ADAPTIVE EQUALIZATION

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INTRODUCTION TO EQUALIZATION

Equalization is a technique used to combat inter symbol interference (ISI).

An Equalizer within a receiver compensates for the average range of expected channel amplitude and delay characteristics.

Equalizers must be adaptive as the channel is generally unknown and time varying.

ISI has been recognized as the major obstacle to high speed data transmission over mobile radio channels.
Equalizers

- The goal of equalizers is to eliminate intersymbol interference (ISI) and the additive noise as much as possible.
- **Intersymbol interference** arises because of the spreading of a transmitted pulse due to the dispersive nature of the channel, which results in overlap of adjacent pulses.
- In Fig. 1, there is a four-level pulse amplitude modulated signal (PAM), $x(t)$. This signal is transmitted through the channel with impulse response $h(t)$. Then noise $n(t)$ is added. The received signal $r(t)$ is a distorted signal.
Categories of Equalization

• Equalizers are used to overcome the negative effects of the channel. In general, equalization is partitioned into two broad categories;

1. **Maximum likelihood sequence estimation (MLSE)** which entails making measurement of channel impulse response and then providing a means for adjusting the receiver to the transmission environment.
   (Example: Viterbi equalization)

2. **Equalization with filters**, uses filters to compensate the distorted pulses. The general channel and equalizer pair is shown in Figure.2.
Depending on the time nature

- These type of equalizers can be grouped as **preset or adaptive equalizers**.
- **Preset equalizers** assume that the channel is time invariant and try to find $H(f)$ and design equalizer depending on $H(f)$. The examples of these ADAPTIVE EQUALIZERS are zero forcing equalizer, minimum mean square error equalizer, and decision feedback equalizer.
- **Adaptive equalizers** assume channel is time varying channel and try to design equalizer filter whose filter coefficients are varying in time according to the change of channel, and try to eliminate ISI and additive noise at each time. The implicit assumption of adaptive equalizers is that the channel is varying slowly.
Block diagram of Adaptive equalizer

As the mobile fading channels are random and time varying, equalizers must track the time varying characteristics of the mobile channel, and thus are called adaptive equalizers.
Working principles of adaptive equalizers

The working principles of adaptive equalizers are in the following:

• The received signal is applied to receive filter. In here, receive filter is not matched filter. Because we do not know the channel impulse response. The receive filter in here is just a low-pass filter that rejects all out of band noise.
• The output of the receiver filter is sampled at the symbol rate or twice the symbol rate.
• Sampled signal is applied to adaptive transversal filter equalizer. Transversal filters are actually FIR discrete time filters.
• The object is to adapt the coefficients to minimize the noise and intersymbol interference (depending on the type of equalizer) at the output.
• The adaptation of the equalizer is driven by an error signal.
Operation mode of adaptive equalizers

- There are two modes that adaptive equalizers work;

1. **Decision Directed Mode**: This means that the receiver decisions are used to generate the error signal.
2. **Decision directed equalizer adjustment** is effective in tracking slow variations in the channel response.

- However, this approach is not effective during initial acquisition.

- **Training Mode**: To make equalizer suitable in the initial acquisition duration, a training signal is needed. In this mode of operation, the transmitter generates a data symbol sequence known to the receiver. The receiver therefore, substitutes this known training signal in place of the slicer output. Once an agreed time has elapsed, the slicer output is substituted and the actual data transmission begins.
Operating modes of adaptive equalizer

1) Training mode

2) Tracking Mode
Training mode

- Initially, a known, fixed length training sequence is sent by the transmitter so that the receiver’s equalizer may average to a proper setting. The training sequence is a pseudo random signal or a fixed, prescribed bit pattern. Immediately following the training sequence, the user data is sent.

Tracking mode

- When the data of the users are received, the adaptive algorithm of the equalizer tracks the changing channel. As a result of this, the adaptive equalizer continuously changes the filter characteristics over time. Equalizers are widely used in TDMA Systems.
training sequence (Contd..)

The training sequence is designed to permit an equalizer at the receiver to acquire the proper filter coefficients in the worst possible channel conditions. Therefore when the training sequence is finished. Therefore filter coefficients are near their optimal values for reception of user data. An adaptive equalizer at the receiver uses a recursive algorithm to evaluate the channel and estimate filter coefficients to compensate for the channel.
Block Diagram of Adaptive Equalizer

\[ f(t) = \text{combined impulse response of transmitter, multipath radio channel, and receiver RF/IF} \]

Original Baseband Message \( x(t) \)

\( \rightarrow \) Modulator

\( \rightarrow \) Transmitter

\( \rightarrow \) Radio Channel

\( \rightarrow \) Detector Matched Filter

\( \rightarrow \) IF Stage

\( \rightarrow \) RF Receiver Front End

\( \rightarrow \) Equalizer \( h_{eq}(t) \)

\( \rightarrow \) Decision Maker

\( \rightarrow \) Reconstructed Message Data \( d(t) \)

\( \rightarrow \) Equalizer Prediction Error

\( \rightarrow \) \( \sum \)

\( \rightarrow \) \( e(t) \)

\( \rightarrow \) \( \hat{d}(t) \)

\( \rightarrow \) \( + \)
A Generic Adaptive Equalizer
Transversal filter with \( N \) delay elements, \( N+1 \) taps, and \( N+1 \) tunable complex weights. These weights are updated continuously by an adaptive algorithm.
Algorithm for Adaptive Equalization

- Performance measures for an algorithm
  - Rate of convergence
  - Misadjustment
  - Computational complexity
  - Numerical properties
- Factors dominate the choice of an equalization structure and its algorithm
  - The cost of computing platform
  - The power budget
  - The radio propagation characteristics
Algorithm for Adaptive Equalization

- The speed of the mobile unit determines the channel fading rate and the Dopper spread, which is related to the coherent time of the channel directly.
- The choice of algorithm, and its corresponding rate of convergence, depends on the channel data rate and coherent time.
- The number of taps used in the equalizer design depends on the maximum expected time delay spread of the channel.
- The circuit complexity and processing time increases with the number of taps and delay elements.
Algorithm for Adaptive Equalization

• Three classic equalizer algorithms: zero forcing (ZF), least mean squares (LMS), and recursive least squares (RLS) algorithms
• Summary of algorithms (see Table 1)
Conclusion

• Summary
  – Linear equalizers: suffer from noise enhancement
  – DFE: Error propagation
  – MLSE
    • Optimal method
    • Viterbi equalizer implements MLSE with much lower complexity
Summary of algorithms

<table>
<thead>
<tr>
<th>Algorithm</th>
<th>Number of Multiply Operations</th>
<th>Advantages</th>
<th>Disadvantages</th>
</tr>
</thead>
<tbody>
<tr>
<td>LMS Gradient DFE</td>
<td>$2N + 1$</td>
<td>Low computational complexity, simple program</td>
<td>Slow convergence, poor tracking</td>
</tr>
<tr>
<td>Kalman RLS</td>
<td>$2.5N^2 + 4.5N$</td>
<td>Fast convergence, good tracking ability</td>
<td>High computational complexity</td>
</tr>
<tr>
<td>FTF</td>
<td>$7N + 14$</td>
<td>Fast convergence, good tracking, low computational complexity</td>
<td>Complex programming, unstable (but can use rescue method)</td>
</tr>
<tr>
<td>Gradient Lattice</td>
<td>$13N - 8$</td>
<td>Stable, low computational complexity, flexible structure</td>
<td>Performance not as good as other RLS, complex programming</td>
</tr>
<tr>
<td>Gradient Lattice DFE</td>
<td>$13N_1 + 33N_2 - 36$</td>
<td>Low computational complexity</td>
<td>Complex programming</td>
</tr>
<tr>
<td>Fast Kalman DFE</td>
<td>$20N + 5$</td>
<td>Can be used for DFE, fast convergence and good tracking</td>
<td>Complex programming, computation not low, unstable</td>
</tr>
<tr>
<td>Square Root RLS DFE</td>
<td>$1.5N^2 + 6.5N$</td>
<td>Better numerical properties</td>
<td>High computational complexity</td>
</tr>
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Table 1 Comparison of various algorithms for adaptive equalization
Adaptive decision feedback equalizer

**Adaptive Decision Feedback Equalizer**

- A decision feedback equalizer (DFE) is a nonlinear equalizer that uses previous detector decisions to eliminate the ISI on pulses that are currently being demodulated.

- The basic idea of a DFE is that if the values of the symbols previously detected are known (past decisions are assumed to be correct), then the ISI contributed by these symbols can be canceled out exactly the output of the forward filter by subtracting past symbols values with appropriate weighting. In
Block diagram of Adaptive DFE
Adaptive decision feedback equalizer (Contd..)

If we look at Fig. 18, we see that the estimated signal sequence becomes,

\[ q_k = \sum_{i=-N+1}^{0} c_i R_{k-i} - \sum_{i=1}^{M} d_i A_{k-i} \]

\( \{c_i\} \)s are coefficients of the precursor equalizer,
\( \{d_i\} \) are coefficients of the postcursor equalizer.
N is the number of precursor equalizer coefficients and M is the number of postcursor equalizer coefficients.
Adaptive DFE algorithm is similar to stochastic gradient algorithm, with the important difference that the input to the causal portion of the filter is the decisions rather than the output of the precursor equalizer filter.
This difference will obviously change the desired tap coefficients as well as reduce the noise enhancement due to equalization.
Adaptive decision feedback equalizer (Contd..)

The derivation of a stochastic gradient algorithm for the DFE is a simple extension of the stochastic gradient algorithm for linear case. First, define an augmented vector of N+M coefficients,

$$\mathbf{v}^T = [c_{-(N-1)} \ldots c_0 - d_1 \ldots - d_M]$$

(4.2)

and an augmented input signal vector

$$\mathbf{w}_k^T = [R_{k+(N-1)} \ldots R_k A_{k-1} \ldots A_{k-M}]$$

(4.3)

DFE slicer error can be expressed as,

$$\hat{E}_k = \hat{a}_k - \mathbf{v}_k^T \mathbf{w}_k$$

The adaptation algorithm becomes,

$$\mathbf{v}_{k+1} = \mathbf{v}_k + \beta \hat{E}_k \mathbf{w}_k^*$$
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