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Modern Digital-Analog Jamming Transmitter

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Abstract- Principle analysis results provide construction new digital-analog jammers from the point of view minimizing out-of-band undesirable electromagnetic radiation. Reviewed methods of compensating for nonlinear distortions, which are suitable for use in power amplifiers of jammers, based on linearization methods of the characteristics of powerful amplifying devices. The substantiation of a new compensation method for eliminating undesirable combination components in the spectrum of jamming transmitters with quadrature modulation (the method of depletion of the input signal spectrum).

Keywords: power amplifier, amplifier linearization techniques, jamming transmitter.

1. Introduction:

The progress occurred in almost all of communication equipment requires improving radio transmitters construction principles for jamming advanced models of radio electronic devices with various purposes. [1, 2, 3].

Unfortunately, so far, transmitter power amplifier with high energy could be implemented only on the analog element base; therefore the inspired new approach is digital-analog power amplifier. The most complex and expensive issues in the jamming transmitter is the modulator and power amplifier. Nowadays modulators are completely digital devices where a variety of digital signal modulation techniques are implemented. The most significant requirements of jammer transmitter

power amplifier are high power output as well, high efficiency [1, 2, 3, 4, 5, 7], and ensuring linear signal amplification. Linearity and efficiency are contradictory [6, 7].

High efficiency power amplifier could be provided only through the nonlinear amplifier behaviour. This fact is due to an inherited property of the amplifying device: non-linearity of its current-voltage characteristic curve. Consequently, the emission spectrum of the transmitter provide out of band spectrum, which complicate the task of ensuring the electromagnetic compatibility of radio equipment.

As well analog power amplifier should be able for amplifying signals with digital modulations. These features determine the relevance to the use of the tools and methods in the reduction of nonlinear distortion in high potential jammers transmitters restricted to its cost and performance.

The objectives of this study are to review power amplifier linearization techniques, identifying their strengths and weaknesses, as well as the justification of new technical approach in this area of study that are suitable for practical use in advanced radio transmitters.

1.1 Related Work:

Modern methods of power amplifier linearization are not essentially unfamiliar. Historically, the first was the idea of using a negative feedback circuit in a power amplifier in order to it’s linearization. It was implemented by engineer Black as a technique of reducing distortion in telephone repeaters in 1923. This method is commonly applied directly on radio frequencies (frequencies of the transmitter radiation). The block diagram of adaptive feed forward is shown in Figure 1. This architecture has been used successfully for many power amplifier linearization.

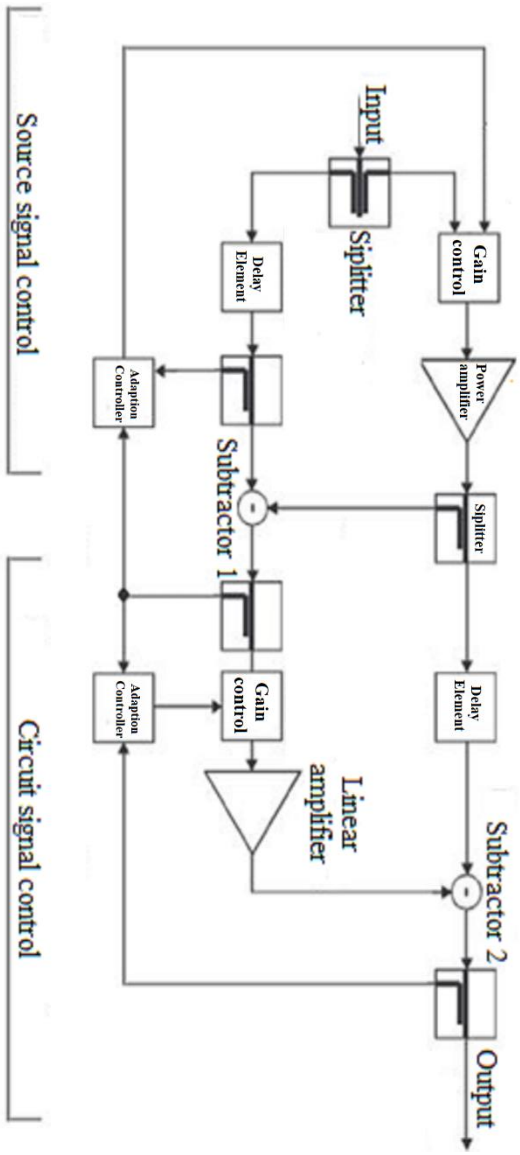

Figure 1: Adaptive feed forward block diagram

1.2 The Essence Of The Black Method

It is also called (forward connection method) consists in obtaining an error signal, which is deducted from the distorted signal that has already passed the amplification tract. The undistorted signal branches out in the splitter and goes to a non-linear power amplifier, as well as to a delay element with a controlled phase and gain. Two signals reach subtractor 1, the first signal has passed the amplification path, and the second is the original signal delayed by a certain number of clock cycles. At the output of the subtractor, we obtain harmonics that were not in the spectrum of the original signal. These harmonics will be the distortions of the original signal, called the error signal. The error signal passes through a compensator (gain control) and a linear amplifier. At the output of the linear amplifier is obtained error signal amplitude equal to the amplitude of the original signal distortion. After that, the error signal is subtracted from the distorted signal that passed the nonlinear power amplifier. Adaptation controllers in front of each gain control adjust the amplitude and phase. Thus, the only way to minimize the power at the output of the circuit is to suppress the distortion of the power amplifier.

Efficiency of the feed-forward system is reduced due to the power consumption of the linear amplifier. Correspondence of amplitude and phase is problematic due to the tendency of the amplifier characteristics to change under the effect of temperature and time, as well as due to the variation in production tolerances. However, despite these defects, adaptive technology can ensure the operability of the system, and the use of a digital processor will allow for the implementation of an adaptive operation algorithm.

Among the methods that make it possible to increase power efficiency when using linear modes of operation amplifier elements, the most widely used method is the automatic voltage control (the method proposed by Rassadin) [9,10]. It relies on varying the supply voltage of the cascade power amplifier output in accordance with the modification law of the single-band modulation envelope, which allows maintaining a critical mode in the entire amplitude range of the amplified signal.

As shown by theoretical and experimental studies of the characteristics of a push-pull linear cascade of class AB, its efficiency at an acceptable level of nonlinear distortion really, as a rule, does not exceed 50% in the continuous power mode when operating at a purely resistive rated load. Under conditions of mismatch of the load, the real efficiency is reduced to 40%. Another significant drawback of this method is the need to use a powerful regulated power source (RIP), which leads to a decrease in the overall efficiency of the device and additional nonlinear distortions due to the delay of the envelope signal in the RIP path.

In Quantization Method [9], the amplified signal with a variable amplitude is divided into several channels by power, depending on its instantaneous value. This is done in such a way that the signal powers in these channels always coincide with pre-selected discrete (quantized) levels. At the same time, the amplification paths of most of these quantized channels can operate in slightly overstressed or in key mode. In the adder connected to the outputs of these channels, the original signal with a variable amplitude is restored.

The disadvantages of this method should also include the complexity of the forming quantizing device and multi-level adder. The practical construction of adders of high-frequency signals with different amplitudes showed a significant complexity of their implementation and a very low efficiency.

An amplifier according to Doherty Method was developed to increase the efficiency in amplifiers operating at powers lower than the saturation power. This method uses an amplifier on a carrier (or main) frequency operating in class B and tuned to a power 4 times (6 dB) less than the maximum power and a peak (or additional) class C amplifier, the output of which begins to arrive at a level signal minus 6 dB from the maximum level in saturation mode (Figure 2). [11]

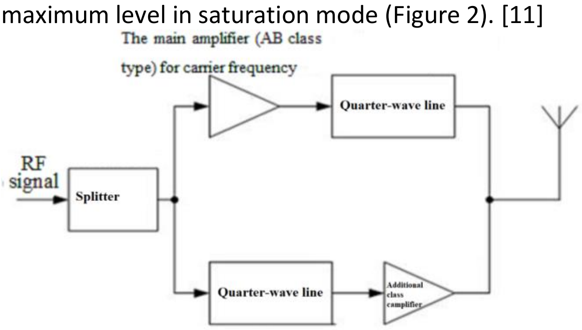

Figure 2: Doherty Method block diagram

Since the carrier frequency amplifier is connected through a quarter-wave line, its input impedance begins to decrease due to a decrease in the output impedance of the peak amplifier. For this reason, within six decibels to the maximum power, the carrier frequency amplifier operates at the maximum output voltage level (which corresponds to the critical mode), with a theoretical efficiency reaching 78.5%. The result is two power peaks with the indicated efficiency - at the detuning level minus six dB and at maximum power when both amplifiers operate in the maximum class B efficiency mode (the cutoff angle for the peak amplifier will be close to 90 degrees with a large output signal).

Class C and Class B modes in real conditions lead to strong nonlinear distortion of the signals in the amplifier. Therefore, usually the carrier amplifier is shifted to class AB mode with a small quiescent current, and the peak amplifier bias circuit is tuned to the optimum bias, which ensures constant gain and phase of the entire system. In this case, some improvement in efficiency can be obtained, but it is difficult to provide a high level of gain linearity. In addition, it is necessary to take into account potential changes in the supply voltage and technological variations in the parameters in the range of ambient temperatures. This can be done only with the help of a digital control unit, which makes such a system very complex, and in real operating systems it still finds application only in the microwave range.

Thus, the simplest classical methods for increasing the linearity of amplifiers are considered and Their main advantages and disadvantages are shown in table 1.

It is important to note that all the methods considered do not allow achieving a high linearity of the power amplifier when amplifying signals with complex types of modulation.

Consider modern promising linearization methods for power amplifiers and analyze them in terms of suitability for use in transmitters of radio interference stations. In recent years, developers have increasingly begun to consider linearization methods by introducing pre-emphasis into the transmission path and using adaptive linearization

Table 1: The Advantages and disadvantages of the simplest methods of linearization

Method	Advantage	Disadvantage
Class A linear amplifier mode	Stable efficiency of the amplifier and the ability to maintain performance when the load changes from idle to short circuit	Low continuous efficiency characteristic of class A (less than 50%)
Automatic Voltage Regulation Method (Class A)	Amplifier efficiency up to 50% in continuous mode and purely active load	Requires the use of a powerful regulated power supply, IMD ~ 30 dB
Quantization method	High efficiency corresponding to the key mode of operation of the transistor	The application is limited by the class of high-power transistor transmitters, the complexity of constructing a quantization device and a multi-level adder
Doherty Method	Theoretically achievable efficiency is 78.5%	The ability to implement only using a digital control unit, which complicates the system

methods.

This is because the considered circuit solutions using relatively simple methods for constructing transmission paths provide linearity in the transmission of signals with only simple types of modulation (AM, PSK, FSK). However, already with complex types of modulation (QPSK, QAM, WCDMA, OFDM, etc.), which turn out to be more susceptible to non-linear distortions, these methods do not provide the required quality. When transmitting such signals, in addition to intersymbol interference (ISI), which increases the error rate, due to the influence of nonlinear effects, the spectrum of the transmitted signal expands, which leads to interference with other radio means. Due to the specifics of their application, interference transmitters are subject to very stringent requirements for spurious and out-of-band emissions, therefore, we will consider in more detail modern promising linearization methods for power amplifiers suitable for use in interference transmitters.

Currently, three main methods of linearizing signal power amplifiers with digital signal modulation methods are widely used [12]:

- The predistortion method.
- Feedback method.
- Feed forward method.

These linearization methods for power amplifiers provide spectral efficiency, which helps to use complex types of modulation and high data rate. Based on these fundamental methods, linear architectures are formed, which include:

linear implementation using nonlinear components - Linear amplification using nonlinear components (LINC);

combination of a universal modulator with an analog feedback loop - Combined analog locked loop universal modulator (CALLUM);

- Envelope suppression and restortion - Envelope elimination and Restoration (EER);
- feedforward to suppress intermodulation distortion - Cancellation third order intermodulation feedforwarding (CTOIF).

The general structure of methodological approaches to achieve high linearity of signal amplifiers with complex types of modulation, as well as the existing linear architectures, can be represented, as shown in Figure 3, and in a systematic form - in table 2.

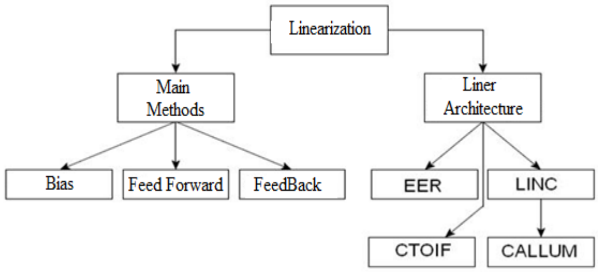

Figure 3 A review of the linear power amplifier

Below is a brief overview of these methods and linear structures with an analysis of their advantages and disadvantages.

1.3 Polar Feedback Scheme

Polar feedback is an improved version of the feedback on the envelope. The operating principle of the polar feedback circuit shown in Figure 4. Polar feedback is a broadband type of feedback.

The main feature of the scheme is that the phase and amplitude components operate independently.

The essence of the method is that the signal that passed the amplification path is removed from the splitter and transferred to the intermediate frequency using a local oscillator. The phase and amplitude are extracted from the received signal using a limiter and a demodulator, respectively. At the initial signal, the phase and amplitude are distinguished in a similar way. The amplitude components of the original signal and the signal that passed the amplification path are compared using a differential amplifier, called the error signal amplifier, at the output of which the resulting error signal is generated.

Table 2:The classification of the methods of linearization

Feedback	Feedforward	Predistortion
Radio Frequency Feedback	Adaptive Forward Link	Radio frequency predistortion
Envelope Feedback, Polar Feedback, Quadrature Feedback	Envelope Elimination and Restoration (EER)	LINC/CALLUM
Adaptive predistortion	Cancellation third order intermodulation feedforwarding (CTOIF).	Adaptive predistortion

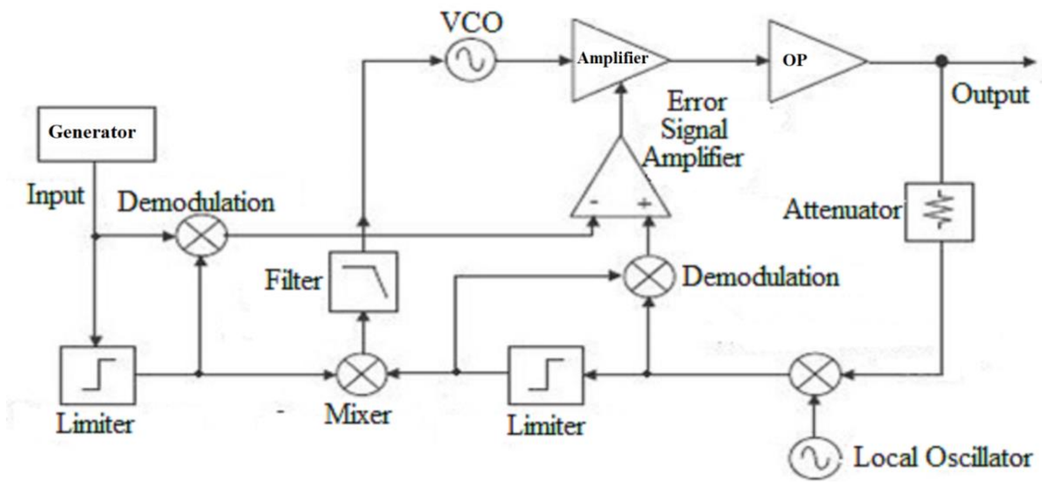

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Figure 4: Polar feedback circuit

The phase components of the signals are multiplied in the mixer. As a result of multiplication, an output signal is generated that controls the VCO (voltage controlled oscillator). Based on the error signals coming from the VCO and the error signal amplifier, new phase and amplitude are generated in the amplifier [13].

The polar feedback circuit provides a relatively high efficiency of the power amplifier (which can be completely non-linear) and stability due to the presence of two types of feedback. Since both the amplitude and the phase are corrected in the polar feedback system, the circuit reduces the effect of temperature and load on the operation of the amplifier.

The main disadvantage of polar feedback is the presence of two different control circuits having different bandwidths for the phase control circuit and the amplitude control circuit. This leads to additional adjustments to the amplitude and phase characteristics, which means that deviations of the latter from the characteristics of the equivalent circuit with a Cartesian (quadrature) feedback loop appear.

Quadrature feedback was first proposed by Petrovich. The main idea of the method is to use an I-Q modulator to modulate the carrier frequency before applying it to a nonlinear but highly efficient amplifier, the Petrovich circuit is shown in Figure 5. The main circuit of the system contains a main controlled gain loop with compensating filters, a synchronous I-Q modulator and an antenna as an output load [13]. The signal that passed the amplifier path through a radio frequency splitter is fed into the feedback circuit. branched signal is then synchronously demodulated and subtracted from the original signal using differential amplifiers. The characteristics of the control loop depend on the gain and compensating filters. Reducing the level of intermodulation distortion essentially depends on the gain of the loop, and compensation stabilizes the behavior of the controlled system. The synchronization of the modulator and demodulator is provided by dividing the common radio frequency signal of the carrier frequency.

Due to differences in the forward and reverse circuits, phase control is necessary in order to maintain the correct relationship between the input and feedback signals. Quadrature feedback can automatically track temperature changes in non-linearity and changes in supply voltage. Despite its advantages, this scheme is only conditionally stable and its regulation is one of the key problems. A non-linear amplifier also affects the stability of the signal, as it creates phase shifts in an excessively wide band. Another limiting factor of the system is the nonlinearity of the mixers when transferring the frequency spectrum to the intermediate frequency and vice versa. But the main disadvantage of this scheme is the narrow bandwidth, which limits its use in broadband systems, which include interference transmitters. Based on the quadrature feedback loop shown in Figure 5, the adaptive digital predistortion method is also constructed (Figure 6).

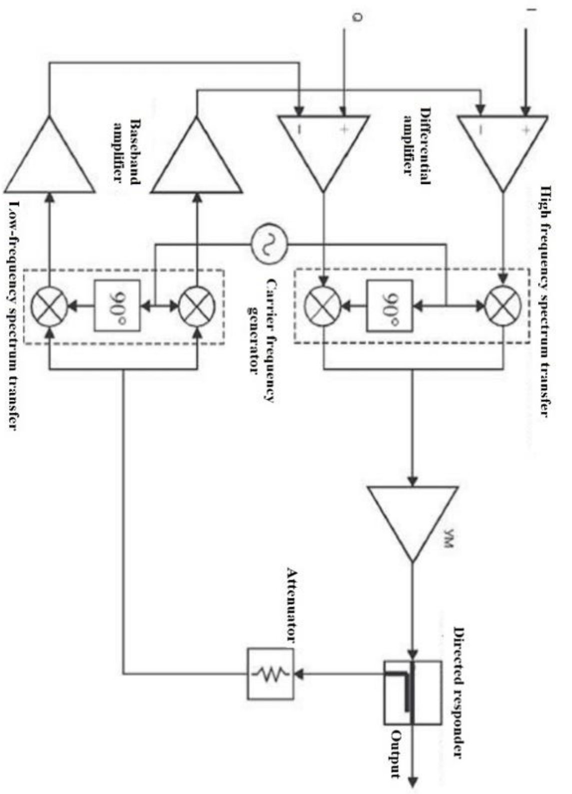


Figure 5: Petrovich circuit

Digital pre-emphasis can be used to control analog and digital broadband input signals, analog and digital input signals of intermediate or radio frequency (as opposed to a simple Cartesian feedback loop). Despite the fact that in recent years have developed many versions of the digital predistortion, they can be classified according to several criteria, which boil down to two main groups. One of them - a LUT (look up table) before- and another group - a parametric predistortion based on Volterra series [13].

The basic principle on which adaptive digital pre-distortion is based is the quadrature feedback loop with the addition of digital signal processing. A diagram characterizing the operation of the device is shown in Figure 6.

In general, the operation of digital pre-distortion schemes is based directly on both groups of pre-distortion devices and the adaptation algorithm.

The original signal is input to the digital signal processing unit. The signal that has passed the amplification path is separated in a directional coupler and transferred to a lower frequency using a converter. Next, the converted signal is also fed to the input of the digital signal processing unit (DSP).

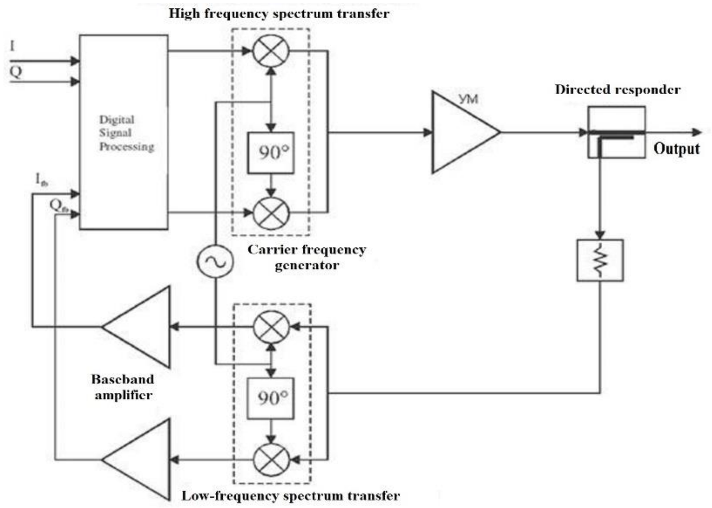


Figure 6: The adaptive digital predistortion method

The DSP unit calculates the coefficients, which are the amplitudes of the individual samples. In order to obtain a pre-distorted signal, the calculated coefficients are stored in the tables of the DSP block. In the memory of the DSP block, the coefficients of the signal received via the feedback circuit are also stored. At the output of the DSP block, a pre-delayed signal is generated, which is transferred to the high frequency region and amplified in the power amplifier. The speed and complexity of adaptive pre-distortion is one of the most important problems for their practical implementation. The inertia of such devices makes it difficult to use them in interference transmitters, which are subject to stringent performance requirements.

All of the above methods of increasing the linearity of amplifiers dramatically complicate the circuit design of high-frequency paths of transmitters, but although they are mainly aimed at increasing linearity, their application can give even a small gain in efficiency.

The total distortion of the multistage amplifier dΣ can be determined by the formula [5]:

$$d\Sigma = 20\log(10^{\frac{d1}{10}} + 10^{d2/10} + \ldots) \quad .....(1)$$

where d1 and d2, etc. are the products of the mutual modulation of the 1st, 2nd, and subsequent stages of the amplifier in dB.

For example, for an amplifier with two amplification stages (preliminary stage and output stage), it is possible to determine by how many decibels the mutual modulation at the output of the device is degraded if the difference in the values of "A" between the level of mutual modulation of the output stage and the preliminary stage is known. The formula by which the calculation can be made is given below:

$$B = 20\log(1 + 10\text{-}A/20) \quad ......(2)$$

where A is the absolute value of the difference in the distortions of the preliminary stage and the final stage, and B is the increase in distortion at the output of the amplifier.

Table 3 shows the calculation data by the formula (2).

Table 3: The calculation data

The difference between the IMD values (dB) of the preliminary stage and the end stage	Increase IMD (dB) amplifier output
0	6.0
5	3.9
10	2.4
15	1.4
20	0.8

An analysis of the active element operating modes used in power amplifiers shows that only class A amplifiers satisfy the established requirements for high-linear power amplifiers of radio stations of jamming transmitters.

The output stage operating in class A can provide a mutual modulation level of no more than minus 40 dB with an efficiency not exceeding 25%, while the preliminary stages should work in an even more under-stressed mode and with a lower efficiency (not more than 15%). The linearity requirements for stationary transmitters (no more than minus 45 dB) can be satisfied by class A amplifiers operating with an efficiency of ~ 15% and providing a level of 3rd order intermodulation products of no more than minus 55 ... 60 dB.

Due to the low efficiency of class A amplifiers with high demands on the energy efficiency of jamming devices, this pure mode of operation of the transmitters becomes unacceptable for practical use in interference transmitters. The use of highly efficient classes of operating modes of amplifier cascades such as AB, B, C, and E without the use of special methods for reducing distortion is not acceptable when designing power amplifiers of interference transmitters due to significant distortion of the signals at the amplifier output - of the order of minus 30 dB or more.

2. Methodology

This methodology suggestions for improving the linearity of the amplification of signals in transmitters interference. Here is the rationale for the new linearization method of the amplifier, suitable for use in interference transmitters. In the linearization methods discussed above, harmonics resulting from nonlinear distortion are obtained by subtracting the undistorted input signal from the distorted signal that has passed the amplification path and amplifying it in a linear amplifier. The result is harmonics that were not in the spectrum of the original signal (error signal). Subsequently, these harmonics subtract from the amplified signal. In this case, it becomes necessary to amplify the amplitude of the error signal in the linear amplifier to the level of the distortion amplitude of the input signal. Therefore, at the output of the linear amplifier of the error signal, nonlinear distortions can occur, which will be part of the amplified input signal.

In accordance with the proposed linearization method, non-linear distortions are compensated by artificially depleting the spectrum of the input signal, taking into account the detected distortions in the amplifier.

In the closest in essence adaptive method for reducing out-of-band radiation of a power amplifier [10], to improve the linearity of a power amplifier, the amplitude-amplitude characteristic (AAX) and phase-amplitude characteristic (PFC) of a power amplifier are preliminarily measured, and the values of the correction table for digital correction of AAX and FAX are calculated, choose and implement the adaptation method. Moreover, to calculate the amplitude and phase correction coefficients, it is necessary to solve a complex nonlinear equation, approximate the experimentally measured AAX and PFC of the power amplifier. Moreover, in addition to computational difficulties, approximation errors of the curves AAX and FAX are inevitable. To improve the accuracy of approximation, high-order polynomials (up to the ninth) are used, but even in the region of small amplitudes they do not correctly reflect the characteristics of the power amplifier. In addition, the prototype method involves the procedure of adapting a linearized amplifier to changing the amplification conditions by a linear method or a secant method. As a result, the implementation of amplification is complex and time-consuming.

At the same time, there are both analog and digital predistortion correction systems. Digital correction systems have advantages over analog ones: high speed and accuracy of predistortion correction. Reducing the time of signal formation in a power amplifier is especially important in transmitters of active jamming stations, for example, when real-time suppression of suppressed radio electronic means is required.

In the proposed method, the improvement of the linearity of the power amplifier is achieved through the application of subtracting from the spectrum the spectrum of the original signal distorted during amplification of the signal, the formation of the inverse spectrum with respect to the difference spectrum, and the addition of the spectrum of the original signal with the inverse spectrum. Moreover, the signal processing processes do not contain a large number of complex computational procedures and are carried out in the frequency domain. Operations on signals are replaced by operations on their spectra. All of the above provides a simplification of the implementation of quasilinear amplification of signals in a power amplifier in accordance with the proposed method in comparison with the known.

The justification of the possibility of reducing the processing time of the amplified signal in the frequency domain is given, for example, in the source [11]. This monograph shows that the convolution of two signals (functions) can be performed in the frequency domain by multiplying their spectra based on the convolution theorem. Using standard Fast Fourier Transform (FFT) packages, this approach can reduce the computation time by hundreds of times. The indicated sequence of actions allows linearizing the transfer frequency response of the power amplifier with a shorter calculation time.

Figure 7 shows a block diagram of a device that implements a new linearization method. It is indicated on it: 1 - the first block of the direct FFT, 2 - digital-to-analog converter, 3 - modulator, 4 - power amplifier, 5 - coupler, 6 - demodulator, 7 - analog-to-digital converter, 8 - second block of direct FFT, 9 - block subtracting spectrum, 10 - shaper of the inverse spectrum, 11 - block addition of spectrum, 12 - block fast inverse Fourier transform.

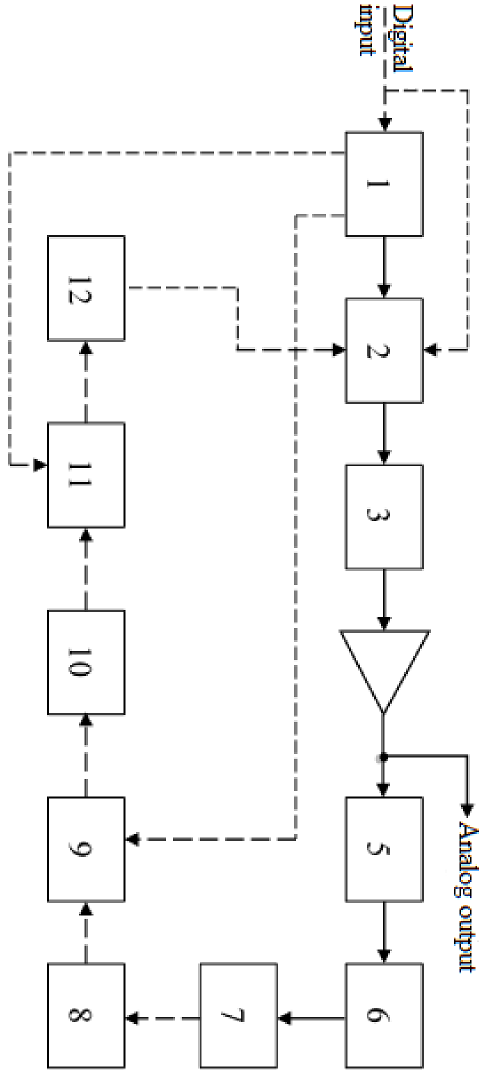


Figure 7: Block diagram of a device that implements a new linearization method.

1- The first block of the direct fast Fourier transform (FFT); 2- digital-to-analog converter (DAC); 3 - modulator; 4 - power amplifier; 5 - coupler; 6 - demodulator; 7 - analog-to-digital Converter (DAC); 8 - second block of direct fast Fourier transform; 9 - block subtraction of spectrum; 10 - shaper inverse spectrum; 11 - block addition spectrum; 12 - block fast inverse Fourier transform.

The purpose of the elements of the amplification device is clear from their name. All of them can be made on the basis of well-known commercially available radio engineering elements. So, for example, the unit of subtraction of spectrum can be performed on the basis of a technical solution proposed in [12]. The inverse spectrum is formed, for example, as shown in [13]. The addition of spectrum can be performed according to the algorithm presented in the source [11]. In the aggregate of interconnected elements, the proposed device is a quasilinear filter for signal processing.

Thus, in general, it was proposed in this way to artificially change (deplete) the spectrum of the amplified signal, so that after its amplification with inevitable nonlinear distortions in the power amplifier (their consequence is the enrichment of the spectrum), these distortions are eliminated. This reduces the time for signal processing due to processing not in the time domain (as in the prototype), but in the frequency domain.

This result in the new method is achieved by the following new sequence of actions on the signal in digital form:

- subtraction from the spectrum of the amplified signal of the spectrum of the original signal (obtaining the spectrum of the difference signal);
- the formation from the spectrum of the difference signal of the inverse spectrum (symmetric in amplitude with respect to the frequency axis);
- addition of the inverse spectrum with the spectrum of the input signal (obtaining the depleted spectrum of the input signal);
- repeated and already final amplification of the signal with a depleted spectrum in the power amplifier.

A device for amplifying a signal power with compensation for non-linear distortions works as follows. The input primary amplified signal in digital form in the form of a digitalized temporal implementation is fed to the input of the first direct fast Fourier transform unit 1, where the amplitude-frequency spectrum of the input amplified signal is calculated and stored. The temporary array of the input signal is also fed to a digital-to-analog converter 2, where it is converted to analog form, and fed to a modulator 3 and a power amplifier 4, where it is amplified in terms of power with inevitable non-linear distortions. The amplified powerful distorted high-frequency signal through the coupler 5 is fed to the input of the detector 6. The coupler 5 attenuates the signal to the desired level, the detector 6 transfers the signal spectrum to the low-frequency region. In the analog-to-digital converter 7, the signal is digitized in the form of a temporal sequence of symbols and passes through the second direct fast Fourier transform unit 8, where the amplitude-frequency spectrum of the distorted amplified signal is calculated. This spectrum, together with the stored spectrum of the original input signal, enters the unit for subtracting spectra 9. In the unit for subtracting spectra 9, the spectrum of the original signal is subtracted from the distorted spectrum and a difference spectrum corresponding to the nonlinear distortion introduced by the power amplifier is obtained. The difference spectrum in digital form is converted into the inverse spectrum in the inverse spectrum shaper 10. The inverse spectrum is fed to the input of the addition of spectra 11, the second input of which also receives the stored spectrum of the input primary amplified signal from the output of the first block of the direct fast Fourier transform 1. As a result of addition spectra, the spectrum of the depleted signal is obtained, which in the fast inverse Fourier transform unit 12 is converted into a temporary implementation. This implementation of the artificially depleted spectrum signal is again fed to the input of the digital-to-analog converter 2, where it is converted to an analog signal, transferred to a high frequency in modulator 3, amplified in power amplifier 4, and fed no longer to coupler 5, but to the consumer stage, for example , into the antenna for radiation into space or into a filter, or into a matching device, or into the next amplification stage.

Thus, the final amplification of a specially converted digital amplified input signal into a truncated analog signal eliminates undesirable non-linear distortion during amplification in an analog power amplifier. Signal pre-processing is completely performed in the frequency domain, which reduces processing time. These advantages are important in the design of jamming transmitters; therefore, the method is supposed to be widely applied in the technique of electronic suppression of radio electronic equipment. This new compensation method for eliminating undesirable combinational components in the spectrum of interference transmitters (the method of depletion of the input signal spectrum) can be applied in advanced analog-to-digital signal power amplifiers obtained using quadrature modulation.

3. Amplifier performance indicators in power amplifiers

To assess the nonlinearity of the PA, dependences of characteristics are traditionally used when the input power changes, for example, the dependence of the output power relative to the input. This dependence determines the compression gain of 1 decibel. However, only a small deviation from linearity is characteristic of a highly linear CM, and the gain is almost unchanged. Such data do not allow accurate amplitude distortions to be recorded, and also to select the input power level at which the maximum allowable criteria for changing the gain and phase shift should be set. For this reason, along with the mentioned dependencies, the following UM quality indicators are used:

THD (non-linear distortion coefficient,) - a value for the quantitative assessment of non-linear distortions equal to the ratio of the rms sum of the spectral components of the output signal that are not in the spectrum of the input signal to the rms sum of all spectral components of the input signal;

SNR (signal to noise ratio) is a dimensionless quantity equal to the ratio of the useful signal power to the noise power;

CNDIN (coefficient of nonlinear distortion including noise, THD + N) is a dimensionless quantity equal to the ratio of signal power to the sum of distortion and noise powers;

SFDR (dynamic range free from spurious components) - dimensionless quantity equal to the ratio of the power of the useful narrow-band signal (carrier) to the power of the most powerful spurious frequency component (harmonic);

SINAD (signal-to-noise ratio including distortion) - dimensionless quantity equal to the ratio of the sum of signal powers, distortions and noise to the sum of distortion and noise powers;

ENOB (effective bit depth) is the real number of different levels of the output signal that is achievable on a given ADC / DAC.

Along with the above, more often in practice, the power factor indicator in the adjacent ACPR (Adjacent Channel Power Ratio) is used A measure of expansion, spurious hit of a signal in adjacent channels. Typically, this parameter is used to assess the degree of distortion caused by the nonlinearity of power amplifiers and, in general, transmission paths of RF units. We can say that the ACP coefficient characterizes their predisposition to interfere with devices using the adjacent RF channel of a radio communication system. In general, the ACPR coefficient is defined as the ratio in decibels of the power value P1 in a certain frequency band (BW1) at the center frequency of the working channel to the power value P2 concentrated in a certain band (BW2) for a given detuning (F) from the carrier frequency of the working channel (fc) (Figure 8).

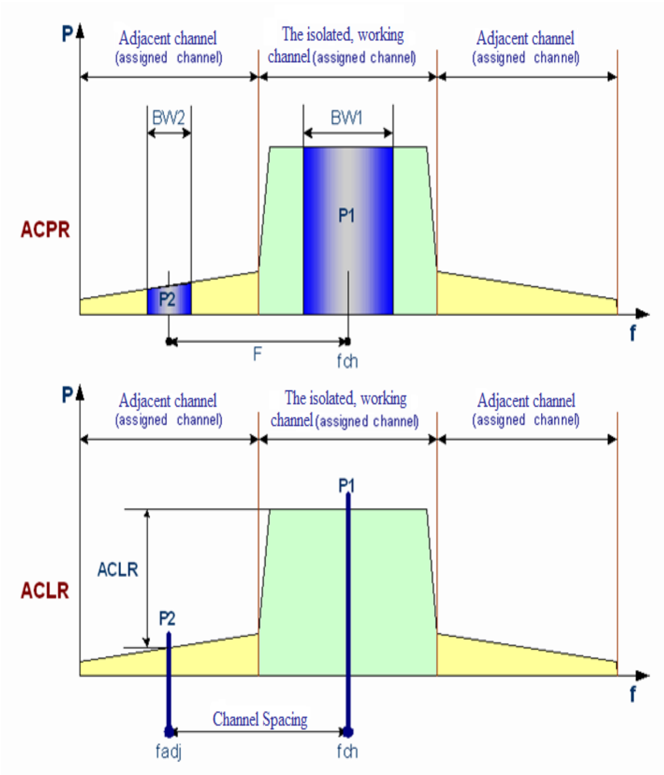
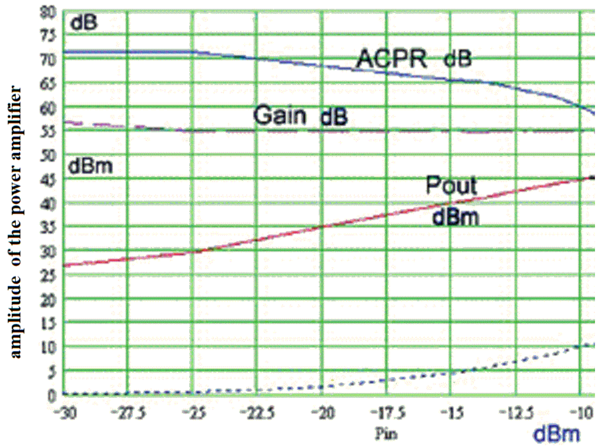


Figure 8: Measuring of coefficients ACPR vs. ACLR

The concept of measuring the ACPR coefficient is illustrated using Figure 8. The specific values of the frequency bands, detunings, and capacities used in the measurements for various standards and technologies can be found in the relevant regulatory documents.

To characterize the quality of the transmitters of the third generation WCDMA communication systems, the ACLR coefficient was introduced similarly to ACPR. This coefficient is defined in WCDMA 3GPP standards, where the following definition is given: the coefficient (ratio) of power leaking into the adjacent ACLR (Adjacent Channel Leakage power Ratio) is the ratio of the average power concentrated on the assigned channel frequency to the average power, focused on the adjacent channel frequency [3GPP TR 21.905]. In both cases, the average power is measured using a measuring filter that has the Root Raised Cosine (RRC) characteristic with a slope coefficient of a = 0.22, in a frequency band equal to chip rate.

Modern, specially designed generators with an arbitrary waveform and spectrum analyzers allow you to remove the amplitude characteristics taking into account more subtle nonlinear effects (Figure 9).



Pout (AM-AM), ACPR, efficiency (%)

Figure 9: The measured amplitude characteristics of the power amplifier: depending on the output power Pout (AM-AM), ACPR, efficiency (%)

So, the ACPR indicator is defined as the relative level of out-of-band signal emissions through the ratio:

ACPR=10·log10 (P(main channel)/P(main channel)),

where P(main channel) is the average signal power in the main channel, P(main channel) is the average signal power in the adjacent channel. This value takes into account both nonlinear and amplifying properties of the system.

For the cellular standard IS-95, ACPR is measured as the ratio of power in the 30 kHz band at a detuning of 750 kHz from the carrier to the total signal power in the 1.23 MHz channel:

$$ACPR = \frac{Power_{\Delta f=30kHz, offset=750kHz}}{Signal\_Power_{\Delta f=1{,}23MHz}} \quad \ldots (3)$$

Given the ratio of the frequency band in which the products in the ACPR channel are measured and the signal bands, it turns out that ACPR is related to the IMD intermodulation distortion level as 1.23 MHz / 30 kHz, i.e. -50 dB corresponds to -23.9 dB for IMD.

ACPR is a generalization of such characteristics as the level of intermodulation distortion IMD on a broadband signal. More subtle criteria for modulated signal distortion, including non-linear ones, are RHO (waveform quality), EVM (vector modulus error), and IQ Offset (stellar offset).

ACPR is similar to the characteristic of the IMD intermodulation distortion level, but not for 3rd order nonlinear distortions, but for a broadband signal, since its measurement summarizes the products of odd-order distortions in the 30 kHz ACPR measurement band. A parameter similar to IP3, or a third-order intersection point, when evaluating ACPR-type distortions is the saturation power according to a given ACPR criterion, for example, with an acceptable level of ACPR = -50 dB.

The EVM parameter, like RHo, is a measure of the quality of a digital communications system. EVM in general has the meaning explained by the formula

$$EVM = \frac{rms\_error\_vector}{symbole\_magnitude} \cdot 100\% \qquad \ldots\ldots(4)$$

where rms_error_vector is the average error vector for the analysis time (characterizes the area of blurring of the spot on the star chart);
symbol_magnitude - the module of the vector from the origin to the point on the star chart, numerically equal to the symbol module. It characterizes the level of in-band distortion of the signal and shows how the position of the modulation points on the constellation diagram changes when non-linear distortions occur:

$$EVM = \frac{\sum_i \sqrt[2]{(I_{in}(i)-I_{out}(i))^2+(Q_{in}(i)-Q_{out}(i))^2}}{\sum_i \sqrt[2]{I_{in}^2(i)+Q_{out}^2(i)}} \cdot 100\%, \qquad \ldots.(5)$$

where Iin, Iout, Qin, Qout are the quadratures of the signal constellation at the input and output of the system.

4. Conclusion

An analysis of the linearization methods of power amplifiers shows that despite a large number of methods for reducing the products of nonlinear conversion at the output of a power amplifier, the problem of improving the quality of output stages of RES transmitters remains relevant. There can be no general recommendations on the choice of a linearization method for the design of radio systems for various purposes. In each case, the specificity of the principles of construction of the transmitter should be taken into account. Taking into account the specific features of the functioning of interference transmitters, in order to improve them in the direction of ensuring electromagnetic compatibility with other RES, it seems advisable to use the new compensation method proposed in this article to eliminate unwanted combination components in the spectrum of interference transmitters with quadrature modulation (the method of depletion of the input signal spectrum).

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