Pre-distorter for SSB broadcast transmitters

Broadcasting

Pre-Distorter for SSB Broadcast Transmitters

This paper describes a device, which can be used in order to make SSB transmitters compatible with AM receivers. The device pre-distorts input audio signal of the SSB transmitter in accordance with a special mathematical algorithm, which compensates the distortion of the output audio signal of AM receivers. The application of such pre-distorters in broadcasting can accelerate the conversion of HF broadcasting to the SSB operation, because the replacement of home AM receivers by special SSB receivers will not be necessary.

The present-day popularity of short-wave broadcasting causes the congestion of HF broadcast bands. A shortage of frequencies enforces stations to share the same frequency bands. As a result co-channel interferences are present.

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The application of SSB transmission instead of AM transmission can substantially decrease the overcrowding of the HF broadcast bands. Due to that fact, the International Telecommunication Union (ITU) has made some recommendations to pave the way for wider use of SSB broadcasting. Nevertheless a substantial progress is not reached in that field yet. The technical essence of the problem lies in the fact, that a single-sideband radio signal with carrier amplitude of a half of the peak value of the radio signal (such value of the carrier was recommended by ITU for beginning of conversion to the SSB broadcasting) cannot be received by conventional AM receivers without a high level of non-linear distortion. Particularly the non-linear distortion factor of the output signal of AM receiver is equal to 19,3% when a sinusoidal audio signal is transmitted with modulation factor equal to 0,9. Such a high level of non-linear distortion aggravates the fidelity of musical transmissions. ITU offered to use special SSB receivers with phase lock loop [1] in order to eliminate the non-linear distortion. But that method necessitates replacement of broadcast receivers and is not realized yet.
The idea of elimination of non-linear distortion at the outputs of AM receivers by means of digital pre-distortion of the input audio signal of the SSB transmitter was described in previous article [2]. That method allows to ensure fidelity of AM reception and consequently allows to continue usage of common home AM receivers. Manufacturing of any special SSB receivers becomes not required.
This article describes the block diagram of such pre-distorter and the mathematical algorithm of digital signal processing that is used in the predistorter.

Cause of non-linear Distortion

Let us consider the cause of distortion of the AM receiver output audio signal when a common SSB radio signal (with carrier) is transmitted.

If the audio signal $S(t) = \cos\Omega t \quad (1)$

is applied on the input of the SSB transmitter (with carrier), the output signal of the transmitter can be expressed as

$$u(t) = A_1\cos\omega_0 t + A_2\cos(\omega_0 t + \Omega)t \quad (2)$$

where ω_0 is transmitter carrier frequency and Ω is frequeney of the audio signal. The envelope of such radio signal can be determined as

$$E(t) = [(A_1 + A_2\cos\omega_0 t)^2 + (A_2\sin\Omega t)^2]^{1/2} \quad (3)$$
$$E(t) = [A_1^2 + A_2^2 + 2A_1A_2\cos\Omega t]^{1/2}$$

The audio signal at the output of AM receiver is in direct proportion to the envelope E(t) because the ampiltude detector of the receiver picks up the envelope of the radio signal. But it is easy to see that function (3) is not directly proportional to function (1). In other words function E(t) is distorted in comparison with function S(t). Consequently output audio signal of the AM receiver is distorted in comparison with input audio signal of the SSB transmitter (with carrier).

Pre-Distorter

A digital signal processing unit called as a pre-distorter can be used to pre-distort input audio signal of the SSB transmitter in order to compensate the distortion of the audio signal at the output of AM receiver.
The pre-distorter can be connected to the SSB transmitter as it is shown in fig. 1.
In the lower position of the switch the pre-distorter is disconnected and the SSB transmitter operates as a common transmitter. In the upper position of the switch, the pre-distorter is connected and SSB transmitter radiates the signal which can be received by AM receivers without distortion.
In accordance with [2] the mathematical description of the pre-distorter can be expressed as follows:

$$U(t) = R(t)^{1/2}\cos\{(2\pi)^{-1}{}_{-\infty}\!\int^{\infty}\ln[R(x)](x - t)^{-1}dx\} \quad (4)$$

where

$$R(t) = \pi^{-1}{}_{-\infty}\!\int^{\infty}\sin\Omega_{max}x(\Omega_{max}x)^{-1}[1+S(t - x)]^2dx \quad (5)$$

where S(t) is audio signal and Ω_{max} is maximum frequency of its spectrum.
The SSB transmitter shown in fig. 1 converts signal U(t) in accordance with the following law:

$$u(t) = U(t)\cos\omega_0 t + \hat{U}(t)\sin\omega_0 t \quad (6)$$

where $\cos\omega_0$ is the carrier frequency, $\hat{U}(t)$ is Hilbert transform of U(t). The industry already produces broadcast SSB trasmitters with digital Hilbert transformers [3].

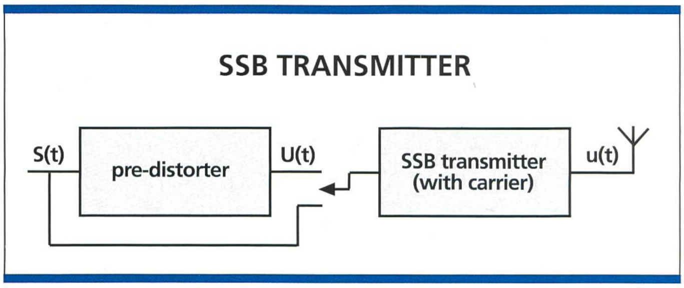

Fig. 1. SSB transmitter with pre-distorter.

Probably that is the most perfect type of SSB broadcast transmitters. SSB broadcast transmitters that do not use digital signal processing methods can be used too, but that transmitters often have substantially non-linear amplitude (-frequeney) and phase (-frequency) characteristica and therefore require other mathematical algorithms of predistortion. Such algorithms are not considered in this article.

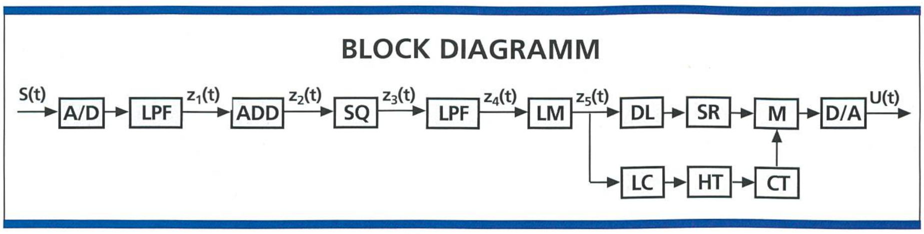

Fig. 2. Block diagram of the pre-distorter. A/D: analog-to-digital converter; LPF: low-pass filter; ADD: adder; SQ: squarer; LM: limiter; DL: delay line; LC: logarithmic converter; SR: square rooter; HAT: Hilbert transformer; CT: Cosine transformer; M: multiplier; D/A: digital-to-analog converter.

Implementation

The block diagram of the digital pre-distorter is shown in fig. 2.

The a.f. signal S(t) to be transmitted is digitized in an A/D converter before being led to the first low-pass filter. The application of that filter is necessary when spectrum width of the a.f. signal S(t) exceeds Ω_{max}. Mathematical description of the low-pass filter can be as follows:

$$z_1(t) = \sum_{i=0}^{N} m_i S(t - \Delta t i) \qquad (7)$$

where Δt – sampling period, $N \cdot \Delta t$ – "length" of delay line of the filter, m_i – weight coefficients of the filter.

Weight coefficients mi can be calculated by means of the formula:

$$m_i = \sin[\Delta t \Omega_{max}(i - N/2)][\pi(i - N/2)]^{-1}(0,54 + 0,46\cos[\pi(i - N/2)N/2] \qquad (8)$$

Note: $m_i = \Delta t \Omega_{max} \pi^{-1}$ when $i = N/2$

When weight coefficients m_i are chosen in accordance with formula (8), the cut-off frequeney of the low-pass filter is equal to Ω_{max}. Multiplicand

$$(0,54 + 0,46\cos[\pi(i - N/2)N/2)]$$

is necessary to ensure flat amplitude (-frequency) characteristic of the filter near cut-off frequeney.

The adder ADD adds unity to the audio signal:

$$z_2(t) = z_1(t) + 1 \qquad (9)$$

The squarer SQ is used to square the signal:

$$z_3(t) = [z_2(t)]^2 \qquad (10)$$

The second low-pass filter is analogous to the first one:

$$z_4(t) = \sum_{i=0}^{N} m_i z_3(t - \Delta t i) \qquad (11)$$

The weight coefficients of that filter are determined by the formula (8). The output signal of that filter corresponds to the signal R(t) which is described by the expression (5).

The limiter LM is used to exclude the overmodulation of the transmitter. If a 90% modulation is necessary, the mathematical description of the limiter can be expressed as

$$z_5(t) = z_4(t) \text{ if } z_4(t) \geq 0,01 \text{ and } 0,01 \text{ if } z_4(t) < 0,01 \qquad (12)$$

The mathematical expression (4) can be realized by means of a circuit which contains a logarithmic converter LC, a Hilbert transformer HT, a cosine transformer CT, a square rooter SR and a multiplier M. Delay line DL is necessary to compensate the delay of the signal in the digital Hilbert transformer HT. The mathematical description of the circuit is as follows:

$$U(t) = [z_5(t - \Delta t N/2)]^{1/2}\cos\{\sum_{i=0}^{N} l_i \ln[z_5(t - \Delta t i)]\} \qquad (13)$$

where l_i – weight coefficients of the digital Hilbert transformer HT. These coefficients can be calculated by means of the formula:

$$l_i = [1,08 + 0,92\cos\pi(i - N/2)(N/2)^{-1}][\pi(i - N/2]^{-1} \qquad (14)$$

if i = 1, 3, 5...
0 if i = 0, 2, 4, 6...

If the SSB transmitter has a digital input, the digital-to-analog converter D/A can be excluded from the circuit of the pre-distorter.

Experimental Testing

The described mathematical algorithm was tested by means of the following experiment. Testing sinusoidal and musical audio records were processed by computer in accordance with algorithm (7) – (14). It was assumed that $\Omega_{max} = 2\pi \cdot 4500$ Hz, N = 200, $\Delta t = 1/44100$ s. Resulting pre-distorted audio signals were used as input signals for a small-power SSB transmitter. The results of experimental spectrum analysis of the output radio signal of the SSB transmitter completely coincided with results described in [2], i.e., confirmed practicability of the method. The listeners did not find any distinction when they received pre-distorted SSB radio signal and AM radio signal by common home AM receivers.

Conclusion

The proposed method of pre-distortion of the audio signals allows sharply decrease non-linear distortion of the output audio signals of AM receivers when SSB signals are received. That makes it possible to realize the conversion to the SSB broadcasting without replacement of home receivers. The proposed pre-distorters can be manufactured as additional devices for existing SSB broadcast transmitters. 6

Bibliography

[1] Rec. ITU-R BS.640-1 "Single sideband (SSB) system for HF broadcasting", 1990.
[2] S. A. Chekeheyev, "Compatible SSB Broadcasting System: New Method of Modulation", IEEE Trans. on Broad., Vol. 43, No. 4, pp. 378–382, December, 1997.
[3] M. Gautsch, W. Tschol, "A New Method of Single Sideband Modulation for High-Power Broadcasting Transmitters", ABB publication No. CH-IT 1014 88E.

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