DIGITAL FILTERS AND THEIR ELECTRONIC NOISE

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INTRODUCTION

Data processing and information processing an important role belongs to the *filtration of signals* by frequency - surgery, performed with analog devices and circuits (active or passive filters) or with early positive and logic circuits in this case called the *filtrate numerical* signal. It may be noted that currently, the analog filtering and digital filtering are devoted enormous works a lot and publications, general-cognitive profile. But chronologically speaking, a digital filtering is a „younger", but definitely not so: one of the first publications dedicated to digital filtering can be considered work you appeared in the 1968 – 1973 IEEE authores A.Oppeenheim, R Schafer and all. Unlike their analog filters (rassive or active) the signal is processed in its true form and unchanged, digital filters have spectrum signal processing systems, represented by *sequences of numbers*, taken at *discrete* intervals of time. These filters use logic circuits while a signal processing is *linear*, resulting in *change* to the form of digital filter input signal, anyway – that as the analog. Easy to see that currently there is a crossover and more profound and effective influence of the classical theory of filters (active first) and „computer science" activity. But see, how good are digital filters, in terms of their *electronic own noise*.

1. FUNCTIONAL COMPARISON : ANALOG FILTERS – FILTERS DIGITAL

Emergence of digital filters can be explained by the tendency to *simulate* analog filters using a digital computer. An example is the creation of speech processing systems (in ex-urss these systems is called *speech synthesis*, the years 1965 t0 1968), which significantly contributed to the perception of digital filtering technique. Filter characteristics of these systems in general can unexpectedly affect signal processing. Country where computer simulated a filter characteristics can be adjusted and a quality of the whole system can be assessed befor implementation of analog filters.Thus, computer numerical synthesis offers considerable advantages in flexibility of the final system. Note however that no signal processing occurs in real time, so the computer will be used to numerically *approximate* or *simulate* an analog filter.As a result of this idea – the simulation of analog filters began to be used **filters numbers** – a way to achieve analog folters programmed computer. The case consists of an *analog – digital* (CAN), a digital filter and *digital – analog converter* (CNA) and in fact approximates an **analog filter**.

The last decade of the twentieth century is notable that in digital system have achieved impressive technological progress, leading to highlight applications where digital filtering techniques used them extensively: integrated circuits with high and very high integration (LSI & VLSI–ULSI) devices coupled through the load (CCD), etc. Based on these circuits was implemented a variety of structures and parameters of digital filters. A wide range of possibilities in the use of computer numerical filtering technique – for processing huge volumes of information – general purpose calculating for effective and rapid resolution of specific problems – small computer specialize strictly on your specific directions under the control of stored programs (so-called „stitched" or „cabled"). More reasons underlying trend microprocessors use digital filter realization, especially those operating in real time:

- small size and high reliability;
- flexibility and high efficiency of digital filters realized;
- contribution to increasing use of digital communication systems and data.

Of linear analog circuit theory is known that the frequency selection signals can be achieved using simple circuits, containing passive elements, L – inductance and C – condenser. It is also well known that the theory and practice of passive LC filter synthesis is sufficiently well established. The major disadvantage of these devices for the signals selecting by frequency that have been used successfully for decades in a row is deadlock, which found themselves in the era of miniaturization and especially of composite microminiaturization the signal processing electronic equipment: manufacture incompatibility of inductance L with a resistor R. Such a passive RC circuit can achieve a frequency dependent transfer function, ie an analog filter. Only that such a filter feature (figure 1) is unsatisfied: it has a very low quality of factor Q ($Q < 1$) due to loss of signal [1]. For this circuit in accordance with a theorem of Kirchhoff we can write a differential equation of order 1 the R and C elements covered by the current i :

$$Ri + u_C = u_1 \qquad (1)$$

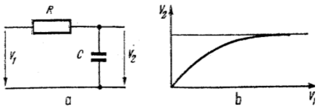


Figure 1. Passiv RC filter.

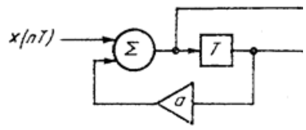


Figure 2. Digital filter.

or
$$Ri + \frac{1}{C}\int idt = u_1 \,. \qquad (2)$$

Performing the necessary transformations and taking into account that $u_C = u_2$, we can write:

$$\frac{1}{RC}\frac{du_2}{dt} + u_2 = u_1 \,. \qquad (3)$$

If the input signal u_1 is a unit step jump, the output signal u_2 will be composed of two components: the forced and free (exponential), ie

$$u_2 = 1 - e^{-\frac{t}{RC}} \,. \qquad (4)$$

Consider now a simple digital circuit shown in figure 2. In digital systems generally are not continuous variables, they intervals which are equal to each other, noting with T. In figure 2 is clear that performs operations: addition – a logical summation element Σ, multiplication by a constant, T the time dealy. Intermediate size $y_1(nT)$ is equal to the amount of output $y(nT)$, but delayed for time T, ie

$$y_1(nT) = y\,[(n-1)T] = y\,(nT - T). \qquad (5)$$

Considering this relationship, for circuit from figure 2 we can write:

$$y\,(nT) = x(n\,T) + a\,y_1\,(nT), \qquad (6)$$

or
$$y\,(nT) = x(n\,T) + a\,y\,(nT - T). \qquad (7)$$

Last equation can be presented in another form, namely:

$$a\,[y\,(nT) - y\,(nT - T)] + (1 - a)\,y\,(nT) = \\ = x(n\,T) \qquad (8)$$

or
$$a\,\Delta\,y\,(nT) + (1 - a)\,y\,(nT) = x(n\,T), \qquad (9)$$

where Δ shows the *difference operator*, ie

$$\Delta\,y\,(nT) = y\,(nT) - y\,(nT - T). \qquad (10)$$

Expression (9) is checked by *finite difference equation*, first order of the circuit of figure 2. This equation can be compared with the differential equation (3) analog circuit figure 1, the same role both equations are linear and can be applied the principle of superposition.

Reaction (or response) circuit in figure 2 to a unit step input quantity can be calculated for $n = 0$; 1; 2; 3; ....So, if $n = 0$ then from (7) results

$$y(0) = x(0) + a\,y(-T); \qquad (11)$$

the $n = 1$

$$y(T) = x(T) + a\,y(0) = x(T) + a\,[x(0) + a\,y(-T)] = \\ = x(T) + a\,x(0) + a^2\,y(-T); \qquad (12)$$

etc.

That,

$$\boldsymbol{y(nT)} = \sum_{i=0}^{n} a^i x(nT - iT) + a^{n+1} y(-T). \qquad (13)$$

Expression (13) is the *solution* of equation (7). If the size of unit step input is equal to 1 , so for $n \geq 0$ $y(-T) = 0$ and output size is equal to:

$$y(nT) = \sum_{i=0}^{n} a^i x(nT - iT), \qquad (14)$$

or
$$\boldsymbol{y(n\,t\,T)} = \frac{1 - a^{n+1}}{1 - a}. \qquad (15)$$

It is obvious that the relation (15) is observed (figure 3) that is similar to the graph in figure 2 – RC circuit response to a unit step. It is easy to prove that digital circuit in figure 2 with sizes $x(nT)$ and $y(nT)$ reversed, responds like a RL circuit, the impute stage which is applied to a jump drive.

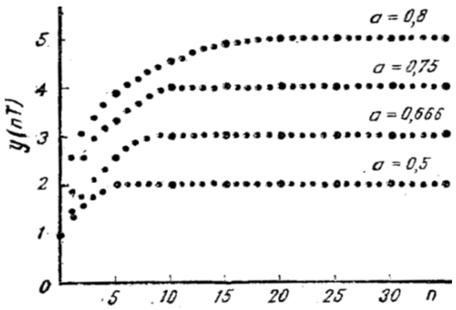


Figure 3. Response of digital filter to a unit step input.

Digital filters are increasingly being used currently in practice very widely. This is due to the advantages offered by digital signal processing to process analogic, which is important especially when are data processed in numerical form.

Among the advantages of digital to analog filters can be mentioned:

- high performance, precise and stable;
- great flexibility because the response filter;
- can be strictly controlled by changing their ratios

- absence of impedance adaptation problems;
- implementation in time division operating systems;
- very small dimensions which meet the requirements of submicron technologies.

Digital filtering technique just ask of meet to modern communications, computer facilities and the achieved performance. Digital filters advantage become dominant, the high performance filters are required, for example to achieve a *low–pass* filter with cut-off frequency $3 \cdot 10^{3}$ Hz and a slope of 90 dB at 3005 Hz frequency, most probably digital filter can only achieve this performance, especially when factors extremely filter can not be controlled strictly. These advantages are necessary and if the system requires a large number of filters. If digital filters are used, a common structure and performs all necessary filters, so the cost of implementation is a constant factor. When using analog filter, they have each done separately with different structures, the cost increases with the theirs number.

Another important advantage of digital filters occurs when a filter requires a great complexity of such an analog filter with more than 20 poles is almost impossible to achieve practically. For a digital filter that condition is not a problem, even if a larger number of poles.

Using digital filters is preferred if the system structure, in addition to filtering out other operations, the division operation time. Entering into a digital filter of a new lead, unlike analog systems, the emergence of new possibilities of operation or even the appearance of completely new systems. From the above it becomes obvious now why the use of digital filters is increasingly expanding range – their performance.

2. EVALUATION OF DIGITAL FILTERS IN TERMS OF THEIR ELECTRONIC NOISE

It can be seen with certainty that currently achieve business systems theory and practice of numerical signal processing has reached the optimal level. Are undeniable and important many advantages of digital filters compared to analog filters. It remaince to convince us that in terms of their electronic noise, these devices are in advantage over traditional analog filters, primarily to active *RC* filters. It seems (so far – it seems!) that precisely in this digital filters are losing ground: we are left to determine if so to what extent.

First of all, we clarify the notion of *their electronic noise*. It is not any noise and has appeared everywhere from home most diverse sources – disturbances of all kinds and beauty, to prevent capture, selection and signal processing such as electromagnetique nature, for example: the influence of external fields electrostatic and magnetic, electrical discharge, local radio and cosmic background, the presence of strong light sources, etc. The own electronic noise has another nature, he is a „pathology" of electronic elements and devices, their „roots" to start of operation principles, of physical and chemical properties of material of which are built, of corpuscular character of electricity (as a result of displacement of the polarized charge carriers). Therein lies the diversity of electronic noise, the conditions in which it appears and how it manifests (see figures 4 and 5). And if the nature of electronic noise types, such as Johnson noise (or *heat*, the noise spectrum „white") is relatively clear, you cannot say about the same noise at low frequencies - *flikker noise* (1/*f*), steady patter - *shot noise*, *burst noise* [1,2], *high-frequency noise* whose nature remains unclear, although there are various hypotheses [3]. It is evident from the above mentioned that the interferences outer of devices and equipment for frequency selection of signals can be effectively counteracted by electrostatic and magnetic shielding, using well-shielding wires connecting or using „twisted pairs" of wires and equipment chassis and power supplies – put to the „mass" safe to exclude disturbing impulses entering the mains.

All these measures do not solve anything, if the talk was about for own noise of electronic devices and equipment. Because in this case the noise coming from the inside of devices and is, figuratively speaking - a her disease. This „disease" are all treats for about 60 to 70 years, ever since were first electronic devises - diodes and other vacuum tubes. During these decades, led „hard fought" against its own noise, primarily for equipment destined to selection signals by frequency, capture and subsequent processing of these signals. Sure, it is the talk about electromagnetic signals very little intensity or power (of the order μV or μW) captured by the antennas of installations planetary or cosmic for radio and telecommunications, or the antennas of radio telescopes cosmic or for related with satellites and space ships that cross the interplanetary end cosmic space (*Hubble* Space Telescope example).

From signal capture systems to require the extreme sensitivity that is naturally determined by its noise level threshold, below which signals cannot be received and processed so. Here was given – and continues to give – the battle against noise, their first.

With respect to such noise can say that everyone understands what it is, looks like it feels like, and

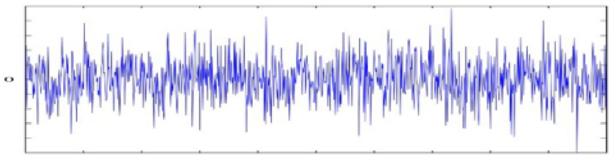


Figure 4. Oscillogramm „*white noise*”.

looks about as in figure 4, and its manifestation – be it a radio without the useful signal (a transmitting station, concrete) hear a dull murmur in speaker unit. This noise has a uniform spectrum called *white noise*. This spectrum is characteristic for the thermal noise (Johnson) which appears across a resistor in a circuit which can be presented in analytical form as a generator of electromotive force $\overline{E_{Zi}^2}$:

$$\overline{E_{Zi}^2} = 4\,k\,\mathrm{T}\,r_i\,\Delta f, \qquad (17)$$

where k is constant Boltsman; T - °K; r_i - active part of complex resistance Z_i , Ω; Δf - frequency band of calculated (or measured) noise. Rms voltage noise of a resistor with value $r = 1$ Ω, in the frequency $\Delta f = 1$ Hz and a temperature 20°C is equal to 0,127 nV. It seems that very little value, but keep in mind that the real values of resistors are thousands, tens thousands of times bigger and frequencies band makes up teens and hundreds of kHz. For example, if a resistor is 10 kΩ and a frequency 10 kHz, rms voltage noise is already ≈ 1,3 μV. It is known that the signals received from the Hubble Space Telescope had values between 0,01 ÷ 0,05 mV, ie only 10 times stronger than the noise of a resistor! Own noise of other passive elements - L and C can be ignored in comparison with noise resistors, is 2-3 orders lower. The major interest is present, of course the own noise of active elements of the electronic circuits: all diodes, transistors - bipolar and field effect (FET) in the production version as discrete devices and circuits integrates (as a finished product) which now form an impressive functional range: operational amplifiers (all kinds) and signal generators, logic gates and circuits locks, alimentation sources, current and voltage stabilizers, etc.

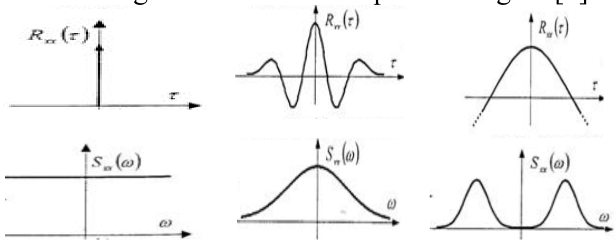
It was noted just above the Johnson noise (thermal) is associated with *"white noise"* – concept rather theoretical. According to Wiener–Hincin theorem [6], the link between the characteristic time is determined, R_{xx} (T) and frequency characterized bottles S_{xx} *(w)* of a random signal

$$S_{xx}(\omega) = F\{R_{xx}(\tau)\} \qquad (18)$$

and

$$R_{xx}(\tau) = F^{-1}\{S_{xx}(\omega)\}. \qquad (19)$$

In relations (18) and (19) F and F^{-1} shows, of course, direct and inverse Fourier transform, respectively. The function $R_{xx}(\tau)$ is the characteristic time of a random signal and is called *the correlation function*; $S_{xx}(\omega)$ – is a frequency characteristic of a random signal and is a *power spectral density function.* In a case of random signal x (t), the correlation function $R_{xx}(\tau)$ has a form δ pulse and power spectral density $S_{xx}(\omega)$ is constant, which means it contains all frequency components. Just this signal which **not exist in nature** – would be infinite power – is called *"white noise"* (figure 5, *a).* Real random signal noise could be called *"colored"* (in specialty publications is frequently called *"pink* noise"); they have different forms for correlation functions and power spectral density (figures 5, b and 5, c). The signals with a narrow correlation function have the band of power spectral density more large and such signals are closer to white noise; for the correlation function wide, a band of power spectral density is narrow and such signals are closer to a periodic signal [6].



a *b* *c*

Figure 5. With noise (*a*), the bandă range (*b*) and narrow (*c*).

Let's see, what about **their electronic own noise** of electronic devices, analog and digital. We start the analysis with extremely simple *RC* circuit, figure 1, *a.* If on input circuit to applies an input voltage V_1 single step, the response of circuit, V_2 will be a curve exponent reflecting the transient load capacitor *C.* It is not talk for the present in a filter circuit, because if its entry would apply a sinusoidal signal of variable frequency, the amplitude –frequency characteristic will be represented as the mirror image of the curve figure 1, *b.* It was noted that this exponent can be **modeled** using a digital device, shown in figure 2 with **digital filter** name.

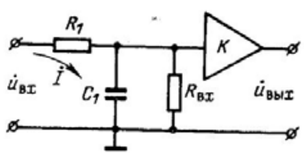


Figure 6. Floor LPF of order one.

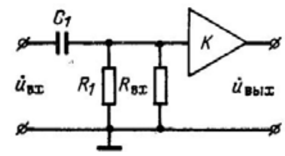


Figure 7. Floor HPF of order one.

The transfer function of LPF and HPF are, respectively:

$$W(p)=\frac{\dot{U}_{out}}{\dot{U}_{in}}=\frac{\frac{K}{T_1}}{p+\frac{1}{T_\Sigma}} \quad \text{and} \quad W(p)=\frac{\dot{U}_{out}}{\dot{U}_{in}}=\frac{Kp}{p+\frac{1}{T_\Sigma}}.$$

What is important to emphasize at this point: that this feature can be obtained with an *RC* circuit (analog) **elementary**, or digital circuit, but ... not as elementary; the latter contains a device assembly (Σ – Adder), a device for multiplying by *a* constant (an amplifier) and one with a delay time *T* (could be a *trigger* – toggle flip – flop logic circuit, CBBS). All three devices together represent tens of passive components – resistors and capacitors, diodes and transistors, levels of amplification or operational amplifiers! Becomes very clear that this set will be a true "band" compared with analog *RC* circuit, in terms of their electronic own noise.

Measured by the measuring system [3], a noise similar *RC* was equal to $(1.25 \div 1.3) \times 10^{-6}$ V. Own noise of digital filter trivial (as in figure 2) measured under the same conditions of experiment, was equal to 97.7×10^{-6} V, *id est* close to 100 times!

We will be continue a comparative analysis of own noise levels any devices on the order of 1 and 2 which performing functions of the filter transfer *low – pass* (LPF) and *high – pass* (HPF), analog and digital respectively.

A floor plan of order 1 LPF and HPF, as well as their corresponding transfer functions [7] are represented in figures 6 and 7. It is clear that functionally these devices cannot achieve high quality transfer function (lower quality factor which determines the slope away from vertical). For this reason the practice of order 2 floor are used, most often connected in cascade, so achieving a high quality factor and satisfactory slope (by approximation Butterworth, Cebyshev or Jakoby elliptic functions). The project process of digital filters based on the same principles of approximation [9]. *K* in both schemes is an amplifier (discrete or operational). As in previous schemes $\dot{U}_{in}=\dot{U}_{вх}, \dot{U}_{out}=\dot{U}_{вых}, R_{вх}=R_{in}$.

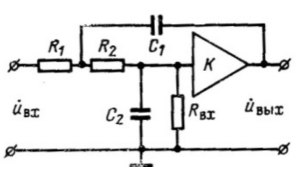


Figure 8. Floor 2 order LPF.

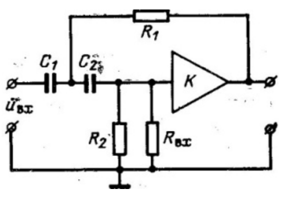


Figure 9. Floor 2 order HPF.

Certainly, no one floor of order 2 cannot provide a transfer function amplitude – frequency characteristic satisfactory LPF or HPF. For this purpose a few floors of order 2 are connected in cascade (in practice are usually sufficient three floors, which is a filter LPF of order 6), obtaining a

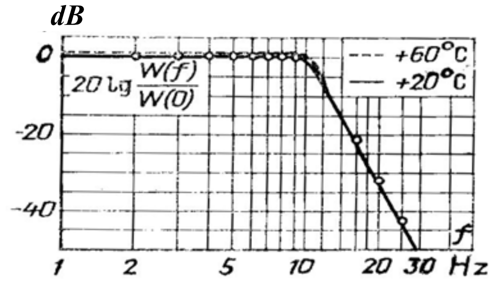


Figure 10. Amplitude–frequency characteristic LPF Butterworth filter of order 6.

frequency characteristic as shown in figure 10.

Consequently, to achieve the required transfer function LPF it is necessary 3 floors, each is an amplifier (operational), two resistors and two capacitor (noise which can be ignored), the total for filter – 3 amplifiers and six resistors. Number of the resistance for filter can increase to 9 in a realization an active *RC* band – pass filter. Previously it was noted that a digital filter can simulate any characteristic, including the represented in figure 10. To see but will "cost" this modelling.

3. BLOCK DIAGRAM AND BASIC EQUATIONS OF DIGITAL FILTERS

Suppose that is necessary to process a signal $x_1(t)$ with a digital filter. Block diagram of such a processing system is given in figure 11, and diagrams of time - in position 11, *b.* Analog signal $x_1(t)$ is applied to input (point *A)* and the sample is obtained as $x_2(nT)$ to point *B,* to input of converter an analog – digital. The analog – digital converter (ADC – *CAN*) transforms each sample into a number (*code*) as the *word number* – a sequence that is coded pulse amplitude $x_2(nT)$. The greater the length of the *word,* the greater the accuracy of representation. ADC output signal is obtained from

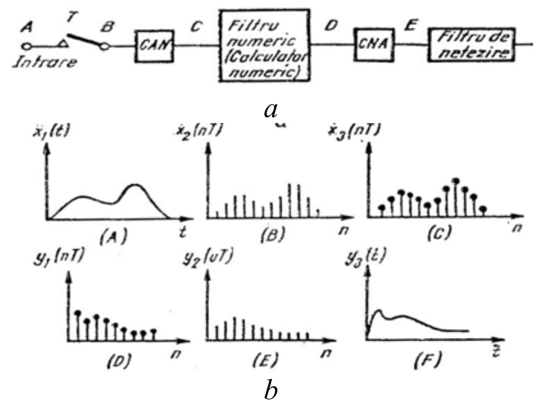


Figure 11. Diagram of a digital signal processing system *(a)* and diagrams of time *(b).*

x_3 *(nT)* to be applied to digital filter inlet. It stressed that the latter is only now "come into play", which is important in terms of its noise, because these devices are not ideal digital filter above, and 2: it is clear that digital filter to processing only strong enough signals, which in principle can overcome the own noise. This means that digital filters cannot be used as devices for signals selection by frequency and to be include in the primary reception floor of installation for capture and processing the weak signals, such has a electromagnetic nature (radio, TV, Telecommunications, etc). Returning to the diagram of figure 5, *a* the digital filter processing the signal $x_3(nT)$ to based on algorithm *P:*

$$y\,(nT) = P\,[x\,(nT)]\,, \qquad (20)$$

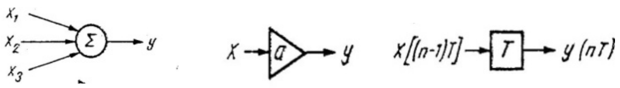
P is a mathematical operator describing digital filter that transforms the input quantity $x\,(nT)$ in the value of output $y(nT)$. Thus, after digital – analog conversion (DAC) and the smoothing filter, we get the analog signal $y_3(t)$. The algorithm *P* describes a digital filter and it is a finite difference equation. In paragraph 1 of this paper was presented equation of order 1, expression (9). But in general, a digital filter can be described by a finite difference equation of order *m:*

$$\boldsymbol{y(nT)} = \sum_{i=0}^{r} L_i x(nT - iT) - \sum_{r=1}^{m} K_i y(nT - iT), \qquad (21)$$

the K_i and L_i are constant real. This algorithm consists of three arithmetic operations: addition (figure 12, *a*), multiplied by the constant (L_i , K_i , figure 12, *b*) and delay time *T* (figure 12, *c*). Relationship between input and output values are represented in the figure 12. Conformed to the algorithm (21), a digital filter must also include memory of constants L_i and K_i. In most cases this algorithm is reduced to two private algorithms, corresponding finite difference equations of order 1 and 2:

$$y\,(nT) = K\,y\,(nT - T) + x\,(nT) + W\,(nT - T); \quad (22)$$

$$y\,(nT) = K_1\,y\,(nT - T) + K_2\,y\,(nT - 2\,T) + \\ + x\,(nT) + L_1\,x\,(nT - T) + L_2\,x\,(nT - 2\,T). \quad (23)$$



$y(nT) = x_1(nT) + x_2(nT) + x_3(nT)$ — *a*; $y(nT) = a\,x(nT)$ — *b*; $y(nT) = x[(n-1)T]$ — *c*

Figure 12. Graphic symbols of algebra logic elements.

The equations (22) and (23) correspond to the recursive filter *of order 1* and *order 2,* respectively. By definition [9], in *recursive* digital filter the output current quantity is determined by its previous values and current and past values of the input quantity; this dependence is expressed by the formulation (21), recursive filters is called *infinite memory filters.* Unlikely that the memory is really infinite, but these filters require additional units of memory is a fact. Therefore, we can talk about an

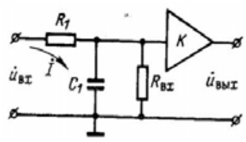


Figure 13. Analog LPF of order 1.

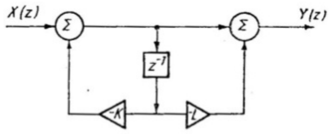


Figure 14. Digital low-pass filter of order 1.

The transfer function in a complex form (analog)

$$H(s) = \frac{1}{1 + sRC}$$

The transfer function (Z-standard transformed) :

$$H(z) = \frac{Y(z)}{X(z)} = \frac{1 - Lz^{-1}}{1 - Kz^{-1}}$$

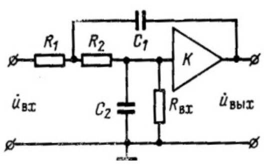


Figure 15. Analog filter LPF of 2nd order (scheme Sallen-Key).

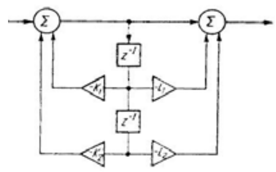


Figure 16. Recursiv filter LPF of 2nd order (canonical scheme).

obvious **redundancy** logic circuit type CBBS and therefore a high level of noise on their own. In filters *norecursive* the output quantity is determined explicitly by present and past values of the input size, *id est*

$$y(nT) = \sum_{i=0}^{r} L_i x(nT - iT).$$

Because of their structure, these filters are called *transverse*, with *finite memory.* But in this case, the excess of logic circuits is present and their high level of own noise cannot be avoided. This development can be justified by comparing various schemes to achieve the transfer function of order 1 and 2 with analog and digital means, which are show in the following figures (figures 13, 14 and 15, 16).

The transfer function of recursive digital filter of order 2 is

$$H(z) = \frac{1 - L_1 z^{-1} - L_2 z^{-2}}{1 - K_1 z^{-1} - K_2 z^{-2}}.$$

The analog filter transfer function of order 2 is

$$H(s) = \frac{\dfrac{K}{T_1 T_2}}{s^2 + s\dfrac{(1-K)T_1 + T_1\dfrac{R_2}{R_{in}} + T_2 + T_c}{T_1 T_2} + \dfrac{R_1 + R_2 + R_{in}}{T_1 T_2 R_{in}}}$$

Here

$$\dot{U}_{in} = \dot{U}_{вх}; T_1 = R_1 C_1; T_2 = R_2 C_2 \; T_c = R_1 C_2 .$$

Comparing the diagrams represented in figures 13, 14 and 15, 16 can be seen that the analogue schemes contain a single active element – stage amplifier or operational amplifier and some passive elements; the digitals are made up of adders, multipliers and delay circuits, with a large number of active elements – amplification floors or operational amplifiers. Digital filters, even the simplest requires additional memory devices also contain bipolar transistors or field effect (FET); about the set of passive elements do not deserve discussion. For example, in figure 17 present a 6-order *RC* active filter mounted version of the classic 60-70 years (twentieth century), based on discrete elements – resistors, capacitors and transistors type *pnp* and *npn* from soviet production; this filter provides a characteristic LPF.

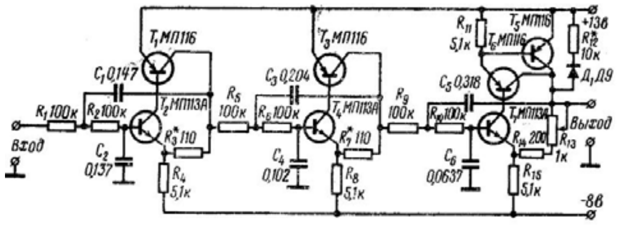


Figure 17. Active *RC* analog filter of order 6(LPF).

In the figure 18 is the block diagram of two floors of the six numerical order of a filter 6, which can produce (simulate) and amplitude-frequency characteristic of LPF.

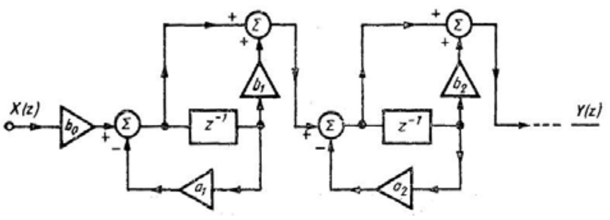


Figure 18. Digital filter in cascade (2 floors) of order 6.

The corresponding transfer function is:

$$H(z) = b_0 \cdot \left(\frac{1 + b_1 z^{-1}}{1 + a_1 z^{-1}} \right) \cdot \left(\frac{1 + b_2 z^{-1}}{1 + a_2 z^{-1}} \right) \cdot \left(\frac{1 + b_3 z^{-1}}{1 + a_3 z^{-1}} \right) \cdot$$

$$\cdot \left(\frac{1 + b_4 z^{-1}}{1 + a_4 z^{-1}} \right) \cdot \left(\frac{1 + b_5 z^{-1}}{1 + a_5 z^{-1}} \right) \cdot \left(\frac{1 + b_6 z^{-1}}{1 + a_6 z^{-1}} \right) . \quad (25)$$

4. FINAL CONSIDERATIONS IN COMPARATIV EVALUATION OF ANALOG AND DIGITAL FILTERS OWN NOISE

If active *RC* filter (Sallen-Key scheme) only just made LPF transfer function, a digital filter order number 6 can simulate any characteristic and not only by means of selection signals by frequency, but a ... regulators, given a fact that the general notion of digital filter includes the notion of *digital regulator.*

A *z* transfer function of each system with sampling correction can be treated by numerical filtering concepts. Thus, a digital filters can perform any required controllers behaviour - proportional P*,* proportional integral *PI*, proportional integral and differential *PID,* delay etc. Excellent! Only that can all be achieved in systems with well-defined signals strong enough, when the noise of their tens and hundreds active and passive elements does not affect the proper functioning of devices and equipment, as a whole.

Therefore, a digital filters can be used anywhere, except the floors of equipment for capture and primary processing of low power signal (tenths of μW), which can "lose" in the own noise of this facilities. This can be confirmed by the results of measurements made by measuring installation proposed in [3]; own noise of active *RC* filter in figure 17, measured at the output filter with input terminals shorted, was equal to 3.7×10^{-5} V / Hz ^{½}.

The own noise of numeric filter only of order 2 (two floors, figure 18), measured under the same conditions, was equal to 21×10^{-5} V / Hz ^{½}, ie five times higher, which can be explained by excessive number (redundancy) of active devices and circuits.

Although a digital filters convert the analog signal into digital code, thus ensuring advanced security and certain independence of all kinds of external disturbances (electrostatic and magnetic fields, lightning, interference and jamming, interference by alimentation mains, etc…), its own noise leave mark on the final output quality – images for TV programs, voice and music clips for radio links, telecommunications and mobile telephone.

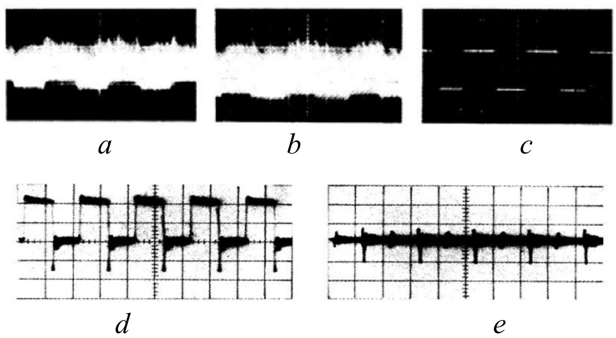
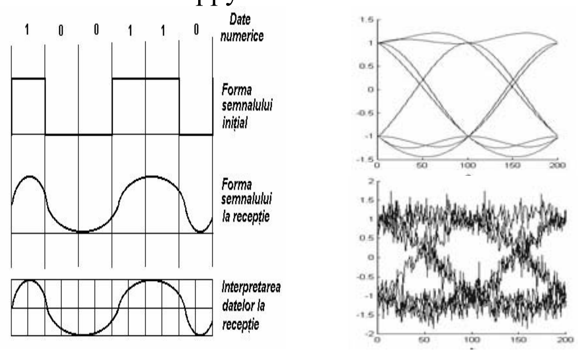


Figure19. Own noise influence on the functioning of CBBS.

Own noise overlaps to the useful signal (figure 19, *a* and *b)* and necessity to "cleanse" it (figure 19, *c),* which is not always successful in simple terms – in figure 19, it is shown that the noise reduction trend was mitigated ... the useful signal, noise continued and happy!



a *b*

Figure 20. Errors in signals interpreting.

Even if the relationship signal / noise a ratio is in favour of the numerator, this does not guarantee absolutely exact interpretation and in digital systems (not necessarily filters), which can be seen in figure 20, *a* and *b*. Clearly, as undesirable and unwanted is a influence of own electronic noise in numerical data processing systems.

How paradoxical isn't seems, but the problem is not given due attention, if judging by the low number of publications devoted to this problem and the approach surface level. For example, in [1] we read: in complementary transistors with field effect and *pn* junction, 2SJ72 and 2SK147 (firm Toshiba) **cell geometry** is used (our checking) of the grid, allowing to obtain a phenomenal low noise voltage e_Z = 0,7nV/$\sqrt{Hz}$, which correspond to the thermal noise of a 30 Ω resistor (quoted concluded).

Formidable, only too laconic: no method of calculation (if exist any) or references to other sources. Sure in a few decades there have been dozens of works dedicated to calculation of noise parameters of components, devices and electronic circuits. Just only a high signal/noise correlation can guarantee of numerical circuits (and filters) success forever.

5. CONCLUSIONS

1. In the information processing discern an important paper a filtering (selection) of signals (by frequency). This can be done analogic, using a oscillating *R-L-C* circuits or numeric, from digital filters.
2. A digital filters have advantages over analog (including active *RC* filters) – characteristics accurate and stable, small dimensions, impedance which not required to adopt and – very importantly – outstanding flexibility: the filter type can be modified by simply change of coefficients.
3. Comparative analysis of digital filters and active *RC* filters highlights a particular feature of digital filters: redundancy, *id est* the excessive number of active elements for logic circuits, used for the transfer functions *low - pass, high - pass* and *passband.* The active elements (bipolar or field effect transistors, discrete or integrated) presents main source of devices and equipment own *electronic noise*. So, in this regard are not impeccable digital filters: its noise level exceeds own noise active *RC* filters similar tens or even hundreds of times.
4. Digital filters are not influenced by all kinds of external disturbance – magnetic or electrostatic fields, jamming and lightning, accidental impulses by the alimentation circuit. But their noise can influence the correct interpretation of the processes, so numerical filters require sufficiently strong signals. Therefore their use in input floors of installations for capture and processing of information in the form of electromagnetic signals is not justified, are preferred in such cases analogic, active *RC* filters.

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