t)}$ is the received waveform for a symbol. If a single symbol, say the symbol b_0, is transmitted, the optimal demodulator is the one that employs the matched filter, i.e., we can pass the received signal through the matched filter $\mathrm{v(t)} = \mathrm{v(-t)}$ and then sample the matched filter output at time t=0 to obtain the decision statistic. When a sequence of symbols is transmitted, the matched filter is used to perform demodulation. One way is to sample the matched filter output at time t = mT to obtain the decision statistic for the symbol b. At t = mT, the output of the matched filter is

$$Z(m) = \sum_{n=0}^{N} \mathrm{b} * \mathrm{v} * \mathrm{v}(\mathrm{mT}-\mathrm{nT}) + \mathrm{n(t)}$$

Where n(t) is a zero-mean Gaussian random variable. The first term is the desired signal contribution due to the symbol b and the second term contains contributions from the other symbols. These extra symbols are called intersymbol interference (ISI). The causes of ISI are multipath propagation and band limited channels.

2.1 Multipath Propagation
One of the causes of inter-symbol interference is what is known as multipath propagation in which a wireless signal from a transmitter reaches the receiver via many different paths. The causes of this include reflection , refraction (such as through the foliage of a tree) and atmospheric effects such as atmospheric ducting and ionosphere reflection. Since all of these paths are different lengths - plus some of these effects will also slow the signal down - this results in the different versions of the signal arriving at different times. This delay results in the spreading of a symbol to all the other subsequent symbols, interfering with the detection of these symbols .Multipath propagation often distorts the amplitude or phase of the signal causing interference with the received signal.

2.2 Band Limited Channel
Another cause of intersymbol interference is the transmission of a signal through a band limited channel, i.e., one where the frequency response is zero above a certain frequency (the cutoff frequency). Passing the signal through a band limited channel results in the removal of frequency .The amplitude of the frequency components below the cutoff frequency may also be attenuated .

The ISI is caused at the transmitter due to the presence of pulse shaping filters at the transmitter end. Three types of pulse shaping filters have been used

Root cosine, Root raised cosine and Gaussian.

3. DESIGNING

3.1 Channel Estimation

The small-scale variations of a mobile radio signal can be directly related to the impulse response of the mobile radio channel. The impulse response can be considered as a wideband channel characterization and can be used to represent all information necessary to simulate or analyze any type of radio transmission through the channel. The filtering nature of the channel is caused by the summation of amplitudes and delays of the multiple arriving waves. The impulse response is used to predict and compare the performance of many different mobile communication systems and transmission bandwidths for a particular mobile channel. The channel impulse response relates the transmitted and received signals as

$$Y(n) = X(n) * h(n)$$

From, the above mentioned equation the channel impulse response can be calculated as

$$h(n) = y(n) * x(-n)/p(n)$$

Where y(n) is the received signal, x(-n) is the transmitted signal and p(n) is a constant.

For estimating the channel impulse response,
A computationally efficient method is used **[5],**in accordance with the IEEE 802.11b WLAN standards that utilizes DSSS.

This method comprises of the auto-correlation and cross-correlation of the 11 chip barker sequence with the received signal.In WLAN the type of modulation used is CCK modulation [**4**]. In CCK modulation ,the sampling rate is fixed at M times the bandwidth of baseband signal..The channel estimation is done by using the technique of training bits. In this technique first the received signal is demodulated using the barker sequence .The demodulated signal is then used to find the strongest signal. The strongest signal is spreaded using the barker sequence and samples is then correlated with the received samples. The correlated output is divided by a constant which is, the autocorrelation of the barker sequence. A separate unit for obtaining the symbol information may be used and the symbol information can be fed to the channel estimator. But in this implementation, we have merged the unit for obtaining the symbol information into the channel estimator. This would provide us the flexibility of performing channel estimation anywhere in the preamble without bothering much about synchronizing the unit for obtaining the symbol information and the channel estimator.

RESULTS

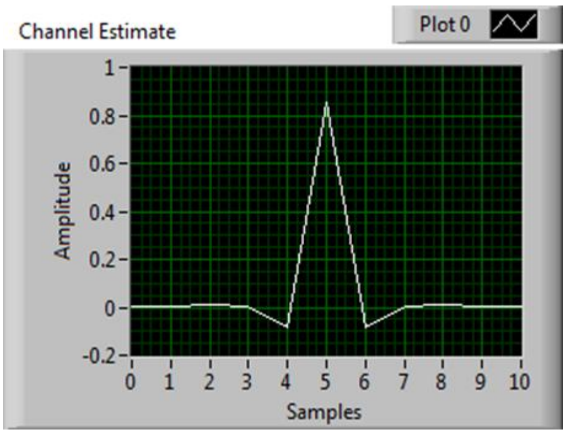


FIG 1.Channel Estimate

3.2 Equalizer Design

Two types of equalizers are designed for the purpose of equalization

1. Adaptive equalizer

2. Linear equalizer

3.3 Adaptive Equalizer

An adaptive equalizer is an equalization filter that automatically adapts to time-varying properties of the communication channel. It is implemented for performing tap-weight adjustments periodically or continually. In general, for the case of first call or reset adaptive equalizer uses a training sequence but for this case as we have the channel estimate in Fig 1. so we will be using these co-efficient as the training sequence. Periodic adjustments are accomplished by periodically transmitting a preamble or short training sequence of digital data known by the receiver. Continual adjustment are accomplished by replacing the known training sequence with a sequence of data symbols estimated from the equalizer output and treated as known symbols. When the adaptive filter functions continuously and automatically for a period of time it is called decision directed. Adaptive filters depend on recursive algorithms to update their coefficients and train them to near the optimum solution. In this case The LMS algorithm is used for continuously updating the filter coefficients as done in [14]-[17]. The equation is given by

$$C(n+1) = C(n) + U * e(n) * x(n)$$

First the received signals is compensated for its gain. This is done by comparing its power level with the ideal case and compensating for the power loss. As CCK modulation have been used, so the symbol rate is 8 samples per symbol. The symbol rate is changed to sample rate by passing the signal through a multi-rate filter. The oversampling is removed before passing the signal through the equalizer. The result of the adaptive filter in terms of its co-efficient is shown below .

RESULTS

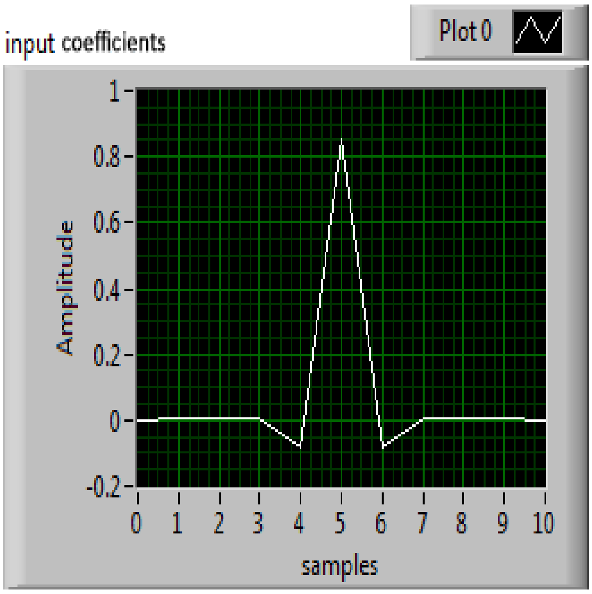


FIG 2. INPUT COEFFICIENTS

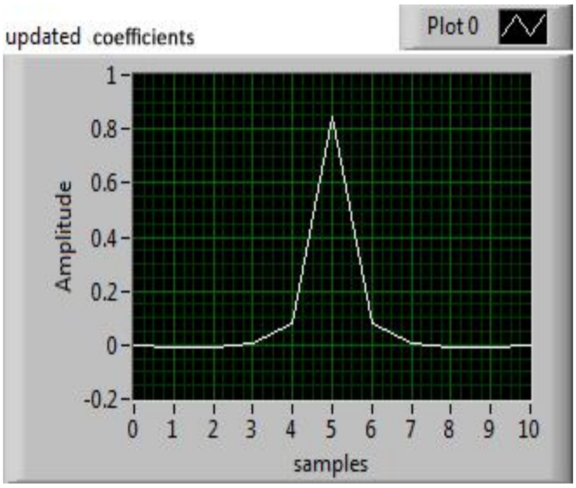
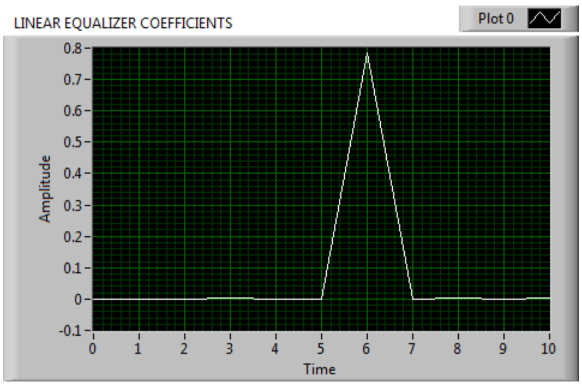


FIG 3.OUTPUT COEFFICIENTS

3.4 Linear Equalizer

A linear equalizer works on the concept that, if the transfer function of the equalizer is made the inverse of the channel, then Inter Symbol Interference (ISI) can be completely eliminated at the receiver. Linear Equalizers are easy to implement and are highly eff ective where ithe ISI is not severe (like the wire line telephone channel). The major issue in designing is the problem of stability. The equalizer co-efficient are not stable as the region of convergence(ROC) is beyond the unit circle as some of the poles are beyond the unit circle.. This process of filter stability check be done in number of ways [18]-[22] . To stabilize the filter the whole process of taking Z-transform and Inverse Z-transform is taken on an unit circle. So, that the resultant poles are inside the unit circle and the filter is both causal and stable.

RESULTS



4. RESULTS AND ANALYSIS

The output results can be measured by the help of the error vector magnitude or EVM (sometimes also called receive constellation error or RCE).It is a measure of the performance of a digital transmitter or receiver. The transmitted signal is represented by constellation points. Various imperfections in the implementation like carrier leakage, phase noise cause the actual constellation points to deviate from the ideal locations. EVM can be defined as a measure of how far the points are from the ideal locations **[23]-[25].**

Transmitter EVM can be measured by specialized equipment, which demodulates the received signal in a similar way to how a real radio demodulator does it. One of the stages in a typical phase-shift keying demodulation process produces a stream of I-Q points which can be used as a reasonably reliable estimate for the ideal transmitted signal in EVM calculation.

An error vector is a vector in the I-Q plane between the ideal constellation point and the point received by the receiver. In other words, it is the difference between actual received symbols and ideal symbols.

The average power of the error vector, normalized to signal power, is the EVM. For the percentage format, root mean square (RMS) average is used. The EVM can be measured mathematically as

$$EVM = \sum_{n=0}^{N} X(n) - X^{\wedge}(n)$$

The error vector magnitude is equal to the ratio of the power of the error vector to the root mean square (RMS) power of the reference. It is defined in dB as:

$$EVM(dB)=10\log_{10}(P_{error}/P_{reference})$$

Where Perror is the RMS power of the error vector. For single carrier modulations, the power of the outermost (highest power) point in the reference signal constellation.

EVM is defined as a percentage as :

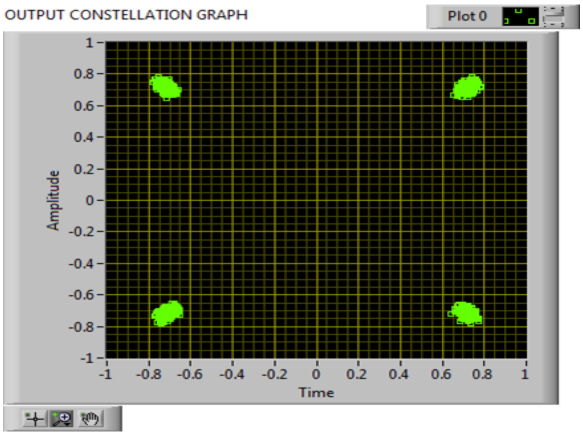
$$EVM(\%)=(\sqrt{P_{error}}/P_{reference})*100\%$$

EVM, as conventionally defined for single carrier modulations, is a ratio of a mean power to a peak power. Because the relationship between the peak and mean signal power is dependent on constellation geometry, different constellation types (e.g. 16-QAM and 64-QAM), the same mean level of interference, will give different EVM values.

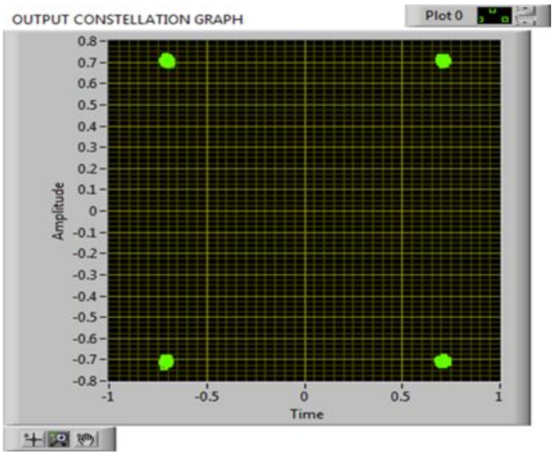
The equalized signals are represented by the constellation graph by:

- Adaptive equalizer

AT SNR=30dB

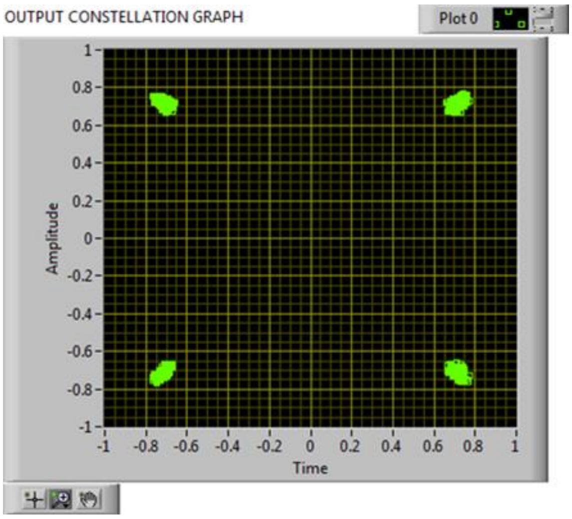


AT SNR=35dB

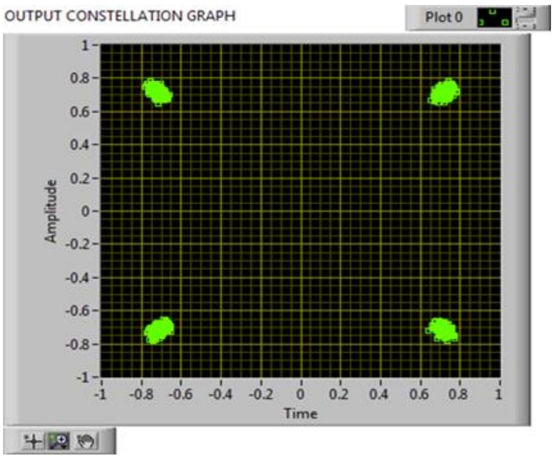


- Linear Equalizer

AT SNR=35 dB



AT SNR=30 dB



5. CONCLUSION

In this paper, we have shown how it is possible to implement the equalization of a IEEE 802.11b signal in Labview. A channel estimation code is designed first and on the basis of that Adaptive and linear equalizer are designed. The Adaptive equalizer is designed on the basis of a recursive algorithm, while linear equalizer is designed on the concept of inverse filter .The output is analysed with EVM and results are matched with the industry standards. There are a number of areas where future developments can be made .The channel estimation can be improved by taking more number of averages. The convergence factor of the adaptive filter can be improved by continuously changing the convergence factor with respect to the SNR of the signal.

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