EXAMPLES OF DIGITAL EQUALIZERS FOR SINGLE CARRIER COMMUNICATION

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***Abstract*:** Equalization is crucial in digital communication. The transmitted signal can be severely affected by phenomena occurring in the channel, resulting in Intersymbol Interference. Equalization provides the means for correcting such negative effect. Being such a wide area of study, our interest is centered around a particular type of equalizers: adaptive equalizers. Theoretical concepts related to Adaptive Equalization are discussed in the second part in order to understand the datasheets of real commercial devices presented in part three. It is indeed the goal of this paper to present these examples of equalizers: a multi-mode QAM demodulator (TDA8046) manufactured by Philips Semiconductors and a Digital Filter (GC2011) manufactured by Graychip Inc. used in a Digital PSK and QAM demodulator.

***Keywords*:** digital equalizers, equalization, demodulator, PSK, QAM

INTRODUCTION

Corruption and transformation of the transmitted signals in a given channel are present in digital communication systems. Intersymbol Interference (ISI), caused by multipath propagation is, among others, one of the undesirable effects caused by the channel due to its own characteristics. This specific corruption of the signal can be defined as the distortion caused by the resulting overlap of received symbols. As a result, the received signal experiments a high bit error rate. In order to handle this problem, equalization tries to compensate for those unwanted effects. Even if the characteristics of the channel cannot always be known “a priori” (for example, a telephone call in which the call route will be always different), equalization can provide the means to adaptively compensate for varying conditions. [1]

The purpose of this study is to search for examples of real, commercial digital equalizers in the industry, or as part of more complex applications or devices, such as digital demodulators. In this paper, it is our particular goal to focus on Adaptive Equalization and Linear Equalization. For this matter, we present the datasheets of two commercial devices that make use of digital equalization: a QAM demodulator (TDA8046 Philips) which uses a Decision Feedback Equalizer (DFE) and a Digital Filter chip (GC2011 Texas Instruments) which in turn uses a Fractionally Spaced Equalizer (FSE). The first part of this paper consists of a description of several key concepts of equalization. In the second part we provide the description of the commercial uses of Adaptive Equalization.

Adaptive Equalization Principles

Intersymbol Interference

The signal at the receiver side consists of the direct signal and the reflected signals, which are caused by elements such as buildings or trees in the vicinity of transmission lines, causing multipath propagation. As a result, the signals overlap and produce a distorted signal. With the distortion and the interference between symbols considered as noise, making the communication having a high error rates. This phenomenon is known as Intersymbol Interference (ISI) which causes frequency selectivity channel. To overcome this, equalizers compensate for distortions in the received signal, hence reducing ISI. [1].

To have a better understand on ISI phenomenon, the “eye diagram” is used. In this type of diagram, as shown in figure 1, we can visualize the superposition of multiple bits having different delays, gains and phase, transmitted to the receiver. [7]

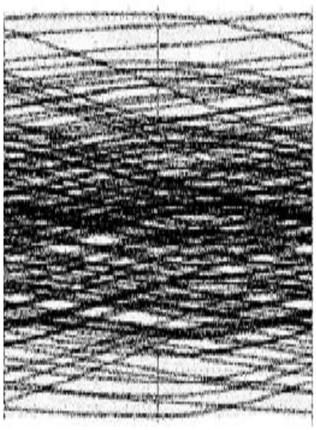
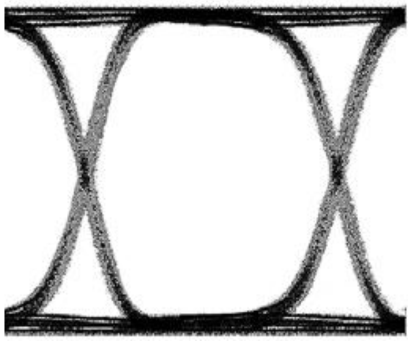


Figure 1. a). Eye diagram of signal with ISI



b). Eye diagram of signal without ISI [6]

Structure of Equalizers

Linear Transversal Equalizers

The transversal (tapped-delay-line) equalizer is the simplest structure in equalization. Its basic working principle consists of considering the current and past values of the received signal and weighting them by coefficients (or tap gains) c_n and performing a sum in order to produce the output. The samples of the received signal at the symbol rate are stored in digital shift registers and the equalizer output samples z_k are computed digitally once per symbol according to:

$$z_k = \sum_{n=0}^{N-1} c_n\, r(t_0 + kT - nt) \qquad (1)$$

Where t_0 corresponds to the sample timing and N is the number of equalizer coefficients. [1]

The coefficients can be chosen in order to force the samples of the channel and equalizer impulse response to zero, all but one of the N T-spaced instants in the span of the equalizer. This is known as Zero Forcing equalizer. [1] A diagram of a Linear Equalizer is shown in figure 2.

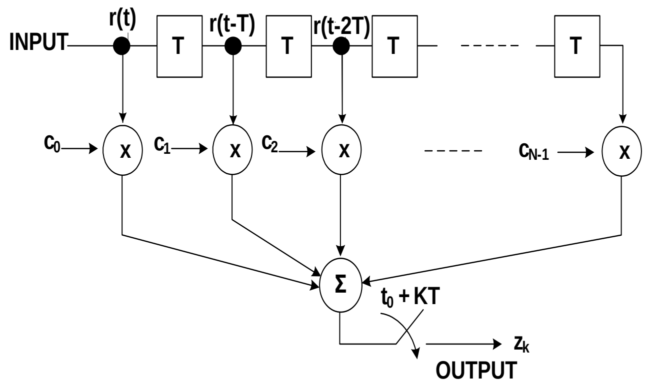


Figure 2. Linear Equalizer [1]

Another approach is to choose the coefficients so that the mean-square error (MSE), this means, the sum of squares of all the ISI terms plus the noise power at the output of the equalizer, is minimum. LMS is considered always as Adaptive Equalization, which will be explained later in this document. In this case, it is important to note that the delay introduced by the equalizer depends on the position of the main or reference tap. Usually, this tap's gain has the biggest magnitude. [1]

Decision Feedback Equalizer

The Decision Feedback Equalizer is a type of adaptive equalizer consisting of a Feedforward filter (FFF) and a Feedback filter (FBF). There are three main parts of this type of equalizer. The first part is the feedforward filter, the second part is the feedback filter, and the third part is the decision device. Both of the feedforward and feedback filter have delay line taps which are spaced at the symbol interval *T*. [1]

The output of the feedforward and the feedback filter is subtracted in order to get the decision value of the equalizer z_k. Afterwards, the signal is taken to the decision device which produces the symbol $\hat{x}_k$. By assuming this past decision value to be correct, this value is fed back into the feedback filter as the input signal. [1]

Together with the training signal or the *k*th symbol decision $\hat{x}_k$, the output of the decision device derives the error e_k as the parameter to get the feedback coefficients b_m according to:

$$b_m(k+1) = b_m(k) + \Delta e_k \hat{x}_{k-m}, \qquad (2)$$

With m = 1, …, M. [1]

A diagram of a Decision Feedback Equalizer is shown in figure 3.

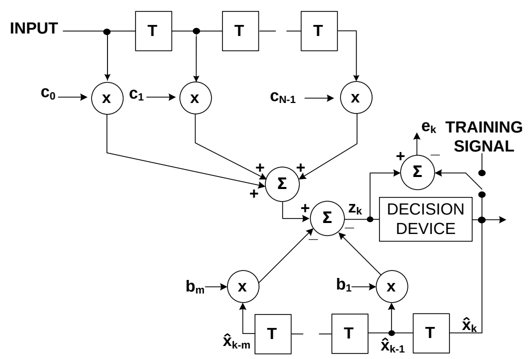


Figure 3. Decision feedback filter [2]

Fractionally Spaced Equalizer

The Fractionally Spaced Equalizer (FSE) is a type of linear equalizer. The difference between the linear transversal equalizer and the fractionally spaced equalizer is at the coefficients. In the FSE equalizer the τ-spaced delay line tap is a fraction of, or smaller than, the symbol interval T. The bandwidth of the input of the FSE has to be equal to |f| < ½ τ, which can be done by carefully choosing the value of τ. [2]

To easily depict the working principle of FSE, we use the example of the transmitted signal pulse that has a cosine spectrum with roll-off factor β, then the spectrum for Fmax will be expanded to (1 +

β) /2T. And the sampling rate at the receiver is $2F_{max} = \frac{1+\beta}{T}$ (3)

By using the FSE, the signal will be sampled at a rate at least equal to the Nyquist rate with tap spacing T/ (1 + β). For example, if β = 1 then the tap is 1/2 T-spaced equalizer. If β = 0.5, then the tap is 2/3 T-spaced equalizer and so on.[2]

From the above explanation the tap spacing for FSE can be described as MT/N, where N and M are integers and N is greater than M. ½ T-spaced equalizer often used in many applications of FSE. [2] A diagram of FSE is shown in figure 4.

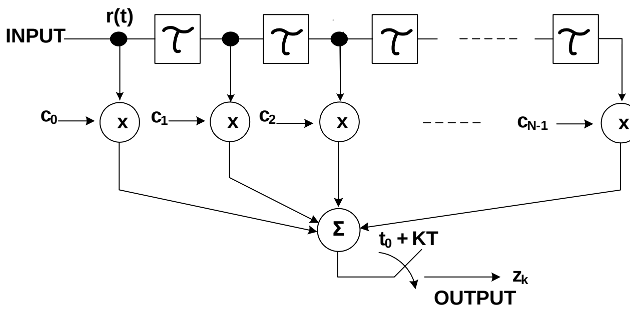


Figure 4. Fractionally Spaced Equalizer [2]

Adaptive Equalization

Since the characteristics of the channel may be time-variant, the coefficients of the equalizer can be updated adaptively after the training sequence, given that there is one. This way, the error signal is computed using the estimate that the receiver generates from the transmitted sequence. Adaptive Equalization is a means or algorithm that performs the adjustment of the coefficients of the equalizer. Normally, the error estimates are correct enough to allow the adaptive equalizer to maintain precise equalization. The larger the step size, the faster the equalizer tracking capability. In practice, the value of the step is selected for fast convergence during the training period and then reduced for fine tuning during the steady-state operation. [1]

LMS

A particular algorithm for this matter is LMS (Least Mean-Square) which uses the steepest descent method in order to find the coefficients that minimize the error. In this iterative procedure, arbitrary initial coefficients are chosen. The gradient of the function formed by the coefficients is computed in these inital coefficients, and each tap weight is modified in the direction opposite to its corresponding gradient component. [2] Succeeding values of the coefficients are obtained according to:

$$\boldsymbol{C}_{k1} = \boldsymbol{C}_k - \Delta \boldsymbol{G}_k \quad (4)$$

Where $\mathbf{C}_k$ is the column vector of equalizer coefficients, $\mathbf{G}_k$ is the gradient vector and Δ is a positive number chosen small enough to ensure that the iterative process will converge. If the minimum MSE is reached for some $k=k_0$ then $\mathbf{G}_k=0$ (or at least close to 0), so that no further changes are done to the coefficients. [2]

Training Sequence

In order to acquire information about the channel characteristics, a known signal has to be transmitted so that the receiver can generate a synchronized version of it. This is what is known as the training period. The process carried out in this period involves the iterative solution of sets of simultaneous equations corresponding to the type of equalizer being used, i.e. LMS or ZF. The training sequence, which can be periodic isolated pulses or a continuous sequence, must be at least as long as the length of the equalizer. A diagram showing the Training Sequence generator is shown in figure 5.

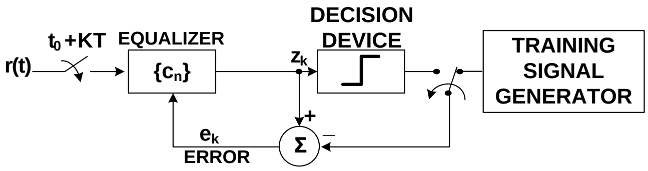


Figure 5. Training Sequence [1]

Once the receiver has generated the synchronized version of the transmitted signal, a sequence of error signals can be computed at the output of the equalizer. This can be used to adjust the coefficients in order to reduce the sum of the square errors.

$$e_k = z_k - \hat{x}_k \qquad (5)$$

Each tap gain can be adjusted each symbol interval. The idea is to move the coefficients closer to the unique optimum set of coefficients corresponding to the minimum MSE. The tap gains are updated according to:

$$c_n(k+1) = c_n(k) - \Delta e_k r(t_0 + kT - nT)$$
(6)

n = 0, 1, . . . N-1

Where $c_n(k)$ is the *nth* tap gain at time *k*, e_k is the error signal and Δ is a positive adaptation constant or step size. [1]

Data Sheets Of Equalizers

TDA8046 Multimode QAM demodulator

Application and Features

This is a Multi-mode QAM demodulator for different modulation schemes (4, 16, 32, 64 and 256 QAM), with a digital demodulator and square root cosine Nyquist filter with roll-off of 15% or 20%, a high performance adaptive equalizer, symbol rate of 7 Msymbol/s, digital detectors for generation of required control voltages for carrier and clock recovery and an I^2C-bus interface. The frequency of the IF signal (IFQAM) is down converted to a frequency that equals the symbol rate (rs) by a mixer which is driven from a local oscillator with a frequency of $f_{CAR} = f_{IF} + r_S$. Its main application is to serve as a demodulator for digital cable TV and cable modem. [3]

Equalizer description

The equalization function in the TDA8046 is carried out by means of two filters, a Feed Forward Equalizer and a Feedback Filter. The number of T-spaced taps (12 or 14) of the adaptive filter can be selected through the I^2C bus. Moreover, the equalizer is based on a Decision Feedback Equalizer (DFE) structure with Least Mean Square (LMS) coefficient updating algorithm. This equalizer does not need any training sequence. [3] The block diagram of the equalizer is shown in figure 6.

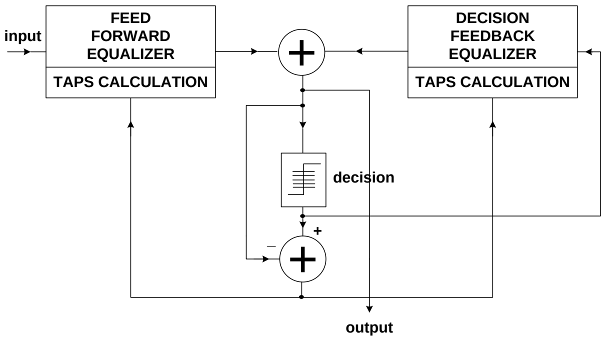


Figure 6. TDA8046 DFE equalizer structure [3]

The main tap of the equalizer can be adjusted for fine Automatic Gain Control

(AGC) function. The settings of the equalizer taps can be read via the I^2C-bus. If the equalizer diverges, an automatic reset of the taps can be performed. [3]

The convergence steps of the FFE/DFE parts of the equalizer are programmable via the I^2C-bus. When the system locks, the steps are automatically modified for optimum performances. Besides reading the equalizer tap values, the main tap of the equalizer can also be programmed. After setting the main tap, the other coefficients can be set to zero. The equalizer settings can also be frozen via the I^2C-bus. [3]

The following figure represents the QAM spectrum seen by the equalizer. It corresponds (in the frequency domain) to the multiplication of a full Nyquist spectrum by the impulse response of the channel. [3] Hence, the equalizer must compensate for the frequency selectivity of the channel, which is shown in figure 7.

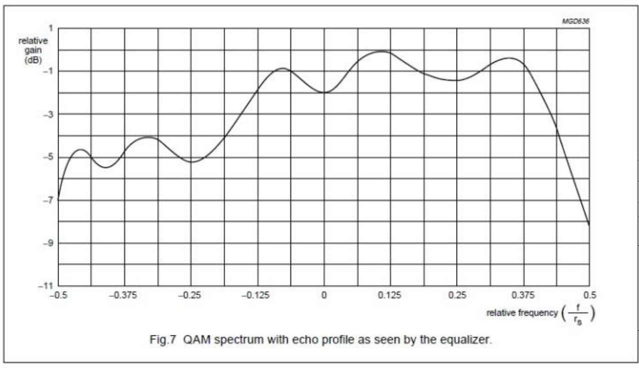


Figure 7. QAM spectrum seen by the equalizer [3]

QAM demodulator using the GC2011, GC3011 and GC3021 chips

Application and Features

This is a BPSK, QPSK, and QAM radio signal demodulator. The application of this demodulator is for microwave links, cable TV decoders (either QAM or VSB modulations), and satellite receivers. The symbol rates of this demodulator are up to 17.5 MSymbol/s with 70 MHz demodulator's clock. [4]

Equalizer description

The digital filter GC 2011 is proposed as a passband equalizer in the digital demodulator for PSK and QAM signals, using Fractionally-spaced Forward Equalizer (FSE). [4] Figure 8 shows the equalizer used for passband demodulator.

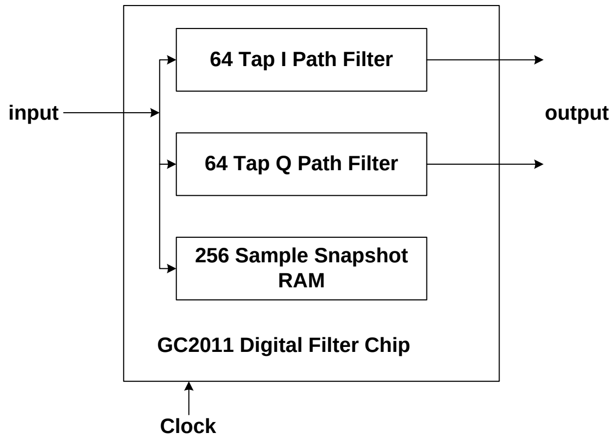


Figure 8. Passband equalizer [4]

Before carrying out the equalization process, the signal is digitized at the nyquist rate and resampled at a rate equal to N times the symbol rate. The clock rate of this equalizer has to be equal to the clock rate of the output of the resampler. The FSE uses a 64 T/4 spaced equalizer, which has the same delay value as a 32 tap T/2 spaced equalizer. The real part of the sampled signal is used for the equalizer's input, and the complex part of the sampled signal is used for the FSE coefficient. The snapshot RAM stores the input samples for the updating of the FSE coefficients and it can acquire blocks up to 128 samples. The output of this equalizer is a complex sample with reduced intersymbol interference that will be used at a carrier removal chip to obtain the data symbols. [4]

CONCLUSIONS

Even when information on Digital Equalization theory is widely available in many publications and books, it is not easy to find manufacturers that will make public the exact working principles of the devices making use of digital equalizers that they produce. The description that is available is rather general and doesn't always allow the general public to verify the structure of the equalizer. Normally, the specific technical documentation about a given device is kept private for internal use within the company that manufactures the device. However, it is important to highlight that, even if the available description is not very detailed, it actually allows us to learn the most relevant characteristics of the equalizer.

In addition, we observed that, as a part of most QAM, BPSK and QPSK demodulators, an equalizer can be used to reduce intersymbol interference which causes frequency selectivity channel on digital communication. Datasheets of "standalone" equalizers were not found, only as part of a more complex application which is demodulators.

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