A perceptual approach on equalization

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This article is about equalizing, with special attention for the digital domain. On the beginning, we are going to consider the filters in the analog domain, trying to find useful relations between the system function in the s complex variable and the sonic effects that it produces. Then we will study the conversion of the filter in the digital domain. The object of this article is finding the main elements that can improve the quality of an equalizer.

1 In the analog domain

In this section, we are considering ideal and perfectly linear filters, so they can be represented by a polynomial system function in the s domain.

1.1 Second order filters

Few peoples know that many analog equalizers and almost all the digital equalizers in commercial applications have only a second order IIR stage for each band. This is particularly true for the mixer channel equalizers, because its circuit (or algorithm in the digital ones) should be as simple as possible because it is replicated one time for each channel. But often, this is also true for totally dedicated processors.

The standard form for a second order IIR stage is:

$$G(s) = \frac{a'_2 s^2 + a'_1 s + a'_0}{b'_2 s^2 + b'_1 s + b'_0}$$

and after normalizing

$$G(s) = K \frac{s^2 + a_1 s + a_0}{s^2 + b_1 s + b_0}$$

K does not affect the frequency response; so, excluding K, a second order IIR filter is identified by 4 parameters. And so, choosing suitably these four parameters, it is possible to recreate the response of almost all the available equalizers. Some of these equalizers are warm, crispy and have a pleasant sound, other sound muddy, unpleasant and remove definition from the musical program. But all these equalizers are based on the same second order structure, so all these differences are only in four parameters! This means that the filter's sound quality mainly depends on its adjustment, that is how the user parameters are translated into the filter's coefficient. Saying ”mainly” we mean that there are other elements (see the next chapter) that can affect the sound quality, but their relevance is not primary.

1.2 The influence of the parametrization

Let's give a definition of parametrization: Let $p_0, p_1, \cdots, p_n$ be the user parameters (i.e. the parameters that the user can set on the panel of the equalizer). A parametrization is a function $(a_0, a_1, b_0, b_1) = F(p_0, p_1, \cdots, p_n)$ that determines the 4 biquadratic filter coefficient, given all the user parameters. A very simple parametrization could be:

$$\begin{cases} a_0 = p_0 \\ a_1 = p_1 \\ a_2 = p_2 \\ a_3 = p_3 \end{cases}$$

but a user will have serious difficulties finding a useful setting for the filter! That's because the parametrization has been choosen according to mathematical criteria and not to acoustic parameters. This is a more useful parametrization:

$$\begin{cases} p_0 = Frequency \\ p_1 = Gain \\ p_2 = Q \end{cases}$$

This is typical for a parametric equalizer, where its 3 parameters cover 3 of 4 degree of freedom of the filter. The remaining parameters will be implicitly set into the F function. It is clear that the sound quality of an equalizer is strictly related to its parametrization; so, let's try to understand which are the primary elements involved in the sound quality. To do this, consider a peaking filter (the one used into the mid-bands of equalizers). Its transfer function is:

$$G(s) = K \frac{s^2 + a_1 s + a_0}{s^2 + b_1 s + a_0}$$

the filter has only 3 degree of freedom. If the user parameters are:

$$\begin{cases} p_0 = Frequency \\ p_1 = Gain \\ p_2 = Q \end{cases}$$

the user has the control over all the 3 degree of freedom and so, by these three parameters, he is able to reach every possible setting of the filter. But why are there equalizers with wonderful mid-bands, and other eq whose mid-bands sound nasty? If you have two different parametric equalizers and give a setting to the first, is possible to obtain on the second eq the same sound canvas of the first, by finding the right value of the parameters. But why one equalizer is better than the other? To answer this question, let's suppose we have a particular equalizer in which the three user parameter are directly connected to the coefficients of the filter. Close this eq in a black box and give it to a user. Surely he will define horrible this equalizer, because the parameters have no musical mean. Even so, this strange eq can recreate the sound of all the second order mid-band in the world, the ones of the best equalizers too! So, where is the problem? Now the answer is easy: using that strange eq, when we find a useful setting if we change one of the three parameters we loose all the musicality of the setting; for a musical useful change, we have to modify all the three parameters at the same time; if not so, we loose the "character" of that setting. This is also true if the user parameters are Frequency, Gain and Q: if we move one of these controls, a good parametrization should change the other two. This is called a good progressivity; it means that if we change one parameter starting from a good sounding setting, what we get is another good sounding setting that doesn't loose the "character" and the musicality of the first. For example, if we act on the Frequency parameter, the Gain and the Q have to be modified because the filter operates on a different frequency range, where the human ear has a different sensibility. So, the main feature that gives a good sound to an equalizer is the progressivity of the parameters.

1.3 Increasing the order

If we want a wider sound palette than second order filters, we could increase the order to third or greater. For simplicity, let's consider a third order filter: by normalizing its system function in the same way of the second order case, we obtain 6 degree of freedom. Now we can follow two different approaches: give the user all the parameter (or a great part of them), or give him only a few parameters and fix the other degree in the parametrization function as intrinsic constants. Both solutions have some disadvantages. If the user has all the parameters, every time he wants to setup the filter, he have to deal with an excessive number of controls; unlike frequency and gain, many of these parameter have not a well defined purpose and only describe the character of the filter; these are those parameter typically described with strange and fanciful names. So the user have to deal with many not well defined parameters, and he have to make several attempts to reach the wanted setting. On the other side, giving the user only some parameters, it will be very difficult to find a parametrization that fixes all the other degree of freedom giving a good sound quality. In these cases, it is useful to design the filter starting from the schematics of hi-end analog equalizers.

2 Digital domain

Many people know that hi-end analog equalizers can give a sound character that generally we cannot obtain using digital eq. Which features are involved in this difference?

2.1 Numeric approximation

The discretization method affects the sound of the filter; this is not as evident as the design of the transfer function, but can introduce some audible artifacts. We can divide this topic into two parts: the quantization of the samples and of the filter coefficient, and the time discretization. About the quantization, we suggest a specialized text like [1]; we now suppose that the samples and the coefficient are represented using floating point numbers, so we do not consider quantization issues. Let's talk about time discretization: the most used method is the bilinear transformation. The advantage of this method, over the simpler finite difference method, is that the order of convergence doubles but the order of the digital filter (i.e. the number of multiplications and additions for each sample) remains unchanged. That's why the bilinear transformation approximate the integral by the trapezoidal formula, that has a order of convergence double that the order of rectangular formula. The trapezoidal formula is an implicit method, but for linear differential equation (like the ones of an equalizer) doesn't create non-computability problems. Do higher order methods bring to better sonic results? The answer, after many listening tests, is that there are no relevant sound quality enhancements. Instead, the computational cost is much higher, at least by a factor of 2. If we want to double the computational cost of the filter in order to enhance its sonic response, there is a much more efficient method: the oversampling. A factor of 2 can greatly reduce all the artifact of the bilinear transform. Near Nyquest frequency, digital filters have an inaccurate behaviour and create artifacts in the signal. This phenomenon is quite audible in many digital mixers where the great number of channel multiplies the negative effects of each filter. The deterioration is like a loose of definition in the high frequencies; analog mixers do not have the Nyquest frequency limitation, so they are much more transparent. If the filter works in oversampling, the Nyquest frequency is placed far away from the frequency range of the filters, so the artifact are greatly reduced.

2.2 Nonlinearities and aliasing

Another important feature of analog gear is that, unlike their mathematical models, real electrical component are not ideal and linear. The nature of the nonlinearities are so different that we cannot study them with an analytic approach: they are not limited to nonlinear transfer functions (waveshapers) like the ones of the active stages, but there are many other classes of nonlinearities that affect the signal in its path: for example the magnetic hysteresis, the dependence of the component's value on the temperature, the presence of microfonic components, etc It isn't possible to create a model for each class of nonlinearity, so we are going to describe two of the most useful ones. The first is the mostly used because it is very simple to implement: it is the distortion of the active stage next to the passive equalization circuit. It can be simulated with a static nonlinearity (a time invariant transfer function, also known as waveshaper) or with a dynamic one (like a tube stage with bias shifting [2]) . This model is simple because does not affect the linearity of the filter that can be implemented using standard techniques. The second model is more complex and simulates the nonlinearities of passive components; the resistance, capacitance and inductance values are not contant, but depend on their state variable: the resistance depends on the current, the capacitance depends on the charge and the inductance depends on the magnetic flow. Now, the equation of such a model is not a linear differential equation; in the preceding model all the nonlinearities were out of the differential part, but now the non linear elements are inside the filter equation. The time discretization is much more difficult: using an implicit method we need to solve a nonlinear equation of each sample, so the computational cost are very high. Using an explicit method, like Runge-Kutta, we must use a high factor of oversampling, in order to maintain the accuracy of the solution.

Every nonlinear model generates aliasing; the only solution is oversampling. The choice of the factor depends on the strength of the nonlinearity's action. For a light nonlinear action, we can use a x 2 factor. If the nonlinear effects are evident, a x 4 factor can give good results. In the case of heavy distortions, like in a guitar preamp simulation, at least a x 8 factor is needed. Oversampling is not only useful to reduce aliasing; it can give all the advantages described in section 2.1. So, working in oversampling, is a suggested practice where the sound quality is a primary requirement.

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