Impulsive Noise Suppression in OFDM Based Communication Systems

Sergey V. Zhidkov

Abstract — *Orthogonal frequency division multiplexing (OFDM) is a technique used for terrestrial digital video broadcasting (DVB-T) and many other modern applications. The longer OFDM symbol duration provides an advantage in a presence of weak impulsive noise, because impulsive noise energy is spread among simultaneously transmitted OFDM sub-carriers. However, it has been recently recognized that this advantage turns into a disadvantage if the impulsive noise energy exceeds certain threshold. In this paper the algorithm for impulsive noise suppression in OFDM receivers is proposed and investigated. Whereas traditional methods for impulsive noise suppression are implemented in a time domain before OFDM demodulation, proposed algorithm compensates impulsive noise in a frequency domain after OFDM demodulation and channel equalization. The method is applied to DVB-T and its performance is studied by means of simulation[1].*

Index Terms — DVB-T, Impulsive noise, OFDM.

I. INTRODUCTION

Orthogonal Frequency Division Multiplexing (OFDM) is a multicarrier modulation scheme that can cope with high degree of multipath distortions. This technique has been used in digital audio broadcasting [1] and has been chosen for European digital terrestrial video broadcasting (DVB-T) [2]. In general, OFDM systems are less sensitive to impulsive interference than single-carrier systems. This is due to fact that the OFDM symbol has longer duration and impulsive noise energy is distributed among simultaneously transmitted OFDM sub-carriers. However, under certain circumstances impulsive noise can significantly affect performance of OFDM systems. For example, it has been recently noted that DVB-T system that uses 64-QAM (quadrature amplitude modulation) could be seriously affected by impulsive interference [3].

Traditional methods for impulsive noise suppression in multicarrier receivers are based on time domain signal processing before conventional OFDM demodulator [4]. These methods provide some degree of protection against impulsive noise, but performance of traditional methods is still far away from theoretical bounds. Another group of methods exploits specific properties of OFDM signals. It is known fact that OFDM modulator with zero sub-carriers in several consecutive positions can be viewed as Reed-Solomon coder in a complex field. This can be used for impulsive noise compensation as shown in [5]. The problem is that conventional error correction decoders cannot cope with the signal affected by both impulsive interference and white Gaussian noise. Recently, iterative algorithm for decoding complex number codes in impulsive noise channels was described and close relationship between OFDM and complex number codes was shown in [6].

In this paper we present practical algorithm for impulsive noise suppression in OFDM receivers based on principles similar to [6]. Analysis of the algorithm and results of simulations demonstrating effectiveness of the algorithm in DVB-T system under various channel conditions are also presented.

II. SYSTEM AND CHANNEL MODEL

First, let us introduce the model of the OFDM system and communication channel. In the OFDM transmitter set of information bits are first mapped into baseband symbols $\{S_k\}$ using modulation schemes such as phase-shift-keying (PSK) or quadrature-amplitude-modulation (QAM). In every OFDM symbol interval symbols $\{S_k\}$ are transformed by means of inverse discrete Fourier transform (IDFT) and digital-to-analog conversion to the baseband OFDM signal as

$$s(t) = \sum_{k=0}^{N-1} S_k e^{j2\pi k \Delta f t}, \; 0 < t < T_s, \tag{1}$$

where N is the number of sub-carriers, Δf is separation between adjacent sub-carriers, and T_s is the OFDM symbol interval.

The received signal (in the time domain) after down-conversion, analog-to-digital conversion, guard interval removing, and synchronization can be represented as

$$r_k = \sum_{l=1}^{L} h_l s_{k-l} + w_k + u_k, \; k = 0,1,..., N-1, \tag{2}$$

where $s_k = s(kT_s/N)$, h_l is the channel impulse response, L is the length of channel impulse response, w_k is additive white Gaussian noise (AWGN), and u_k is the impulsive noise. Analytical characterization of impulsive noise is quite difficult task. Several analytical and empirical impulsive noise models were proposed in a literature in last twenty years [7]-[11]. Some of these models characterize impulsive noise from particular sources in certain conditions with high degree of accuracy. However, in many practical situations it is very difficult to predict statistical properties of impulsive

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interference. On the other hand, impulsive noise from any source in any system has two common properties: (a) impulsive noise energy is concentrated into short periods; (b) impulsive noise energy is much higher than that of background (thermal) noise.

In this study we assume that impulsive noise sequence $\{u_k\}$ can be characterized by these two properties. We also assume that impulsive noise duration is much shorter than active period of the OFDM symbol T_s.

III. Impulsive Noise Compensation

A. Conventional Methods

A simple method of reducing the adverse effect of impulsive noise is to precede conventional OFDM demodulator with a limiting nonlinearity [4]

$$r_k^{(comp)} = \begin{cases} r_k, \text{ if } |r_k| < A_0 \\ A_0 e^{j\arg(r_k)}, \text{ otherwise} \end{cases}, \quad k = 0,1,...,N-1, \tag{3}$$

or blanking nonlinearity

$$r_k^{(comp)} = \begin{cases} r_k, \text{ if } |r_k| < A_0 \\ 0, \text{ otherwise} \end{cases}, \quad k = 0,1,...,N-1 . \tag{4}$$

The non-linearity reduces the effect of large received signal values as these are assumed to be the result of impulsive noise. The blanking nonlinearity (4) gives a slightly better results, because in average replacing affected samples by zero is likely to introduce the least error energy. A major advantage of this method is that it is very simple to implement and gives an improvement over conventional OFDM demodulator in impulsive noise.

B. Proposed Algorithm

Whereas conventional methods for impulsive noise suppression are implemented in a time domain before OFDM demodulation, proposed algorithm compensates impulsive noise in a frequency domain after OFDM demodulation and channel equalization. Received signal after fast Fourier transform (FFT) can be expressed as

$$R_k = H_k S_k + W_k + U_k, \quad k = 0,1,...,N-1 \tag{5}$$

where $\mathbf{H} = [H_0, H_1, \ldots, H_{N-1}]$ is the discrete Fourier transform (DFT) of channel impulse response, $\mathbf{S} = [S_0, S_1, \ldots, S_{N-1}]$ is the DFT of transmitted signal, $\mathbf{W} = [W_0, W_1, \ldots, W_{N-1}]$ is the DFT of AWGN term, and $\mathbf{U} = [U_0, U_1, \ldots, U_{N-1}]$ is the DFT of impulsive noise, respectively. In order to develop practical receiver, first we assume ideal channel estimation (i.e. $\hat{H}_k \equiv H_k$). Now, after frequency domain equalization received signal can be expressed as

$$R_k^{(eq)} = R_k \hat{H}_k^{-1} = S_k + W_k \hat{H}_k^{-1} + U_k \hat{H}_k^{-1}, \quad k = 0,1,...,N-1 \tag{6}$$

The main idea of proposed algorithm is to estimate impulsive noise term $U_k \hat{H}_k^{-1}$, and subtract it from equalizer output. This can be done as described below. First, preliminary estimation of transmitted baseband symbol $\hat{S}_k$, $k = 0,1,\ldots,N-1$ is derived from equalizer output as follows:

1. Sub-carriers, which should be silent, are set to zero (e.g., DVB-T signal consist of 343 zero sub-carriers in 2K mode, and 1375 zero sub-carriers in 8K mode),
2. Sub-carriers, which are used as pilots, are replaced by known values (e.g., DVB-T signal includes 45 and 177 continues pilots in 2K and 8K mode, respectively, and each 12th sub-carrier is used as scattered pilot),
3. Sub-carriers, which are used for data transmission, are demapped to nearest positions in constellation plot.

Hereinafter, this procedure is referred to as "demapping and pilot insertion" procedure. Explicit parameters of the "demapping and pilot insertion" procedure depend on the number and position of zero and pilot sub-carriers used in a system.

Now, estimation of total noise term $D_k = W_k + U_k$, $k = 0,1,\ldots,N-1$ is performed according to following expression

$$\hat{D}_k = \hat{H}_k \left(R_k^{(eq)} - \hat{S}_k \right), \quad k = 0,1,...,N-1 . \tag{7}$$

The total noise term $\mathbf{D} = [D_0, D_1, \ldots, D_{N-1}]$ is a frequency domain representation of impulsive noise corrupted by AWGN. In a frequency domain impulsive noise can be represented by the sum of complex sinusoids

$$U_k = A_1 e^{j2\pi k t_1/N} + A_2 e^{j2\pi k t_2/N} + \ldots + A_M e^{j2\pi k t_M/N}, \tag{8}$$

where M is the number of samples affected by impulsive interference, $t_1, t_2, \ldots, t_M$ are positions of these samples and $A_1, A_2, \ldots, A_M$ are complex amplitudes of these samples. The goal of impulsive noise compensator is to estimate parameters M, $t_1, t_2, \ldots, t_M$, and $A_1, A_2, \ldots, A_M$, and reconstruct impulsive noise vector $\hat{\mathbf{U}} = [\hat{U}_0, \hat{U}_1, ..., \hat{U}_{N-1}]$. This reconstruction performed in four steps:

1. First, vector $\hat{\mathbf{D}} = [\hat{D}_0, \hat{D}_1, ..., \hat{D}_{N-1}]$ is transformed into time domain $\hat{\mathbf{d}} = [\hat{d}_0, \hat{d}_1, ..., \hat{d}_{N-1}]$ by means of IDFT;
2. Now, vector $\mathbf{d}$ consist of white Gaussian noise samples and high amplitude impulsive samples on several positions ($t_1, t_2, \ldots, t_M$). In order to detect impulsive noise samples and obtain time domain estimation of impulsive noise $\hat{\mathbf{u}} = [\hat{u}_0, \hat{u}_1, ..., \hat{u}_{N-1}]$ variance of $\hat{\mathbf{d}} = [\hat{d}_0, \hat{d}_1, ..., \hat{d}_{N-1}]$ is estimated by expression

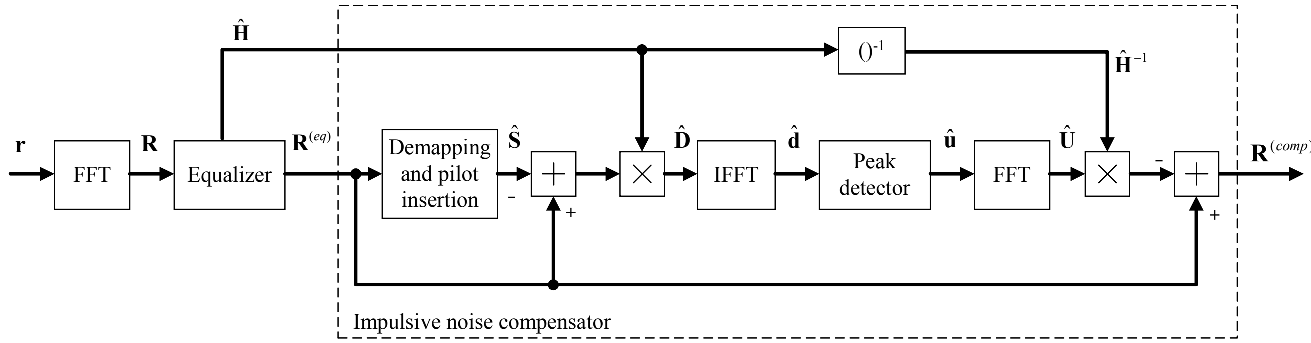

Fig. 1. Basic block-scheme of proposed impulsive noise suppression algorithm

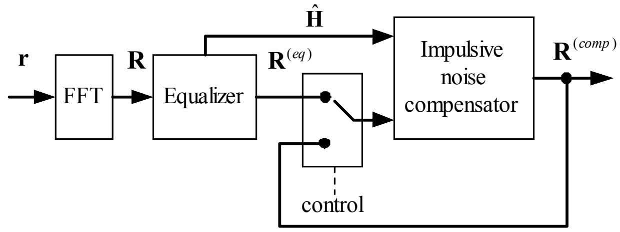

Fig. 2. Multistage (iterative) variant of proposed algorithm

$$\hat{\sigma}^2 = \frac{1}{N}\sum_{k=0}^{N-1}\left|\hat{d}_k\right|^2 ; \tag{9}$$

3. After that, the time domain representation of impulsive noise $\hat{\mathbf{u}} = [\hat{u}_0, \hat{u}_1, ..., \hat{u}_{N-1}]$ is reconstructed by following rule

$$\hat{u}_k = \begin{cases} \hat{d}_k, \text{if } \left|\hat{d}_k\right|^2 > C\hat{\sigma}^2 \\ 0, \text{otherwise} \end{cases}, \quad k = 0,1,..., N-1, \tag{10}$$

where C is threshold value that corresponds to small probability of false detection;

4. The frequency domain representation of impulsive noise $\hat{\mathbf{U}} = [\hat{U}_0, \hat{U}_1, ..., \hat{U}_{N-1}]$ is derived from $\hat{\mathbf{u}} = [\hat{u}_0, \hat{u}_1, ..., \hat{u}_{N-1}]$ by means of forward DFT or directly by expression (8).

In a final step of this procedure, estimated impulsive noise vector $\hat{\mathbf{U}} = [\hat{U}_0, \hat{U}_1, ..., \hat{U}_{N-1}]$ is multiplied by inverse channel frequency response $\hat{\mathbf{H}}^{-1} = [\hat{H}_0^{-1}, \hat{H}_1^{-1}, ..., \hat{H}_{N-1}^{-1}]$ and subtracted from equalizer output

$$R_k^{(comp)} = R_k^{(eq)} - \hat{U}_k \hat{H}_k^{-1}, \quad k = 0,1,..., N-1. \tag{11}$$

The block-scheme of proposed impulsive noise compensation algorithm is shown in fig. 1. Here, peak detector performs operations described by (9) and (10).

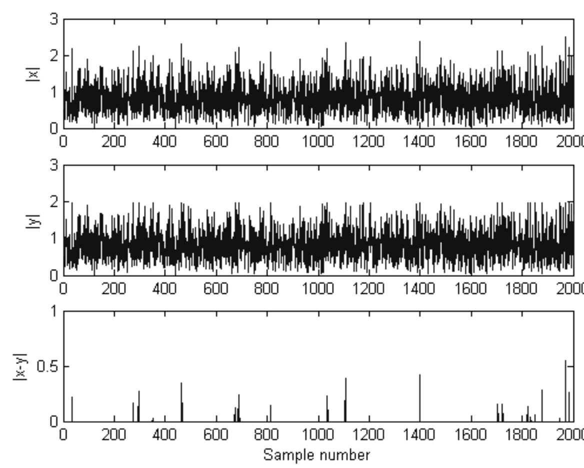

Fig. 3. Illustration of impulsive nature of clipping distortion; upper graph: absolute values of original OFDM signal, middle graph: absolute values of clipped OFDM signal (clipping ratio is 7 dB), lower graph: absolute values of difference (clipping noise)

In order to improve impulsive noise suppression above described procedure can be recursively applied to the compensated signal $R_k^{(comp)}$, k = 0,1,…,N–1. This leads to multistage (iterative) impulsive noise compensation procedure as shown in fig. 2.

It is worth to note that performance of proposed algorithm is getting worse when amplitude of impulsive noise is increased. In this case demapping process becomes less reliable and reconstructed impulsive noise vector $\hat{\mathbf{U}} = [\hat{U}_0, \hat{U}_1, ..., \hat{U}_{N-1}]$ differs significantly from the actual impulsive noise vector $\mathbf{U} = [U_0, U_1, ..., U_{N-1}]$. In this situation it is advantageous to use conventional time domain methods (e.g. limiting or blanking nonlinearity) before demodulation process to reduce energy of extremely strong impulses and improve overall performance of the algorithm.

It is interesting to note that proposed algorithm could improve system performance in the presence of clipping distortion at the transmitter, because this kind of distortions can be viewed as additive impulsive noise (fig. 3).

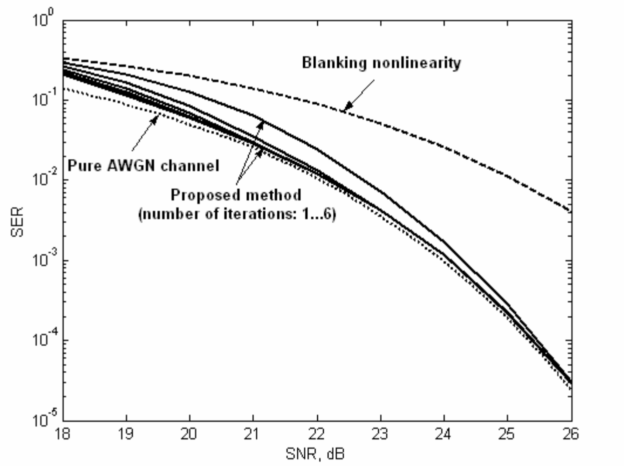

Fig. 4. Simulated performance in the Bernoulli-Gaussian impulsive noise and non-multipath channel

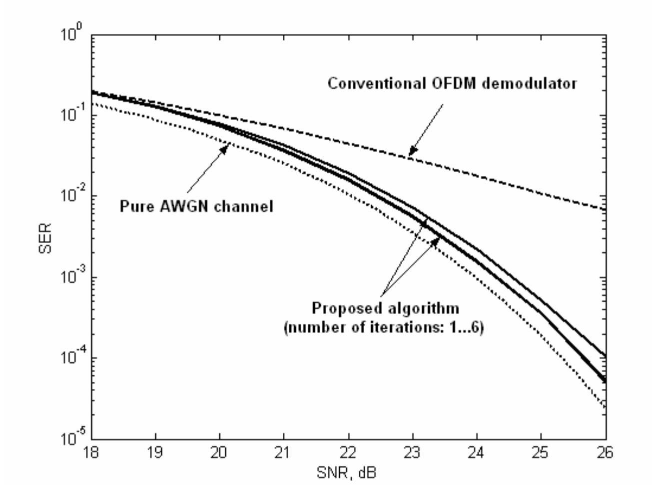

Fig. 6. Simulated performance in the channel with clipping at the transmitter (clipping ratio is 5.5 dB)

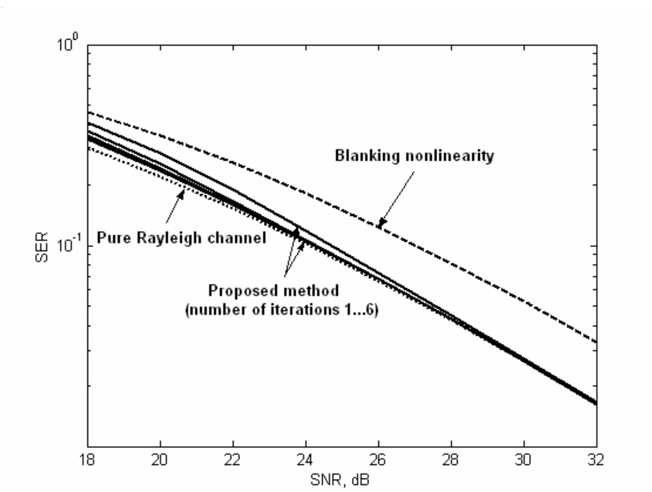

Fig. 5. Simulated performance in the Bernoulli-Gaussian impulsive noise and static Rayleigh channel

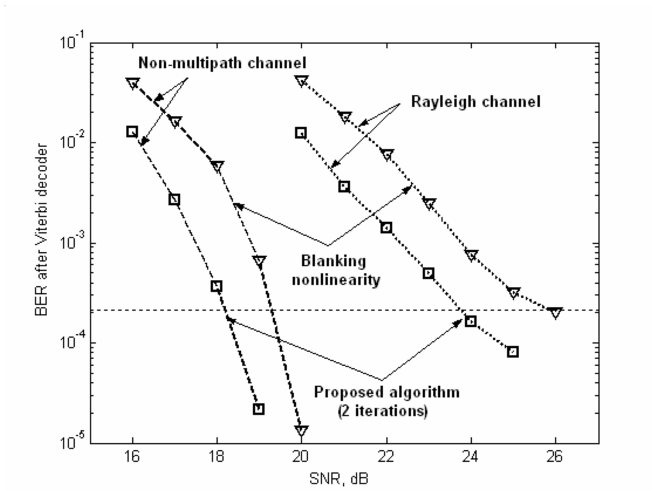

Fig. 7. Performance of proposed algorithm applied to DVB-T system with noise recorded from hairdryer

IV. SIMULATION RESULTS

The performance of proposed method has been studied by means of computer simulation. Two cases were considered. In a first case we simulated performance of uncoded OFDM system with ideal channel information. In a second case impulsive noise from hairdryer was recorded and performance of DVB-T compliant receiver was simulated.

A. Uncoded System with Ideal Channel Information

The simulation results for uncoded OFDM system with ideal channel estimation are presented in the fig. 4-6. These results are obtained for 64-QAM OFDM system with 2048 sub-carriers and 171 pilot tones (pilot sub-carriers are equally spaced in a frequency domain, i.e. each 12th sub-carrier is used as pilot). In order to simulate impulsive noise simple Bernoulli-Gaussian model was used [12]

$$u_k = b_k g_k , \qquad (12)$$

where b_k is the Bernoulli process, i.e. independent and identically distributed sequence of zeros and ones with probability $P(b_k=1) = p$, and g_k is complex zero mean white Gaussian noise. In this study we fixed standard deviation of g_k to $10\sigma_W$, where σ_W is standard deviation of background noise w_k. Impulse probability was fixed to p=0.01. Two types of channels were simulated. First one is non-multipath channel, whereas second is Rayleigh channel defined in DVB-T specifications [2]. In order to reduce amplitude of strong impulses blanking nonlinearity (4) was applied to the time domain signal before OFDM demodulator.

As one can see (fig.4 and fig.5) proposed algorithm shows large performance improvements with 1-3 iterations, and for high signal-to-noise ratio values the performance of proposed method in impulsive noise channels is nearly the same as the performance in channels without impulsive noise. Simulation results for channel with clipping at the transmitter (fig.6) also show significant improvements comparing to conventional OFDM demodulator.

B. *DVB-T Example*

In this section we present the results of simulation for DVB-T compliant receiver. In this study, the DVB-T transmitter and receiver were configured for 2K, 64-QAM mode operation, guard interval was set to 1/32 and inner code rate was 2/3. Typical impulsive noise from hairdryer was recorded on hard disk and used as a noise source for these simulations. Number of iterations in proposed impulsive noise compensation algorithm was fixed to two. As in the previous study blanking nonlinearity was applied to the time domain signal before OFDM demodulation. Bit error rate (BER) after Viterbi decoder was measured and plotted vs. signal-to-noise ratio (SNR). The results of simulations for two channel types (non-multipath and Rayleigh channel) are presented in fig. 7. From the fig. 7, it is seen that for this particular case quasi-error free condition (BER$\approx 2\cdot 10^{-4}$) can be archived with 1.1 dB (non-multipath channel) and 2.2 dB (Rayleigh channel) lower signal-to-noise ratio if proposed algorithm is used.

V. CONCLUSION

Impulsive noise can cause serious problems in reception of OFDM signals, for example, DVB-T signals modulated with 64-QAM symbols. In this paper algorithm suitable for suppression of impulsive noise in OFDM based communication systems is presented. We have presented the results of simulation, which characterize the performance improvements in a DVB-T receiver. Combination of proposed algorithm with error correction coding can further improve performance of impulsive noise compensation. For example, error correction coding/decoding process can be included in the demapping and pilot insertion block in order to improve preliminary estimation of transmitted symbols. Also, frequency domain estimation of impulsive noise can be used to provide reliable channel state information for soft Viterbi decoder. Finally, the implementation of proposed impulsive noise compensator is computationally intensive, which would require significant amount of silicon area. Reduced complexity implementations would therefore be of interest for future work.

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