TWO-STAGE DIGITAL FILTERING OF A SIGNAL HAVING AN UNKNOWN FREQUENCY

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Two principal methods (see [1]) are currently in use for the solution of a number of practical problems in the measurement of the unknown carrier frequency of a packet of radio-frequency pulses which appear at a constant repetition frequency F_r. The first method consists in constructing a multichannel system containing narrow-band filters for each of the comparatively narrow regions Δf_i, i = 1...N of the spectrum $S(f)$ of the expected frequencies (Fig. 1a), while the ensemble of all the channels covers the spectrum completely ("parallel" processing). The second method consists in searching for the expected signal according to its frequency and is implemented by means of a one-channel tunable system ("sequential" processing). The resolving power of the systems is determined by the passband of each of the "elementary" filters Δf and their frequency diversity δf which, in turn, depends on the statistics of the expected signal (see [1]). An excess of the voltage above the threshold level at the output of the i-th "elementary" filter is evidence of the fact that the frequency of the processed signal $f_s = iF_r/N$, where N is the order of the processing system and is equal to the number of channels in the system.

The principal shortcoming of the first method is the cumbersome nature of the system. A long processing time, which is proportional to the number of channels used, is inherent in the second method. These shortcomings may be almost completely eliminated by two-stage filtering consisting of a broad-band (crude) stage and a narrow-band (refined) stage. However, narrow-band filters must be tuned over wide limits, and this involves difficulties for analog implementation. If the filter parameters are altered mechanically for tuning purposes, then it is impossible to satisfy the filtering requirements of signals having a rapidly varying frequency in real time. At the same time, the tuning range of electronically controlled elements (varicaps, etc.) is very restricted, while the laws governing the variation of the control voltages are essentially nonlinear. Therefore, the purpose of the present work is to investigate the possibilities and methods of two-stage filtering of signals using tunable digital filters.

The essence of two-stage filtering consists in the following. Assume that it is required to measure an unknown signal frequency in a band 0 to $F_r/2$ with a maximum error not exceeding δf, it being true that $F_r/(2\delta f) = N$. During the stage of crude measurement of f_s (Fig. 1b), $M \ll N$ "elementary" filters having the bands $\Delta f_1 \gg \delta f$ are used. During the stage of refined measurement (Fig. 1a), both the bands $\Delta f = \delta f$ and the center frequencies of these M filters are varied. Under these conditions the range Δf_M is covered.

Let us consider the comparative responses of parallel, series, and two-stage digital filtering systems. The number of "elementary" filters in the parallel system must be N, while the processing time is $t_{pro} \simeq (3\text{-}4)/\Delta f$. In the series system one filter is required, but the processing time increases to the value $t_{pro} \simeq (3\text{-}4)N/\Delta f$. For two-stage measurement of the signal frequency only $M \ll N$ "elementary" filters are required, while $t_{pro} \simeq (3\text{-}4)M/(N\Delta f) + (3\text{-}4)/\Delta f \simeq (3\text{-}4)/\Delta f$ because $M \ll N$. Consequently, in a two-stage filtering system one may achieve economy of hardware and energy expenditures while preserving the required speed of response.

Let us stipulate the following initial conditions for determining the quantity M for known F_r and Δf: a separate comb of M "elementary" amplitude-frequency responses (AFR) during the stage of refined measurement of f_c must cover the frequency range occupied by two "elementary" AFR during the stage of crude measurement (Fig. 1a and 1b); a comb of "elementary" AFR must completely cover ΔF during crude measurement; the device must ensure simultaneous measurement of the frequencies of L independent signals. Starting from these conditions, one may write $2F_r/(LM) = \Delta f M$, $LM = \gamma[2LF_r/\Delta f] = \gamma[2LN]$.

Thus, the speed of the response of the device and the required hardware expenditures for implementing it for stipulated F_r and Δf are determined by the quantity L.

The use of the two-stage filtering method with a variable resolving power imposes specific requirements

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