EQUALIZER FOR MULTIPATH CHANNEL

Yuri V. Andreyev* and Andrey A. Kletsov**

*Institute of RadioEngineering and Electronics, Russian Academy of Sciences
Mokhovaya st. 11-7, GSP-9, 125009, Moscow, Russia
E-mail: chaos@mail.cplire.ru

**Moscow Institute of Physics and Technology
Institutskii st. 9, 141700, Dolgoprudnyi, Moscow region, Russia
E-mail: chaos@mail.cplire.ru

Abstract – In this report a problem of channel equalization of multipath propagation channels is discussed. Sophisticated IEEE 802.15.3a standard channel is equalized. Equalizer is designed as a FIR filter inverse to the channel.

Index terms – Multipath propagation, ultrawideband communications, chaotic communications, channel equalization, filter design.

I. Introduction

One of the development trends of wireless communications is increase of the communication rate due to application of novel communication schemes, transition to higher frequency bands, use of wideband signals, etc. Design of high-rate communication systems requires more attention to the phenomenon of multipath propagation. The transmitted signal comes to the receiver in several beams reflected from different objects. These beams interfere at the receiver input. This phenomenon is especially prominent by indoor communications. The multipath propagation is basically exhibited in two effects — interference in frequency domain (e.g., signal fading) and in time domain (e.g., intersymbol interference). The fading is observed in the case of two beams with close frequencies arrived with opposite phases, resulting in a sharp signal fading down to disappearance. The second effect — interference in time domain — is connected with delay of some beams with respect to others, which can result in synchronization errors and essential distortions of the form of the received signal.

Let us consider for certainty a scheme for transmission of binary information, in which the transmitted signal is represented by a sequence of radio pulses, and the presence of pulse on a given time frame corresponds to transmission of information symbol "1", and its absence to symbol "0". With increasing pulse repetition rate (increasing data rate) the pulse duration decreases, and the superposition of pulses of one beam onto positions of other pulses of another beam becomes more and more probable. The phenomenon of interference of several beams with delays larger or of the order of the pulse duration is called intersymbol interference (ISI). Numerically it can be characterized by the number of adjacent pulses affected by the given pulse. The length of ISI depends on pulse repetition period T and on the impulse response of communication channel. In practice it can reach 30 symbols and more. For example, for pulse duration T = 10 ns (which in the considered communication scheme corresponds to the rate of 100 Mbps) one beam is delayed by one pulse against another beam at the path length difference Δl = 3 m.

II. Design of equalizer

As was already mentioned, the signal in multipath channel undergoes distortions both in frequency and in time domain. Effective operation of the receiver requires that these distortions be compensated, if possible. This task is solved by a device called channel equalizer.

Not all channels can be equalized, even theoretically. For example, an ideal low-pass channel cannot be equalized if frequencies above cutoff are transmitted. However, in practice the channel is usually matched with the frequency range of the transmitter.

Let us build equalizer in the form of a linear finite-impulse response filter (FIR filter). It must adjust to the concrete channel, e.g., using the knowledge of the channel response. It must be a filter inverse by its properties to the channel filter [1].

Consider a communication channel as a FIR filter with response $\boldsymbol{h}$. Mathematically, in frequency domain the equalizer's frequency response $E(f)$ must be reciprocal to the channel response $C(f)$: $E(f) = 1/C(f)$. In time domain the equalizer must restore the signal "spread" in time by the channel by means of collecting taps into required time frames.

Let the equalizer's impulse response be vector $\boldsymbol{b}$, which length $2K$ is no less than that of the channel filter. As a result of filtration, we must obtain a single sample from the impulse response $\boldsymbol{h}$. This can be written in the form of a system of linear equations:

$$q(mT) = \sum_{n=-K}^{K} b_n h\left(m\tau_1 - n\tau_2\right) = \begin{cases} 1, & m = 0 \\ 0, & m = \pm1, \pm2, \pm3, \ldots, \pm K \end{cases},$$

where τ_1 is the period of output signal sampling and τ_2 is the input signal sampling period of the equalizer. In the case of $\tau_1 = \tau_2$ equalizer is called *symbol spaced equalizer*; while in practice *fractionally spaced equalizers* are more often used, with $\tau_2 = \tau_1/2$ [1].

In matrix form the above relation is

$$\mathbf{H}\boldsymbol{b}=\boldsymbol{q}, \tag{1}$$

where $\mathbf{H}$ is the matrix with elements $h(m\tau_1\text{-}n\tau_2)$, $\boldsymbol{b}$ is the inverse filter coefficients vector, and $\boldsymbol{q}$ is a column vector with single non-zero element. The equalizer coefficients $\boldsymbol{b}$ are solution to matrix equation (1)

$$\boldsymbol{b} = \mathbf{H}^{-1}\boldsymbol{q}. \tag{2}$$

III. Simulation of equalizer

Design of the channel equalizer was simulated in Matlab in order to test a possibility of signal recovery after its distortion in a simple channel. Let T be the pulse repetition period of the transmitted signal. We take a channel in the form of a FIR-filter with the impulse response spread over 8 pulses

$$\frac{1}{\left(1+\left(\dfrac{t}{2T}\right)^2\right)},\ t=(-4T,4T).$$

If the pulse duration $T = 10$, the channel impulse response $\boldsymbol{h}$ has $K = 81$ taps (period of input signal sampling $\tau = 1$) (Fig. 1).

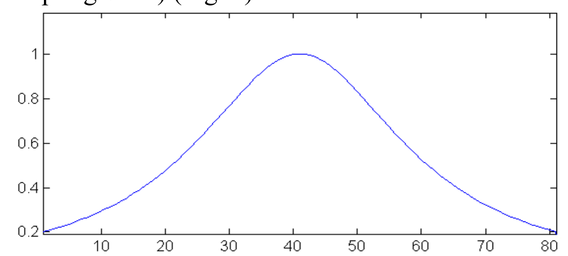

Fig. 1. The channel impulse response $\boldsymbol{h}$

The information signal is represented by a sequence of symbols "0" and "1". Let each symbol be encoded in the channel by $N = 10$ samples, the symbol "1" be represented by a sequence of random samples and symbol "0" by zero samples (Fig. 2).

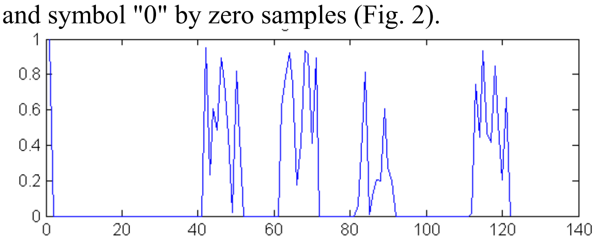

Fig. 2. Transmitted signal waveform

Let the channel impulse response be unknown in the receiver. In order to obtain it in the receiver, the signal of the transmitter is specially formed as follows. A unit sample is set in the beginning of the transmitted signal, followed by a spacing (zero) interval, long enough to ensure no ISI between the sample and the first bit pulse, i.e., here at least by 81-sample zero interval (Fig. 2). Then after the channel this sample will be spread into the channel impulse response $\boldsymbol{h}$.

The signal in the receiver $x(t)$ is shown in Fig. 3. As can be seen, visual extraction of information from this signal is impossible here.

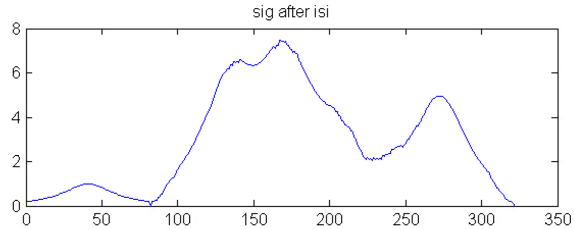

Fig. 3. Signal in the receiver $x(t)$, distorted by channel

Since the single unit sample was put in the beginning of the transmitted signal, the distorted received signal begins with the impulse response characteristic of the channel. So, this response $\boldsymbol{h} = x(k)$, $k = 1, \ldots, 81$, is picked out and the coefficients of the inverse filter $\boldsymbol{b}$ are calculated. Vector $\boldsymbol{b}$ is given by relation (2), which means that a single nonzero sample is restored from the channel impulse response. The filter must accumulate enough input samples before giving output signal, so vector $\boldsymbol{q}$ is set as $\boldsymbol{q}$ = (0... 010... 0) with nonzero 41-st position. This means that the restored signal is delayed with respect to the input signal by 40 samples. Matrix $\mathbf{X}$ is made of samples of the channel impulse response $X_{m,n} = x(2m\text{–}n)$. Note, that here $\tau_1 = 2$ and $\tau_2 = \tau_1/2 = 1$, i.e., the output of the equalizer is sampled with twice less period that the input signal.

The original signal is restored by means of convolving the received signal with the equalizer response: $y(n) = b(1)x(n) + b(2)x(n\text{-}1) + \ldots + b(K\text{+}1)x(n\text{-}K)$, $n = 1, 2\ldots 160$. The restored signal is shown in Fig. 4. In the absence of external channel noise, the restored signal coincides with the transmitter signal within the calculation accuracy.

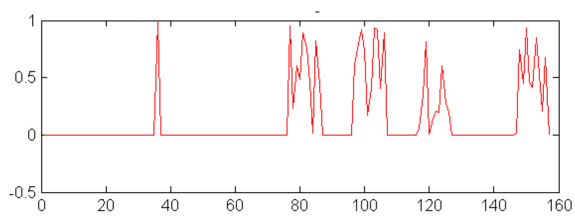

Fig. 4. Restored signal $y(t)$

This simple example demonstrates that the procedure for channel equalization using the filter inverse to the channel [1] is successful in the case of "good" smooth channels.

IV. Equalizer for IEEE 802.15.3a standard channel

Consider now a more complicated channel which model is accepted in IEEE 802.15.3a standard (awaiting adoption) [2]. This standard describes systems of ultrawideband communications in 3–10 GHz range. The model of multipath propagation for this standard is continuous, i.e., instead of several beams it deals with many clusters of beams each consisting of a number of beams with log-normal distribution. The total number of beams is not fixed: the better the resolution, the more beams are distinguished. This model corresponds to conditions of indoor multipath propagation, with many reflecting surfaces.

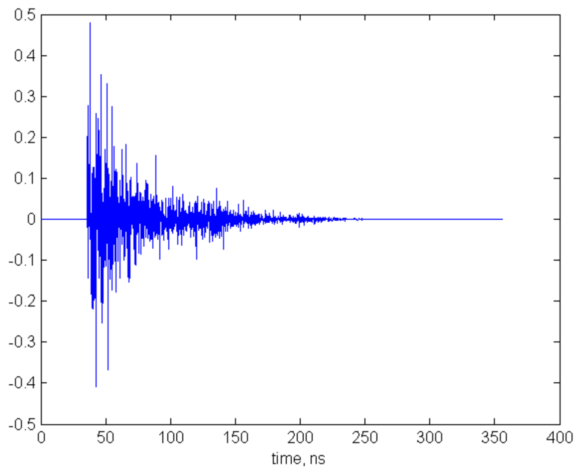

Fig. 5. Impulse response characteristic of CM-4 multipath channel

In this standard, four channel models are given for simulation of ultrawideband communication systems, one LOS model CM-1 and three NLOS channel models (CM-2–CM-4). Signal mean delay in these models increases from 5 to 25 ns, respectively. The channel affects the transmitted signal at the time interval 60 to 250 ns, respectively. A typical impulse response for CM-4 is shown in Fig. 5.

In IEEE 802.15.3a standard the channel is considered varying, and the following calculation rules are proposed. 100 different channel realizations are considered and for each of them 200 packets of 1024 bytes are transmitted. 15 worst channel realizations are discarded.

Here, we consider a system of ultrawideband digital communications, in which information bits are encoded by radio pulses (Fig. 3). Symbol «1» is transmitted by chaotic radio pulse, whereas symbol «0» by zero radio pulse. Pulse period T is determined by information rate R, so that $T = 1/R$. At the rate R = 100 Mbps the pulse repetition period T = 10 ns, and at R = 500 Mbps T = 2 ns. Consequently, for this communication scheme an increase of transmission rate leads to shorter pulses, hence, the ISI effect increases.

Let for certainty the communications be in 3–5 GHz range with R = 200 Mbps rate. The original chaotic signal is filtered within 3–5 GHz band, sampled with 4 GHz sampler, and modulated by information signal. Duty cycle is 1, the duration of the pulse encoding one information bit is 5 ns or 80 samples. Here we discuss equalization of the worst CM-4 channel model.

In all four models CM-1–CM-4 the signal is badly distorted in the channel. As can be seen in Fig. 6, the signal spreading leads to such a strong ISI that retrieving information becomes practically impossible.

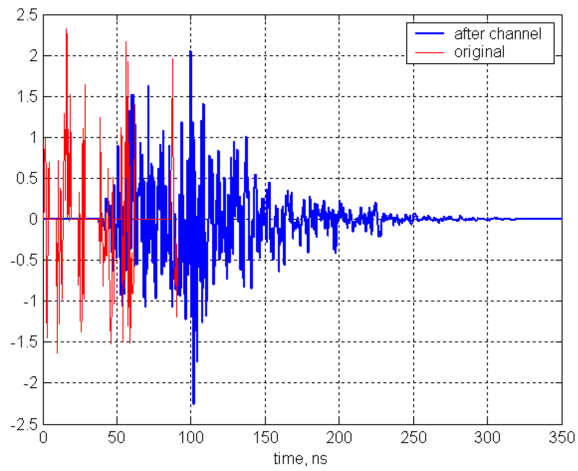

Fig. 6. Signal distortion in CM-4 channel

Let the channel impulse response h_n, $n = 1, \ldots, N$, be known in the receiver, as well as the received signal samples. As in the previous simulation, the impulse response is used to design an equalizer of the kind of FIR filter inverse to the channel. However, the procedure here is different.

The signal $y(k)$ at the output of a filter can be obtained by means of convolving the input signal $x(k)$, $k = 1, \ldots, M$, with the pulse response h_n, i.e., $\boldsymbol{y} = \boldsymbol{x} \otimes \boldsymbol{h}$. The input signal represented by a single unit-magnitude sample, i.e., δ-impulse, is spread in time domain into h_n, so that $\delta \otimes \boldsymbol{h} = \boldsymbol{h}$. The task of the equalizer is to gather this vector back to δ-impulse. Let the required inverse filter response be b_n, $n = 1$,

…, N. Filtering $\boldsymbol{h}$ with the inverse filter must give a vector $\boldsymbol{e}$ with a single nonzero sample, i.e.,

$$\boldsymbol{b} \otimes \boldsymbol{h} = \boldsymbol{e}. \tag{3}$$

As is known, discrete convolution of two vectors $\boldsymbol{b}$ and $\boldsymbol{h}$ of lengths M and N gives a vector of length $L = M+N-1$. The convolution is calculated as

$$e(i) = b(1)h(i) + b(2)h(i-1) + \ldots + b(i)h(1),\ i = 1, \ldots, N+M-1.$$

This result can be obtained by means of multiplying vector $\boldsymbol{b}$ by $N\times(2N-1)$ matrix of the following kind, composed of the direct filter coefficients

$$\begin{pmatrix} h_1 & & & & \\ & \ddots & & 0 & \\ & \cdots & & & \\ h_N & \cdots & h_3 & h_2 & h_1 \\ & \ddots & & \cdots & \\ 0 & & h_N & \cdots & \\ & & & h_N & \\ & & & & h_N \end{pmatrix}. \tag{4}$$

Vector $\boldsymbol{e}$ of length $2N-1$ must be zero with the single nonzero sample at Nth position. So, we obtain a overdetermined system of linear equations with b_n unknown. The system has $2N-1$ equations for N variables. The solution to this system is sought in the sense of the least square approximation.

The quality of solution can be controlled by means of convolution of direct and inverse filters (3). The result is shown in Fig. 7. As can be seen, the result is not ideal: the central sample is not unit but ~0.9, and a certain noisy fringe is observed with the magnitude of the order of 0.01–0.03.

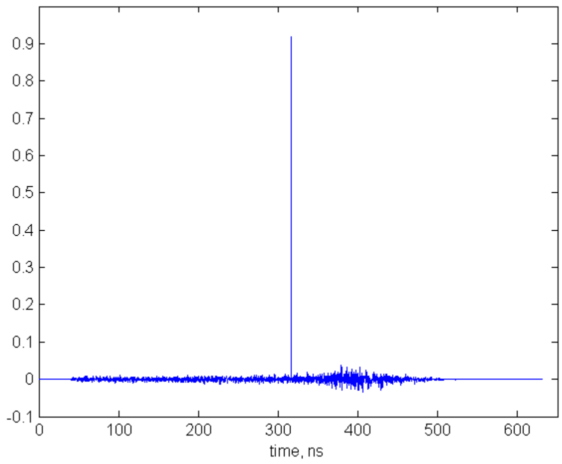

Fig. 7. Convolution of channel and equalizer for CM-4.

Application of the equalizer to the signal in Fig. 6 is shown in Fig. 8. As can be seen, the equalizer gives a slightly noised signal, though the signal waveform is restored rather good. The quality of equalizer can be estimated by signal-to-noise value (SNR), which here is 17.9 dB.

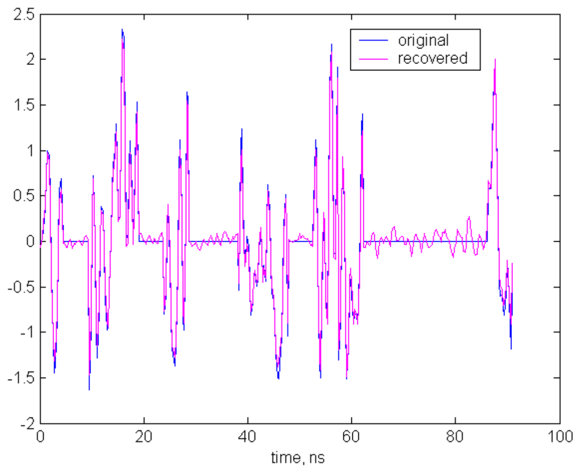

Fig. 8. Signal recovered by equalizer of channel CM-4.

Equalization of other channel models was also simulated. Practically the same results were obtained.

V. Conclusions

In this report a problem of channel equalization was considered in the case of multipath propagation channels. Sophisticated IEEE 802.15.3a standard channel models were analyzed. The equalizer was designed as a FIR filter inverse to the channel filter. As was shown, equalization of IEEE 802.15.3a multipath channel was possible. A procedure for filter design was proposed. Equalization of the channel with the designed filter gives ~ 15–20 dB SNR of the recovered signal.

Acknowledgment

This report is supported in part by a grant from the Russian Foundation for Basic Investigations (No. 02-02-16802).

References

[1] J.G. Proakis. “Digital communications”,. McGraw-Hill, Inc., NY, 1995.

[2] IEEE P802.15 Wireless Personal Area Networks, Channel Modeling Sub-committee Report (Final)