USING NONLINEAR AMPLIFIER SIMULATION IN DYNAMIC RANGE CONTROLLERS

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ABSTRACT

Amplifying devices where the gain is automatically controlled by the level of the input signal performs dynamics processing. Non-linear components simulating tube amplifiers can be used in these devices to make musical signal audibly dense [1]. This paper deals with the simulation of tube amplifiers using the power polynomial approximation of transfer characteristic and their use in dynamic range controllers. The influence of various non-linear amplifying devices simulating tube amplifiers on the output signal spectrum of dynamic effects is presented as well.

1. INTRODUCTION

Dynamics processing is based on level detection system called envelope follower, an algorithm to derive a gain factor from envelope follower, and a multiplier to weight the input signal (see Fig. 1). Dynamic range controller consists of a direct path for delaying the input signal and a side chain path that performs a level measurement and gain factor g(n) calculation. The level measurement is followed by a static function and a part for the attack and release time adjustment.

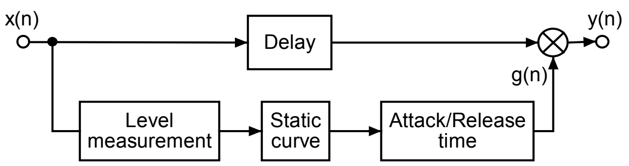

Figure 1: *Block diagram of dynamic range controller.*

Static function determines type of the dynamic processor – limiter, compressor, expander, ducking, noise gate, or combined systems [3]. This paper not deals with side chain path and gain factor calculation. See [3] or [4] for details.

2. NON-LINEAR AMPLIFIER IN DYNAMIC RANGE CONTROLLER

There are four positions in direct signal path where non-linear amplifier can be placed (see Fig. 1):

- before dynamic controller
- after dynamic controller
- before or after delay block
- in place of product block (voltage-controlled non-linear amplifier)

A non-linear amplifier placed before the dynamic effect adds higher harmonics into input signal of the dynamic controller. Higher harmonics ratio varies in dependence on the input signal amplitude. Changes of the dynamic controller parameters do not affect higher harmonics ratio.

A non-linear amplifier placed after the dynamic effect makes no sense with the dynamic range controller. In addition to this it can increase level of the dynamic controller output signal and degrade the limiter and/or compressor operation.

A non-linear amplifier placed before or after the delay block causes that the level measurement block follows envelope of signal that differs from signal entering the multiplier to weight the input signal. Fig. 2 shows static transfer characteristic of the envelope follower without any non-linear processing (dash line) and with a non-linear amplifier with transfer characteristic according to Fig. 4 placed before the delay block (solid line).

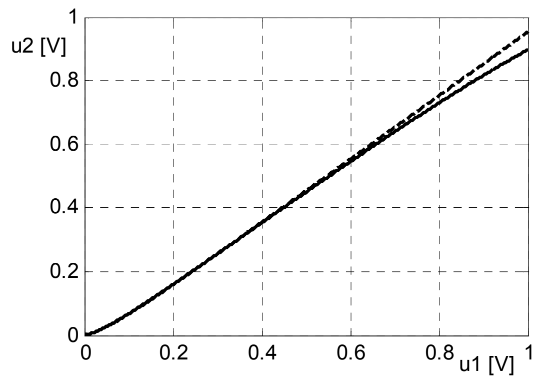

Figure 2: *Deviation of static transfer characteristic of envelope follower caused by non-linear amplifier placed in direct path.*

2.1. Non-Linear Transfer Characteristic Approximation

For analytic solution of non-linear circuits we must have analytic expressions that describe relations between particular varying quantities. They are mainly analytic representations (approximations) of characteristics of non-linear circuit elements.

Approximation is an operation that is always a compromise between accuracy and simplicity. For the solution, we always use the simplest approximation that offers the possibility of sufficiently true representation of the effect examined. The power polynomial approximation, linearization, approximation by a broken line and by an exponential function are mainly used.

The power polynomial approximation is most suitable for transfer characteristic approximation of tube amplifiers with receiving tube operating in starved-plate mode [5].

2.2. Finding of Polynomial Coefficients

We can find power polynomial coefficients using an approximation method, e.g. the least mean square method, from static transfer characteristic of the system measured in a sufficient number of points. The order of polynomial is determined experimentally on the basis of the change in maximum deviation of the real transfer characteristic from its approximation. If the maximum deviation does not change markedly, increasing the polynomial order does not make sense.

However, for computer DC analysis we need a model of amplifier's electric circuit. Measuring the static transfer characteristic of real analog model is time-consuming and in many cases it is not possible due to the coupling capacitors.

We can also determine the coefficients of the polynomial that approximates the static transfer characteristic of the system from the higher harmonics ratio when generating the harmonic signal. See [5] for details.

2.3. Non-linear Amplifier Saturation

Saturation is a term representing a non-linear amplifier parameter, which causes sound to be more "tubed". This parameter can influence the signal boost, tube heating, operating point, etc. Anyway, this parameter changes higher harmonics ratio of the output signal. There are many possibilities how to change higher harmonics ratio in the non-linear amplifier model that uses the power-polynomial approximation. One possibility is the system in Fig. 3. This system keeps higher harmonics mutual ratios unchanged with change of saturation. It changes only higher harmonic amplitude ratios with respect to the amplitude of the first harmonic. See [5] or [7] for details.

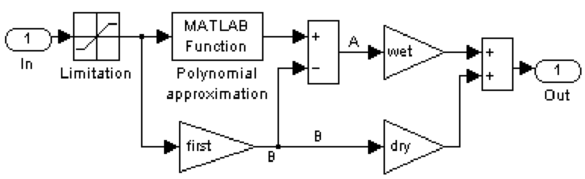

Figure 3: *Simulink model of non-linear system with change of saturation.*

3. ALGORITHM OPTIMIZATION

Evaluating polynomial of higher order is a very power-demanding process. Optimization of non-linear amplifier using power-polynomial approximation with adjustable saturation is described in [6] and [7]. Further increment of power demands are over-sampling or band-limiting techniques [4] that are used to avoid aliasing distortions. Especially over-sampling method is not very useful for high-order polynomials because the approximation polynomial must be evaluated for all samples of over-sampled input signal. Especially for band-limiting techniques the double Harvard architecture, vector, or SSE optimization can be used.

4. SIMULATION RESULTS

The non-linear amplifier model from Fig. 4 was used for simulation. The "Tube Simulator! block is subsystem for model from Fig. 3. Coefficients of approximation polynomial for the first testing were obtained from output signal spectrum of real analog tube amplifier with the 12AX7 tube operated at low plate voltage. Transfer characteristic of this amplifier is in Fig. 5. The characteristic is transformed to the "VST plug-in" amplitude range <-1;1> with 3 dB overloading of input signal. Fig. 6 shows higher harmonics ratio of this system with harmonic signal generation with nominal amplitude.

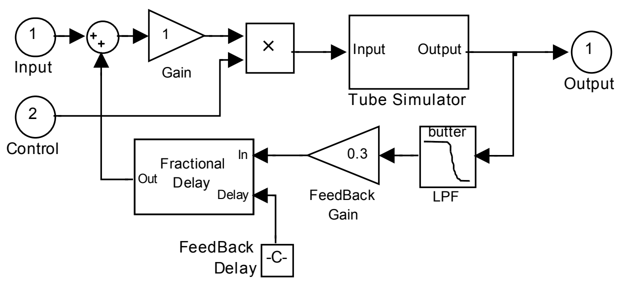

Figure 4: *Simulink model of simple non-linear amplifier using power-polynomial approximation.*

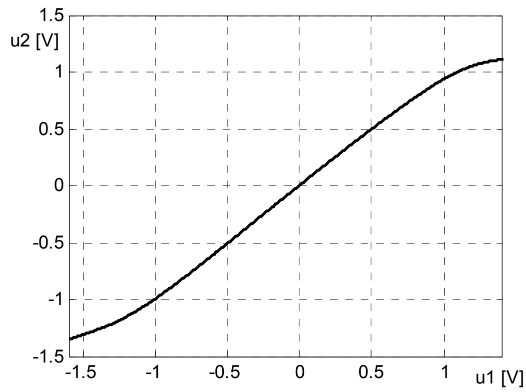

Figure 5: *Transfer characteristic of non-linear amplifier used for simulation.*

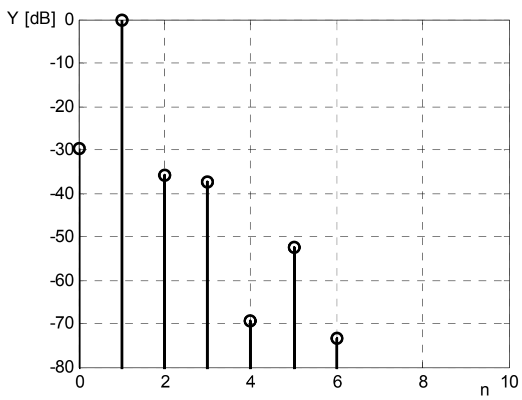

Figure 6: *Higher harmonics ratio of non-linear system with transfer characteristic according to Fig. 5 with harmonic signal generation with nominal amplitude.*

A combined compressor-expander system described in [3] or [4] and a special harmonic input signal with envelope according to Fig. 7 were used for simulation.

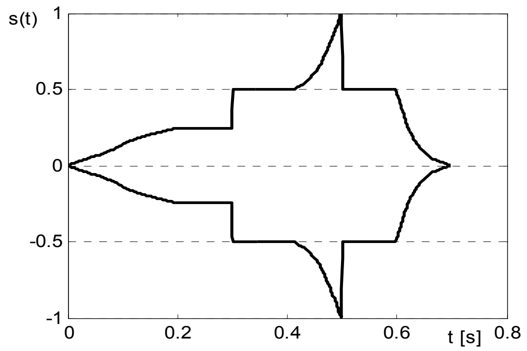

Figure 7: *Test signal envelope.*

The release time of 5 ms and the attack time of 1 ms were used. Averaging time was set exactly to 25 periods of test signal to minimize signal distortion caused by ripple of the envelope follower output signal. The expander and compressor threshold levels were set according to the test signal – first part of test signal falls into expander stage, second part into linear stage and impulse into compressor stage.
The output signal spectrum in each part was counted from exactly 80 periods of the input signal after end of the transients. No weighted window was used - frequency of the test signal was set to 1/20 of sampling frequency, so each period consists of exactly 20 samples. Every block of samples used for discrete Fourier transform of output signal starts and ends at the first sample of period (zero value).
You can see some simulation results in Fig. 9 to 16. The output signal envelope and spectrums with compressor ratio (CR) set to 5:1 and expander ratio (ER) set to 1.5:1 are in Fig. 9 to 12. The output signal envelope and spectrums with compressor ratio set to 2:1 and expander ratio set to 3:1 are in Fig 13 to 16. As can be seen from Fig. 9 and 13, the output signal envelope (solid line) is almost the same as the output signal envelope when no non-linear amplifier is used (dash line). Fig. 8 shows variation of the input non-linear amplifier (dash line) and control non-linear amplifier (solid line) transfer characteristic from linear model.

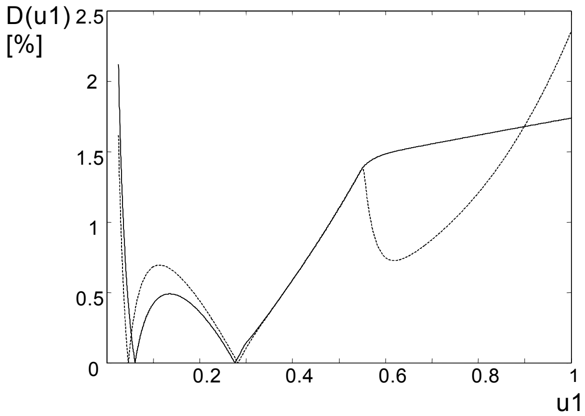

Figure 8: *Variation of transfer characteristic from linear model.*

Higher harmonics ratios differ in each stage (expander, linear, compressor) and also depend on the ratio parameters. Contrariwise, the gain parameter does not affect the output signal spectrum. It follows from simulation results that higher harmonics ratios differ in each stage (expander, linear, compressor) and also depends on the ratio parameters. Contrariwise, the gain parameter does not affect the output signal spectrum. It also follows from simulation results that the output signal envelope is almost the same as the output signal envelope when no non-linear amplifier is used and static transfer characteristic does not differ from static transfer characteristic of dynamic controller without non-linear amplifiers.

5. VST PLUG-IN IMPLEMENTATION

The described model was implemented as an VST plug-in module without its own graphic user interface. The VST plug-in allows setting of common compander parameters as well as the all parameters of the non-linear amplifier model – i.e. the higher harmonics ratio, saturation, feedback time, feedback level, and filter parameters. The plug-in uses oversampling technique to avoid aliasing distortions. Order of upsampling and downsampling is derived from order of power polynomial used for approximation.

6. CONCLUSION

The model described in the paper can be used for the digital dynamic effects implementation, especially for digital companders or maximizers working on real-time. Resulting sound is more expressive and ”audibly dense” than sound of standard dynamic controllers.

7. ACKNOWLEDGEMENTS

The paper was prepared within the framework of FRVS project N°2125/2003/F1 of the Ministry of Education, Youth and Sports.

8. REFERENCES

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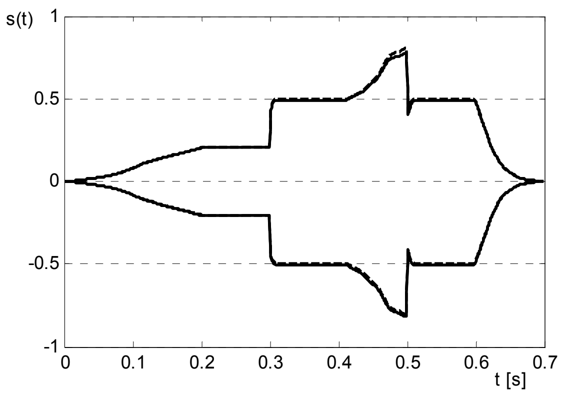

Figure 9: *Output signal envelope (ER = 1.5, CR = 5).*

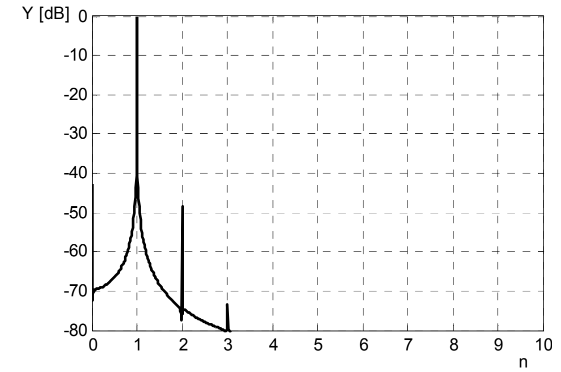

Figure 10: *Output signal spectrum – expander (ER = 1.5).*

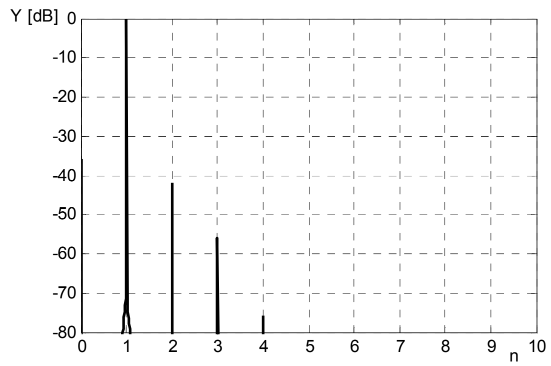

Figure 11: *Output signal spectrum – linear (Gain = 2 dB).*

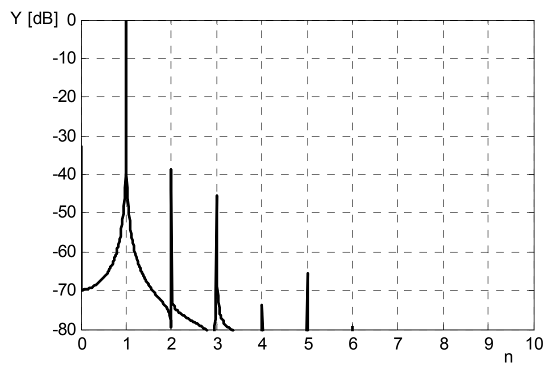

Figure 12: *Output signal spectrum – compressor (CR = 5).*

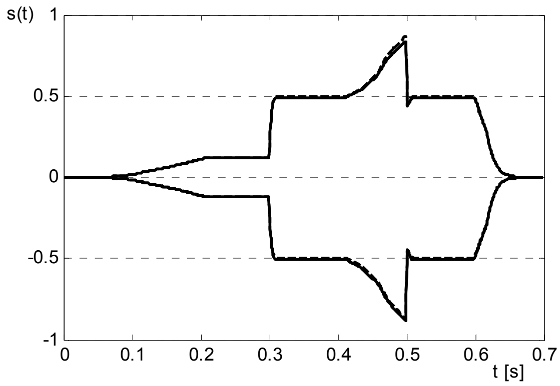

Figure 13: *Output signal envelope (ER=3, CR=2).*

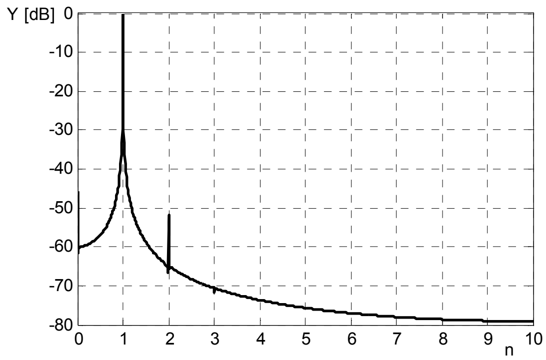

Figure 14: *Output signal spectrum – expander (ER = 3).*

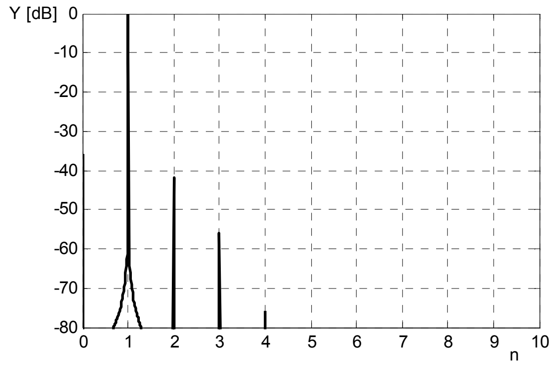

Figure 15: *Output signal spectrum – linear (Gain = 0 dB).*

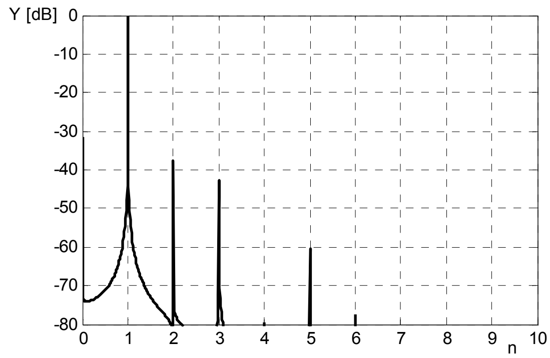

Figure 16: *Output signal spectrum – compressor (CR = 2).*