New Methods of Stereo Encoding for FM Radio Broadcasting Based on Digital Technology

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Abstract. *The article describes new methods of stereo encoding for FM radio broadcasting. Digital signal processing makes possible to construct an encoder with properties that are not attainable using conventional analog solutions. The article describes the mathematical model of the encoder, on the basis of which a specific program code for DSP was developed. The article further deals with a new method of composite clipping which does not cause impurities in the output spectrum, and at the same time preserves high separation between the left and right audio channels. The application of the new method is useful mainly where there are unwanted signal overshoots on the input of the stereo encoder, e.g., in case of signal transmission from the studio to the transmitter site through a route with psychoacoustic lossy compression of data rate.*

Keywords

Stereo encoder, stereo generator, composite clipping, digital encoding, digital signal processors, DSP.

1. Introduction

Currently, for the transmission of an audio signal between the central site of the radio station and the FM transmitter site, a channel with psychoacoustic lossy compression of the data rate (e.g. using MPEG or Musicam algorithm) is usually used. Such a channel transmits the signal of the left and right audio channels. Its advantages include economical operation due to lower required bandwidth compared to a channel with no data rate compression. The disadvantage is poor transmission of the waveform of the audio signal from the signal source to the transmitter site. Algorithms for lossy data rate compression work with the psychoacoustic characteristics of the human ear. This means that changes made to the signal are almost inaudible. They were designed to make the audible spectrum of the signal, with regard to the masking thresholds of human hearing, correspond to the original sound. However, this processing and transmission method modifies the waveform of the audio signal. The rate of modification depends on the specific reduction algorithm used, on the transmission data rate and other channel properties. Depending on these parameters, the modification of the signal shape leads to increased peak levels by approximately 1 – 3 dB compared to the signal level on the channel input. The effective values of the signals on the channel input and on the output are practically identical. Therefore, if the output of such a transmission channel is connected to the input of a stereo encoder at the transmission site, the encoder will encode these short-term "overshoots" as well. As a result, the maximum admissible peak frequency deviation of the FM transmitter is exceeded for short periods of time.

In standard configuration there are two ways of solving the problem. The input sensitivity of the stereo encoder can be reduced so that the peak frequency deviation is not exceeded. This will, however, reduce the effective value of the transmitted signal, which will negatively impact the subjective loudness of station. The other option is using a so-called composite clipper, which clips the peaks right at the output of the stereo encoder. There is no exceeding of the peak deviation, but the price for it is distortion of the encoded stereo signal, mainly of the 38 kHz subcarrier. As a result of this distortion, the bandwidth of the encoded stereo signal broadens. Such a signal then contains spectral components in the 57 kHz band, interfering with the RDS signal. The consequence is lower coverage with the RDS signal, or even interference with adjacent FM channels.

The aim of this article is to describe a model of digital stereo encoder with composite clipping which causes no distortion of the 38 kHz subcarrier. This prevents the above problems and at the same time does not affect the overall loudness of station. The article does not deal with a detailed stereo encoding process. It focuses on the process of composite clipping and the relating selection of the optimum sampling rate for the encoded stereo signal. MPX clipping method described in this article was originally developed by the author. Note that different manufacturers use other methods of smart MPX clipping. "Half-cosine interpolation" is the best known one and it is described in the US patent [1]. In contradiction with this filtration and subtraction method [1], the described method is simpler and uses less DSP power due to the choice of optimal sampling frequency and also due to the direct look-ahead synthesis in the time domain. A great advantage of the described method (in comparison with [1]) is less DSP computing power needed to achieve comparable results. A disadvantage is slightly less control over newly generated spectral components inside the 0 - 53 kHz band. For a deeper understanding of the issue, it is recommended to learn more about FM broadcast audio processing (for instance in [2]). The article does not discuss the creation of preemphasis (emphasizing high frequencies before stereo encoding), because this is usually already included in the previous phase of signal processing.

The spectrum of the encoded stereo signal in a pilot tone system is shown in Fig. 1; an example of a time domain signal for natural excavating signals of the left and right audio channels (without a pilot tone) is depicted in Fig. 2.

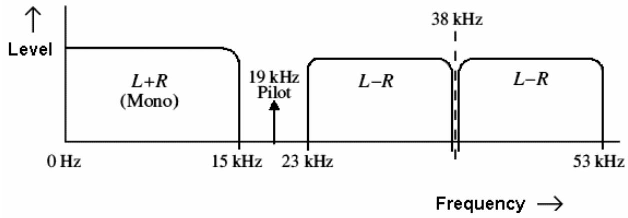

Fig. 1. Spectrum of encoded stereo signal.

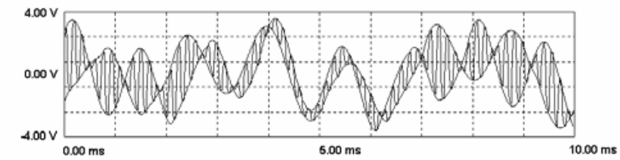

Fig. 2. Time domain of encoded stereo signal.

The encoded stereo signal is created from the left and right audio signals in a stereo encoder. The essence of stereo encoding consists of mutual chopping of both audio channels into one output channel. This is carried out synchronously with the frequency of the 38 kHz subcarrier. Transitions between signals of the left and right audio channels represent sections of the 38 kHz subcarrier. The subcarrier disappears if the left and right channel signals are identical. In the end, a 19 kHz pilot signal on the level of –20 dB, which is derived from the 38 kHz subcarrier by dividing it by two, is added to the encoded signal. The pilot signal is used to synchronize the stereo decoder of the receiver. Since stereo encoding is based on the time multiplexing principle, the encoded stereophonic signal is sometimes also called a composite signal, a multiplex or MPX. The FM transmitter is modulated directly by the resulting composite signal. The CCIR standard allows a peak frequency deviation of +/- 75 kHz. The frequency deviation is proportional to the signal voltage on the output of the stereo encoder. The proportionality constant is the input sensitivity of the FM transmitter.

The critical requirements on a stereo encoder are mainly the quality of input and output signal filtration, the ensuring of complete separation of the left and right channels, and the stability of all parameters in time. As explained above, the requirement of composite clipping is also relevant; it should be done in such a way that the emergence of any spectral components of the signal outside the allowed band, i.e., mainly in areas above 53 kHz, may be prevented. The composite clipping may not touch the audio signal which is orthogonal to the signal being clipped. This would cause crosstalk between the left and right audio channels during clipping. This requirement can only be met in the digital domain through a process which has no direct counterpart in executable analog processes. First it was necessary to create a suitable mathematical description, before the actual design and implementation of the encoder could be realized. The output of the description specifies requirements on input and output sampling rates and composite clipping algorithms. This article does not discuss the creation of the pilot signal in detail.

2. A Model of the Stereo Encoding Process

Let there be two independent signals of the left and right audio channels L and R on the system input. These signals come from the A/D converter on the input and are represented by equidistant sequence of samples with sampling rate 320 kHz with 24-bit resolution and linear quantization. The sampling rate on the encoder input is selected with regard to the possibilities of further filtration. Before the actual encoding begins, the bandwidth of the left and right audio channels must be limited to 15 kHz. The sampling rate of 320 kHz enables correct digitization of analog signals with a frequency range up to its half, i.e., up to 160 kHz. It creates sufficient reserve for high quality input filtration. The stereo encoder makes the frequency filtration first and then proceeds with the encoding into output samples.

Let there be two sample sequences on the encoder input, $\prod_{LOUT(N')}$ and $\prod_{ROUT(N')}$, i.e., the sequence of samples of the left channel and the sequence of samples of the right channel. N' means order of original input samples. Before the actual encoding process begins, the sequences must be transformed to other sequences of samples with a reduced sampling rate. We have to carry out filtration (FLTR) in order to limit the bandwidth of the left and right audio channels to 15 kHz (the reduction must be made before the actual stereo encoding; see Fig. 1) and to get rid of all higher products. Then the sampling rate conversion (SRC) is also made, in symbolic notation:

$$\begin{aligned}&\Pi(320\ \text{kHz}) \to \text{FLTR}(15\ \text{kHz}) \to \\ &\to \text{SRC}(320\ \text{kHz}/38\ \text{kHz}) \to \Pi(38\ \text{kHz})\end{aligned} \tag{1}$$

The signal is converted to the sampling rate of 38 kHz in order to make the sampling synchronous with the subcarrier. The preceding digital filtration works with the upper frequency limit of 15 kHz. In choosing a suitable type of digital filtration, we consider the requirements placed on filter parameters. Two filtration structures are possible, IIR and FIR. IIR is not very suitable; its group delay characteristics are not consistent. The group delay characteristics rise sharply in the area of the highest transmitted frequency, rather independently of the method of filter approximation. This causes overshoots in the time domain signal. These overshoots would devalue the signal being processed and, consequently, raise the transmitter deviation.

If the FIR structure is used, even group delay characteristics across the filter pass band can be attained. This creates the conditions for faithful reproduction of the audio signal waveform on the filter output.

The whole process of filtration and sampling rate conversion can be symbolically denoted as follows:

For the left channel:

$$\begin{aligned}&\Pi_{\mathrm{LOUT}}(N') \to \mathrm{FIR} \to \Pi_{\mathrm{FIRLOUT}}(N') \to \\ &\to \mathrm{SRC}(320\ \mathrm{kHz} / 38\ \mathrm{kHz}) \to \Pi_{\mathrm{L}}(N)\end{aligned} \tag{2}$$

For the left channel:

$$\begin{aligned}&\Pi_{\mathrm{ROUT}}(N') \to \mathrm{FIR} \to \Pi_{\mathrm{FIRROUT}}(N') \to \\ &\to \mathrm{SRC}(320\ \mathrm{kHz} / 38\ \mathrm{kHz}) \to \Pi_{\mathrm{R}}(N)\end{aligned} \tag{3}$$

where N' is the order of original samples at f_{sr} = 320 kHz, N is the order of converted samples at f_{sr} = 38 kHz.

3. The Encoding Process

The actual stereo encoding follows after the input filtration and sampling rate conversion. It is symbolically shown in Fig. 3.

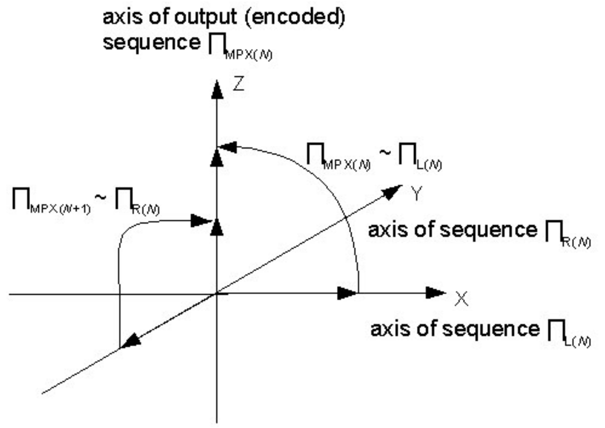

Fig. 3. Symbolic illustration of the stereo encoding process.

In 3D notation, the axes X and Y correspond to the input sequences $\prod_{L(N)}$ and $\prod_{R(N)}$. The Z axis corresponds to the output sequence $\prod_{MPX(N)}$, which represents the resulting encoded signal. All three axes are perpendicular to each other: they make up an orthogonal system.

The figure illustrates the requirements placed on the stereo encoding process. We can see that orthogonality of signals $\prod_{L(N)}$ and $\prod_{R(N)}$ is required. Orthogonality is defined as zero mutual power in the resulting encoded stereo signal. This is the only situation where the audio channels do not interfere with each other. Let us now explore a sampling rate suitable for the creation of output sequence $\prod_{MPX(N)}$. It is suitable that the sequence is sampled synchronously with the 19 kHz pilot signal. Another requirement is to respect the bandwidth of the signal carried by sequence $\prod_{MPX(N)}$. This bandwidth is expressed by the following relation (see also Fig.1):

$$BW(\Pi_{\mathrm{MPX}}(N)) = 2f(\mathrm{PILOT}) + BW(\Pi_{\mathrm{L}}(N), \Pi_{\mathrm{R}}(N)) \tag{4}$$

where $BW(\prod_{MPX(N)})$ is the resulting bandwidth of the signal carried by sequence $\prod_{MPX(N)}$, i.e., the complete encoded stereo signal (kHz), $f_{(PILOT)}$ is the pilot signal frequency of 19 kHz, $BW(\prod_{L(N)}, \prod_{R(N)})$ is the bandwidth of the input signals expressed by sequences $\prod_{L(N)}, \prod_{R(N)}$, i.e., 15 kHz (the bandwidth of the input FIR filter) and hence:

$$BW(\Pi_{\mathrm{MPX}}(N)) = 53\ \mathrm{kHz}\,. \tag{5}$$

According to the sampling theorem, this requirement determines the minimum sampling rate $f_{sr\,(MPX)min}$:

$$f_{sr}(\mathrm{MPX})_{\min} = 2BW(\Pi_{\mathrm{MPX}}(N)) = 2.53\ \mathrm{kHz} = 106\ \mathrm{kHz}\,. \tag{6}$$

At the same time, synchronous sampling relative to the pilot signal frequency is required. According to the sampling theorem, the minimum sampling rate $f_{sr\,(PILOT)\,min}$ for the expression of the pilot signal is equal to twice its frequency, so that:

$$f_{sr}(\mathrm{PILOT})_{\min} = 2f(\mathrm{PILOT}) = 2.19\ \mathrm{kHz} = 38\ \mathrm{kHz}\,. \tag{7}$$

Consequently, a set of sampling rates can be considered which satisfy the following condition for all integer positive n:

$$f_{sr}(\mathrm{MPX}) = n.38\ \mathrm{kHz}\,. \tag{8}$$

Because of the requirement for synchronous sampling relative to the 19 kHz pilot frequency, we will only consider even multiples of 19 kHz which concurrently satisfy the condition:

$$f_{sr}(\mathrm{MPX}) > 106\ \mathrm{kHz}\,. \tag{9}$$

Both conditions are satisfied for $n \geq 3$, i.e., for sampling frequencies of the series 114, 152, 190, 228, 266, 304, 342, 360,…. kHz.

Higher sampling rates lead to higher requirements on the speed of the D/A converter, which must be used on the ultimate output of the stereo encoder. In the circuit structure, DSP Analog Devices ADSP 21161 are used with D/A converter AD1852 (for more details see [3]), which can work with a maximum sampling rate of 192 kHz. Therefore, from the above series, only the sampling rates for n = 3, 4 and 5 can be taken into account; namely:

$$n = 3 \quad f_{sr} = 114\ \mathrm{kHz}, \tag{10}$$

$$n = 4 \quad f_{sr} = 152\ \mathrm{kHz}, \tag{11}$$

$$n = 5 \quad f_{sr} = 190\ \mathrm{kHz}. \tag{12}$$

4. The Composite Clipping Requirement

A specific requirement is placed on a stereo encoder: the ability to clip the composite signal. Such clipping in the existing structures was coupled with the extension of the encoded signal bandwidth – interfering spectral components over 53 kHz appeared. Besides that, crosstalk between the left and right audio channels got worse in the moments of clipping. The goal is strong reduction of such effects.

In order to attain such goals, in addition to the actual encoding, other processes must also be performed in the encoder, focusing on composite clipping. The requirement for simple implementation of these processes then affects the selection of the optimum sampling frequency. Let us illustrate a situation where the output composite signal is made up only of the signal of one (e.g., left) channel (here a sine wave signal, indicated by the "envelope of subcarrier 38 kHz", dotted line, see Fig. 4).

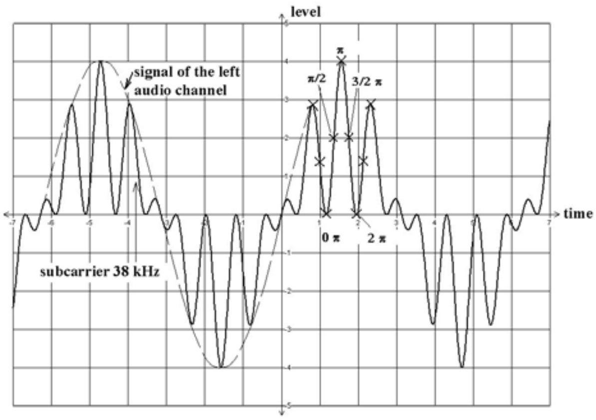

Fig. 4. MPX signal made up only of the left channel.

The optimum situation will be for $n = 4$, i.e., for sampling rate f_{sr} = 152 kHz. Only in this case is the signal of the 38 kHz subcarrier sampled synchronously and always at the same place, at angular positions 0 π, π/2, π, 3/2 π, and then again at 0 π at the beginning of the next period. This creates optimum conditions for the calculation of the signal in these points, which are phase constant relative to the 38 kHz signal. In this way, optimum utilization of the power of the used DSP is ensured.

5. Implementation of the Composite Clipping

A completely new way of clipping the composite signal is the essential property of the stereo encoder. What we have in mind is a method of clipping which does not corrupt the spectral purity of the output signal and mutual orthogonality of the signals, or sequences $\prod_{L(N)}$ and $\prod_{R(N)}$. In such a case the signals must satisfy the following energy conditions:

$$E_{\Pi MPX} = E_{\Pi L}(N) + E_{\Pi R}(N) + 2E_{\Pi RL} \tag{13}$$

where $E_{\prod MPX}$ is the energy of the sums of signals represented by input sequences (encoded signal), $E_{\prod L(N)}$ is the energy of the signal represented by the input sequence of the left channel, and $E_{\prod R(N)}$ is the energy of the signal represented by the input sequence of the right channel.

If the signals are orthogonal, it is required that:

$$E_{\Pi RL} = 0\,. \tag{14}$$

If the left channel is expressed by the function $f_L(t)$ and the right channel by the function $f_R(t)$, the requirement may be written as:

$$E_{\Pi RL} = \int_{-\infty}^{+\infty} f_L(t) f_R(t) dt = 0\,. \tag{15}$$

For discreet sequences it means:

$$E_{\Pi RL} = \sum_{N=0}^{N=\infty} \Pi_L(N) \Pi_R(N) = 0\,. \tag{16}$$

Common ways of clipping the signal time value lead to violation of the requirements, so that:

$$E_{\Pi RL} \neq 0\,, \tag{17}$$

$$E_{\Pi RL} = \sum \Pi_L(N) \Pi_R(N) \neq 0\,. \tag{18}$$

Moreover, spectral components on frequencies over 53 kHz emerge due to distortion of the signal. Fig. 5 shows this situation on the time course of the clipped signal.

It is evident why both requirements are violated. Conventional clipping causes the emergence of points in which the resulting function is not smooth, i.e., it does not satisfy the condition of a continuous first derivative. For these points the following holds:

$$\underset{\lim t \to tclipL}{f'(t)} \neq \underset{\lim t \to tclipR}{f'(t)} \tag{19}$$

where *tclipL* and *tclipR* are limit approximation points from the left and right, respectively, to the point of putting on and putting off, respectively, of the clipper (where the signal touches on the clipping threshold, CT).

An example of the discontinuous shape of the first derivative $f\,'(t)$ of the clipped function is shown in the bottom part of Fig. 5.

The violation of orthogonality between the channels and consequently the emergence of crosstalk is caused by the distortion in the neighborhood of the clipping point. The sine shape of the 38 kHz subcarrier is corrupted and, therefore, spectral components which have non-zero mutual power with the signal of the other channel emerge.

The reason for this phenomenon is that clipping causes the creation of a lot of spectral components, not only on the odd multiple frequencies but also under the 38 kHz subcarrier frequency. In general, for instance, certain DC components may emerge, which subsequently affects the other audio channel as well. This happens mainly when the same half of the wave is clipped systematically, e.g., when the clipping affects only the positive half-waves of the signal in a certain moment.

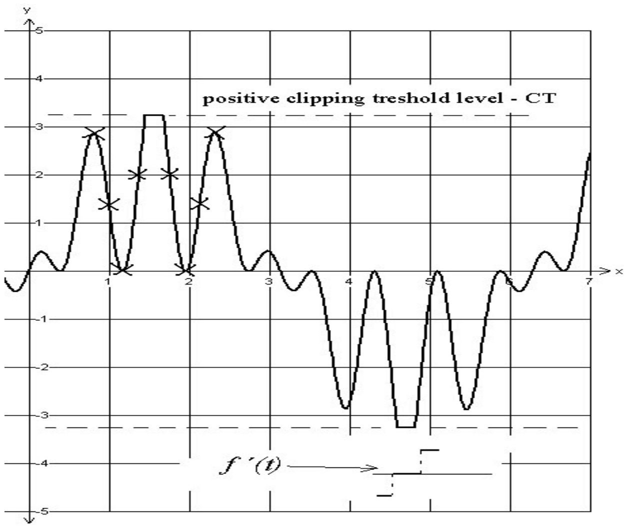

Fig. 5. Conventional clipping.

6. The DSP Solution

Both problems, namely the extension of the spectrum of the resulting stereo signal and the violation of orthogonality of the audio channels, can be solved in a special procedure. Through digital processing in DSP, the signal development over time can be monitored and so the risk of sharp clipping can be prevented. A technique called „look ahead“ is used to this end. DSP stores the incoming sample sequences in a cache memory. Thus signal analysis can be carried out before there is the need to generate the output signal. The procedure is characterized by the introduction of a small additional delay to the signal path, since the signal is analyzed first before it is encoded.

The output signal is first processed in the same way as in any encoder without composite clipping capabilities. The program subsequently compares the maximum signal values with the preset clipping threshold. If the threshold is exceeded, the difference between the maximum signal value and the threshold value for the given sample is calculated (size of the "clipped part of the signal").

This yields the amount of clipping which would be caused by a conventional clipper. Then the value of the given signal sample is reduced by this clipping size. At the same time, the samples adjacent to the given sample from left and right also need to be modified. If the clipped sample is N, then the following samples need to be modified (see also Fig.6.):

$$\Pi_{MPX(N-1)}, \Pi_{MPX(N)}, \Pi_{MPX(N+1)} . \tag{20}$$

Let the threshold value be expressed as CT (Clipping Threshold). The value of each sample $\prod_{MPX(N)}$ must be compared with CT in the absolute value, i.e.:

$$\left|\Pi_{MPX(N)}\right| > \mathrm{CT} . \tag{21}$$

If the condition is satisfied, then the following modification of the sample size is made:

$$\Pi_{MPX(N)} = \Pi_{MPX(N)} - \left(\Pi_{MPX(N)} - \mathrm{CT}\right) = \mathrm{CT} . \tag{22}$$

Here a "programming" notation is used, so that the original and the newly generated sequences both have the same notation, although it is now a new, modified sequence.

Then the sample has to be expressed with the proper sign.

The preceding sample $\prod_{MPX(N-1)}$ and the following one $\prod_{MPX(N+1)}$ also need to be recalculated. The 38 kHz subcarrier should have a cosine shape and if no modification is applied to the samples, a distortion would result. That would violate orthogonality and extend the spectrum. The values of samples $\prod_{MPX(N-1)}$ and $\prod_{MPX(N+1)}$ are found by calculating the value of the cosine function for the required sampling phase angles α for $\prod_{MPX(N-1)}$ and β for $\prod_{MPX(N+1)}$. In case of optimum sampling, n = 4, these angles will always be equal to (see Fig. 4):

$$\alpha = \pi/2 , \tag{23}$$

$$\beta = 3\pi/2 . \tag{24}$$

The values $\prod_{MPX(N-1)}$ and $\prod_{MPX(N+1)}$ for n = 4 can be calculated from the relation:

$$\begin{aligned} &\Pi_{MPX(N-1)} = \\ &= \left(\Pi_{MPX(N)} - \Pi_{MPX(N-2)}\right)\left(\cos(\pi+\alpha)+1\right)/2 \\ &+ \Pi_{MPX(N-2)} \end{aligned} \tag{25}$$

and

$$\begin{aligned} &\Pi_{MPX(N+1)} = \\ &= \left(\Pi_{MPX(N)} - \Pi_{MPX(N+2)}\right)\left(\cos(\pi+\beta)+1\right)/2 \\ &+ \Pi_{MPX(N+2)} \end{aligned} \tag{26}$$

Since α always equals το $\pi/2$, the above relation simplifies into:

$$\begin{aligned} &\Pi_{MPX(N-1)} = \\ &= \left(\Pi_{MPX(N)} - \Pi_{MPX(N-2)}\right)\left(\cos(\pi+\alpha)+1\right)/2 \\ &+ \Pi_{MPX(N-2)} = \\ &= \left(\Pi_{MPX(N)} - \Pi_{MPX(N-2)}\right)/2 + \Pi_{MPX(N-2)} \\ &= \left(\Pi_{MPX(N)} + \Pi_{MPX(N-2)}\right)/2 \end{aligned} \tag{27}$$

Similarly, for $\beta = 3\pi/2$ we find:

$$\Pi_{MPX(N+1)} = \left(\Pi_{MPX(N)} - \Pi_{MPX(N+2)}\right)/2 . \tag{28}$$

This is optimal because the sample values can be found by simple addition and division. Then DSP is not overburdened by the time-consuming calculation of the cosine function. Fig. 6 shows the new clipping method in contrast to the conventional one.

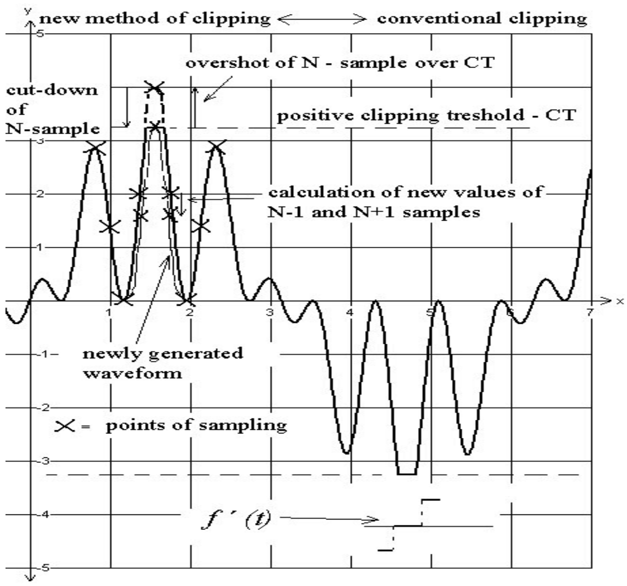

Fig. 6. Comparison between new and conventional method of clipping.

It needs to be added that, due to the properties of the stereo encoding process, the described procedure will also work if both channels are excited, rather than just one. This is because the encoder switches the left and right channel signals in sequence into the output signal. The transitions between the values of the switched channels are carried out at “part cosines.” It follows that the above procedure works correctly for any exciting signals whose spectrum is limited to 15 kHz.

We can see (Fig.6) that the newly generated part is smooth and, since its shape is defined by two cosine parts, there is no undesirable distortion of 38 kHz subcarrier. The newly generated part connects smoothly with the original shape. The connection between the shapes evidently always falls on points where the first derivative of both curves is zero. Due to the fact the only envelope of 38 kHz subcarrier is affected by clipping, new spectral components are generated primarily inside the 0 - 53 kHz band.

On the encoder output, the resulting sequence of samples must be converted to analog. Analog Devices AD 1852 converter with built-in FIR filter and oversampling is used. The converter properties ensure that the signal conversion to analog is performed with minimum distortion.

7. Conclusion – Results

The newly developed stereo encoder exhibits properties that are not attainable by conventional analog solutions. It can carry out composite clipping while maintaining high separation between the channels and without giving rise to interfering spectral components over 53 kHz.

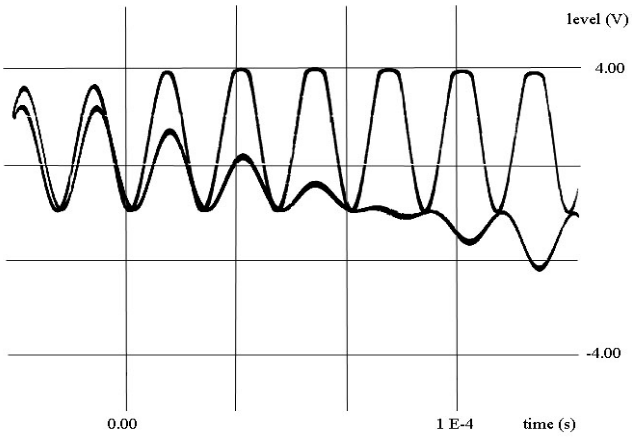

Fig. 7. Time domain for 2 dB clipping, conventional method.

The results measured on an encoder prototype are shown in Fig. 7-10. Fig. 7 depicts the time domain of a signal with 2 dB clipping for the conventional clipping method. Fig. 8 shows the same for clipping with the new method. Fig. 9 shows the resulting composite signal spectrum for conventional clipping. Fig. 10 shows the same for clipping with the new method. Spectrum of original MPX signal without clipping is shown in Fig. 11.

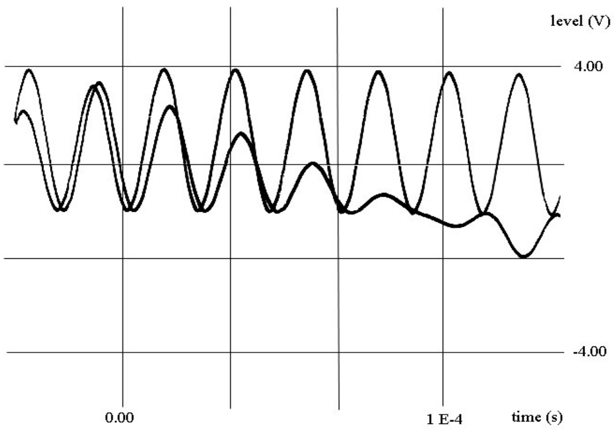

Fig. 8. Time domain for 2 dB clipping, new method.

The spectrum of interfering components outside the 0 - 53 kHz band was reduced by more than 25 dB compared to the conventional method when the new clipping method was used.

The solution is more effective in comparison with older methods of composite signal conditioning, described for example in [4]. On the other hand the method does not protect audio signals against hearable distortion caused by clipping. Due to the fact the envelope of 38 kHz subcarrier is affected by clipping, new spectral components are generated above all inside the 0 - 53 kHz band.

A level of such spectral components depends on an amount of clipping. In real conditions (audio signal with

2 dB casual peak clipping) undesirable spectral components inside the 0 - 53 kHz band are suppressed typically around – 45 dB relatively to full MPX signal level. If no additional filtering is used, the method ensures no overshoots (0 dB) over maximal acceptable MPX level. For better protection of the pilot signal, 19 kHz digital notch filter can be applied on the output of the clipper as option. Introducing of notch filter causes the overshoots around 0.5 dB. It has been examined using 19 kHz notch filter with a stopband ± 500 Hz from 19 kHz in real conditions (audio signal with 2 dB casual peak clipping, without pilot signal). In such way results are similar to [1].

In conclusion, it can be stated that the newly developed clipping method contributes significantly to the frequency spectrum protection of an FM transmitter site. Thus it not only protects the RDS signal itself but also prevents interference with other FM transmitters on adjacent frequencies.

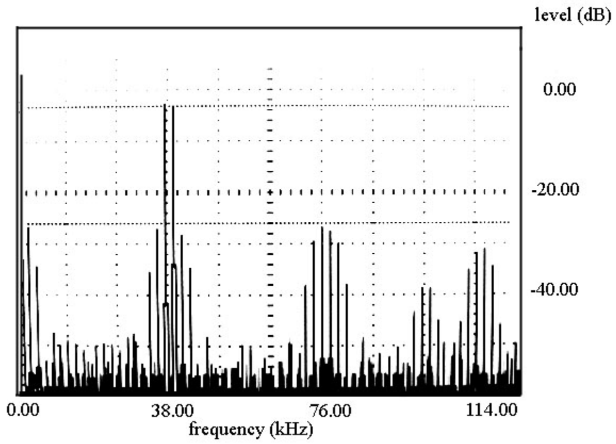

Fig. 9. Resulting spectrum of MPX signal, conventional method.

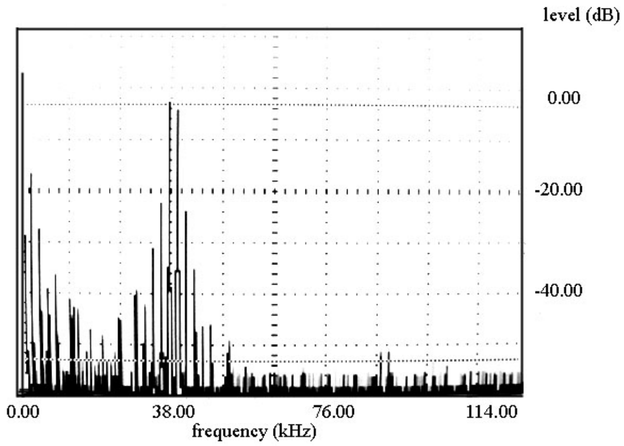

Fig. 10. Resulting spectrum of MPX signal, new method.

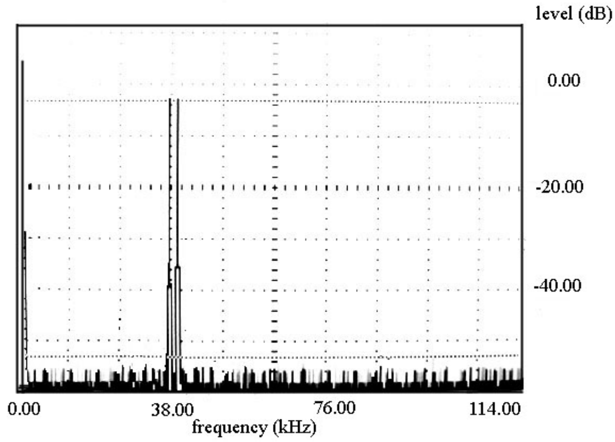

Fig. 11. Spectrum of original MPX signal, without clipping.

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