Nonuniform Sampling Delta Modulation - decoding problems

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Abstract: The problems of reconstructing signals coded with use of adaptive sampling delta modulation have been shown in the paper. Synchronization procedure of the algorithm "with return to the basic sampling instant" has been analyzed. Investigations of the bit error robustness of the delta system with sampling adaptation have been presented. Special interest was aimed at the behaviour of Non-uniform Sampling Delta Modulation system with reference to the transmission errors. The phenomena of the parasitic oscillation and dynamic regulation of the *dc* level in the reconstructed signal, as a result of the channel errors, have been discussed. Parasitic oscillation parameters have been analyzed. It has been found that the algorithm "with return to the basic sampling instant" improves the bit error robustness of the adaptive sampling delta modulation.

Key-Words: Adaptive delta modulation, Non-uniform sampling, Error compensation, Robustness, Parasitic oscillation.

1. Introduction

The idea of non-uniform sampling used in delta modulation in order to reduce the data rate of the coded source is the topic that is still not completely understood. An algorithmic robustness of ADM converters with sampling adaptation (NSDM) has not been the object of numerous simulation and analytical investigations. The compression properties of DM systems are based on a rule that the A/D conversion is accomplished on the removed redundancy input process. The efficient samples decorrelation is usually made using the adaptation procedures [1, 2]. The step size adaptation is the simplest one. The uniform sampling ADM systems allow reaching a wide dynamic range of the constant SNR_{max} ratio, however its value is not higher then in case of **L**inear **D**elta **M**odulation [1]. It is a result of this that coding methods with uniform sampling do not utilize all compression possibilities involved within the ADM systems [2, 3, 4].

At present there are some hopes of achieving the better compression results while converting, in the real time, speech signals, TV, video, by means of a delta converters with sampling adaptation [4, 5, 6]. Earlier analytical investigations theoretically showed an improvement of the quality of non-stationary input process conversion due to the application of non-uniform sampling [4, 7]. The great advantage of the variable-rate delta modulations is high data-protection performance [4, 6]. The NSDM schemes have been proposed and have been studied in [5, 6].

A voice encoding system based on ADM codecs was already used in the Shuttle system, because of its tolerance to channel errors [8]. Today, there are many military and commercial systems using this method [9]. Bluetooth^{TM} [10] employs a low-cost, 64-kbps **C**ontinuously **V**ariable **S**lope **D**elta (CVSD) the modulation scheme. Several semiconductor manufacturers produce specialized telecommunication IC's based on ADM codecs.

An important problem concerning the delta converter is the process of correct reconstruction of the coded signal in the presence of channel noise. Algorithmic robustness of ADM converters with uniform sampling has been the object of numerous simulation and analytical investigations [4, 8, 11].

The adaptation decoders operating on the basis of the Jayant, Abate, Song algorithms have been previously characterized in other papers [1, 2, 8]. It has been proved that converters in which the syllabic algorithm had the greatest robustness [4].

The problems of reconstructing signals that are coded with use of adaptive sampling delta modulation after transmission in noisy line have not been an object of studies so far.

In the present paper the bit error robustness of the NSDM converters with the Zhu algorithm ("with return to the basic sampling interval" [5]) was described. In particular:

- The synchronization and reaction of NSDM demodulator on the burst noise during slow signal changes ("silent gap");

- Dynamic regulation of the *dc*-level that appears in the reconstructed signal after transmission in noisy line;
- Analysis of parasitic oscillation when an erroneous bits take place in the silent gap.

2. Principle of the NSDM technique

The NSDM conversion predicts a value of sampling interval depending on the rate of changes of the input waveform. The quantization step-size q remains constant. Fig.1 shows the functional diagram of the NSDM coder and decoder.

The staircase Y_i in the NSDM modulator estimates the input signal X_i and can be expressed as:

$$Y_{\mathrm{i}} = \sum_{k=1}^{i-1} \varDelta(T_{S\,k})q \qquad (1)$$

Where: $\varDelta(T_{S\,k}) = \mathrm{sgn}[X_{\mathrm{k}} - Y_{\mathrm{k}}]$ (2)

q - the quantization step-size.

The sampling instant $T_{S\,i} = t_i - t_{i-1}$ varies according to the characteristics of X_{i} and is equals:

$$T_{S\,\mathrm{i}} = \begin{cases} K1 \cdot T_{\mathrm{S\ i\text{-}1}} & \text{dla} \quad \mathrm{MIF} > 1 \\ T_{\mathrm{s\_start}} & \text{dla} \quad \mathrm{MIF} = 1 \\ K2 \cdot T_{S\ \mathrm{i\text{-}1}} & \text{dla} \quad \mathrm{MIF} < 1 \end{cases} \qquad (3)$$

Where: $K1 < 1 < K2$ (4)

Formula (3) represents the 3-bit Zhu adaptation algorithm of the sampling interval changes, with the MIF logic, that has been shown in the Tab.1 [5].

As can be seen, the NSDM output coder stream carries the information about the sampling instant of the modulator. So that in the demodulation process the irregular staircase signal can be recovered [5].

Table1. MIF (*Modified Interval Function*) Logic

b_{i}	$b_{\mathrm{i-1}}$	$b_{\mathrm{i-2}}$	MIF	Description
0	0	0	<1	T_{Si} ↘
0	0	1	=1	$T_{\mathrm{Si}}=T_{\mathrm{s\_start}}$
0	1	0	>1	T_{i}↗
0	1	1	=1	$T_{\mathrm{Si}}=T_{\mathrm{s\_start}}$
1	0	0	=1	$T_{\mathrm{Si}}=T_{\mathrm{s\_start}}$
1	0	1	>1	T_{Si}↗
1	1	0	=1	$T_{\mathrm{Si}}=T_{\mathrm{s\_start}}$
1	1	1	<1	T_{Si} ↘

3. Synchronization procedures of the NS-DM decoders

Based on observation and analysis of the NSDM decoding mechanism, it was affirmed, that synchronization processes after start of the decoding in any position relative to the input data beginning, and after one bit distortion in a data transmission stream, has the running in a similar way [3].

Depending, whether the decoding beginning in the silent gap or out of it, the great differences in the operating of the NSDM decoder appear. The silent gap is the part of speech when its value is constant or changes very slowly. Analysis presented in the

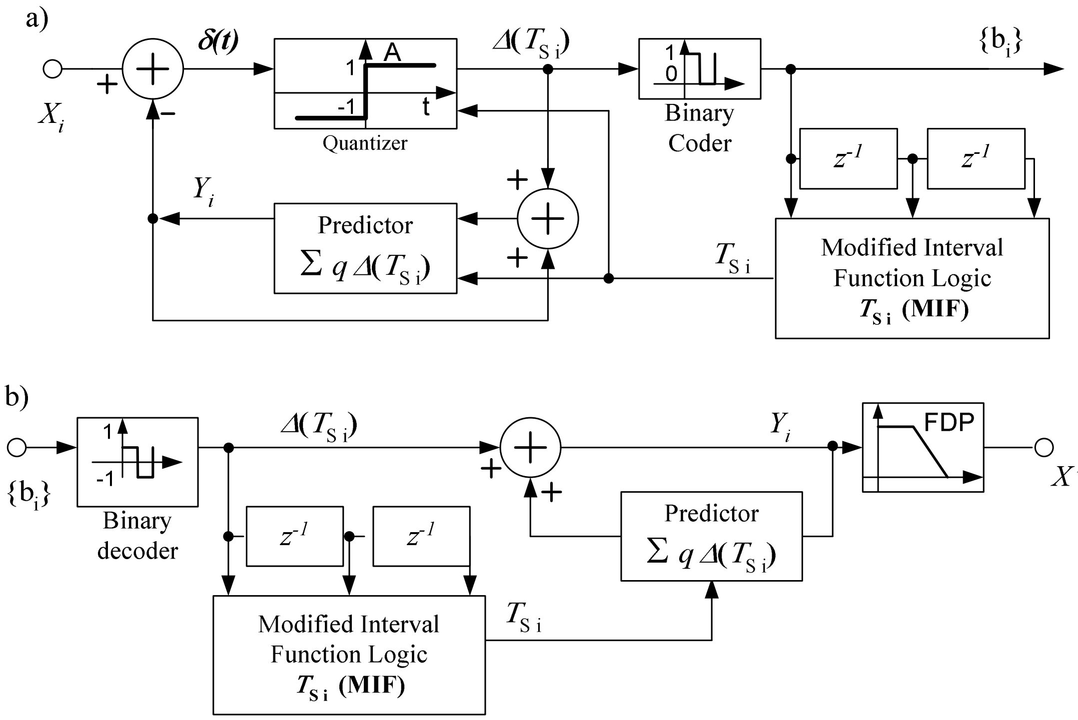

Fig.1. Block diagram of NSDM modulation: a) coder; b) decoder.

chapter 3 and 4 are related to the behavior of the decoder being started in the “signal existence”.

The quality estimation of the signal reconstructed by the non-uniform sampling delta decoder that starts in the moment arbitrarily chosen by the user was the first task of the research.

The Figure 2a shows NSDM decoder block diagram implemented as standard logic cells, so it is technologically independent, and simulated in PSPICE.

The NSDM decoder (Fig. 2a) comprises D flip-flop, MIF adaptation logic, CD clock divider, DAC converter and LPF low pass filter. These elements, except low pass filter, are substantially the same and operate in the same manner as the corresponding elements in NS-DM coder. The flip-flop with recovered sampling clock CLK_VARout functions merely to retime incoming data stream [12, 13].

Synchronization procedures of the NSDM decoders with the Zhu algorithm [Tab.1] are determined by two basic features [14]:

- The appearance in input stream $\{b_i\}$ of the time interval T_{S_start} (for 3-bits Zhu algorithm T_{S_start} appears after each sequence of bits: 110, 001, 011, 100)
- The time relations between bit streams $\{b_i\}$, CLK_VARout, $\{Q_i\}$ resulting from the NS-DM adaptation algorithm.

At the beginning of the decoding process the input streams $\{b_i\}$ and the sequence of bits $\{Q_i\}$ are not synchronous, but because of the mentioned earlier NSDM decoder features, the synchronization between $\{b_i\}$ and $\{Q_i\}$ streams always comes up to.

Generally, the resynchronization process consists of two stages (Fig. 2b):

1) The appearance of the same sign bit in the streams $\{Q_i\}$ and $\{b_i\}$,

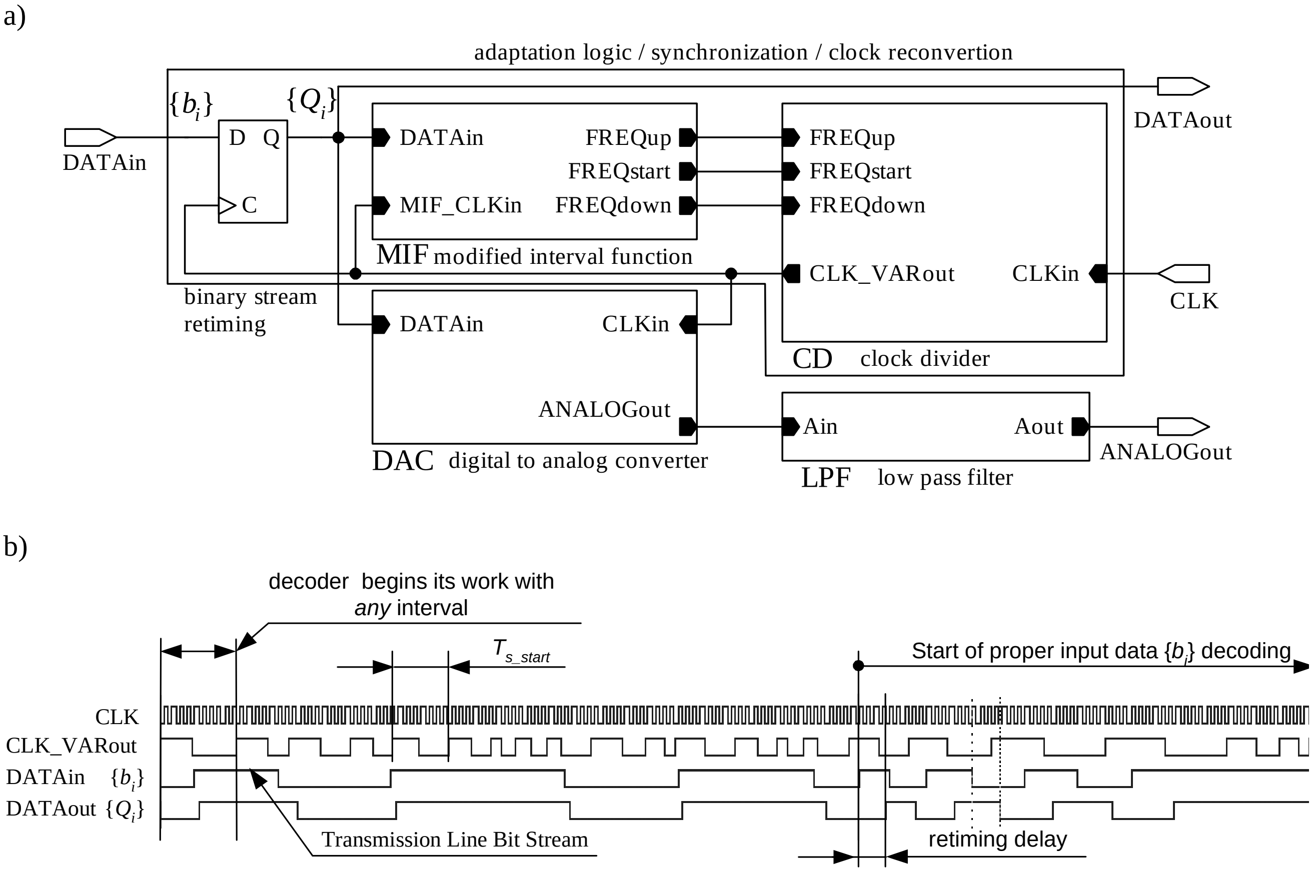

Fig.2. Data synchronization mechanism: a) block diagram of decoder, b) time waveforms.

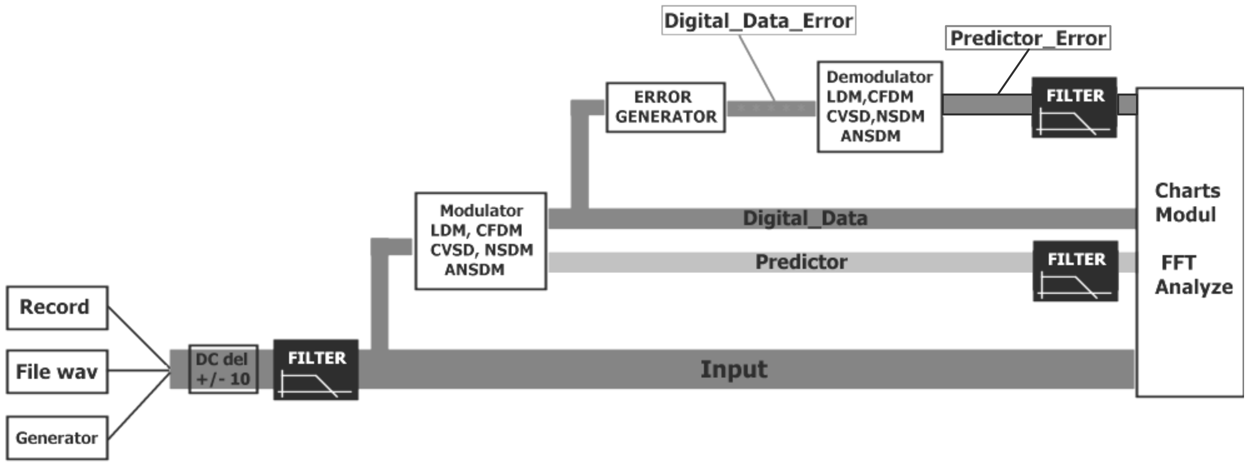

Fig.3. Block diagram of the 1-bit delta converters emulation system.

2) The beginning of the T_{S_start} bit in the $\{b_i\}$ stream with the opposite sign to the $\{Q_i\}$ sign, what caused that decoder forces the T_{S_start} time interval in $\{Q_i\}$ stream.

If the both stages appear, the timings $\{b_i\}$ and $\{Q_i\}$ come up to the full synchronization (Fig.2.b).

In the monograph [3] has been shown that the NSDM decoder with the Zhu algorithm after disappearance of channel errors always leads to resynchronization.

4. Transmission gaps influence on the reconstructed signal quality

Describing advanced delta adaptation coders by means of the clear analytical model is very difficult. Computations lead into the iterative solving of complex sets of equations with using the computer-aided methods for calculating [11]. In the face of a lot simplifying assumptions which must be assumed to their solving, the analytic method does not provide in this case more accurate results than algorithmic simulations.

The simulators are very useful for monitoring and measurement the channel errors influence on the received voice quality. It provides a valuable possibility to insert the error bits into the Transmission Line (Fig.4), makes the calculation of *SNR* value for all delta converters and average bit rate BR_{avg} for NSDM. The simulator allows the monitoring and sound reproduction of the input source (sine, triangle, rectangle, and speech source approximated by a white noise signal having the integrated power spectrum and Gaussian amplitude distribution), predictor waveform, and reconverted signal (Fig.3).

The error robustness investigation of the NSDM decoders with return to the basic sampling interval has started from analysis of the decoder reaction on the transmission gap.

The transmission gap should be treated as a particular case of a channel bit errors (burst series), in which a certain numbers of successive output bits assume the same value. The effect of a transmission gap on the signal reconstruction by the NSDM

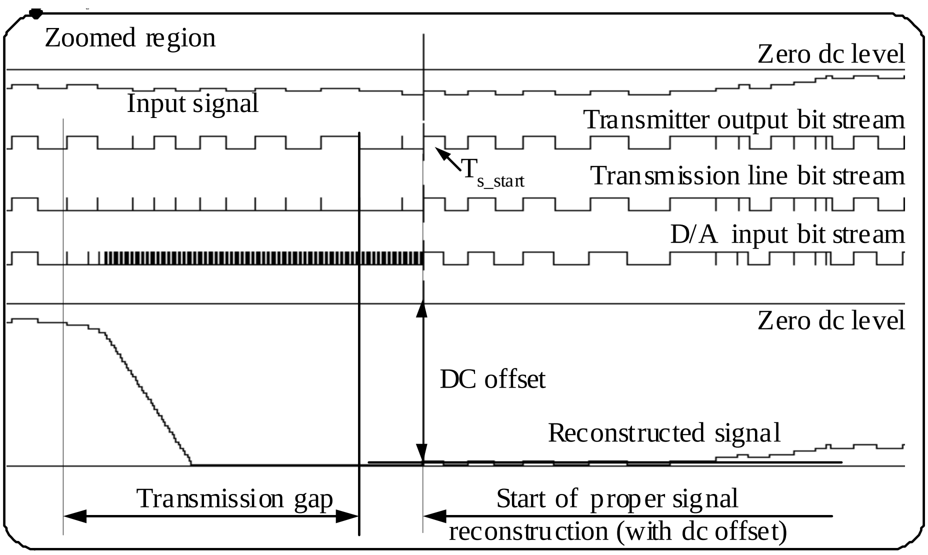

Fig.4. Effects of the transmission gap - detailed view.

demodulator has been presented in the Fig.4. According to the MIF logic (Tab.1), the value of the reconstructed signal is increased (or decreased - Fig.4) during the gap instant. The rate of approximation slope is greater and greater because adaptation increases the sampling frequency. After the time period depending on the decoder parameters, the approximating curve attains the maximum (or minimum - Fig.4) value. If the transmission gap is continued, the approximating curve keeps, all the time, the boundary value.

When the transmission gap is terminated, the bits in the „Transmission line" have the values corresponding to the modulated signal. Next, the well-known process of the dynamic shift of the *dc* level begins and continues until the moment when a variable component (*ac)* of the reconstructed signal occurs without distortions (Fig.5).

The process of the *dc* level shifting can be simply converting of the positive and the negative jumps is continued until the moment when the *dc* level increased so much that all negative jumps can be correctly reconstructed. In particular case, when the maximum amplitude of the input signal has the amplitude equal to the half of the range, the *dc* level (in the established state) returns to the zero value. The described behavior of the *dc* level (eliminating the cutting off the peaks of the signal) takes place in each method of the delta conversion (with negative feedback loop).

The rate of changes of the *dc* level being a result of a transmission gap depends mainly on the signal parameters.

The NSDM decoder with Zhu algorithm after disappearance of bit errors always leads to resynchronization [3], i.e. correct reconstructing variable component (*ac*) of the signal. The time of the NSDM decoder synchronization after the bit

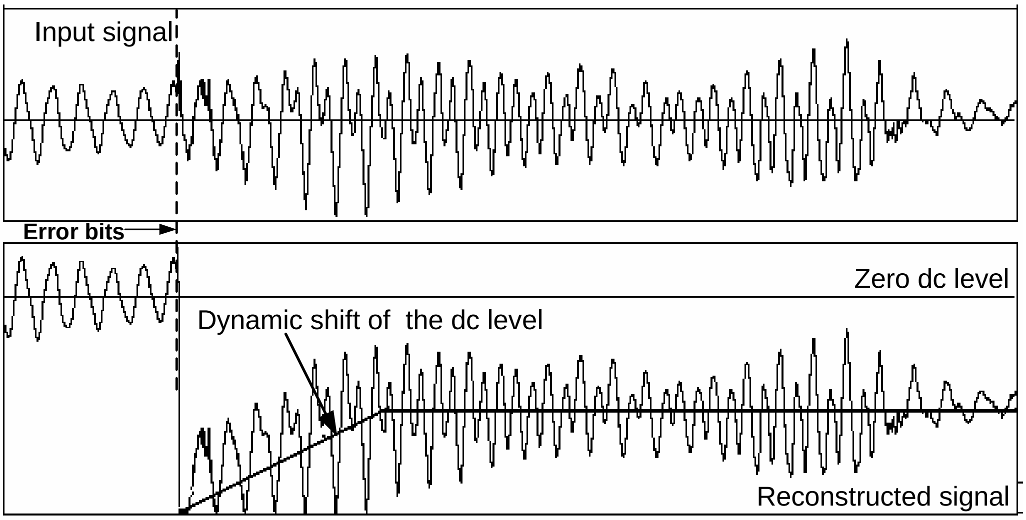

Fig.5. The dynamic shift of the *dc* level after the transmission gap is terminated.

explained. Let us assume that a transmission gap caused a shift of the approximating signal to the minimum value (Fig.5). If now the new coder output bit has the logical value 0, the subsequent value of the approximating signal will remain equal to the minimum value[1]. If however, the subsequent bit has the logic value "1", than the value of approximating signal will increase. As a result, the positive jumps of the output stream will be converted correctly, whereas from among the negatives ones only those for which the approximating signal does not exceed the minimum value. This non-balance in the proper

errors termination does not depend on the fact how long these errors lasted and which bits were interfered[2].

It can be proved by the following argumentation:

Algorithms „with return of the sampling interval to the basic value" cause that every equalization of the approximating and input signals value forces the return of the sampling interval to its basic value (Tab.1). Consequently, the process of the input signal reconstructing consists of many independent

[1] Similarly values smaller than the negative supply voltage cannot be obtained in the hardware version.

[2] Time of regaining the synchronization by the decoder depends on the fact which bit was interfered. This time depends, above all, on the sequence of bits $\{b_i\}$, i.e. also on the input signal (Fig.2b).

parts whose start and end are determined by the basic value of the sampling interval. After the bit errors are terminated, each sequence of the bits indicating the return of the sampling instant to the basic value (T_{S_start}) begins the fragment of correct reconstructing, independent of the previous fragments (Fig.4). Bit errors that appeared in the previous fragments do not affect the correct decoding of the *ac* signal component in the new fragments.

In a situation when the end of the transmission gap occurs beyond the silent gap the resynchronization time is very short. The full synchronization takes place at the moment of the valid[3] T_{S_start} appearance in the „Transmission line stream" (Fig.4). In the opposite case, resynchronization takes place only after the end of the silent gap. A detailed description of the principles of resynchronization has been presented in [3].

5. Effects of the bit errors occurrence in the silent gap

Great differences in the operating of the NSDM decoder are observed in dependence on the fact whether the erroneous bits take place in the silent gap or out of it. When a false bit occurs in the silent gap the demodulator produces parasitic oscillations (Fig.6).

Oscillation is caused by decoder interpretation of error bits. During the transmission gap the synchronization between transmitter and receiver is lost. The “Line bit stream” (Fig.6) that represent series of maximal sampling intervals T_{S_max} is interpreted by decoder as a series of the same sign bits, because $T_{S\_start} << T_{S\_max}$. So according to decoder algorithm, the sampling instant can quickly reach its lowest value T_{S_min}. When the transmission gap (errors) is finished, the bits in the “Transmitter output” and in the “Line bit stream” are again the same logic value (Fig.6). While bit on the “Transmitter output” (Fig.6) goes to opposite logic value (successive T_{S_max} interval starts) the direction of the prediction is also changed. In result, the oscillation into reconstructed signal is introduced (Fig.6). Its magnitude is limited by the decoder parameters. The waveform (Fig.4) shows that false reconstruction occurs until appearance of the sampling instant T_{S_start}. Next synchronization

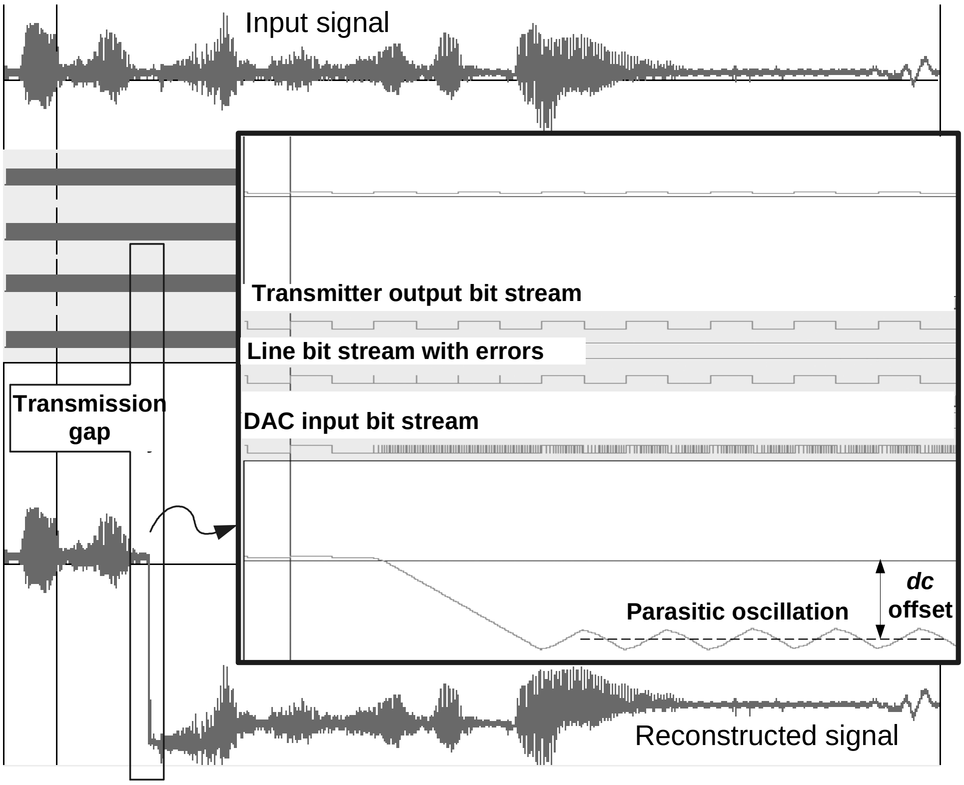

Fig.6. Parasitic oscillation and dc offset.

[3] The logic value of the T_{S_start} have to become the opposite to the continuing bit in the "D/A input bit stream" (Fig.4).

process starts, and the *dc* offset is the only noticeable effect of errors.

6. Frequency and amplitude of the parasitic oscillation

Based on the decoder algorithm (Fig.7), a frequency f_{osc} equals:

$$f_{osc} = \frac{1}{2 \cdot T_{S\_max}} \quad (5)$$

Where: T_{S_max} – maximal value of sampling instant.The f_{osc} value is almost constant, similar as to peak to peak amplitude A_{pp} of the parasitic oscillation.

Parasitic oscillation waveform

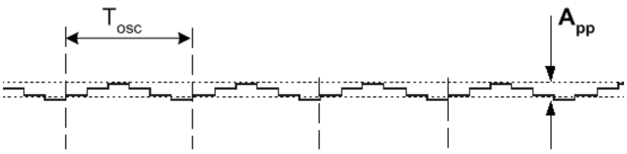

Fig.7. Example of the parasitic oscillation waveform for 3-bit Zhu algorithm.

It maximum is determined by all others decoder internal parameters (Fig.8)

$A_{pp} = f(T_{s\_start}, T_{s\_max}, T_{s\_min}, K1, K2):$

$$A_{pp} = q \cdot n_{tot} = q \cdot (n_{a\lg} + n_{adp} + n_{\min}) \quad (6)$$

Where:

n_{alg} –the number of algorithm bits ($t_o = n_{alg} T_{S\_start}$)

n_{adp} –the number of sampling instants during adaptation time (t_{adp}) i.e. from T_{s_start} to T_{s_min}

n_{min} –the number of T_{S_min} periods during t_{min} time,

n_{tot} – total number of sampling instant during T_{s_max}.

Taking into consideration relation between T_{s_max}, T_{s_start}, and T_{s_min}, the total number of sampling instant during parasitic oscillation can be found. Thus the maximal value of A_{pp} is:

$$A_{pp} = q \cdot \left\lceil n_{a\lg} + \left\lceil \frac{\log \frac{T_{s\_min}}{T_{s\_start}}}{\log K1} \right\rceil \right\rceil + \\ + q \left\lceil \frac{T_{s\_max} - n_{a\lg} \cdot T_{s\_start} - T_{s\_start} \dfrac{1-(K1)^{\frac{\log \frac{T_{s\_min}}{T_{s\_start}}}{\log K1}}}{1-K1}}{T_{s\_min}} \right\rceil \quad (7)$$

Where: symbol $\lceil \bullet \rceil$ means rounding to the first nearest integer.

7. Summary

Simulation investigations of the synchronization procedures in the adaptive delta demodulators with the non-uniform sampling have been carried out.

It has been proved in the present paper that the NSDM decoder with the Zhu algorithm, after disappearance of channel errors, always leads to resynchronization.

After finishing of the bit errors (eg. transmission gap) an appearance of the T_{S_start} interval in the Transmitter output, an active edge of the CLK_VARout (Fig.2b) is the necessary and sufficient condition for resynchronization of the NSDM demodulator. After the resynchronization, automatic regulation of the *dc* level takes place, so the variable component (*ac* signal) can be reconstructed without distortions. It results from the algorithm operation and negative feedback loop of the delta converter. This behavior is analogical to the automatic regulation of the quiescent point in amplifiers, as well as to the systems with the dynamic biasing.

The change of the *dc* level in the reconstructed signal is the major consequence of transmission gaps (or another kind of bit errors) for all delta systems. Therefore its proper filtration at the delta demodulator is needed [6, 15, 16].

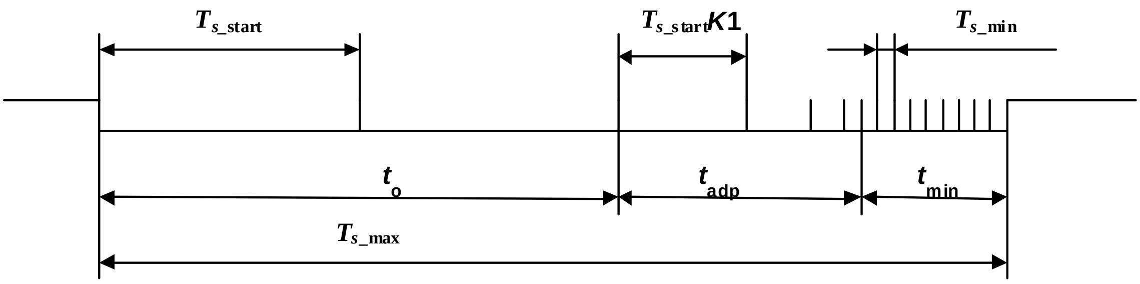

Fig.8. Diagram of the sampling instant adaptation procedure.

Distortions of the *ac* envelope last until the time of *dc* level is suitably changed. The time of the *dc* level automatic regulation depends primarily on the properties of the input signal.

Based on simulation studies it was found that the algorithm with returning to the basic sampling interval decreases slightly the accuracy of signal reconstruction, but reduces the distortions resulting from the gaps (or other channel errors) occurred during the transmission.

After an error bits occurrence in the silent gap, NSDM demodulator can produce parasitic oscillations (Fig.6). The effective way of elimination of them is assuming such value of f_{S_start}, that:

$$f_{S\_start} < 2 f_{S\_min}$$

Other ways lead only to the minimization of the oscillation amplitude. There are the following ways:

- The applying of the suitable output filter in the decoder;
- The applying of minimal acceptable values of the quantization step-size.

The optimization of the NSDM converter parameters with respect to parasitic oscillation minimization is opposed to those, which pursues the maximization of the *SNR* ratio and the minimization of the BR_{avg}.

Problems of decoding signals of bits stream with variable clock frequency, in presence of the channel noise have not been an object of studies so far.

The investigations executed so far do not fully solve out the problems concerning all features of the NSDM modulators. Further studies of new coding algorithm Continuously Non-uniform Sampling Delta Modulations (CVNS-DM), which joins the robustness of the CVSD algorithm and coding efficiency of the NSDM algorithm, are still needed [2, 3, 4].

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