Simulation analysis of the transmission gaps influence on the decoding processes in the NS-DM system

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Abstract: Investigations of the bit error robustness of the delta systems with sampling adaptation have been presented. In particular, the behaviour of **N**on-uniform **S**ampling **D**elta **M**odulation system with reference to the transmission gap has been described. The phenomena of overloading and dynamic regulation of the *dc* level in the reconstructed signal, as a result of the transmission gap, have been discussed. The Zhu algorithm has been analyzed. It has been found that in the presence of the channel noise, the Zhu algorithm improves the quality of the decoded signal.

Key-Words: Delta modulation, Variable-rate sampling, Error compensation, Robustness.

1 Introduction

The idea variable sampling use in delta modulation 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 an essential 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, 2]. It is a result of this that coding methods with uniform sampling do not utilize all compression possibilities involved within the ADM systems [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 [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. Today, there are many military and commercial systems using this method [8, 9, 10]. Bluetooth^{TM} [11] 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, 12, 13, 14].

The adaptation decoders operating on the basis of the Jayant, Abate, Song algorithms have been characterized in detail in the papers [1, 12]. It has been proved that converters in which the syllabic [4] or the instantaneous Zhu algorithm [5] are revealed the greatest robustness.

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 many researches so far.

The bit error robustness of the NSDM converters with the Zhu algorithm ("with return to the basic sampling interval" [5]) has been presented in the paper. In particular, the reaction of NSDM converter on the transmission gap has been described.

The overload and dynamic regulation of the *dc*-level that appear in the reconstructed signal as an effect of the burst noise (the transmission gap) have been analyzed.

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 k remains constant. Fig. 1 shows the functional diagram of the NSDM coder and decoder.

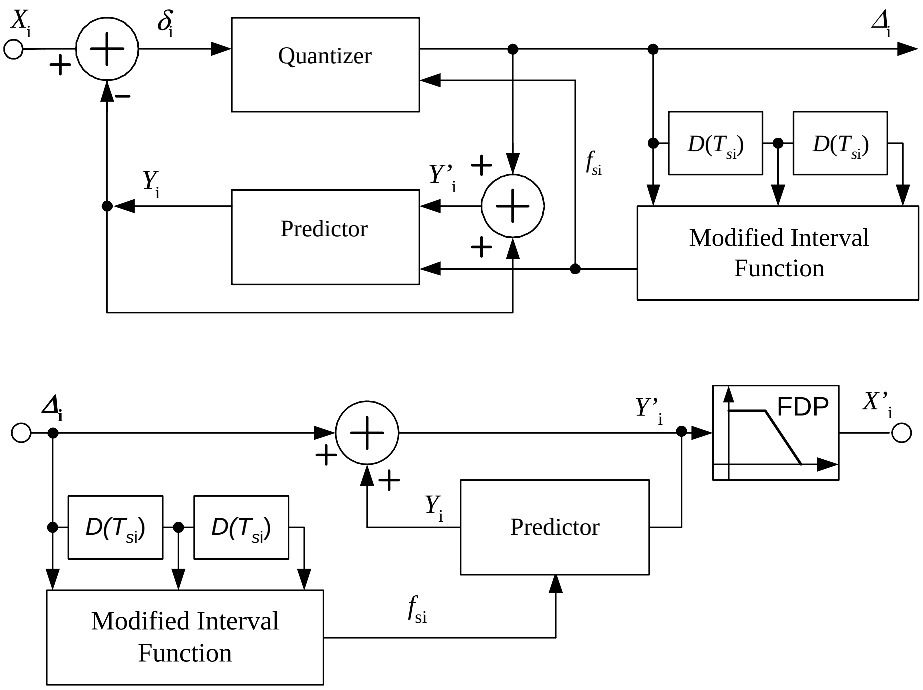

Fig.1. Block diagram of NSDM modulation: a) coder; b) decoder

For the input signal X_i the staircase in the NSDM modulator can be expressed as:

$$Y_\mathrm{i} = \sum_{k=1}^{i-1} \Delta_\mathrm{i} k \qquad (1)$$

Where $\Delta_\mathrm{i} = \mathrm{sgn}[X_\mathrm{i} - Y_\mathrm{i}]$ (2)

k- quantization step-size.

The sampling instant $T_i = t_i - t_{i-1}$ varies according to the characteristics of X_i and can be expressed as:

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

Where $Q < 1 < \mathrm{P}$ (4)

The MIF logic has been shown in the Table 1. Formula (3) represents the 3-bit Zhu adaptation algorithm of the sampling interval change.

As can be seen, the NSDM output coder stream carries the information about the sampling timing 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_i ↘
0	0	1	=1	$T_\mathrm{i}=T_{\mathrm{s\_start}}$
0	1	0	>1	T_i↗
0	1	1	=1	$T_\mathrm{i}=T_{\mathrm{s\_start}}$
1	0	0	=1	$T_\mathrm{i}=T_{\mathrm{s\_start}}$
1	0	1	>1	T_i↗
1	1	0	=1	$T_\mathrm{i}=T_{\mathrm{s\_start}}$
1	1	1	<1	T_i ↘

3. Synchronization procedures of the NS-DM decoders

Based on observation and analysis of the NSDM decoding mechanism, it was affirmed, that resynchronization 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 [15].

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.

Depending, whether the decoding beginning in the "silent gap" or out of it, the great differences in the operating of the NS-DM decoder appear. The "silent gap" is the part of speech when its value is constant or changes very slowly. The timings on Figure2 are related to the behavior of the decoder being started in the "signal existence".

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

The NS-DM decoder (Fig. 2. a) 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.

Synchronization procedures of the NS-DM and ANS-DM decoders with the Zhu algorithm [Table 1] are determined by two basic features:

- 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 NS-DM decoder features, the synchronization between $\{b_i\}$ and $\{Q_i\}$ streams always comes up to.

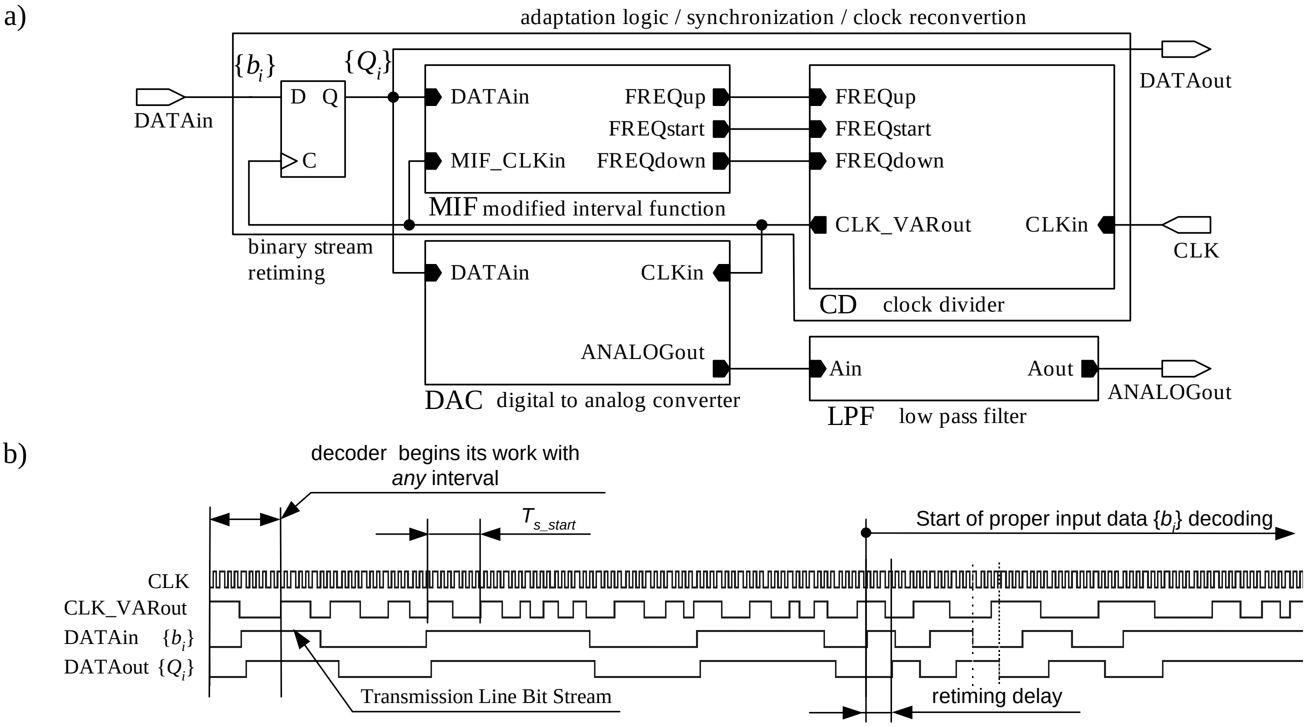

Fig.2. Data resynchronization mechanism: a) block diagram of decoder, b) time waveforms.

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

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

The transmission gap should be treated as a particular case of a channel bit errors, in which a certain numbers of successive output bits assume the same value.

In the Fig. 3 the effect of a transmission gap on the signal reconstruction by the NSDM demodulator has been presented.

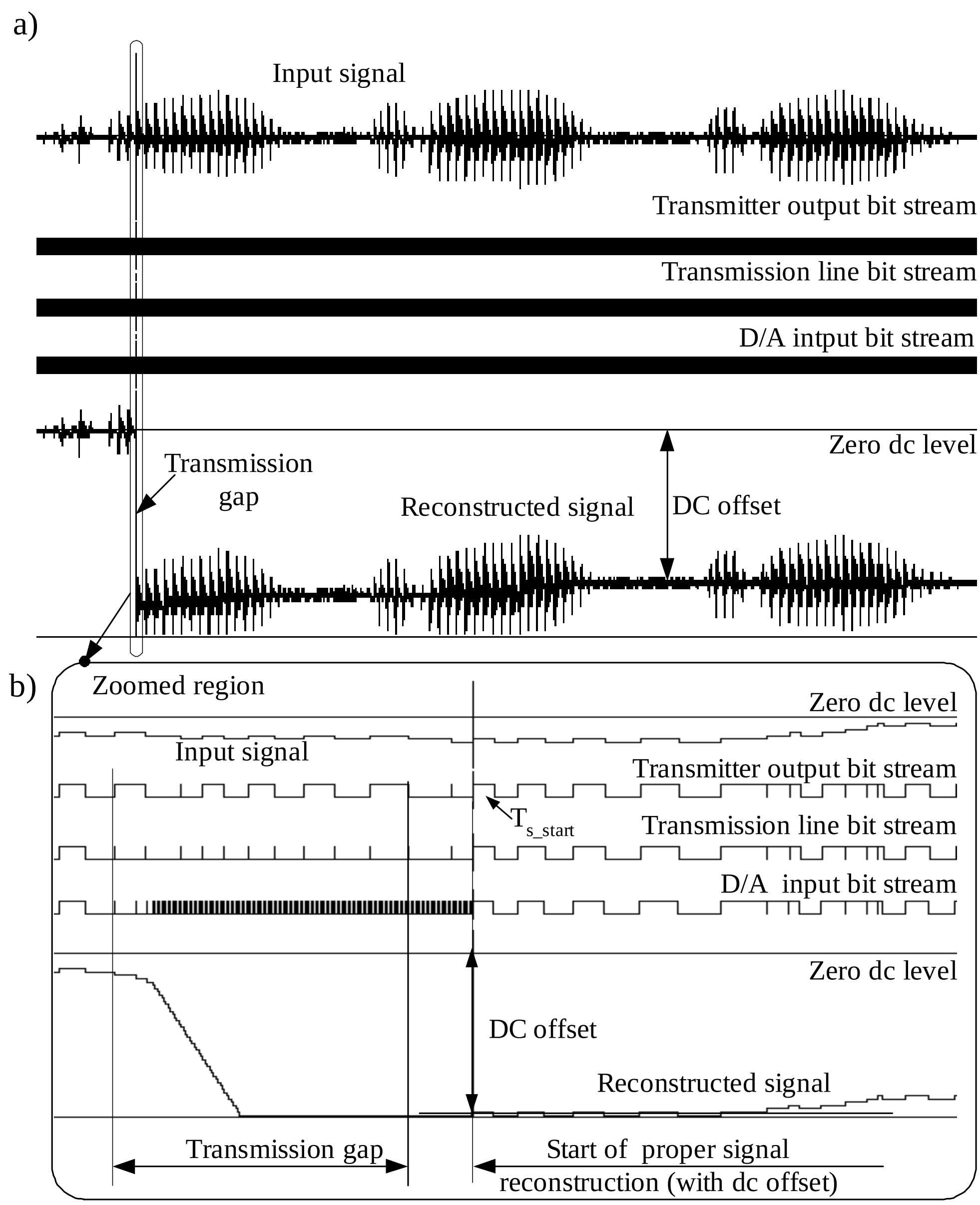

Fig.3. Effects of the transmission gap a) global view, b) detailed view.

In the *Mod Delta* simulator [3] the transmission gap is simulated as a sequence of the mark "0" only or of number "1" only (Fig. 3 b).

Then, according to the MIF logic (Tab.1), the value of the reconstructed signal (Fig.2) is decreasing (or increasing) 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 (Fig.3) attains the minimum (or maximum) 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 in the electronic circuits begins and continues until the moment when a variable component (*ac)* of the reconstructed signal occurs without distortions.

The process of the *dc* level shifting can be simply explained. Let us assume that a „transmission gap” caused a shift of the approximating signal to the minimum value (Fig.3). 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

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

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 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 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 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 begins the fragment of correct reconstructing, independent of the previous fragments (Fig 3.b). 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 so-called „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.3.b). 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. Summary

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

The phenomena of overloading and dynamic regulation of the dc level in the reconstructed signal as a result of transmission gaps have been discussed.

On the basis of simulation investigations it was found that the Zhu algorithm (with returning to the basic sampling interval) decreases slightly the exactness of signal reconstruction, but reduces the distortions resulting from the gaps (or other channel errors) occurred during the transmission.

After finishing of the gap an appearance of the T_{s_start} interval in the $\{b_i\}$ and occurrence in the course of its continuation - an active edge of the CLK_VARout (Fig.2.b) is the necessary and sufficient condition for resynchronization of the NSDM demodulator.

After the resynchronization, automatic regulation of the *dc* level takes place, so that the variable component (*ac* signal) can be reconstructed without distortions. It results from the algorithm of the delta converter operation. 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 for delta systems. Therefore its proper filtration at the delta demodulator becomes an important task.

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

The investigations executed so far do not fully solve out the problems concerning all features of the NS-DM modulators.

[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.

[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" [3].

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 NS-DM algorithm, are still necessary.

This research would allow to state whether the elimination of "the returning to the basic sampling interval" from the NS-DM algorithm keeping the high quality and high tolerance to channel errors, is possible.

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