# Analysis of the effect of non-linearity on the output signal of an amplifier with feedback for non steady-state signals

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## 1. Introduction

Within the community of high-end audiophiles, there is a fierce discussion on the use of feedback. Some state 'you can't have enough feedback' (e.g. ref. 1) whereas others state that feedback leads to 'non-musical' electronics. It has been shown in a previous report (ref. 2), that, contrary to the general opinion that feedback only *reduces* the imperfections of electronics, it can actually *introduce* artefacts, not present in the input signal. The misunderstanding is caused by testing with continuous, steady-state sinusoidal signals. The issue is that music is -fortunately- non steady-state and that memory effects, in combination with the inherent non-linear properties of the electronics, inside amplifiers are the root cause of the introduced artefacts. An important conclusion of ref. 2 is that that the analysis of non steady-state signals is more interesting than of steady-state signals to determine the *perceived* quality of electronics. It has also been shown that the common approach to use the results of steady-state signals to predict how an amplifier will react to music is incorrect and that the widespread equations for amplifiers with feedback are flawed. More details can be found in ref. 2 and these will therefore not be repeated here. So a novel way to analyse the properties of electronics to non steady-state signals needs to be developed. This will enable the study of the effects of non-linear operating lines, feedback and their interaction on the response to music. This is necessary to improve our understanding of the related phenomena and to guide developers to design better sounding amplifiers.

The perceived quality of amplifiers is hard to correlate directly to the specifications. Amplifiers do sound differently, even when specifications are comparable and often these are orders of magnitude better than those of loudspeakers, essential to compare amplifiers. The perceived quality of amplifiers depends on a large number of different aspects and it is not certain that all these have been revealed. The risk is that incorrect conclusions are drawn, because some aspects are overlooked or not taken into account. The confusion is illustrated by the difference of opinion on feedback, as mentioned above. In this paper, we will show that amplifiers with identical feedback factors can have very different properties and that feedback can introduce audible and irritating artefacts. This is not caused by feedback alone, but by the complex interactions between different aspects of the amplifier design. We will also compare several statements from the fora with results from the analysis, presented in this paper to get more clarity on which are correct and which are not.

In sec. 2, the flaws in the common analysis of audio equipment will be clarified. In sec. 3, the effects of the non-linear properties of electronics on non steady-state signals will be illustrated, which will be applied to a modelled amplifier, described in sec. 4. In sec. 5, the non-linear analysis will be explained and in sec. 6, the results of this analysis will be presented. These results will be the input for a discussion and interpretation in sec. 7. The paper will end with conclusions, recommendations and suggestions for future work in sec. 8.

## 2. Background

It is important to emphasize that all the equations, derived to 'prove' that feedback reduces all kinds of misery, have two major basic errors: the first is that the open loop gain $\mu_0$ is a constant, but it is not, as will be shown shortly in more detail. A major consequence is that the derived equations are only approximations, at best. It is not hard to grasp that, loosely speaking, the more $\mu_0$ deviates from a constant, the more the actual results will differ from the approximations. These equations are also used to 'prove' that feedback reduces the distortion of an amplifier. But distortion is caused by a non-constant $\mu_0$ (!) so the 'prove' uses the assumption of a constant $\mu_0$ to reduce the distortion, caused by a non-constant $\mu_0$. Mathematically, this is, as the Dutch say, cursing in church! The second flaw is related to the first: it is assumed that the different types of misery can be analysed in isolation, thus ignoring interactions. This is, however, incorrect as such interactions cause the introduction of artefacts. Neglecting these is therefore a recipe to overlook several severe drawbacks of feedback. These artefacts, related to the interactions, are very likely the root cause of the fierce discussion between audiophiles.

The common approach, to use the analysis of a system using continuous sine waves, is based on the assumption that the response for continuous sine waves determines the response to any signal. It is founded on the Fourier theory (e.g. refs. 3 – 5), but this is not allowed: Fourier theory *may only be applied to linear and time-invariant systems*, but an amplifier is neither. So one should not be surprised that such an approach leads to incorrect outcomes. Therefore, a different analysis is required to determine the response of an amplifier (which always has non-linear properties internally) to a non-steady signal like music. Such inconsistencies between theory and practice have been noticed before (e.g. refs. 6 and 7), but the root cause(s) have neither been identified nor counteracted. In this paper, the response of a non-linear amplifier is determined using the direct temporal response of an amplifier with feedback using similar techniques as is applied to solve differential equations numerically. Such a technique has been used before (ref. 8) and proved to be successful.

## 3. Effects of non-linear properties of electronics

Before the actual analysis is started, a basic understanding of the effect of the non-linear properties is helpful. A simple amplifier stage is shown in fig. 1 and due to the non-linear properties of the base-emitter diode, it distorts. The voltage at the collector, when a 6 cycle tone burst is applied as input signal, is shown in fig. 2, upper trace. This signal is superimposed on the quiescent collector voltage. After subtracting the quiescent collector voltage, it can be regarded as the sum of an AC component and a DC component. The latter is nothing but the average value of the signal over the entire length of the tone burst. (**N.B.** In the ideal case, there should be no DC component!) The DC component is shown in fig. 2, lower trace, in which the value at the left and right hand boundaries represent the quiescent voltage of the collector. So the non-linearity introduces a shift in the average DC value of the collector, which can be regarded as *detection* of the signal. In other words, the *envelope* of the signal is also an output signal of this stage! It is similar to the detection of AM radio signals and it leads to the introduction of frequencies *below* the frequency of the signal, which generates it. This will be discussed in more detail in secs. 6 and 7. Note that when a continuous signal is used, the generation of the envelope is not noticed, as the envelope of a continuous signal is only a 'constant' DC offset. This shows already a severe limitation of the measurements with continuous sine waves. See also ref. 2.

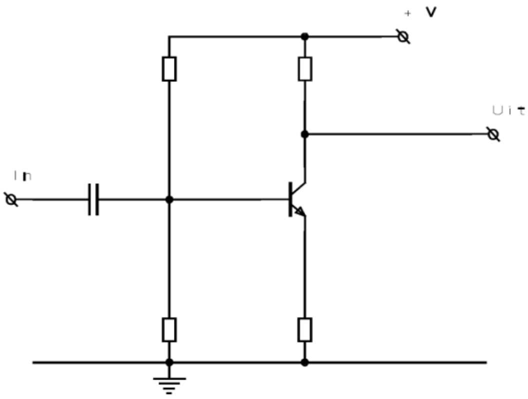


**Figure 1**: *A simple amplification stage will introduce distortion.*

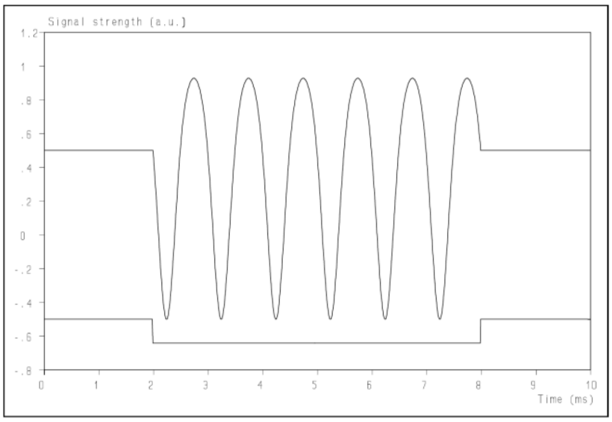


**Figure 2**: *The collector voltage of the stage of fig. 1 when a 6 cycle tone burst is applied (upper trace) and the average value of it (lower trace). Frequency of tone burst is 1 kHz.*

It should not come as a surprise that the deviation of the average collector voltage depends on the strength of the input signal, as is illustrated in fig. 3. This phenomenon might be responsible for the statement that an amplifier needs 'headroom' because due to this effect, the artefacts will increase `more than proportional with increasing signal amplitude. But a well-designed amplifier should not produce such artefacts at audible levels and therefore it should be able to deliver high-quality output up to its clipping level. We will come back to this later.

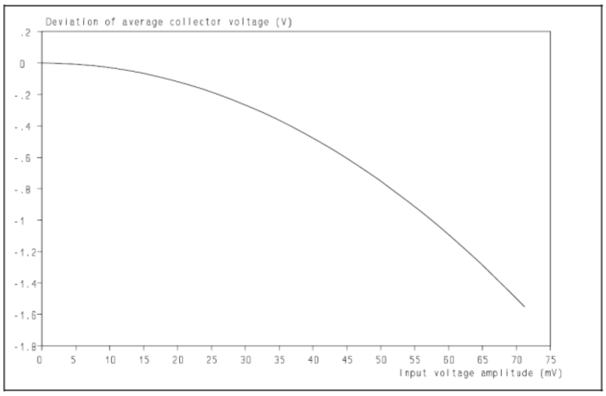


**Figure 3**: *Deviation of average collector voltage as a function of the input signal amplitude due to the non-linear properties of the base-emitter diode.*

<u>Conditions for figs. 2, 3 and 5</u>:

| | |
|---|---|
| Supply voltage | 36 V |
| Collector voltage (rest) | 19 V |
| Amplification | ≈ 240 times |

An example of an artefact, which is introduced in order to be able to apply feedback, is shown in fig. 4. To limit the open-loop bandwidth of an amplifier, a first-order low-pass filter is introduced, in this case by adding a capacitor to the collector load of the single amplifier stage. (**N.B.** We will refer to this filter as the 'low-pass stability filter' in the remainder of this paper.) This capacitor will, in combination with the collector resistor, act as a low-pass filter, which will have a more severe effect on the oscillating (AC) part than on its envelope. The reason is that the demodulation generates frequencies lower than those of the AC part. Thus, depending on the cut-off frequency, the envelope is less reduced by the low pass stability filter than the AC part. This can be seen in fig 5, in which the upper trace is the same as the upper trace in fig. 2, the lower trace the response with the capacitor added. The signal has degraded to a modulation on the average 'DC' value, even though the actual input signal has no DC contribution at all. So the envelope of the input signal is clearly added to the output signal of this amplification stage. The question is: will the feedback be able to eliminate this artefact? To that end, a different approach is required, which will be described in the next sections.

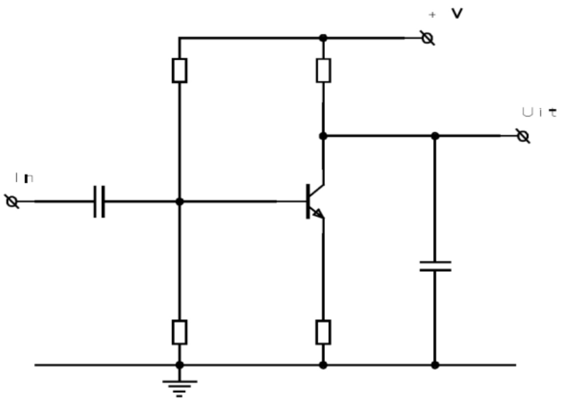


**Figure 4**: *A simple amplification stage will introduce distortion, which is influenced by the cut-off capacitor. See fig. 5.*

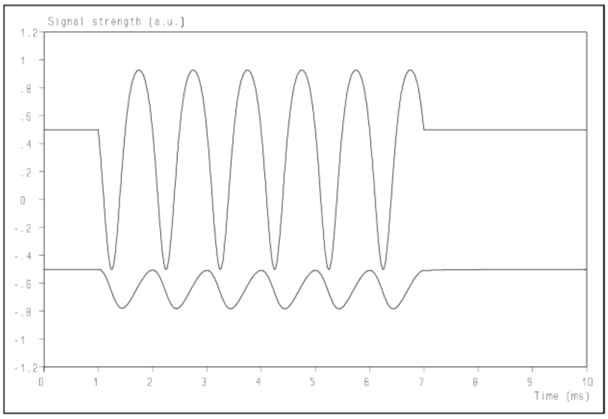


**Figure 5**: *The collector voltage of the stage of fig. 1 when a 6 cycle tone burst is applied (upper trace) and the value of it when the capacitor is added (lower trace). Cut-off frequency of first order low pass filter is 100 Hz.*

## 4. Modelling of amplifier

In order to determine whether similar effects still occur when feedback is applied, a numerical simulation has been performed. It is executed purely in time domain as such a non-linear system cannot be analysed using e.g. Fourier Transformation as shown above. Therefore, a similar approach has been used as with the numerical solution of differential equations. Before details of the simulation will be described, first more details of the simulated amplifier will be presented.

The simplified diagram of the simulated amplifier is shown in fig. 6. It consists of a subtraction stage, assumed to be perfect, followed by the non-linear amplification part. The open-loop amplification is roughly 100, but because the operating line is non-linear, as is shown in fig. 7, this value is only achieved at the operating point (zero volts input signal). At other input voltages, the open-loop gain has a different value as is shown in fig. 8. This shows (again) that $\mu_0$ is not a constant. **N.B.** Note that due to the inherently non-linear properties of active components, any amplifier will have a non-linear open-loop operating line. It is up to the designer to reduce the non-linear properties of the open-loop amplifier as much as possible, which will manifest itself in lowering the variations of the value of $\mu_0$. We will come back to this later in more detail.

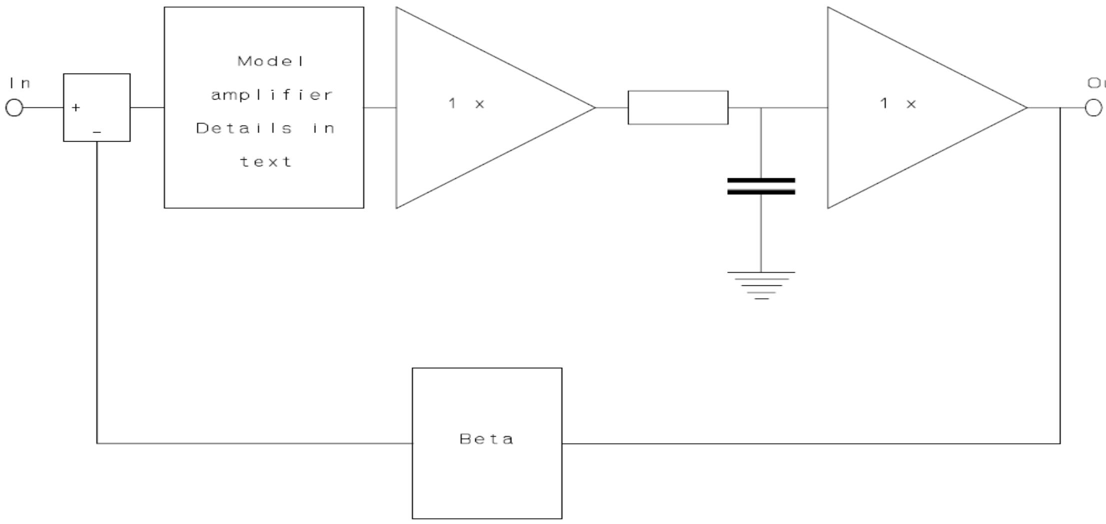


**Figure 6**: *Simplified diagram of the amplifier, as used in the simulation. For details: see text.*

The output of the amplification part is buffered by a perfect 1x amplifier (input impedance infinite, output impedance zero, amplification perfectly 1 without any distortion). Subsequently, a first-order low-pass filter is included, which is the low-pass stability filter in this case. In this simulation, the cut-off frequency of this filter is set at 2500 Hz. The output of the low-pass filter is buffered with an identical 1x amplifier. The output signal is fed back to the subtraction stage via a network with a value of $\beta$, taken in this simulation as 0.1. In the ideal world, the amplification of the simulated amplifier should be 9.091. For very small signals, this will be the case, but what happens when larger input signals are applied?

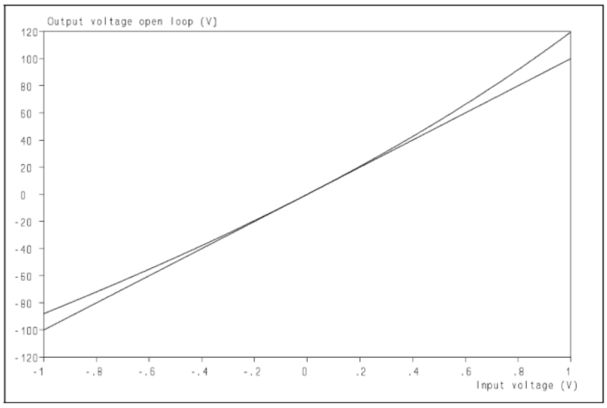


**Figure 7**: *Operating line of the modelled amplifier (curved line, upper trace) and an ideal amplifier (straight line, lower trace). See fig. 8 for the 'local' amplification.*

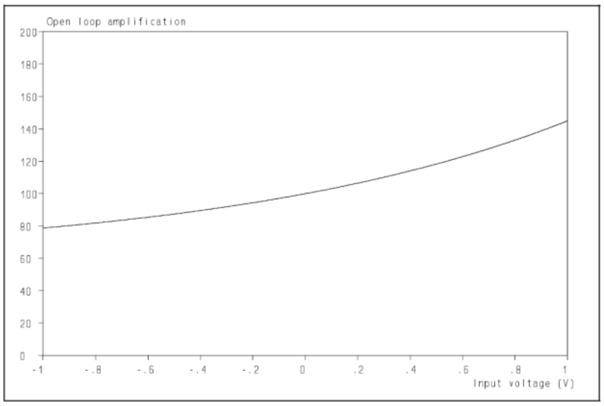


**Figure 8**: *The value of the 'local' open-loop gain of the modelled amplifier, which is basically the derivative of the curve (upper trace) from fig. 7.*

**N.B.** The modelled amplifier is only to demonstrate the technique to solve the non-linear temporal response. It is neither an existing amplifier nor specific for the solver. The technique can be used on any non-linear system, to be described by the user.

## 5. Numerical solution of the temporal response of modelled amplifier

The response of an amplifier is determined using a technique, similar to the numerical solution of differential equations. Therefore, it requires initial values (starting conditions), which is done by putting the charge on the capacitor to zero and so is the output voltage. The input signal is sampled at a high frequency (in this case 2 MHz) and for each sample the response is calculated by the following steps:

- The next sample of the input signal and the output signal of the latest sample are used to determine the output of the differential stage.
- The output of the differential stage is used to determine the output of the modelled amplifier, using the operating line of fig. 7.
- The output voltage of the modelled amplifier is used, together with the voltage across the capacitor, to determine the current flowing in or out of the capacitor and thus calculating the new charge of the capacitor and the voltage across it.
- The voltage across the capacitor is used as the output voltage of the amplifier.
- The output voltage of the amplifier is fed back to the differential stage after multiplication by $\beta$.
- The whole cycle is repeated for the next sample from the input signal.

The whole simulation is therefore rather straightforward and the only assumption needed is that the changes between two successive samples are sufficiently small to allow this approach.

In order to test whether the algorithm yields reliable results, first a simulation was made with a linear system, as this can be calculated using the standard techniques like the Fourier Transform too, so the results can be compared. To that end, a linear system was selected with a second-order low pass stability filter with feedback. Such a system shows overshoot when the parameters are chosen correctly. The results of this comparison are shown in figs.

9 and 10. Fig. 9 presents the result of the algorithm, described in this section and fig. 10 those of the Fourier Transformation. As can be seen, the results are identical, showing that the algorithm provides reliable results.

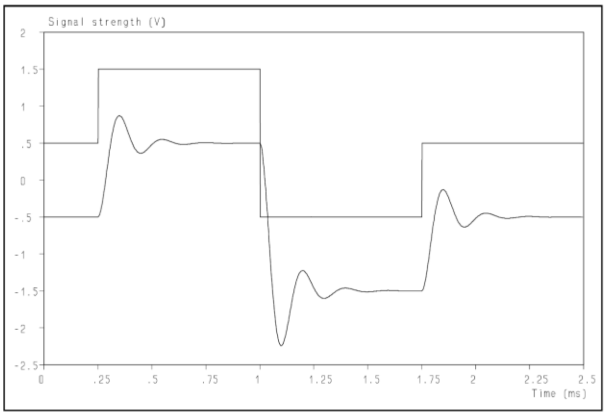


**Figure 9**: *Calculated response of a linear system with a second order low-pass stability filter using the algorithm of sec. 5.*

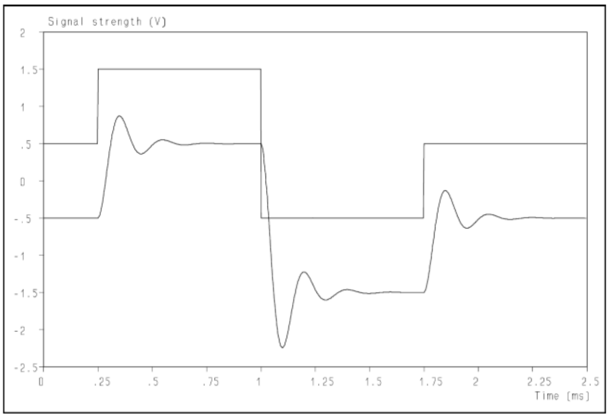


**Figure 10**: *Calculated response of a linear system with a second order low-pass stability filter using the Fourier theory.*

The deviations, introduced by the modelled amplifier, can be revealed by subtraction of the output signal of an ideal, perfect amplifier from the response of the modelled amplifier. The output signal of the perfect amplifier could be determined analytically, but it can also be obtained by using the same algorithm as the non-linear amplifier. The latter approach has the advantage that deviations, introduced by the algorithm itself, will (virtually) be identical in both results and thus cancel when these are subtracted to reveal the deviations, introduced by the modelled amplifier. To that end, the calculation is repeated using a perfectly linear operating line, the straight line in fig. 7, representing a constant $\mu_0$ of 100. This has been used in all the results, presented below.

Note that this technique can be applied to any input signal of any desired length. It is also possible to introduce other imperfections of amplifiers. Because there is only one underlying assumption, the technique is far better suited for the analysis of non-linear systems than the techniques which are mostly used. **N.B.** It is recommended that the value of the input signal at its start is 0 (zero), so it corresponds to the starting values of the voltage across the capacitor and the output voltage. It is also recommended that the last part of the input signal is 0 (zero) over a certain stretch of time to reveal the delayed response of the amplifier / system due to memory effects.

In case of doubt that the assumption is too coarse, one can simply increase the sampling frequency of the input signal and compare the outcomes of the calculations using different sampling frequencies. If the differences are negligible, the lowest sampling frequency is applicable.

**N.B.** A practical remark: it is highly recommended to use double precision variables in the calculations as the round-off errors with single precision variables can degrade the results or even prohibit a numerical solution: when the differences between two successive samples are smaller than the least significant digit of the variables, the calculation will produce nonsense.

## 6. Results for non steady-state signals

The response of the amplifier to a 3-cycle tone burst of 2000 Hz has been calculated both with the non-linear and with the linear operating line. The results are presented in fig, 11, albeit that the signals are shifted in the vertical direction to simplify the comparison, which ‘by the eye' does not reveal any differences. Subtraction of the two signals shows that there are differences, as is shown in fig. 12. It is clear that the difference signal has a strong second harmonic component, but also that it is superimposed on a large 'DC' contribution. This 'DC' contribution is so large that the difference signal does not drop below zero! The 'DC' component is basically the *envelope* of the input signal. Such an envelope consists of frequencies *below* the frequency of the input signal and the signal is not further related to the input signal. This is further complicated by the frequency dependent interaction between the cut-off frequency and the non-linear properties of the amplifier.

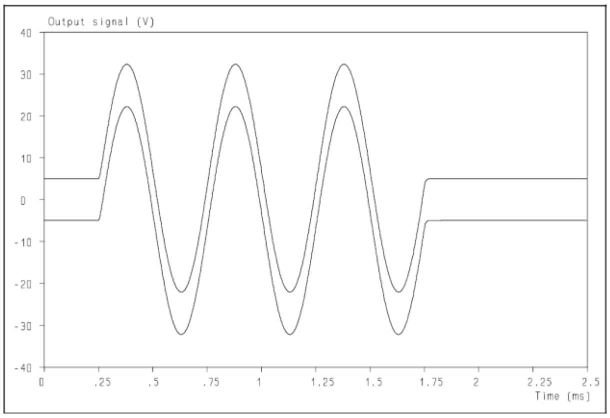


**Figure 11**: *Output signals of the modelled amplifier of fig. 6 (upper trace) and the ideal amplifier (lower trace). The curves have been shifted for clarity. Input signal is 3 cycle tone burst of 2000 Hz*

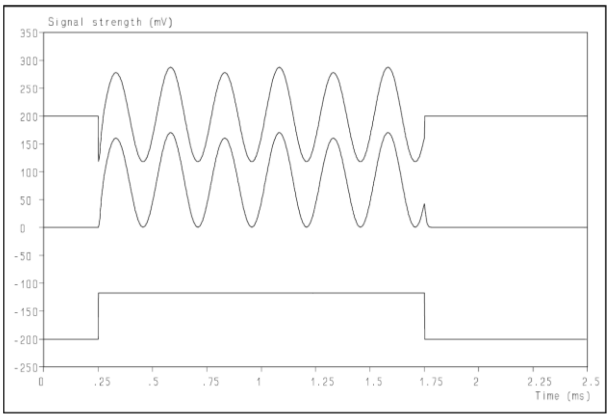


**Figure 12**: *Difference signal between the output of the modelled and the ideal amplifier of the 3 cycle tone burst of 2000 Hz (middle trace). It can be split into an AC part (upper trace) and a 'DC' part (lower trace), which can be regarded as the envelope of the input signal. AC and DC traces have been shifter by 200 mV for clarity. Compare with fig.11.*

When the value of the 'DC' component is calculated as a function of the input amplitude, it shows to increase more than proportional with the amplitude of the input signal, as is shown in fig. 13. This is -qualitatively- in agreement with fig. 3.

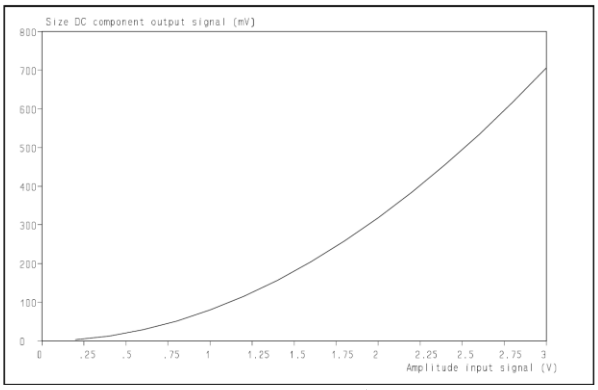


**Figure 13**: *The increase of the 'DC' component as a function of input signal amplitude is stronger than proportional. See also fig. 3.*

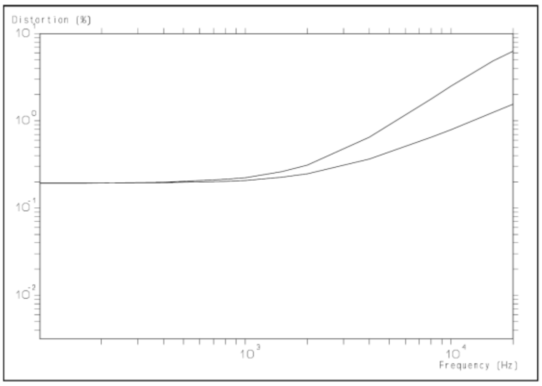


**Figure 14**: *The increase of the distortion (including the 'DC' component) as a function of the frequency of the input signal (upper trace). Lower trace is the inverse of the modulus of the low-pass open-loop stability filter.*

When the 'DC' component is calculated as a function of the frequency of the tone burst signal with a constant amplitude, it increases with frequency, as is shown in fig. 14 (upper trace). As the feedback factor (= $\mu\beta$) decreases due to the open-loop low pass stability filter, the first guesstimate of this increase would be the inverse of the modulus of this filter characteristic, which is shown in fig. 14, lower curve. Although the frequency at which the increase starts is obviously related to the cut-off frequency of the low-pass stability filter, it is clear that the increase of the artefacts goes steeper. The effect for two different frequencies is illustrated in figs. 15 and 16. The explanation and interpretation of these phenomena will be discussed in more detail below.

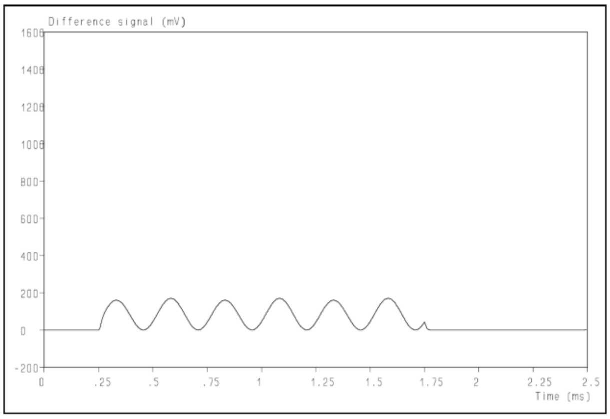


**Figure 15**: *The distortion signal of the amplifier of fig. 6 when the input signal frequency is 2 kHz.*

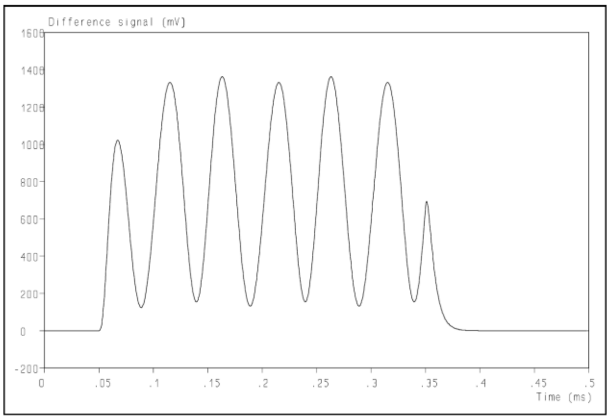


**Figure 16**: *The distortion signal of the amplifier of fig. 6 when the input signal frequency is 10 kHz.*

In general, distortion figures can be reduced by increasing the feedback factor (= $\mu\beta$). However, in order to maintain the closed loop stability, the cut-off frequency of the low-pass stability filter needs to be reduced by the same factor, which will keep the gain-bandwidth product constant. This was used to study the influence of an increased feedback factor by a factor 10 by changing the amplification of the second buffer block of fig. 6 to 10 times, reducing the cut-off frequency of the low-pass stability filter to 250 Hz and keeping $\beta$ the same. The results of this calculation for the frequency dependence of the distortion are shown in fig. 17.

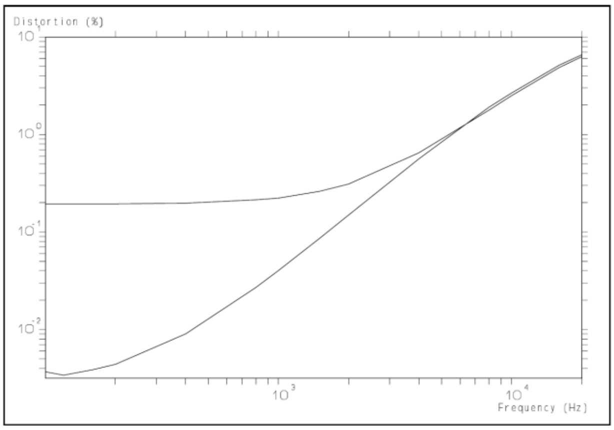


**Figure 17**: *The distortion (including the 'DC' component) of the amplifier of fig. 6 as a function of the frequency (upper trace) and of the amplifier with the increased open-loop gain but reduced open-loop bandwidth (lower trace). Note the rapid increase above the cut-off frequency of the low pass stability filter.*

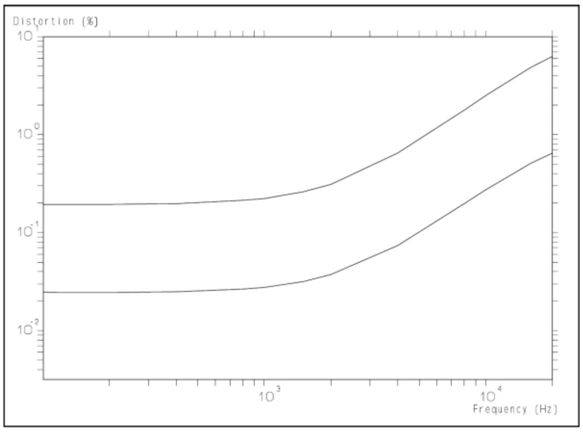


**Figure 18**: *The distortion (including the 'DC' component) of the amplifier of fig. 6 as a function of frequency (upper trace) and of the amplifier with the reduced non-linearity of the operating line (lower trace). Compare with fig. 17.*

Another way to reduce the distortion is to reduce the non-linearity of the operating line. In fig. 18, the results are shown of the amplifier of fig. 6, together with those of this amplifier when the non-linearity is reduced. It is also interesting to compare the amplifier with the reduced non-linearity of the operating line with the amplifier with an increased feedback factor. These results are shown in fig. 19.

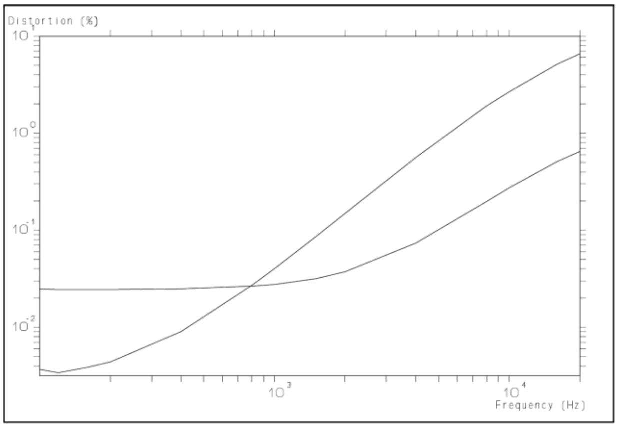


**Figure 19**: *The distortion (including the 'DC' component) as a function of frequency of the amplifier of fig. 6 with the reduced non-linearity of the operating line (upper trace at left) and the amplifier with the increased open-loop gain and reduced open-loop bandwidth (lower trace at left). See also figs. 17 and 18.*

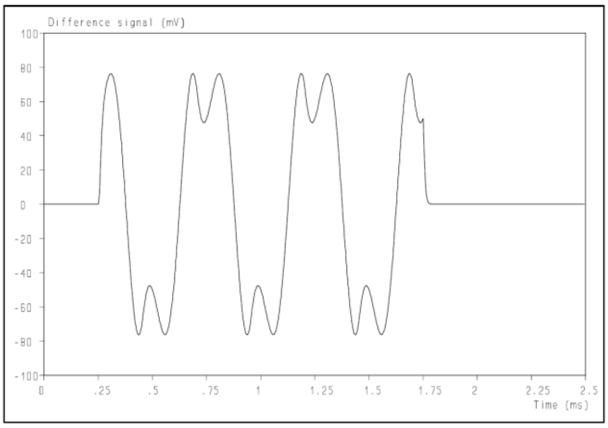


**Figure 20**: *The response of a non-linear system with a symmetric operating line. Note the complete disappearance of the 'DC' contribution. Compare with figs. 12, 15, 16 and 22.*

## 7. Discussion

The 'DC' component in the output signal is caused by the non-symmetric operating line, which reflects itself in -in this case- positive derivative of the 'instantaneous' $\mu$ of the amplifier, see figs. 7 and 8. It is also in agreement with the results in sec. 3, figs. 3 and 5. When this conclusion is correct, the 'DC' component should disappear when the operating line is symmetric. This is confirmed by the results of a simulation, as shown in fig. 20.

It is clear that feedback is able to suppress the 'DC' contribution, but not to eliminate it, as shown in figs. 14, 15 and 16. These figures also show that the 'DC' contribution increases with frequency. This can -partly- be explained by the decrease of the feedback factor (= $\mu\beta$) due to the low-pass stability filter. But the increase goes faster than can be explained by the decrease of the feedback factor alone, illustrated by the inverse of the modulus of the low-pass stability filter. The cause of this more rapid increase is the non-linearity of the operating line of the amplifier: with a smaller feedback factor, the input signal for the amplifier increases, thus requiring the use of a larger fraction of the (non-linear) operating line with the same input signal amplitude. **N.B.** The more rapid increase of the distortion with frequency was noted in ref. 6, but the author does neither provide additional evidence nor an explanation. This is a bit surprising as it contradicts his assumption that the differential amplifier at the input operates in a linear mode due to the small amplitude of the signal it processes. We will come back to this in a short while.

The use of a larger feedback factor reduces the distortion and the 'DC' contribution to the output signal, as one would expect, as can be seen in fig. 17. But when the gain-bandwidth product of the amplifier is kept the same, the cut-off frequency of the low-pass stability filter needs to be reduced by the same factor. As a result, the increase of the 'DC' component starts at a lower frequency and consequently, the contribution at higher frequencies is the same as with the lower feedback factor. This is not surprising as the feedback factors of both will be the same in this region. So the improvement is only effective in de lower frequency range and one could doubt whether this is -from perception- a desirable modification.

The 'DC' contribution is basically a demodulation of the actual input signal, albeit not in a linear sense. This is shown by fig. 13, where the increase of the 'DC' component as a function of the amplitude of the input signal is shown when the input signal is a 3-cycle tone burst. The increase is rather quadratic than linear, which means that the envelope of a more complex signal is distorted. This has been verified by using a tone burst with a triangular envelope as shown in fig. 21. The distortion plus 'DC' component is shown in fig. 22 with a fitted triangle. It is clear that the envelope is not presented as a triangle, which is in agreement with the previous conclusions. It is not unrealistic that this non-linear increase of the artefacts leads to the statement that an amplifier needs 'headroom' to limit the artefacts when the input amplitude increases. The question is whether there are ways to avoid this increase. This will be discussed shortly. First, we will look at the effect of this demodulation on the perceived quality of an amplifier.

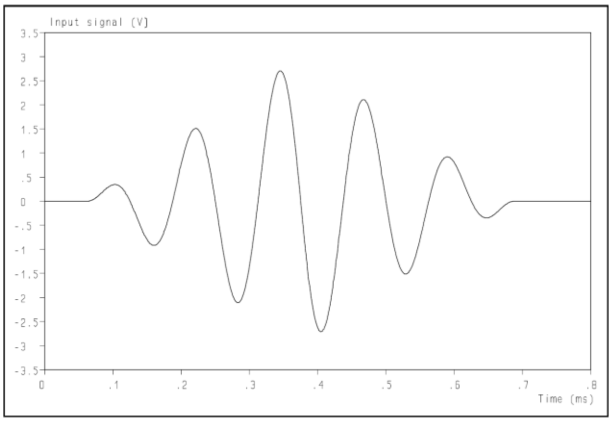


**Figure 21**: *A tone burst with a triangular envelope. Note that the other tone bursts, used in this paper, have a rectangular envelope.*

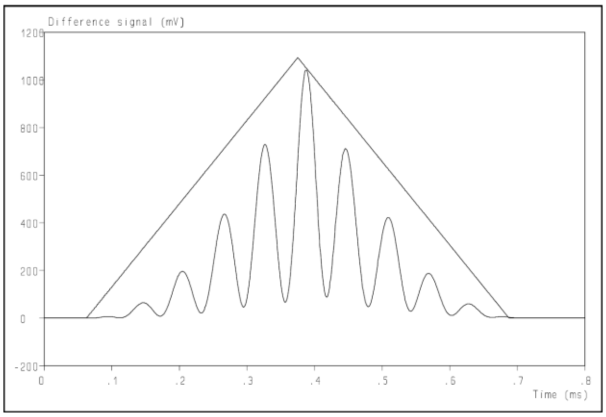


**Figure 22**: *The difference signal of the tone burst with a triangular envelope and the fitted triangle. Frequency is 8 kHz, amplitude of input signal is 3 V.*

The demodulation generates *lower* frequencies, contrary to harmonic distortion, which generates higher frequencies. Due to the decrease of the feedback factor with increasing frequency by the low-pass stability filter, the envelope of higher frequencies is more strongly contributing to the artefacts. But these end up at frequencies in the midrange, where human hearing is most sensitive. It is likely that this is annoying to the ear and it will reduce the detail and transparency of the reproduced sound. To avoid this, the feedback factor should be kept constant in the audio band and the increase of the 'DC' contribution with the input amplitude should be as small as possible.

As can be seen in fig. 16, the improvement with an increased feedback factor was obvious for the low frequencies. If this improvement could be extended to higher frequencies, this would be very helpful. But this runs into other problems, caused by the Law of Conservation of Misery.

An increased feedback factor at higher frequencies can be achieved by increasing the gain-bandwidth product of the amplifier. This would mean that the amplifier is able to process very high frequencies. Let us take an example: a feedback factor of 100 and an open-loop bandwidth of 20 kHz. This would result in a closed loop bandwidth of 2 MHz and indeed, some amplifiers have such a wide closed-loop bandwidth, being able to amplify long and medium wave radio stations. But still, the increase of the distortion with increasing frequency occurs, leading to demodulation of radio stations. It is very hard to avoid such signals creeping into the electronics and demodulation will create artefacts in the reproduced sound, which probably will be experienced as 'unrest' and the like. It certainly will not contribute in a positive way to the listening experience. So, a feedback factor of around 10 at 20 kHz, resulting in a gain-bandwidth product of 200 kHz, is more realistic and desirable. If one wants to keep the feedback factor constant in the audio band, the feedback factor at 20 kHz is the maximum achievable. Although a reduction of the artefacts by, in this example, a factor of 10 is still worthwhile, the level of the artefacts should be very low to start with, as a major improvement from feedback is not feasible.

The obvious way to reduce the artefacts is by reducing the non-linearity of the operating line. The improvements are clearly illustrated in figs. 18 and 19. Note that at higher frequencies this is more effective than an increase in the feedback factor. Especially the amplification stages, close to the input, can introduce misery, which is not, or not fully, suppressed by feedback (ref. 2). So these contribute to a large extent to the artefacts at higher frequencies due to the reduced feedback factor and thus the use of a larger fraction of the operating line. But making the operating lines of all the amplification stages as linear as possible is highly recommended as all contribute to the generation of the undesirable artefacts, thus enabling the limited feedback factor of around 10. Note that such an approach also reduces the increase of the artefacts with increasing amplitude and makes the amplifier less susceptible for HF induced misery as it will remain close to linear at any frequency. Such an approach will also virtually eliminate the 'need' for 'headroom' and will result in amplifiers which reproduce sound with much detail and transparency.

The discussions at high-end audio fora bring several interesting statements to the table. The statement that a constant feedback factor in the audio band leads to better sounding amplifiers can be underpinned by the results of the simulations, reported here. The statement that feedback leads to 'non-musical' amplifiers is, in general, not correct, but it is clear from the results, presented in this paper, that it easy to design amplifiers which generate audible and undesirable artefacts by feedback. A low cut-off frequency of the low-pass stability filter to allow a high feedback factor at lower frequencies to compensate for a strong non-linear open-loop operating line is likely to be a recipe for a 'non-musical' amplifier which reduces the reproduction of detail and transparency. It is also likely to have bad overload recovery properties due to a large charge in the capacitor of the low-pass stability filter. As a high feedback factor is usually applied on amplifiers with strong non-linear operating lines, such amplifiers would need 'headroom'. So when an amplifier would need 'headroom', it is likely to suffer from other artefacts as well. The statement that 'an amplifier which is rather processing error signals than music does not sound musical' is probably correct as this analysis shows that the use of as linear operating lines as possible of all amplification stages, in combination with a moderate feedback factor, generates little artefacts. Such a design will only have to process small error signals.

The power amplifiers of 'Temporal Coherence' do apply a moderate feedback factor which is constant in the audio band. All the amplification stages are given an operating line as linear as possible by a novel approach in the design, different from local feedback. This has proven to be very effective in the creation of an amplifier which is very open, transparent and 'clean' with a very detailed reproduction of the sound. This approach is used in all the equipment, designed and build by 'Temporal Coherence', not only the power amplifiers. This approach is underpinned by the results of this analysis.

## 8. Conclusions, recommendations and future work

The analysis of non-linear systems, using the common techniques with continuous sine waves, is based on the incorrect application of Fourier theory. The commonly used equations to show that feedback reduces all kinds of misery are flawed because the underlying assumptions are incorrect, thus leading to incorrect results and, even more problematic, the neglect of interactions between the different kind of artefacts. Especially these interactions lead to the generation of artefacts which are not present in the input signal and can be very irritating to the ear.

In order to resolve the response of a non-linear system to non steady-state signals in time, a different approach is required. In this paper, a technique is described, similar to the numerical solution of differential equations, which is able to do this for any input signal of any desired length and with any non-linear system. In this paper an example is presented, but the technique is not limited to this example.

The technique has been used to study the effects of feedback on a non-linear system. The results can be used to steer the design of well-sounding electronics and to find out whether several statements on feedback in the high-end audio community are correct, exaggerated or, bluntly, nonsense. The statement that a constant feedback factor in the audio band is recommended has been underpinned by the results of these simulations. But these also point at a relatively low level of feedback which can be applied to avoid the demodulation of radio stations and other HF signals. So the statement that ‘you cannot have enough feedback’ is not supported by these results, quite the contrary. Reducing the non-linearity of the operating lines of all the individual amplification stages is a far better approach to reduce all kinds of artefacts, generated by the amplifier, than a large feedback factor. However, the realisation of this concept requires a rather different approach in the design of amplifiers. But this has been done successfully by ‘Temporal Coherence’.

It can be concluded that the technique, described in this paper, is a useful tool during the development of amplifiers as it enables the determination of its response to any signal in time before it is built. As its application is not limited to any system, it could be used to help resolve a long standing discussion topic in the high-end audio community: why is there so much difference in the perceived quality of vacuum tube amplifiers and semiconductor amplifiers? From a physics point of view, well designed amplifiers should not contribute anything undesirable to the reproduced sound, so the basic active components should not be decisive in the perceived quality of the reproduced sound. This not (yet) the case (ref. 9), but maybe the artefacts, introduced by both types of amplifiers, could provide clues on the underlying causes.

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