The sneaky pitfalls of feedback and feedback theory

Dr. Hans R.E. van Maanen (Temporal Coherence, info@temporalcoherence.nl)

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Summary and conclusions

Feedback seems to be a panacea for all the shortcomings of audio equipment. Yet there are many critics of feedback in the 'high-end' audio community. There is no doubt that the *specifications* of many semiconductor amplifiers are much better than those of loudspeakers and vacuum tube amplifiers, but this does not correlate well with the *perceptual* assessment of the equipment. How is that possible? And why is it possible to hear differences between amplifiers that have distortions two orders of magnitude smaller than the speakers one needs to listen to these amplifiers? Are certain phenomena overlooked and, if so, what can we learn from them to improve the *perceptual* quality of equipment? This article analyses some pitfalls and parasitic effects of feedback and gives indications for improvement of the *perceptual* quality. This was confirmed by the design of amplifiers, using guidelines derived from this analysis, where listening by musicians with 'golden ears' was considered more important than measurement results.

This analysis showed that parasitic effects occur in amplifiers with global feedback, which often escape attention, because they are not shown by the mathematical equations as used in *practice* for feedback systems. These parasitic effects lead to the *introduction* of artifacts, which are *specific* to feedback systems. This is surprising, because the general idea is that feedback *only* suppresses *undesirable properties*, but it is an unequivocal result of the analysis, which also shows that the commonly used equations for feedback are in fact *approximations* that do not describe the properties of amplifiers in sufficient detail. It can also be doubted whether different assumptions regarding the open-loop properties of amplifiers are correct.

Suppression of these parasitic effects requires making the individual amplifier stages as linear as possible and by designing the amplifier so that other open-loop properties of the amplifier are as close as possible to all assumptions, in combination with a constant, but moderate, feedback factor over the entire audio band in order to suppress the *interactions* between the various imperfections of the amplifier.

Testing equipment using continuous sinewaves rarely reveals these parasitic effects, as they only emerge in the *amplifier's dynamic* response to *music*. The usual approach that the sine wave response allows the prediction of the behaviour under all conditions, ignores the conditions under which the Fourier theory can and may be applied and thus unfortunately leads to incorrect results and conclusions. Thus, there is a great need for well-defined *dynamic* test signals, which better encompass the properties of music, but as long as these are not available, human hearing remains the best piece of "measuring equipment" that can be used. Ultimately, it's about how we experience the reproduced sound.

1. Introduction

The theory of feedback in amplifiers has been around since 1934 (ref. 1) and is widely regarded as an important tool to improve the quality of electronics. The equations, which show this, are relatively easy to derive using high school-level math. Yet in the 'high-end' audio community there are many opponents of feedback in amplifiers. Some even argue that feedback should be avoided if one wants to make 'musical' amplifiers. Unfortunately, such statements are rarely underpinned by scientific or technical evidence, but several unexplained phenomena are known: amplifiers with the same (low) distortion figures do 'sound' different, even though the speakers, which have to be used to listen to these amplifiers, have distortion figures that are at least two orders of magnitude higher. Thus, the distortion figures *as such* cannot explain the *perceptual* differences between amplifiers. Of course, other phenomena play an important role in the auditory quality of audio systems as noted earlier (ref. 2), but no satisfactory explanation has yet been given for this either. In this article, we will study these paradoxes and try to identify the underlying causes. Because insight into this is necessary to be able to achieve perceptual improvements.

It may come as a surprise that details of the feedback theory are overlooked because it has been used for so long. The derivation of the various equations is simple, but as usual, the devil hides in the details: in all these equations, the open-loop gain is a simple, single, parameter. But unfortunately, the open-loop gain is not a simple, but a very complex parameter: the gain is non-linear (this causes distortion), decreases with frequency (to keep the amplifier stable), is sensitive to the power rail voltages and variations, interacts with the load like a speaker, etc. In order to obtain the <u>correct</u> result using the feedback equations, a complete, detailed and preferably analytical description of the open-loop properties is therefore required. However, this is very difficult, if not impossible, as will be discussed in this article (sec. 4). Thus, one has to use approximations/simplifications to describe the open-loop amplification. This means that an <u>exact</u> description of the properties of the amplifier after feedback is *no longer feasible*. Some of the implications of the approaches/simplifications will be discussed in this article.

In sec. 3 we will give a more detailed analysis of feedback *within* an amplifier and point out some aspects, which are generally overlooked. In sec. 4, we discuss the Achilles heel of feedback theory, which is the cause of the differences between the calculated and realized closed-loop properties of an amplifier. Some examples of these differences will be discussed, showing that feedback can *introduce* artifacts and not just reduce these. Secs. 6 and 7 discuss the influence of the complex speaker impedance and the requirements for power supplies. In sec. 8, various imperfections of amplifiers will be presented, which *cannot be detected* using continuous signal measurements and their consequences. In sec. 9, the unconventional development strategy of 'Temporal Coherence' will be revealed. This leads to the conclusions and design guidelines in sec. 10 for optimizing the *perceptual* quality of amplifiers. After all, *how* we experience the reproduced sound is the crucial aspect of audio.

2. Nomenclature/definitions used in this paper

In this article we will use a number of variables, which are sometimes defined differently than usual. To avoid confusion, we will define it here:

Open-loop amplification μ:

$$\mu = \frac{V_{out}}{V_+ - V_-}$$

where:

V_{out}	= Output voltage	[V]
V_+	= Input voltage at the non-inverting input	[V]
V_-	= Input voltage at the inverting input	[V]

V_o is also used in a number of places, where V_o is the same as V_{out}.

Difference voltage across the (input) differential amplifier:

$$V_{sub} = V_+ - V_-$$

where:

V_{sub}	= Difference voltage across the differential amplifier	[V]

Open-loop gain at low frequencies (well below the open-loop cut-off frequency) μ_0:

$$\mu_0 = \frac{V_{out}}{V_+ - V_-}$$

The feedback factor β:

$$\beta = \frac{V_-}{V_{out}}$$

N.B. Note that μ is a very complex parameter, because it is a nonlinear function of the input voltage, frequency, supply voltage, the load impedance and probably some other things.....

N.B. Note that β can be a complex number and a function of frequency, μ is both.

The closed-loop gain A:

$$A = \frac{V_{out}}{V_{in}} = \frac{\mu}{1 + \mu\beta}$$

where:

V_{in}	= Input voltage	[V]

Note: This formula is only valid for non-inverting amplifiers.

The feedback margin M:

$$M = \frac{\mu}{A} = 1 + \mu\beta$$

The open-loop bandwidth:
This parameter is determined by the -3 dB (open-loop cut-off) frequency (relative to the maximum value of μ) of the dominant pole of the open-loop gain.

The closed-loop bandwidth f_{max}:
This parameter is determined by the -3 dB frequency of the closed-loop gain A (relative to the maximum value of A).

N.B. Note that the values of these last four parameters depend on the value of μ and therefore depend on the actual conditions.

3. Local and global feedback inside the amplifier

The usual way to work with the feedback equations is to think of the amplifier as a 'black box' with a differential input and a single output. The power supply can also be taken into account, but not in all cases this happens. The 'global' properties are looked at, which means that aspects such as distortion, noise, frequency response, etc. are *attributed to the 'black box',* but *where* in the 'black box' these are generated is not considered important, let alone essential. While this greatly simplifies the derivation of the equations, it also limits finding opportunities for optimization of the amplifier. Therefore, let's 'break open' the amplifier and dig deeper into the different sections and then be able to locate where unwanted by-products are generated.

We assume three separate amplifier stages, each amplifier stage is supposed to be perfect, and after each stage the 'misery' it generates is added.(**N.B. "**Misery" involves more than just distortion, but also noise, hum, signals coming from the power supplies, etc. But for the sake of simplicity, we will usually talk about distortion, but one should keep in mind that it also includes other unwanted signals.) In this way, the approximation of fig. 1 is obtained:

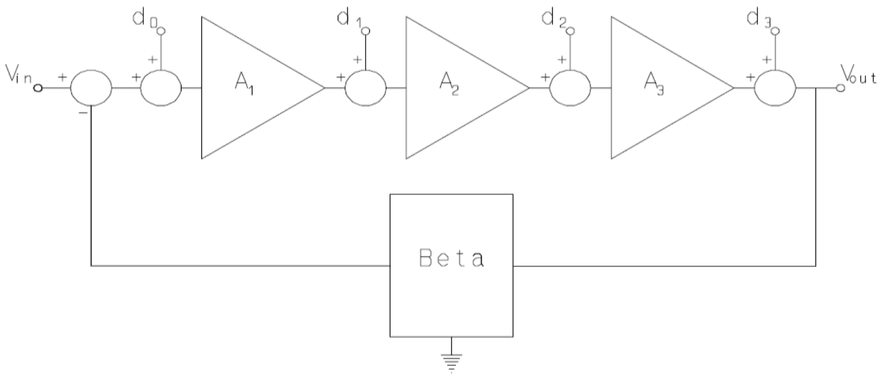


Figure 1: *The multistage amplifier approach to calculate its properties when global feedback is applied.*

The amplification of the i^{th} stage is A_i and its 'misery' is d_i. Immediately after subtracting the input and the feedback signal, d_0 is added. This represents the "misery" that the subtraction circuit, usually a differential amplifier, contributes, because it's not perfect either. It is now easy to see that:

$$\mu = A_1 \cdot A_2 \cdot A_3$$

$$V_{out} = (((V_{in} - \beta V_{out} + d_0) \cdot A_1 + d_1) \cdot A_2 + d_2) \cdot A_3 + d_3$$

Reshuffling yields:

$$V_{out} = \frac{\mu \cdot V_{in}}{1+\mu \cdot \beta} + \frac{\mu \cdot d_0}{1+\mu \cdot \beta} + \frac{A_2 \cdot A_3 \cdot d_1}{1+A_1 \cdot A_2 \cdot A_3 \cdot \beta} + \frac{A_3 \cdot d_2}{1+A_1 \cdot A_2 \cdot A_3 \cdot \beta} + \frac{d_3}{1+A_1 \cdot A_2 \cdot A_3 \cdot \beta}$$

$$V_{out} = \frac{\mu \cdot V_{in}}{1+\mu\beta} + \frac{\mu \cdot d_0}{1+\mu\beta} + \frac{A_2 \cdot A_3 \cdot d_1}{1+\mu\beta} + \frac{A_3 \cdot d_2}{1+\mu\beta} + \frac{d_3}{1+\mu\beta}$$

$$V_{out} = \frac{\mu \cdot V_{in}}{M} + \frac{A_1 \cdot A_2 \cdot A_3 \cdot d_0}{M} + \frac{A_2 \cdot A_3 \cdot d_1}{M} + \frac{A_3 \cdot d_2}{M} + \frac{d_3}{M}$$

The suppression of the distortion of the front stages is therefore *less* than the feedback margin *M* and the distortion of the differential stage is amplified as much as the input signal (**N.B.** This is equivalent to saying it sits outside the feedback loop). This means that the distortion of the previous stages is amplified by the subsequent stages, and the distortion products are also distorted, resulting in an increase in the harmonics of the harmonics. Note that the same applies to the sensitivity to variations in the supply voltage and the others, which contribute to the 'misery', as these are all amplified first before the feedback can intervene. The improvements, attributed to global feedback, must therefore be analysed in more detail, because the indiscriminate application of the well-known equations is 'cutting corners' if only the 'global' properties are used. For optimization, a more detailed internal operation of an amplifier is useful, which is why it will be discussed now.

3.1 The input differential amplifier

The first stage is usually a differential amplifier, which combines the functions of subtraction and amplification. Note again that the 'misery' of the differential amplifier *is not reduced* by the feedback and therefore appears amplified at the output. There are three pitfalls with this circuit. First, the signal level increases *with increasing frequency* due to the *decreasing* open-loop amplification *μ*. This can be easily inferred from the equation of global feedback:

$$V_+ - V_- = \frac{V_{out}}{\mu} = \frac{V_{in}}{1+\mu\beta} = \frac{V_{in}}{M}$$

Assuming that the open-loop gain decreases by 6 dB/oct. (first-order filtering) above the open-loop cut-off frequency, determined by the time constant τ, the voltage across the differential amplifier at the input is equal to:

$$V_i - \beta V_o = \left[\frac{V_i}{1+\mu_0\beta}\right] \cdot \left[\frac{1+j\omega\tau}{1+j\omega \, {}^{\tau}\!/_{(1+\mu_0\beta)}}\right]$$

In fig. 2, two examples are shown for different conditions to illustrate the effect.

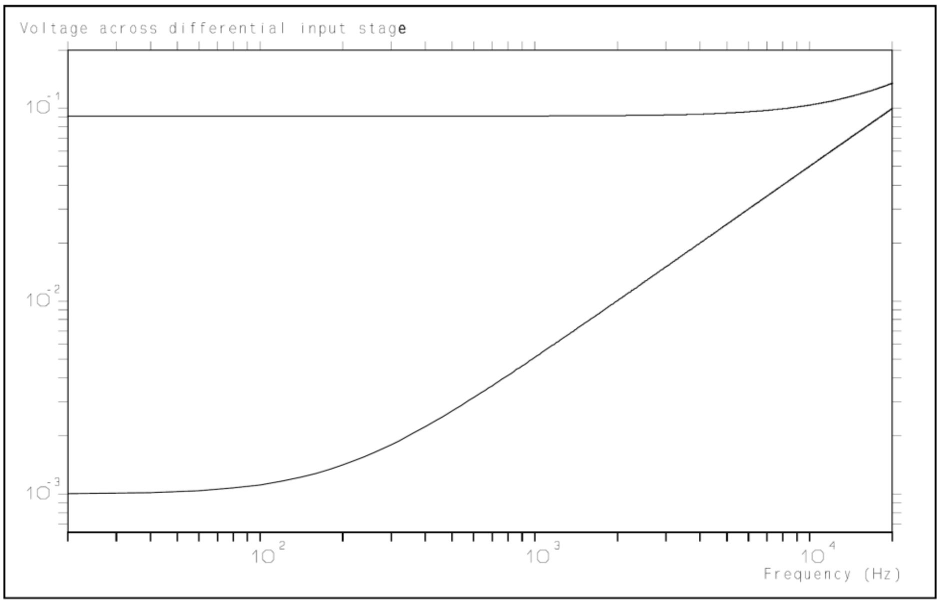


Figure 2: *The differential voltage at the input of the amplifier depends on the open-loop gain at low frequencies* μ_0*, the feedback factor* β*, the closed-loop bandwidth of the amplifier* f_{max} *and -of course- the input voltage* V_{in}*. Bottom trace:* μ_0 = 10 000; β = 0.1; f_{max} = 200 kHz; V_{in} = 1V, *Top trace*: μ_0 = 100; β = 0.1; f_{max} = 200 *kHz*: V_{in} = 1*V*.

Note that the difference voltage across the amplifier input can increase significantly with frequency, so the general assumption that the differential amplifier operates at (very) low levels (ref. 2) may be correct for low frequencies, but incorrect at higher frequencies. It will certainly lead to an increasing distortion with frequency, unless this subtraction stage is perfectly linear or the feedback margin *M* is constant in the audio band. However, this last requirement sets an upper limit to the feedback margin *M*, as will be discussed later. Selecting the input difference amplifier properties according to the requirements set for the signal level at low frequencies is likely to lead to disaster. Since no circuit is perfectly linear, the increase in voltage across the differential amplifier will lead to an increase in distortion (both harmonic and intermodulation) with frequency. It is often reported that this is audibly undesirable, because the *interactions* between the various imperfections of the amplifier will also increase.

Second, in a non-inverting design, the collector emitter voltages (V_{ce}) of the differential amplifier transistors vary when the same (in phase) alternating voltage (AC) is offered on the two bases. The properties of a transistor depend on V_{ce}, so a common mode voltage will generate an output voltage when the transistors are not identical and/or the collector loads are different. **N.B.** Note that at high feedback margins, the common mode voltage becomes very large compared to the differential voltage:

$\frac{V_{in}}{V_+ - V_-} \approx \mu\beta$ and the common mode voltage ≈ V_{in}. The common mode voltage is therefore approximately $\mu\beta$ times larger than the differential voltage.

Both effects are the strongest with differential amplifiers where the emitters are directly coupled to each other. This seems attractive because it maximizes the gain, but the non-linearity of the basic emitter diodes manifests itself, even at small (mV) excitations, in the perceived quality (ref. 3, see also the scale in fig. 2). Due to the exponential characteristic of the base emitter diode, much higher harmonics (higher than the fifth) are generated, which are known to quickly irritate the listener even at low levels (refs. 4 and 5). If, due to the high gain, the cut-off frequency has to be chosen relatively low, the differential voltage increases rapidly with the frequency (see fig. 2) and thus also the distortion of the direct-coupled differential amplifier. Then its output signal is amplified by the following stages. The widespread misconception that the distortion of *all* amplifier stages is suppressed with the feedback margin *M* is incorrect (ref. 2). This can be easily proven by splitting the amplifier into its individual stages, as above, showing that a more detailed description of the amplifier is needed to optimize the performance of the design. (**N.B**. It can be noted that the application of emitter resistors for the transistors of the differential amplifier reduces its distortion, but also the gain. If the open-loop cut-off frequency in the audio band is chosen, V_{sub} increases in the audio band (see fig. 2), so it becomes a complex trade-off between distortion, amplification and frequency dependence. In sec. 5.4, Table 1, is shown that the decrease in distortion is not fully compensated by the lower gain. Only by compensating for the decrease in gain with a distortion-free stage, this would be an improvement. But the distortion-free amplification stage has yet to be invented.)

3.2 The modulation depth of the amplification stages

The third pitfall requires a more detailed description. There are two contributions to the output signal of the differential stage: first, the difference between the input signal and β times the ideal (= misery-free) output signal and secondly the 'misery' at the output of the amplifier, also multiplied by β. In equation:

$$V_{sub} = \frac{1}{1+\mu\beta} V_{in} - \beta \cdot d_t$$

In which d_t is the total misery at the output of the amplifier. Multiplication of the left and right hand sides of the equation with μ results in:

$$\mu V_{sub} = \frac{\mu}{1+\mu\beta} V_{in} - \mu\beta \cdot d_t = V_{out} - \mu\beta \cdot d_t$$

The tipping point is reached when

$$V_{out} = \mu\beta \cdot d_t$$

The contribution of the distortion alone can be significant, compared to that of the input signal. The higher the open-loop gain, the less favourable the ratio between the 'signal' and the 'misery' becomes. Because if

$$V_{out} < \mu\beta \cdot d_t$$

this means that

$$\beta \cdot d_t > \frac{V_{out}}{\mu} \qquad d_t > \frac{V_{out}}{\mu\beta}$$

Because β has, in a given design, a prescribed value, the choice of μ will be important: a *large value* of μ soon means that the amplifier has to process more 'misery' than signal. Also, a higher open-loop gain is often accompanied by a higher distortion.

So, are all the underlying assumptions of feedback theory (such as the quasi-linearity of the individual amplifier stages) still valid? In the popular 'high-end' literature you can read that 'when an amplifier processes more distortion than music, you get a non-musical system'. This doesn't seem to be just hot air, as the above calculation shows.

It is often noted that this is not possible because the 'misery' is suppressed by the feedback margin, but that is incorrect. As we have seen above, the degree of suppression depends on *where* it is generated in the amplifier, and as μ decreases with increasing frequency (see fig. 2), the suppression also decreases. Thus, depending on the design of the amplifier, the above condition can be met with complex, multi-spectral signals. This condition should be kept in mind when an amplifier is designed and it should be used for evaluating a design.

The strength of the difference signal increases with increasing frequency (see fig. 2). All amplifier stages, prior to the cut-off capacitor, will have to process a signal whose strength increases with frequency. This may mean that these stages are operated in a strongly non-linear manner, and the lower the open-loop cut-off frequency is, the sooner the increase in frequency begins. Since the closed-loop bandwidth is determined by the open-loop gain, the time constant of the frequency cut-off and β, these parameters cannot be chosen completely independently of each other, but the choices made do affect the non-linearity of the individual stages and the reduction of their distortion. Below we discuss some options to optimize these choices, but to do so, we must first identify the problems with the *practical application* of feedback.

4. The Achilles heel of the practical application of the feedback theory

The equations, which can be derived from basic theory, are of course, basically correct, **provided that a sufficiently accurate and detailed description of the open-loop amplification is available**. As already mentioned in the Introduction, the *actual* open-loop gain is a very complex and complicated non-linear, frequency-dependent function of at least the input voltage V_{in}, the supply rail voltages, the (complex impedance of the) load and probably a few more contributors and it will be different for every other design. It will be very difficult and probably impossible to determine such a function for several reasons:

1. The nonlinear properties of the operating lines of the different stages can -in theory- be described by a Taylor series, but this will be in the amplitude domain . Each operating line requires a large number of derivatives, and the calculation of the total gain requires multiplication of all these operating lines, resulting in an impractically large number of cross terms.

2. The frequency dependence can -in principle- be described by a complex transfer function, but it is in the frequency domain. This cannot be easily converted to the amplitude domain, so the combination of nonlinear operating lines with the low-pass filtering is very difficult and probably impossible.

3. Describing the interaction of the non-constant output impedance with non-ohmic, complex impedances of, for example, loudspeakers, will be quite a challenge (see secs. 5 and 6).

Thus, although the equations derived from feedback theory are correct in themselves, it is impractical, if not impossible, to *implement* the open-loop properties of the amplifier into these. Even if it were possible, the very complex and complicated results will be so unmanageable that they will prove to be next to useless. As a result, one is forced to use approximations/ simplifications to allow the calculation of the properties of the amplifier after the feedback loop is closed. *However, once approximations/simplifications are used, the theory is no longer exact and the results are also reduced to approximations.* Therefore, one has to wonder what the differences will be between the calculated and actual properties of the amplifier in closed-loop operation as a result of the applied approaches/simplifications. These differences are probably at the root of the inconsistencies between the measured and perceived quality of amplifiers and the completely opposite opinions about the application of feedback in amplifiers.

In the next section, we will discuss these types of differences and try to find ways to improve the *perceptual* quality of amplifiers by looking at artifacts, which no longer show up in the calculated results due to the approaches/simplifications applied. Especially the interaction between the different imperfections will be studied, because they cause perceptually disturbing artifacts.

5. Differences between the calculated and actual properties of amplifiers

The application of (global) feedback should provide a clear improvement of the amplifier properties. In general, the artifacts produced by the amplifier in 'open loop' are reduced by the factor $\mu\beta$. Since $|\mu\beta|$ in general » 1, this leads to a significant improvement. The most frequently mentioned improvements are the reduction of distortion, the wider (closed-loop) bandwidth and the reduced output impedance of the amplifier. All very positive, you might think. So why is there so much controversy about the application of feedback in High-End Audio? Opinions range from 'you can't have enough feedback' (ref. 6) to 'amplifiers with feedback don't sound musical' (high-end fora). It has been shown above that the practical application of feedback theory requires the use of approximations/simplifications. So let's take a closer look at the implications of the approaches/simplifications applied. Maybe we'll find some clues there.

The most important parameters in the derivations are μ and β. A common, simple approach is to use constant values for these. Often, other properties of the amplifier (such as the output impedance) are also *assumed to be* constants. For example, when β consists of only passive components, such as a voltage divider and/or small capacitors, this is a good approximation of reality. But the open-loop amplification μ requires active components such as vacuum tubes or (field effect) transistors. These *always* introduce distortion and distortion is created by the non-linearity of the operating line (the relationship between input and output voltages of the amplifier). An example is given in fig. 3. And although, at first glance, it seems like a fairly straight line, this is ‘trompe l’oeil’. The 'instantaneous' value *of* μ is the derivative of the operational line. And its derivative is far from a constant.....

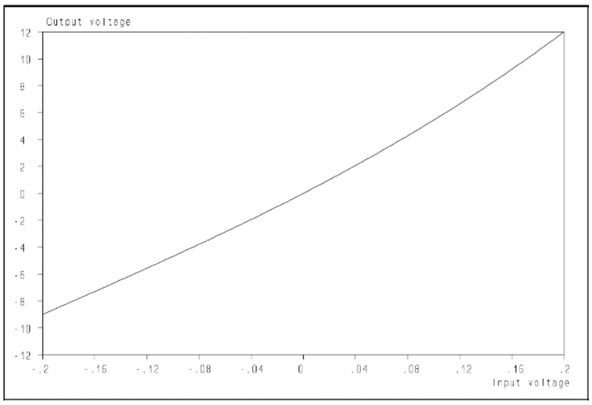


Figure 3: *An example of a nonlinear operating line. The amplification is around 40, the deviations from an ideal, straight line are not serious at first glance.*

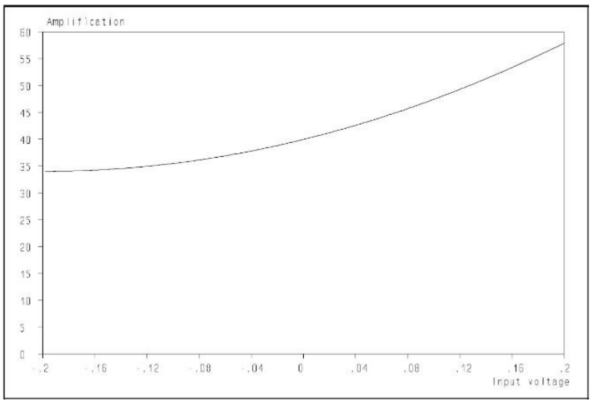


Figure 4: *However, the 'instantaneous' amplification, the derivative of the operating line, shows a strong variation. The gain is about 65% higher at +0.18 V than at -0.18 V. This is not negligible and the straight line approximation is therefore incorrect.*

....as shown in fig. 4! Thus, the use of a constant value for the open-loop gain μ_0 in the well-known feedback equations is not *realistic*: a constant value of the open-loop gain would mean that the amplifier *does not distort.* Thus, the approximation *of* μ_0 as a constant *eliminates* the possibility of determining the distortion properties of the amplifier, both in open and closed loop. It is therefore necessary to develop a nonlinear feedback theory (which requires mathematicians with skills far beyond those of the author). For practical applications, linear theory can be used as long as one keeps in mind that this is only an approximation, and it can *be applied if and only if* the operating lines of *all the stages* in an amplifier hardly deviate from a straight line. (**N.B.** Note that the equations presented in sec. 2 are also incorrect. Wherever μ or μ_0 is used, it should actually be replaced by $\mathrm{d}V_o/\mathrm{d}V_{in}$.). Thus, it requires a more detailed analysis of the undesirable phenomena in an amplifier to optimize its performance.

5.1 The output impedance of the amplifier

The output impedance of an amplifier is an important parameter in audio, especially when it is a power amplifier that drives a loudspeaker, as it determines the 'damping factor'. It is common to use the 'open-loop' output impedance for this value, divided by the feedback factor. But the underlying *assumption* is that the 'open-loop' output impedance has a constant value, but is this realistic? Well, certainly in class AB amplifiers, this is not the case. This is illustrated in figs. 5 and 6, which show the output impedance of a FET (Field Effect Transistor) output stage as a function of the source current. At low source currents, it is even above the 50 Ω, but as can be seen in the magnification (fig. 6), at 100 mA it is still in the order of 2 Ω. This is not negligible compared to the speaker impedance. And when the speaker impedance is not ohmic, but complex (as is usually the case, ref. 8), it becomes even more complicated.

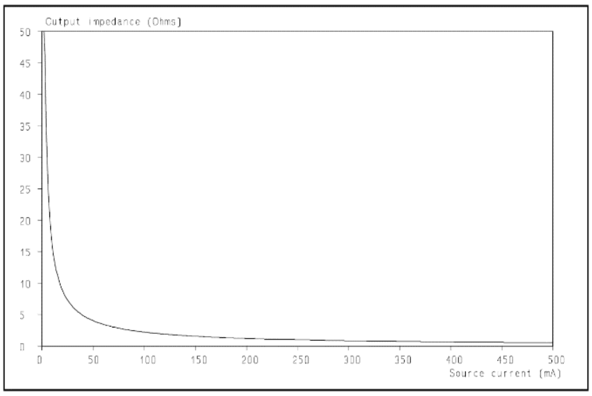


Figure 5: *The output impedance of a FET output stage, configured as a source follower, as a function of the source current. Note that at low source currents, the output impedance is much higher than the impedance of loudspeakers.*

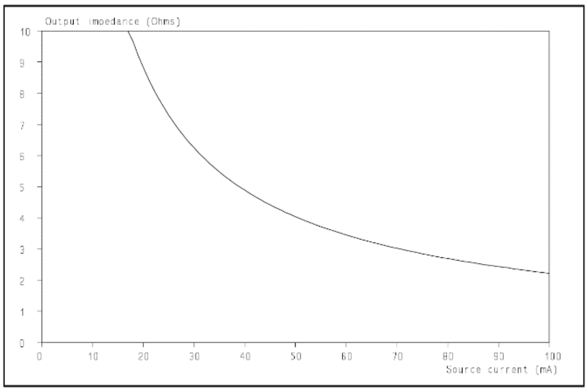


Figure 6: *The output impedance of a FET output stage, configured as a source follower, as a function of the source current. Note that at low source currents, the output impedance is much higher than the impedance of loudspeakers. Enlargement of fig. 5.*

As is clear from the above results, the output impedance of a power stage does -in many cases- *not* have a *constant* value! As a result, the calculation of 'the' output impedance is impossible, but there are more aspects, which are reflected in the feedback. With a varying output impedance around the zero crossing of the output signal, the effective open-loop gain also varies: the output impedance and the load impedance act as a voltage divider, reducing the open-loop gain around the zero crossing. When the load is purely ohmic, this can be calculated, but with a complex load (as of most speakers, ref. 8), it becomes even more complicated, especially with multi-spectral and dynamic signals such as music. The complexity is illustrated in the figs. 7 and 8, which represent the measured impedance of a loudspeaker unit in a housing. Finding a mathematical description of the interaction between the non-constant output impedance (figs. 5 and 6) and the complex, non-constant, frequency-dependent complex load impedance (figs. 7 and 8) is in reality impossible, which further substantiates the problems mentioned in sec. 4. It is also to be expected that this complicated interaction increases the crossover distortion in a class AB amplifier and it could (partially) explain the perceptual differences between class A and class AB amplifiers.

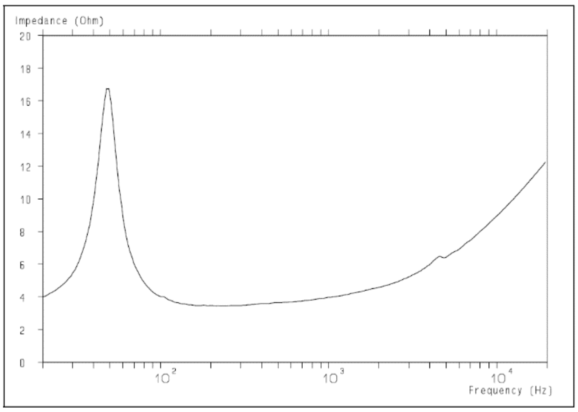


Figure 7: *The modulus of the impedance of a loudspeaker unit in a housing.*

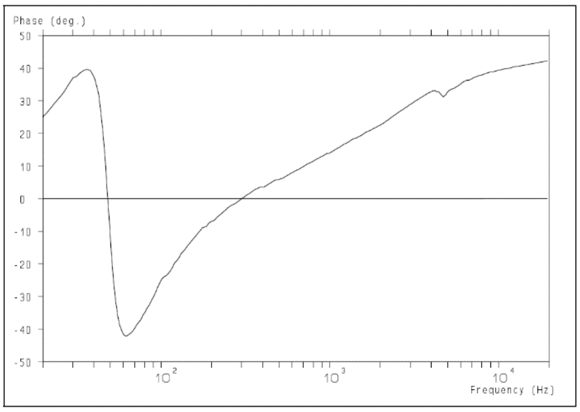


Figure 8: *The phase of the impedance of a loudspeaker unit in a housing.*

Thus, the crossover distortion is suppressed less than one might expect, and the damping is less than calculated, amplifying artifacts that will quickly start to annoy the listener. Where is feedback when you need it? The main conclusion is that the *applied approximations* to constant values of the open-loop gain and open-loop output impedance are *incorrect* and that therefore the approximate results of the commonly used feedback equations are clearly different from the *actual* performance.

5.2 Interacting imperfections

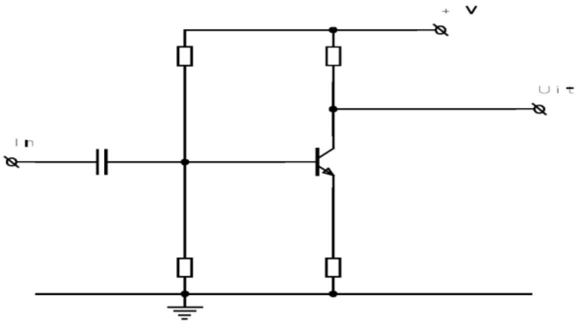
Another problem, which is usually 'swept under the carpet', is the *interaction* between the various imperfections in a closed feedback loop. The usual way to derive the feedback equations is to keep everything perfect except for one property. This greatly simplifies the derivation of the equations, but in reality all imperfections occur *simultaneously,* so it is probably *incorrect* to derive the equations in this way*,* since then it is *not possible* to take the *interactions* into account. That these interactions are important will be shown in the following sections.

5.3 Interaction between the non-linear characteristic and the limited bandwidth

An example of an interaction is illustrated as follows: take a low-frequency signal with a large amplitude and a high-frequency signal with a small amplitude. Intermodulation between the 'low' and the 'high' frequency is generated, but because the 'instantaneous' bandwidth of the closed-loop amplifier varies (modulated by the low frequency, see fig. 3), a *phase modulation* of the 'high' frequency also occurs (**N.B.** Phase modulation bears a close resemblance to frequency modulation). This has been noted before (ref. 5), but the assumptions in that article regarding the closed-loop bandwidth are rather optimistic, as will be explained below. (**N.B.** This effect is a direct result of the use of (global) feedback and thus an example of an artifact, which is *introduced* by feedback.) Similar interactions are to be expected between, for example, variations in the supply voltage and the input signal. Such interactions are difficult to capture in equations, but they cannot and should not be neglected. Numerical simulations are the easiest way to show these and to provide semi-quantitative results, useful for optimization. (**N.B.** These effects will *not* be detected with a sliding, *continuous sinewave signal,* simply because it requires *more complex signals* than simple sinewaves.) Thus, these interactions plea for a linear (open-loop) input-output relationship and a high open-loop cut-off frequency. This is quite obvious, because with this approach the imperfections are kept as small as possible and thus also the *interactions* between the different imperfections. This is further elaborated in the section below.

5.4 Memory effects caused by the non-linear input-output relation

The nonlinear open-loop input-output relationship leads to a parasitic effect: the distortion generates a (small) DC component in the output signal. The cause is best be understood by an example, for which the base circuit of fig. 9 is used. (**N.B.** The examples shown will be exaggerated to some extent to illustrate the phenomena more clearly.)



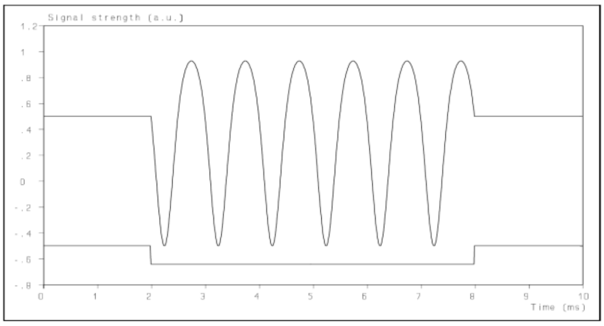


Figure 9: *The basic circuit used to illustrate the generation of a DC voltage due to distortion.*

Figure 10: *Top trace: the distorted output signal of a tone burst. Bottom trace: The average value of the upper trace, averaged over a single cycle.*

The distortion is caused by the nonlinear properties of the base-emitter diode. In the positive part of the input signal, it conducts *more* above the value of the operating point than *below* it in the negative part of it. As a result, the collector voltage drops further below the operating point setting during the positive part of the input signal than it rises during the negative part of it. This is illustrated in fig. 10 (upper trace) and therefore the collector voltage, averaged over a single cycle, is lower than the set value of the operating point (lower trace). Thus, a non-constant signal (as is common in music) is accompanied by a rectified component, which is related to the *envelope* of the input signal. Thus, it does not consist of harmonics of the input signal. (**N.B.** Note that the envelope consists of relatively low frequencies; this envelope signal is *not present in the input signal at all*.) Therefore, the feedback is not able to completely suppress this: in order to appear at the inverting input of the differential amplifier, it must be present at the output. Because the envelope is absent from the input signal and contains many low frequencies, this leads to an unwanted 'restlessness' in the sound. The value of the low-frequency, rectified component will naturally increase with the amplitude of the input signal and this does it more than proportionally. This phenomenon is illustrated in fig. 11.

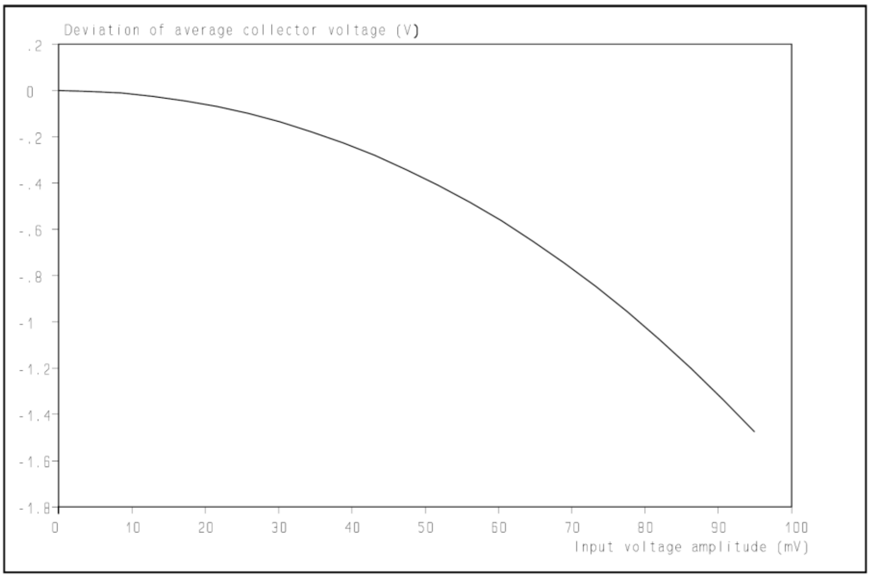


Figure 11: *Deviation from the initial operating point of the average collector voltage as a function of the input voltage. Amplification of the stage ≈ 200 times, supply voltage 36 V.*

N.B. Only if the input-output relationship is perfectly symmetrical, this effect will not occur. That's not going to happen in practice. **N.B**. In almost all modern semiconductor amplifiers, the amplifier stages are DC-coupled. Shifts in the operating point of one stage are passed on to the operating points of the following stages. Shifts in the operating points of the amplifier stages are undesirable, as will be discussed in sec. 6. It can also be the background to high-end listeners' desire for "headroom" (more power than is normally needed), as these phenomena quickly become stronger with the input signal. So if the amplifier is always used well below its maximum design value, these artifacts will be quite small and therefore not as irritating. However, it remains an *artifact*, which can be avoided by a good design of the amplifier.

The effects of this mechanism are amplified by the memory effect of the capacitor, which is used to limit the open loop bandwidth of the amplifier, as shown in fig. 12. Fig. 13 shows that in that case the varying component of the collector voltage is converted into a kind of modulated DC signal, which is not really surprising because the generated low-frequency component is hardly attenuated by the low-pass filtering, while the alternating voltage of the input signal is. In this example, the frequency of the input signal is 1 kHz, and the cut-off frequency is set at 200 Hz. It is clear that the lower the cut-off frequency is chosen, the worse this phenomenon becomes. So this is another argument to keep the open-loop cut-off frequency as high as possible, which also tends towards a design in which the feedback margin in the audio band is constant. As can be seen in fig. 14, when the cut-off frequency is set to 20 kHz, the AC component is still dominant.

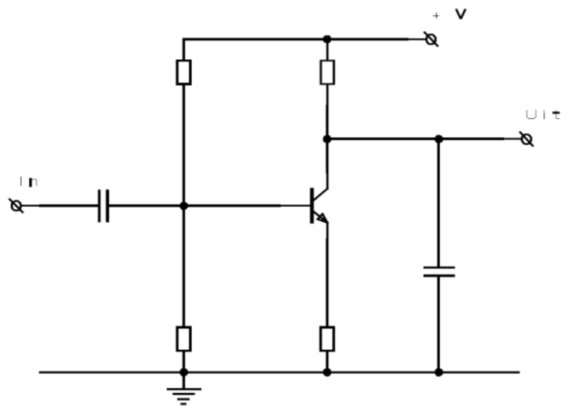


Figure 12: *The introduction of a capacitor to limit the open loop bandwidth of the amplifier creates a memory for the 'DC' signal, generated by the distortion. Compare with Fig. 9.*

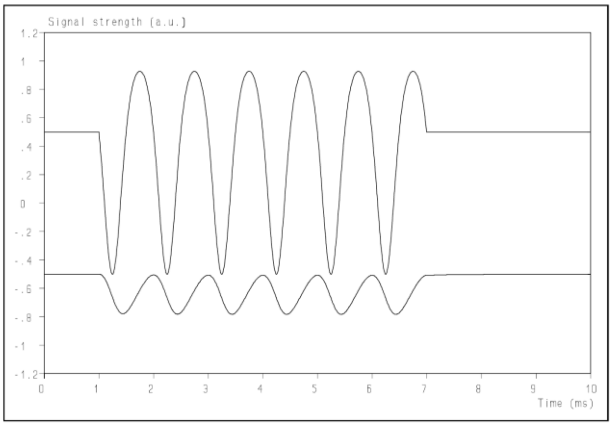


Figure 13: *The collector voltage without (upper trace) and with the capacitor added to the circuit of fig. 9 (lower trace) to achieve the circuit of fig. 12. Note that the collector voltage of the lower trace is rather a modulated DC signal than an AC signal. Input signal 1 kHz, cut-off frequency 200 Hz. See also fig. 10.*

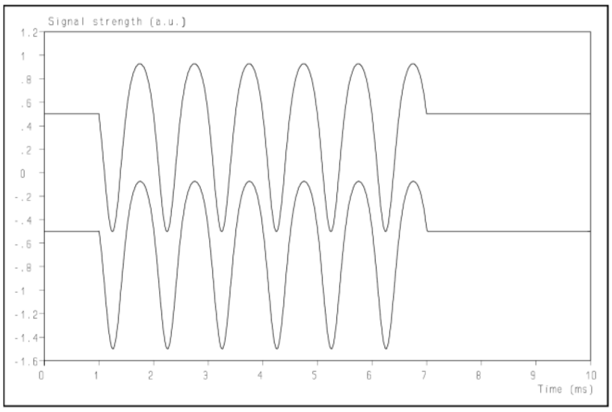


Figure 14: *The same situation as shown in fig. 13 is shown, except that the open-loop cut-off frequency is set to 20 kHz instead of 200 Hz.*

N.B. This effect will also *not be detected* with continuous sinewaves, simply because it is caused by a relatively rapid change in the amplitude of the input signal, as happens with music, but which does *not occur with continuous sinewaves*. A tone burst signal can show it in some, more extreme, cases. **N.B.** Note that this mechanism generates an artifact specifically for amplifiers with global feedback, because the required reduction of the gain with increasing frequency, requires the introduction of the cut-off capacitor. We can therefore conclude again that feedback does not always suppress 'misery' but can also introduce new 'misery'! Another advantage of a high open-loop cut-off frequency is that the 'recovery time' after an overload is short. **N.B.** Note that this phenomenon can also occur with high-frequency (HF) signals (e.g. from mobile phones), which enter the amplifier. Because the carrier frequency of the radio signal will always be above the cut-off frequency of the open-loop amplification, the amplifier is sensitive to this artifact and the only way to suppress it is to linearize ALL amplifier stages as much as possible. Because a linear stage cannot act as a (local) detector.

Fig. 11 shows that the deviation from the operating point increases rapidly with the amplitude of the input signal, a quadratic fit approximates the curve quite well. Because this also gives rise to the varying charge of the cut-off capacitor, and thus results in relatively slow signals due to the envelope at the amplifier output, it can be experienced as a 'slow' response, not related to the rise time/slew rate (the maximum value of dV_{out}/dt). However, if this phenomenon were suppressed, the amplifier would behave a lot better up to its clipping level and a 'normal' rise time/slew rate would be sufficient for playback in the living room.

A common way to linearize an amplifier stage is the addition of an emitter resistor (see fig. 9). Surprisingly, this does not proportionally reduce the parasitic effect. When all the gain obtained can be used for feedback (i.e. below the open-loop cut-off frequency) to reduce the phenomenon, it actually aggravates the situation. This is shown in Table 1 below:

Nominal amplification	Shift of average collector voltage	Normalized shift of the average collector voltage (= shift / amplification)
240	1.6 V	6.67 mV
43	0.46 V	10.70 mV
9.83	0.13 V	13.22 mV

Table 1: *Shift of the average collector voltage of a single amplifier stage as a function of its gain when the gain is set by local feedback, using an emitter resistor.* V_{cc} *= 36 V,* I_c *= 1 mA*.

The linearization of the individual amplifier stages therefore requires different methods than the use of a relatively large emitter resistor. The design team of "Temporal Coherence" has managed to find novel solutions to this problem.

Another limitation, introduced by feedback, is more familiar: the slew rate of an amplifier. This is caused by the current, which is required to charge/discharge the cut-off capacitor by the amplifier stage. The larger the capacitor, the lower the slew rate is. This also pleas for a small capacitor (and thus a wide open-loop bandwidth). Note that the slew rate limitation requires that the signal source is limited in spectral content and/or that an input filter is required to ensure that this limitation is realized under normal operating conditions.

N.B. It should be noted that it is not possible to get rid of the 'misery' introduced within the open loop by the phenomena described above, by global feedback. The best approach, therefore, is to avoid generating these artifacts. See also sec. 10 for the design directives. Prevention is still better than to cure.

5.5 Limitations to the feedback margin

By now, we have encountered several effects that plea for a linear gain of each amplifier stage and a high open-loop cut-off frequency. In popular high-end literature, this is often mentioned, based on listening experiences, but without significant technical or scientific substantiation. In the above, at least some indications can be found that support this and that make it plausible that the exchange of local and global feedback *does not* lead to amplifiers of the same *perceived* quality. Rather, it indicates that it is better to linearize the open-loop amplification as much as possible and to apply moderate global feedback. The additional advantages are i) that the cut-off frequency of the open-loop gain can be chosen high so that a constant feedback margin in the audio range (< 20 kHz) becomes feasible, ii) that the amplifier mainly processes music and little misery, iii) the recovery time after overload is short and iv) that the slew rate is high. A major disadvantage is that a low and constant open-loop output impedance is required, because the moderate global feedback also results in a moderate reduction of the output impedance. When the open-loop output impedance is relatively high and a function of the input signal level (see section 5.1), this will lead to undesirable artifacts. See also sec. 6.

The feedback margin and the closed-loop bandwidth of an amplifier are linked to maintain stability when using global feedback. Optimal stability is achieved when the open-loop gain decreases by 6 dB/oct. above a certain frequency and that the phase shift is -90°. As a result, the maximum feedback margin at 20 kHz is equal to the closed-loop bandwidth divided by 20 kHz. This follows directly from the value of the gain bandwidth product. When a high feedback margin at 20 kHz is required, the amplifier must have a wide closed-loop bandwidth. Although this is not a problem in theory, there are a number of practical problems such as i) the cut-off frequency of the power transistors and ii) the processing of HF signals, such as radio stations (and what else HF clutter is around today, which contributes to the 'misery'), which can lead to disturbing and noticeable artifacts. As the (closed feedback loop) amplifier becomes more and more nonlinear with increasing frequency, it will generate more harmonic and intermodulation products from the HF clutter. But detection can also create low-frequency signals, which can have a detrimental effect on the total sound image as discussed above (**N.B.** Again, note that with linearization of the individual amplifier stages, these effects are smaller!). Therefore, the feedback margin cannot be increased 'ad infinitum', not even to just 'stratospheric heights'. To some extent, the strict relationship between feedback margin and closed-loop bandwidth can be bypassed: if (in the Nyquist diagram) the point (-1, 0) is not within the curve of the $\mu\beta$ product, the amplifier will be stable. With clever tricks, this can be achieved even if the slope of μ is not a nice -6 dB/oct. and/or the phase shift deviates from -90°. But the price is that the stability of the amplifier will depend on the load (a speaker is rarely purely ohmic, see figs. 7 and 8, sec. 6 and ref. 8) and that its impulse response is impaired, introducing additional time smear (**N.B.** Articles about this can be found on www.temporalcoherence.nl). For an extreme case, see figs. 15 and 16. (**N.B.** A similar time smear as in fig. 15, introduced by a tweeter (see fig. 17), was found to be clearly audible and reduce its perceived quality, ref. 7). A lower temporal resolution can also be the basis for the requirement of 'fast' amplifiers, as this leads to the loss of detail, which can easily be interpreted as a 'slow' response. (**N.B**. The time smear, caused by the feedback masks details. Solving this by increasing the slew rate leads to nonsensical requirements for the amplifier: Sometimes 5 V/μs *and up* is mentioned. But with such a slew rate, the amplifier can deliver, undistorted, 200 W at 20 kHz to a 4Ω tweeter. At lower frequencies, this increases very quickly. For both the tweeter and the human audience, these are completely unnecessary, even risky, values. The problem of time smear must therefore be tackled at its root!)

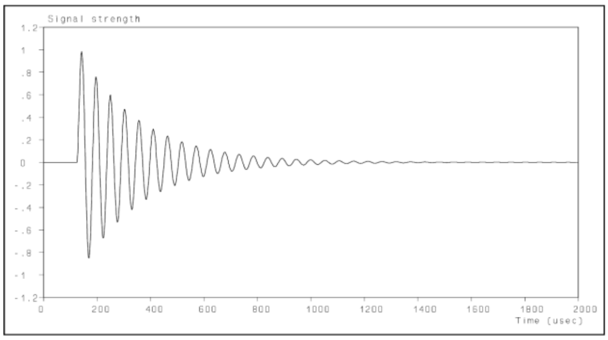


Figure 15: *Impulse response of an amplifier with feedback on the edge of stability.*

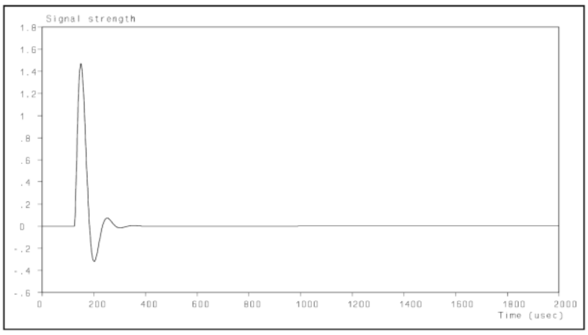


Figure 16: *Impulse response of an amplifier with a lower feedback margin and thus a higher distortion level, but with a higher temporal resolution.*

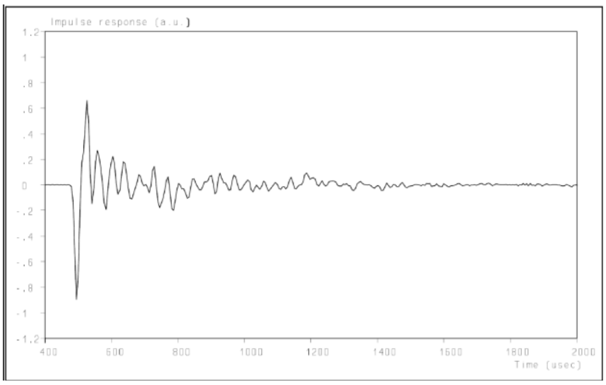


Figure 17: *A tweeter with a similar impulse response as in fig. 15 was found to be responsible for a clearly audible reduction in perceived quality.*

It is therefore possible to realize some extra feedback margin, but there are limits to what can be achieved at 20 kHz. It is possible to increase it at lower frequencies, inversely proportional to the frequency, but with the disadvantages discussed above. Kolinummi (ref. 2) found that above the open-loop cut-off frequency, the distortion increases much more rapidly than can be expected by the slope of -6 dB/oct., possibly caused by the distortion of the distortion products which are less and less suppressed by the feedback and by the increase in the differential voltage at the input stage; thus amplifying the distortion by the differential amplifier (see fig. 2) because it sits effectively outside the feedback loop. Again, it seems better to aim for a constant feedback margin across the entire audio range.

6. The consequences of the loudspeaker impedance

The impedance of a loudspeaker is complex, which means that the voltage and current are not in phase at most frequencies (figs. 7 and 8, ref. 8), which is quite different from the properties of a resistor. So it may happen that the amplifier has to supply, for example, a *positive* output voltage and a *negative* output current. This causes problems with the control voltages for the power transistors, which can lead to audible artifacts because this has to be controlled via the feedback.

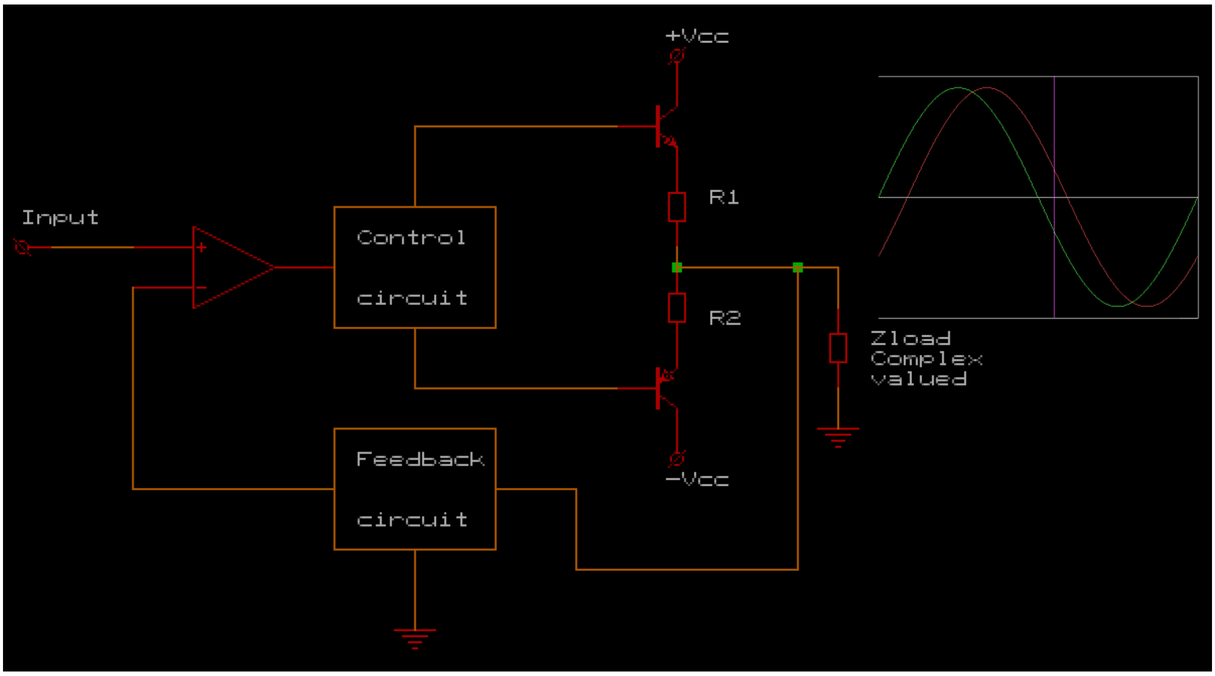


Figure 18: *Simplified block diagram of a power amplifier, which drives a complex load. For explanation, see text.*

In fig. 18, a simplified block diagram of a power amplifier is shown. The graph in the upper right hand corner shows the voltage across (green) and current through (red) the loudspeaker. At the purple line, the voltage is negative, while the current is positive. But to get a negative output voltage, the PNP transistor must be opened, but for a positive output current, the NPN transistor. This contradicts the design criteria of a class AB amplifier: either the NPN or the PNP transistor conducts, but not both, except in a small (current) range, determined by the quiescent current. Thus, the opposite polarity of voltage and current leads to additional error voltages for the amplifier to operate. But error voltages mean distortion at the output, which especially emphasizes the crossover distortion. This phenomenon further enhances the problems of describing the open-loop properties of an amplifier when a non-ohmic load is to be used. Note that the best way to reduce the effects of the complex load impedance is to use impedance compensation (ref. 8), especially with class AB amplifiers.

Related to this is the fact that a speaker is not time-independent/invariant *in dynamic use* because it can store energy (unlike a resistor!), which can be returned to the amplifier at any time. How the amplifier will react to this phenomenon depends a lot on its design. But impedance compensation is always attractive because after impedance compensation, the speaker behaves much more like a resistor and is therefore no longer able to return current to the amplifier. Also, voltage and current are in phase, avoiding the problems mentioned above. **N.B**. More information about the impedance of speakers and how to compensate for their variations can be found in ref. 8 and on the website of "Temporal Coherence" www.temporalcoherence.nl, including a link to a YouTube video on this topic.

7. Power supplies

All amplifiers need a power supply, which must be able to provide the required power and peak currents. But how *stable* should the *voltage* on the power rails be? It can be noted that a varying voltage of the power supply will lead to shifting operating points of the individual amplifier stages. We have already seen that global feedback is less effective for the first amplifier stages. Unknown, and design-specific, is how the variations of the supply voltages interact with the non-linearities of the different amplifier stages. There is a 'memory' effect too: the history of the input signal also determines the supply voltages due to the amount of current previously supplied to the amplifier. This can lead to 'unrest' in the soundstage and a kind of crosstalk when both channels of a stereo amplifier use the same power supply (**N.B.** Again, these effects are *not found by measuring with continuous sinewaves,* simply because it requires a *varying amplitude* of the input signal, which does not happen during a continuous sinewave measurement). The use of stabilized power supplies will solve or prevent most of these problems. The only question is how good that stabilization should be.

(**N.B.** An important working hypothesis, which results from this work, is that shifting of the operating points of the individual amplifier stages is detrimental to the *perceived* quality of the reproduced sound. It leads to 'unrest' in the output signal, probably because it adds things, such as the envelope of the input signal. The operating points of the individual amplifier stages must therefore be kept 'rock steady’ under dynamic conditions.) (**N.B.** In this context, *dynamic* refers to the properties of music-like signals, not AC with a constant amplitude!) Two main sources of such shifts have been identified: i) the nonlinear properties of all individual amplifier stages and ii) a varying supply voltage. Elimination of both results, in general, in a better sounding amplifier.

N.B. One must realize that there is also a frequency dependence, which is often overlooked. The unregulated power supplies are recharged at the (double) mains frequency (50 Hz in Europe, 60 Hz in North America) and this is slow, compared to the usual test frequency of 1 kHz. As a result, the peak currents are averaged out within a charging cycle. But at low frequencies, for example 20 Hz, the opposite happens: the current requested by the amplifier varies slowly, compared to the charging frequency of the unregulated power supplies. Thus, the load on the power supply has rather a DC character when the frequency is low, because at the peaks it almost doubles the average current, compared to a 1 kHz signal of the same strength. This is illustrated in figs. 19 and 20. The crux is that at 1 kHz, the load on the positive power supply vanishes in the negative half of the input signal (and vice versa, of course) so it is not loaded during (almost) half of the charging cycle. But at low frequencies, the positive power supply remains loaded during the entire charging time. This is essential for the dimensioning of the unregulated power supplies.

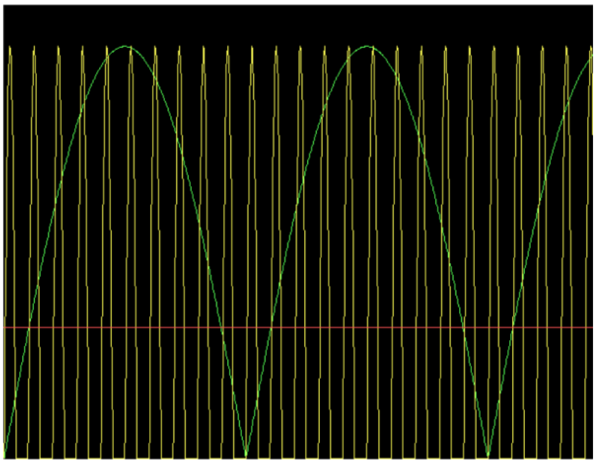


Figure 19: *At a frequency of 1 kHz, the charging frequency is low, which means that the load on the power supply can be averaged out. Shown is the current, requested by the input signal for the positive power supply (yellow, note that half of the sinewave is missing because it includes the negative part), so that averaging (red) over the power supply (green) can happen.*

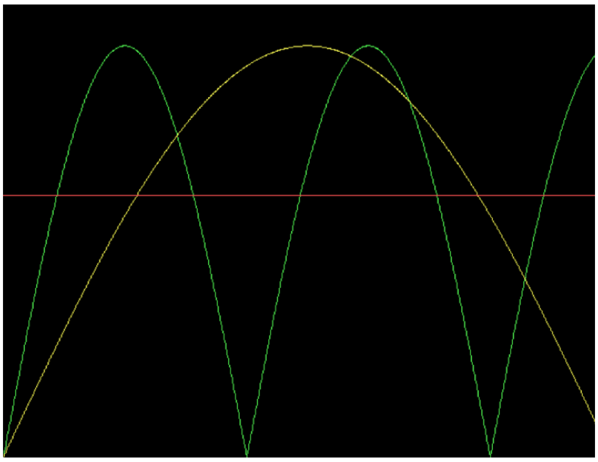


Figure 20: *At a frequency of 20 Hz, the charging frequency is high, so that the load on the power supply can no longer be averaged out. Shown is the current, requested by the input signal for the positive power supply (yellow), so that averaging (red) over the power supply (green) cannot take place and the power supply is effectively loaded more strongly. Compare with Fig. 19.*

8. Phenomena which cannot be determined by using continuous sine waves

Although some have already been mentioned above, there are several phenomena, which cannot be determined using measurements with continuous sinewaves of a constant amplitude, against the general expectation that this should be possible, based on Fourier theory. The following list is not exhaustive, but several important features that remain undetected are:

1. Shifting operating points of the individual amplifier stages
2. Changes in the charge of the cut-off capacitor due to shifting operating points
3. Phase modulation caused by varying closed-loop bandwidth
4. Time smear caused by the amplifier
5. Variations of supply rail voltages due to changing signal strength/power delivery

From the above analysis it should be clear why these cannot be measured with continuous sinewaves of a constant amplitude. The reason why this contradicts the general expectation is that Fourier theory *requires* that a number of *conditions* are met, which, however, is often ignored. It also explains the points mentioned by Kolinummi (ref. 2). This is explained in more detail in a separate article on the conditions for applying the Fourier theory, which can be found at www.temporalcoherence.nl .

Due to the limited usefulness of continuous sinewave measurements, there is a need for more complex measurement signals. Such signals should vary in strength with time scales, which are common in music, and be multi-spectral. A discussion about the required properties should start as soon as possible because it takes time to reach consensus. Another aspect is that the measurements must be carried out under more realistic conditions. A fixed resistor is very different from a multi-way speaker with passive crossover filters and is therefore unsuitable as a simulation: this differs too much from the conditions in which the amplifier will be used. Therefore, the measurement results, especially for class AB amplifiers, will probably be very different from the properties with speakers. Many reviewers note that the combination of amplifier and speaker is critical to the perceptual results, indicating unwanted, but not well understood, interactions between the two. However, these do not occur when an amplifier is loaded with a pure resistor. See also sec. 6.

9. Unconventional development procedure

Focusing on distortion figures alone to optimize amplifier performance is not the best way to steer developments, as there is little correlation between distortion figures and perceived quality. As demonstrated in this article, there are many more things that can go wrong when *music* needs to be processed by an amplifier. But -at the moment- we don't have well-defined testing methods that everyone agrees on, which correlate better with the perceptual quality. That is why the development team of "Temporal Coherence" is supported by a listening team. The members have no technical background and are not able to point out the things they hear in a technical sense. But they have a background in music (as musicians) and are therefore very familiar with 'natural' sounds. So when they develop an opinion about our products, it is always based on the *perceived* quality. Over the years, they have pointed at audible artifacts generated by our equipment. In most cases, the cause was initially a mystery and discovering it was often a challenging quest, but their observations have always been correct!

If the development team thought they had identified the culprit, action was taken to fix the problem, the lessons were in due course used to design amplifiers along the lines described in this article. This resulted in equipment with a significantly better assessment of *perceptual* quality. The improvements manifested themselves in, among other things, a better control of the low frequencies, more dynamics, more detail, better sibilance and, yes, a more 'musical' sound. Or, if you like, it sounded less like reproduced music and more like the real thing. This was confirmed by both our listening team and others (in no way connected to "Temporal Coherence", e.g. reviewers of HiFi magazines, their reviews can also be read on www.temporalcoherence.nl). It is tempting to assume that the explanations given here correctly describe the causes of the improvements, but this is not unambiguous. We do know that there are improvements, but we have no *evidence* that the explanation presented is realistic and complete. This requires additional experiments, which do not necessarily have to be done by ourselves. And if anyone can give other or additional reasons why designing amplifiers along these lines improves the perceptual quality, we will welcome them.

10. Conclusions, further steps and design guidelines

It can be concluded that there is, fundamentally, nothing wrong with (negative) feedback. But the Achilles heel is the impossibility of describing the open-loop properties of the amplifier, including all the details, such as the non-linear operating lines of the individual amplifier stages, the frequency dependence, the non-constant output impedance, the interaction with non-ohmic loads such as speakers etc. This limitation imposes the use of *approximations/simplifications* to calculate solutions of the feedback equations, which, however, are no longer exact, but are reduced to *approximations* of the actual performance of the amplifier. As a result, a number of parasitic effects receive little or no attention because they do not emerge easily due to this *practical application* of the feedback equations. A different approach, such as numerical solutions, is needed to study these. Feedback is certainly *not* a panacea for solving all the imperfections of an amplifier, *it can introduce new imperfections and artifacts*. Control and mitigation of these parasitic effects, often caused by the *interaction* between the various imperfections, require careful design of the amplifier with implicit linearization of the individual amplifier stages, otherwise audible artifacts will degrade the quality of the perceived reproduction. A designer must realize that global feedback can only be applied to a limited extent and that the dynamic behaviour of the amplifier with music is (much) more important for the perceived quality than distortion figures, in line with the findings of refs. 3 and 4. The response to complex signals probably explains some of the audible differences between amplifiers and other electronic audio equipment, which cannot be understood from the distortion figures, but which have given feedback a bad name in certain high-end audio circles. However, such artifacts are hard, in many cases not at all, measurable using *continuous* sinewaves. Since music is a textbook example of such a *dynamic* process, this is probably crucial for the determination of the *perceived* quality of an amplifier. Thus, more complex test signals are needed, representing the non-constant, multi-spectral conditions as these occur in music. As long as there is no agreement on such test signals, human hearing is still the best "measuring instrument" available. The input of the listening team of "Temporal Coherence" confirms this.

Most of the parasitic effects presented in this article are directly or indirectly, caused by the non-linearity of the individual amplifier stages. These effects must therefore be suppressed by linearizing the amplifier stages as much as possible by means of *local* feedback or other methods. By making the amplifier's open-loop amplification as wide-band as possible, but not allowing the closed-loop bandwidth to become too wide, a constant feedback factor can be achieved throughout the audio band. Moderate global feedback will lead to optimal design of electronics for high-end audio applications by improving dynamic response to music. Artifacts, introduced by feedback, cannot be eliminated by global feedback, so this is an important reason to *avoid* generating them in the first place.

The operating points of the individual amplifier stages must be kept "rock steady", and this also requires stabilized power supplies. This is an essential requirement for high-end audio amplifiers.

The interaction between the amplifier and its load should be given more attention, especially with class AB amplifiers, because they have a varying output impedance around the zero crossing, which interacts in a very complicated way with a complex load such as a loudspeaker. Creating a more ohmic load (preferably by impedance compensation) circumvents the majority of the problems, caused by this interaction.

The use of these guidelines within "Temporal Coherence" has resulted in a significant improvement in the perceived quality. Thus, there is circumstantial evidence for its effectiveness.

It can be noted that critical comments from high-end audio enthusiasts are often scornfully dismissed by technical experts as "non-scientific" chatter from freaks who don't understand the theory. The author strongly disagrees with this view, because too often critical comments from people with 'golden ears' did make sense, although initially it was absolutely unclear what its technical or scientific background was. However, such comments did help the development team to further improve the equipment, even though it would have been very difficult to demonstrate the effect of the individual steps with scientific listening tests. But progress over the years is indisputable. The author therefore endorses the view that all findings should be taken seriously.

Acknowledgements

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