The American Journal of Applied Sciences
(ISSN – 2689-0992)
Published: August 20, 2020 | **Pages:** 60-65
Doi: https://doi.org/10.37547/tajas/Volume02Issue08-09

IMPACT FACTOR
2020: 5. 276

Journal Website: http://usajournalshub.com/index,php/tajas

The Role Of Methods And Algorithms Of Sound Processing

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ABSTRACT

This article discusses the issues of methods and algorithms for processing audio signals, the purpose and classification of filters, basic digital filters, low-order and high-pass filters of the first order, as well as the purpose of processing, which can be purely technical tasks, such as matching signal parameters with the characteristics of electro-acoustic tract.

KEYWORDS

Dynamics, mechanisms, digital processing, electroacoustics, plugins, editors, sound recording, acoustics, microphones.

INTRODUCTION

Processing an audio signal means changing its frequency or phase response, narrowing or expanding the dynamic range, applying amplitude, frequency or phase modulation, removing noise, and creating time-delayed fading copies of this signal. The purpose of processing can be both purely technical tasks, such as matching signal parameters with the characteristics of the electro-acoustic path, and art, determined by the sound engineer, in particular, these can be various sound effects (tremolo, vibrato, chorus, echo, reverb, and others).

Currently, the processing of audio signals is carried out mainly in digital form using sound processors. If earlier sound and radio studios were located on several tens of square meters, now they can be replaced by a good computer, which is superior in capacity to ten such studios combined, and in cost it is many times cheaper than one studio. This removes many financial barriers and makes sound recording more accessible to both professional and ordinary amateur.

Modern software allows you to make arbitrarily complex conversions of audio signals and create the most incredible sound effects. In analogue technology, almost every single sound effect is created by using a separate device, when each such device can be very expensive. In digital technology, the quality of signal processing in them is much less dependent on the quality of the equipment. It is very important that for various manipulations with sound a constant change of equipment is not required, it is enough to change the software. That is why digital technology today almost completely replaced the old analog equipment from the studios.

Digital sound processing mechanisms are implemented both in software and in hardware. Most often, sound processing is carried out using sound cards for professional and domestic purposes using sound editors for special purposes. The most widely used editor is Sound Forge, which has the largest package of sound plug-ins; the sound editors Cool Edit Pro and Steinberg WavLab are very popular.[1]

MATERIALS AND METHODS

A particularly high quality of sound processing is provided by the Waves Platinum Native Bundle 4 plug-in package. Sound signal processing can be carried out in real time or applied to an already recorded phonogram. Phonogram processing is used at the stage of mastering or preparing them for replication, when speed is not important, but a meticulous study of all the nuances of sound.

The need for serious frequency correction of sound reproducing equipment is most often due to the poor acoustic characteristics of the premises where a concert is held or a sound recording is heard. If, for example, the hall has smooth, solid surfaces of the stage and floor, concrete or brick walls, a tin roof, then all this can begin to rattle and rattle, and in the best case, the vocalist and listeners will cease to understand the words due to reduced intelligibility. Serious problems with high-quality sound perception arise in the car. The equipment for the frequency correction of sound signals is a connecting link between the sound of a sound reproducing system and the response of the room, and it can largely solve such problems. [2]

Frequency correctors are used in all recording and sound broadcasting studios. With their help, purely technical issues are usually solved, such as limiting the sound path band, suppressing low-frequency noise and network interference, correcting the amplitude-frequency characteristics (AFC) of microphones, loudspeakers, and rooms. Recently, when mixing and mastering, more and more widely, frequency correction is used to solve the creative tasks of sound engineers to create artistic sound effects and give the sound a new color.

This became possible with individual frequency processing of almost every musical instrument and vocalist.

In audio technology, the frequency correction of sound signals is carried out using the following devices and filters, which can be in the form of a separate hardware or software product:

- Bandwidth restriction filters,
- Filters of smooth rise and fall of the frequency response,
- Bandpass filters,
- Presence filters
- Graphic equalizers,
- Parametric equalizers,
- Paragraph equalizers,
- Crossovers.

The filters used in frequency correction, according to the implementation principle, are primarily divided into analog and digital. In turn, analog filters can be performed on both passive and active elements. According to the principle of operation, all filters are divided into linear and nonlinear. Depending on the type of impulse transfer function, the filters are divided into recursive (IIR filters, with infinite impulse response) and non-recursive (FIR filters, with finite impulse response). All analog filters are recursive, digital filters can be both recursive and non-recursive. [3]

RESULT AND DISCUSSION

Among the many recursive filters by the type of the transfer function, the highest quality filters are separately distinguished:

Bessel filters - have the smoothest frequency response and phase response (especially in the passband), however, the steepness of the frequency response decline is the smallest;

Butterworth filters - have a steeper decrease in frequency response (6N dB / octave, N - filter order) and a less linear phase response;

Chebyshev filters - has an even steeper decrease in frequency response, however, their frequency response is not monotonic, but has oscillations of a given level in the passband or in the suppression band. The phase response of the Chebyshev filters is nonmonotonic and has a peak near the cutoff frequency. When lower filter pulsations are set, the steepness of the AFC decay decreases, and the Chebyshev filter turns into a Butterworth filter;

elliptical filters - have the steepest frequency response decay, but have frequency response pulsations both in the passband and in the suppression band. Phase response.

elliptic filters are not monotonous. With increasing pulsation requirements, this filter turns into a Chebyshev filter.

According to what frequencies the filter is allowed to pass through (delayed), the filters used in frequency correction are divided into the following groups.

- Lowpass filter (Lowpass - LP) selects low frequencies to cutoff frequency fc and suppresses frequencies above this frequency.

The Highpass (HP) filter selects frequencies above the cutoff frequency and suppresses frequencies below this frequency. [4]

A bandpass filter (Bandpass - BP) isolates frequencies above the cutoff frequency fcl and below the cutoff frequency fch. Frequencies below fcl and above fch are suppressed. A Band Notch Filter (Bandreject – BR) provides frequencies above the cutoff frequency fch and below the cutoff frequency fcl. Frequencies below fch and above fcl are suppressed. A narrow-band filter (Resonator filter) passes frequencies in a narrow band near the cutoff frequency fc. A notch filter suppresses frequencies in a narrow band near the cutoff frequency fc.

The gain of these filters is determined on a logarithmic scale and can be higher and lower than 0 dB. Typical frequency response of these filters. [5] Quite often, low-cut and high-cut filters are used that limit the frequency range. Most of these filters are recursive, they are designed on the basis of basic filters - low-pass filters, high-pass filters and all pass-through filters (high-pass filters).

If the design of analog filters is based on the use of Laplace transforms, then the calculation of digital filters is carried out using Z-transforms. This is due to the fact that in the first case we have the original signals in the form of a continuous function of time, and in the second, the sound signals are discrete functions of time.[6]

Digital filters are implemented on the basis of only three elements: a delay of one clock cycle with the transfer function z-1, a binary adder and a binary multiplier. When constructing filters on these elements, direct and feedback relationships are used; the coefficients of these relationships are denoted, respectively, as bn and an. Mathematically, the operation of such filters is described by a difference equation (equation in finite differences), as the dependence of the input x (n) and output y (n) signals as a function of the delay time, filter coefficients, and discrete time

nT , where n - sample number, T = 1/ fs , fs - sampling frequency

The transfer function of the filter H (z) is defined as the ratio of the function H (z) Z-images of the output Y (z) and input X (z) signals. The transfer module is the phase response of the frequency response of the filter (AFC), determined by the argument of this function (PFC). Digital filters can work in boost mode when H (z)

> 1 and in the cut-off mode, when H (z) <1. The number of zeros of the filter M is one greater than the number of direct coupling coefficients of the difference equation, and the number of poles N is equal to the number of feedback coefficients of this equation. The order of the filter is determined by the largest of the values of M and N, it is also equal to the order of the polynomial of the difference equation.

Complex filters of 2 and higher orders are built on the basis of links of a lower order. Links 1 and 2 orders can be included in series, in parallel or in combination. When the links are connected in series, their transfer characteristics are multiplied, and in parallel, they are summed. [7]

The simplest filter scheme of the first-order low-pass filter with one zero based on one delay element and the adder is shown in Fig. 1.3. In this scheme, direct communication is used, with which direct and delayed signals are summed. The operation of such a filter is described by the difference equation y (n) = x (n) + x (n -1).

The order of this equation determines the order of the filter. The transfer function of the filter in the form of a Z-transformation has the form H (h) = 1+ z-1, h £ 0.5,

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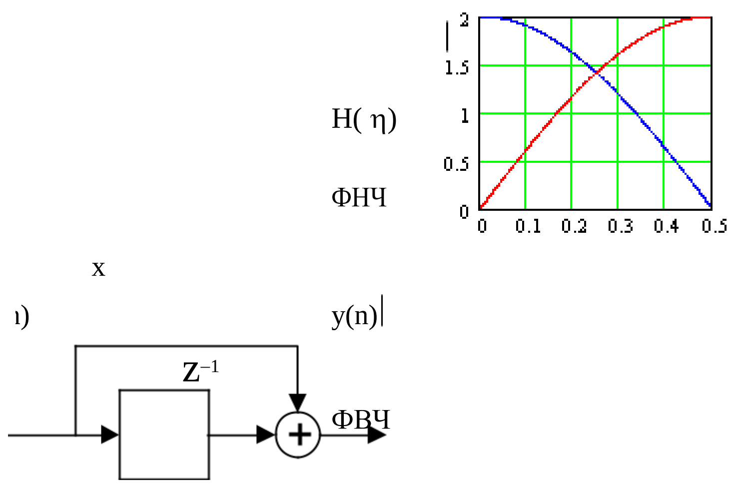

Fig. 1. Low-pass filter with one zero and its frequency response

CONCLUSION

The implementation of such a computational procedure faces two problems. At a sampling frequency of 44100 Hz, a 4410 element filter is required to perform a convolution operation to simulate early reflections delayed up to 100 ms. To simulate late reflections, the duration of which reaches 2 s, 88200 element filter for each sound channel is required. This is not technically feasible. [8]

The length of the impulse responses of the rooms depends on their reverberation time. The reverb perception is most affected by the reverb attenuation from the maximum value to -15 dB. We can assume that attenuation below the level of -60 dB practically does not affect perception. The length of the response pulses of the rooms before they decay to a level of -60 dB can be calculated in a few seconds, i.e. the impulse response length M can be tens or even hundreds of thousands of digital samples.

If the duration of the audio signal is N samples, and the length of the response pulse is M samples, then it is necessary to perform M + N operations of multiplication and addition during this time. With a duration of a musical passage of 1 min and a duration of an impulse response of 3 seconds, about 350 trillion addition and multiplication operations are required during this time. Therefore, the direct calculation of the convolution in real time using the above formula is impossible on modern personal computers due to too high computational complexity. However, convolution can be calculated using the fast Fourier transform, which reduces the number of multiplications per count.

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