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The Active pulse modulated Transducer (AT)
A novel audio power conversion system architecture

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ABSTRACT

A novel audio power conversion system architecture is presented, in the attempt to provide a step forward in overall system efficiency and performance. The Active pulse modulated Transducer system ("Active Transducer") converts power directly from AC mains or from a DC power supply to the acoustic output in one simplified topological stage. New perspectives in audio system design emerge.

1. INTRODUCTION

This paper addresses the fundamental audio power conversion architecture, i.e. conversion from the battery or mains to the actual acoustic power we listen to. A novel audio power conversion system, which we name the Active pulse modulated Transducer (or Active Transducer) is introduced with the objective of improving system efficiency, audio performance, while still reducing the overall system complexity of the audio reproduction system. We believe the new architecture opens new opportunities for audio system designers.

Today, almost exclusively, we still utilize the conventional power conversion architecture as illustrated in Figure 1.

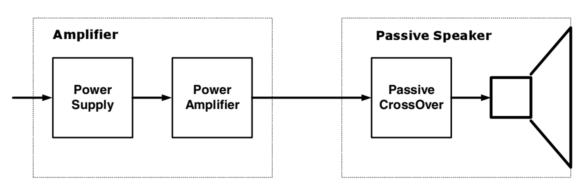

Figure 1 Audio power converting elements in conventional audio chain

It is well known, that this chain is inefficient [1]. The transducer is the fundamental source of the efficiency problems. The conversion efficiency of a classical electro dynamic transducer can be expressed as [2,3]:

$$\eta_{tr} = \frac{100}{1 + \dfrac{2cm^2 R_c}{\pi\rho a^4 (Bl)^2}}$$

Where:

ρ = air density
c = sound velocity
m = vibration system mass
a = effective radius of diaphram
Bl = Force factor
R_c = Coil resistance

Since the magnitude of the second term in the denominator is generally 50 – 100, conversion efficiency is generally 1-2%. Horn speakers have higher efficiency, but other restrictions as linearity and physical space requirements, limiting the applications to professional audio. A general efficiency improvement by an order of magnitude would virtually eliminate the need for power amplification, as we know it today. With classical and equally very inefficient class AB power amplification, total conversion efficiency is typically 0.01% at normal output levels, and only approaching 1-2% for the most efficient systems, operating at high outputs. The heat development causes numerous negative side effects, as voluminous design, heat development, audio performance compromises and a significant Bill Of Materials.

Clearly, there is a theoretical room for rich improvement by reducing this waste of energy and this is a key to change audio system design. Just imagine the potential effects of a system efficiency of 1%, 10%,... It might seem paradoxical, that this level of efficiency is tolerated in the industry, especially since a low efficiency *only* has negative and significant side effects. Some of the answers lie in 1) the engineering challenge where numerous scientific fields need to be combined, 2) actual physical limitations in converting electrical power to acoustic power moving air 3) General market conservatism.

Another paradox in audio design is, that amplifiers and speakers are developed as separate “general purpose” units, amplifiers being designed for a wide range of loads and vice versa for the transducer. Speakers have various resistive and reactive impedance characteristics that the amplifier has handle in order to be competitive and this design criteria significantly complicates the amplifier design. General audio power amplifiers are designed for pure resistive and continuous power specification, although transducers are far from ohmic and music is far from sinusoidal. Crest factors are easily below 1/8 and much lower if the dedicated frequency bands are analyzed individually [5,6] and the amplifier dedicated to a certain bandwidth of reproduction.

It is indeed questionable if the amplification stage and transducer stage should be separately designed from any perspective. There seems to be rich opportunities for topological simplification by integrated and dedicated design, taking a holistic view of the audio power conversion topology limitations, not limited by the conventional break-up and barriers between the power supply, amplifier and transducer. This unorthodox thinking has been another primary motivation in the innovation of the Active Transducer topology.

The active or powered speaker, gaining acceptance in professional audio, is an example of partial dedication on of the amplifier to each speaker. Typically composed of general purpose amplifiers and transducers, the main benefit is the elimination of passive crossovers, and dedicated amplifiers for each band. Generally the potential of full dedication is and cannot be exploited. The system efficiency and performance is as such not improved significantly.

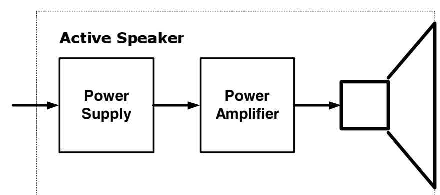

Figure 2 Audio power conversion elements in the active speaker

2. The Active Transducer

The Active Transducer system topology is an electric-acoustic conversion system, converting energy from the AC mains or DC supply to a high quality acoustic output in one topological stage. The objective of this topological simplification of the audio power conversion chain, is to overcome fundamental problems and limitations in conventional and separated power amplifier and transducer design. A concept block diagram is shown in Figure 3

The electric-acoustic conversion stage, preferably based on the electro dynamic principle or other transducer methods with primary inductive high frequency characteristics, is driven directly by an intelligently modulated power pulse train generated inside the transducer structure, using a switching stage that connects directly to the transducer coil. The modulator, the switching stage and the electric-acoustic conversion means are integrated mechanically and electrically in one operational unit, driven directly to from a DC power supply or even directly from the AC mains.

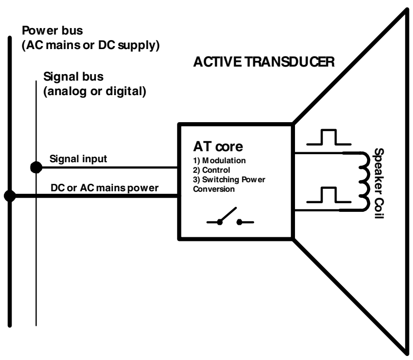

Figure 3 Active Transducer

3. Active Transducer electric cores

We define two variants of the Active Transducer:

- DC Active Transducer (DCAT) – operating on a DC input power supply
- AC Active Transducer (ACAT) – operating directly on AC mains.

The general block diagram of the DCAT core is shown in Figure 4. The main elements are:

1. The pulse modulator
2. The control system
3. The power conversion stage
4. Optional transducer compensation
5. The transducer

The DCAT core consists of a simplified Pulse Modulation Amplifier [1] topology, where the electric DC-AC power conversion stage is dedicated to the application (transducer characteristics, bandwidth, max power level and max crest factor). The integrated modulator and control system converts the analog or digital input source to a pulse modulated signal, which is amplified through a conversion stage, e.g. a H-bridge topology, which operates from the single input DC supply and connects directly to the transducer core. A decoupling capacitor is integrated in order to locally supply the high frequency currents, whereas all audio frequency components are supplied from the DC power supply.

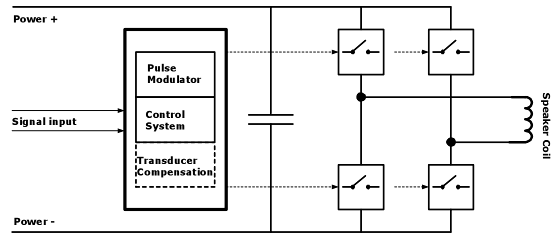

Figure 4 DC Active Transducer (DCAT) core electronic architecture

A further integration step is the implementation of the conversion into a one stage conversion, what we call the AC Active Transducer or ACAT. A block diagram is shown in Figure 5.

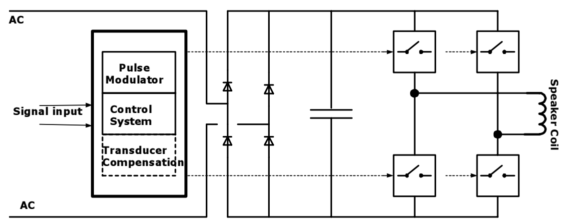

Figure 5 AC Active Transducer (ACAT) core electronic architecture

In the ACAT architecture, the power supply is virtually eliminated by converting directly AC-AC from mains. Rectified mains is a high voltage level, and the system efficiency may be compromised in the amplification stage. However, the elimination of 90% of the power supply block (either transformer based or switching converter) is as tremendous advantage in terms of complexity reduction and complexity improvement. The basic ACAT core is non isolated in the electric single stage AC-AC power conversion chain, and this has to be implemented otherwise if needed.

3.1. AT advantages & perspectives

An intelligent modulation scheme have been developed in order to utilize the inherent qualities of the transducer for filtering of the pulse train and thereby obtaining higher efficiency. Permanent elimination of any passive filtering is a critical step forward, since the most restrictive part of Pulse Modulated Amplifiers (PMA's) / switching Class D amplifiers is the L-C output filter. The filter generally compromises most important parameters as general sound quality, bandwidth, frequency response, efficiency, stability and EMI. The AT will eliminate this limiting element, independent upon application, bandwidth requirement or power requirement.

Another important advantage is the use off the inherent magnet structure for residual heat dissipation from the AT electronic core, hence saving the thermal management system/heat sinking usually used in conventional amplification. Elimination of both L-C filter and thermal management dramatically reduces the AT core physical volume. As such, Active Transducer saves material for packaging, cabling, cooling of amplifier and power supply. Many performance degraders as the filter, connectors and speaker cables are effectively eliminated from the signal path.

The complete mechanical dedication and integration also helps to solve the well-known EMI problems from filter coils and speaker cables containing switching residuals, and the actual EMI noise sources are reduced.

A further dimension is the transducer compensation and equalization techniques that are naturally implemented locally and dedicated as a natural part of the AT electronic core.

In the paper, the basic theoretical and topological foundation for this new audio component and audio architecture are introduced. New opportunities in general audio design emerges and will also be addressed in the paper. A benchmark on all relevant parameters with the conventional power supply / amplifier / transducer topology is given to reveal the advantages and possible limitations of the AT component and AT audio architecture.

The electronic symbol we propose for the Active Transducer is show in Figure 6.

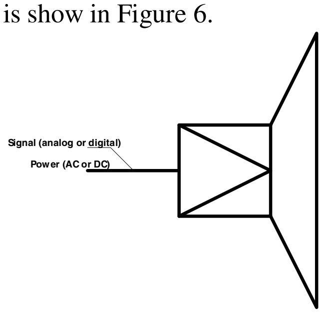

Figure 6 Active Transducer symbol (AT)

4. Pulse Modulated Amplifier limitations

High efficiency and compact power conversion is fundamentally needed in order to reach to total electrical and mechanical integration the Active Transducer paradigm represents.

The PMA topology, shown in Figure 7, realizes drastically improved amplifier efficiency by use of a switching / class D output stage. Recent development in the field [1,8,9,19], has proven that high efficiency and high grade audio performance can be combined. The trick over conventional PWM/class D has been innovation in novel modulation principles and control systems. Today, well accepted high end amplifiers have been introduced with such techniques [16] and PMAs are gaining widespread use in professional audio.

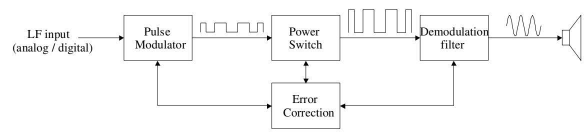

Figure 7 Pulse Modulated Amplifier (PMA)

The passive output LC filter of PMAs serves to demodulate the pulse modulated output power signal, in order to prevent losses from the high frequency residuals in the transducer and equally to reduce EMI from speaker cables. Subsequently, the output filter is critical for the PMA topology to work. Generally, PMAs are implemented with 2nd or 4th order output filters, including RC Zoebel networks to implement a reasonable filter Q in case of open load situations and load steps. A reference simulation of a 250W/80V PMA system implemented with a 2nd order output filter is shown in Figure 8.

4.1. Limitations of the L-C output filter

The passive output filter is probably the most limiting element in the topology – from any perspective it is strongly undesirable as it will be explained below.

Efficiency

Efficiency is degraded since the filter inductors generally adds significantly to the total power loss in the PMA system. The following elements contribute:

- Conductions losses caused by the LF signal current running in the finite DC resistance of the inductor
- Conduction losses residing from the HF ripple current running in the inductor. May be significant from the high frequency part due to the skin- and proximity effects.
- Magnetic core losses due to the LF induction swing from the signal current
- Magnetic core losses caused by the ripple current (a HF induction swing)

In order to achieve acceptable efficiency at all output levels, generally very low permeability cores or air cores has to be used in the application, making inductors quite bulky and inefficient due to numerous inductor turns on the core.

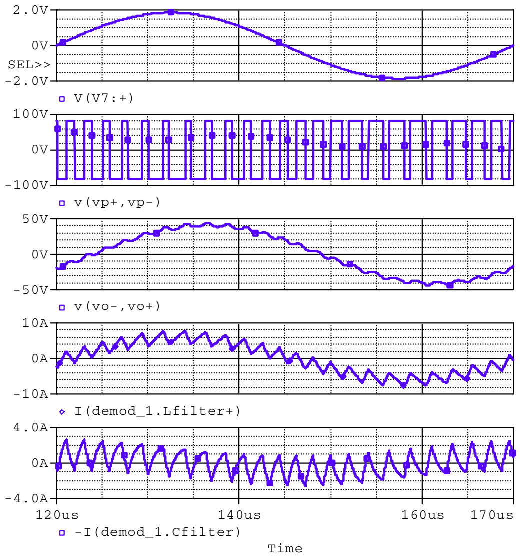

Figure 8 Essential PMA output stage signals in a 250W/80V case example, from top to bottom (1) Input signal, (2) Power stage output (3) Demodulated output (4) Inductor current and (5) Filter capacitor current

Linearity

The inductor and capacitor process both audio and carrier related signals (Figure 8), generally reducing audio specifications since the inductor is not linear. Another limitation is Slew Rate limitation caused the LC filter. SR of PMAs is generally much lower than for class AB amplifiers.

Output impedance

The output filter causes an undesirable frequency dependent output impedance, which only somewhat can be compensated by feedback control, e.g. the MECC topology [1,10,11].

Stability

The resonant nature of 2. or higher order demodulation filters affects system stability, and complicates the implementation of effective feedback control systems. Generally, the LC filter has to be damped by an additional power Zoebel network, in order to prevent damage from load steps and non-loaded operation. The Zoebel network further increases complexity and adds undesirable losses at high frequencies, reducing the system efficiency and power bandwidth.

EMI

LC filtering introduces a ripple current in inductor and capacitor of significant magnitude, decreasing efficiency and causing a significant noise source. The physical size of the inductor causes one of the strongest (radiated) EMI problems within the PMA/class D amplifier topology. In many applications, the output filtering needs shielding.

Space

The filter is potentially the most voluminous part of the amplifier. The LC filter also prevent full silicon integration of the PMA, which is a disadvantages compared to e.g. class AB amplifiers which can be fully integrated up to reasonable power levels.

Cost

The LC filter is one of the most significant contributions on any BOM, hence reducing the competitiveness of Class D over e.g. Class AB. This is worsened by the PCB space and potential shielding required by the output filter.

Overall, it is strongly desirable to fundamentally eliminate any passive filtering, and the Active Transducer architecture does exactly that as one of its most important merits.

5. PULSE DRIVING TRANSDUCERS

The electro dynamic transducer has proven its advantages for nearly a century for telephone and audio reproduction, and is the most widely spread method to convert electric power to acoustic power. Here the implications of driving electro dynamic transducers with a pulse modulated signal will be addressed, with focus on power conversion efficiency. The electronic equivalent in simplified form is well known [3]:

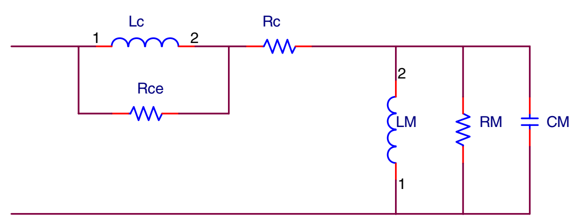

Figure 9 Simplified electrical model of the electro dynamic transducer

One of the inherent elements is the inductive nature, residing from the equivalent coil inductance *Lc*.

Magnetic / Eddy current losses, which are generally frequency dependent are modeled by the parallel inductance *Rce(f)*.

Whereas this lossy inductive nature represents challenge to many conventional class AB designs [2], it is a tremendous advantage in the Active Transducer topology, since the inductive nature can be utilized for demodulation purposes, reducing the contribution losses from the switching frequency related components to neglible levels by proper design of the AT core. Generally, *Lc* may vary from several mH from heavy bass drivers down to 10's of uH in tweeter drivers. Figure 10 shows the impedance of a 15" bass driver, with the characteristics bass resonance frequency and the "lossy" inductor characteristic at higher frequencies.

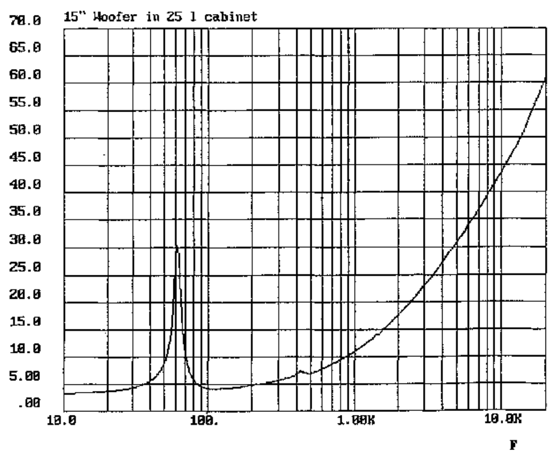

Figure 10 Impedance of 15" bass transducer. Lc = 1000uH.

5.1. Efficiency considerations

One of the major obstacles of pulse driving transducers is the contributions from the high frequency carrier and intermodulation components between signal and carrier, that will introduce loss in the coil series resistance *Rc* and *Rce.*

To investigate this phenomena, it is illustrative to consider the effects of general double sided PWM modulation, where the output waveform can be understood by developing the double Fourier series for conventional NADD PWM [1,7]

$$
\begin{aligned}
V_{O,NADD}(t) &= V_P M \cos(y) \\
&+ 2V_P \sum_{m=1}^{\infty} \frac{J_0(m\pi \frac{M}{2})}{\frac{m\pi}{2}} \sin(\frac{m\pi}{2}) \cos(mx) \\
&+ 2V_P \sum_{m=1}^{\infty} \sum_{n=\pm 1}^{\infty} \frac{J_n(m\pi \frac{M}{2})}{\frac{m\pi}{2}} \sin(\frac{(m+n)\pi}{2}) \cos(mx+ny)
\end{aligned}
$$

For details, see [1]. The time and frequency domain characteristics are shown in Figure 11. The modulated signal is perfectly reproduced in the output waveform, however there is a significant carrier generated high frequency content which will generate losses in Rc and equally generate losses in the magnetic and eddy current losses Rce – this is one of the major concerns by pulse modulated transducers.

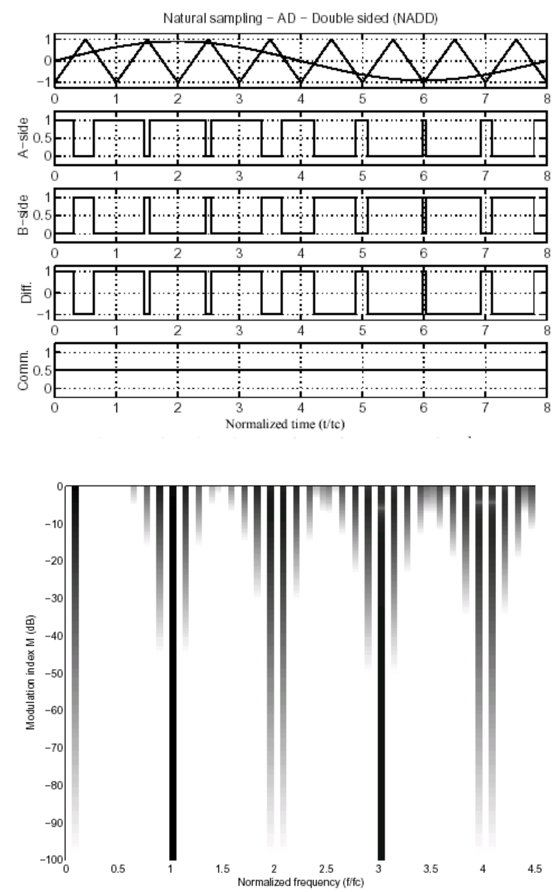

Figure 11 Time and frequency domain characteristics for Naturally sampled AD PWM (NADD) illustrated by HES surface [1]

Assuming that the main contribution is from the first harmonic of the carrier ω_c, the resulting losses from the carrier waveform are will be dependent upon carrier frequency and modulation index M as::

$$PRc(\omega_c, M) = \frac{1}{2 \cdot Rc}\left(2V_P \frac{J_0(\pi \frac{M}{2})}{\frac{\pi}{2}}\right)^2 \cdot \left|\frac{R_c}{R_c + (Rce(\omega_c) \| j\omega_c L_c)}\right|^2$$

Similar, for the lossy inductor represented by the frequency dependent resistor *Rce*:

$$PRce(\omega_c, M) = \frac{1}{2 \cdot Rce(\omega_c)}\left(2V_P \frac{J_0(\pi \frac{M}{2})}{\frac{\pi}{2}}\right)^2 \cdot \left|\frac{Rce(\omega_c) \| j\omega_c Lc}{Rc + (Rce(\omega_c) \| j\omega_c Lc)}\right|^2$$

Defining the *relative loss factor* as the loss related to the maximal output power in percent

$$\Delta Rc(\omega, M) = \frac{PRc(\omega, M) \cdot 100}{Po_{\max}}$$

$$\Delta Rce(\omega, M) = \frac{PRce(\omega, M) \cdot 100}{Po_{\max}}$$

we will have a meaningful measure of the transducer dissipation related to the carrier generated high frequency component. In order to reach the goal of efficiency improvement, accumulated carrier related contributions in core loss should be neglible:

$$\Delta(\omega_c, M) = \Delta Rc(\omega_c, M) + \Delta Rce(\omega_c, M) \le 0.1\%$$

With e.g. a 100W maximal output $Po_{\max}$, the carrier related losses would be less than 100 mW. Considering a worst case but still realistic case example where:

- *Lc=40uH*, corresponding to a small transducer (e.g. tweeter)
- *Rc = 4*
- $Rce(f) = |j\omega_c L_c|$, corresponding to a typical 45 deg high frequency characteristic for the lossy inductor.

The resulting relative loss factor is shown in Figure 12. Clearly, the relative loss factor is much higher than desired at realistic carrier frequencies (300kHz – 600kHz) , with a dominant contribution from the lossy inductor. Conventional 2-level switching is thus *not* a viable approach without severe efficiency compromises. Only in bass transducers with e.g. *Lc>500uH* the relative loss factor approaches acceptable levels.

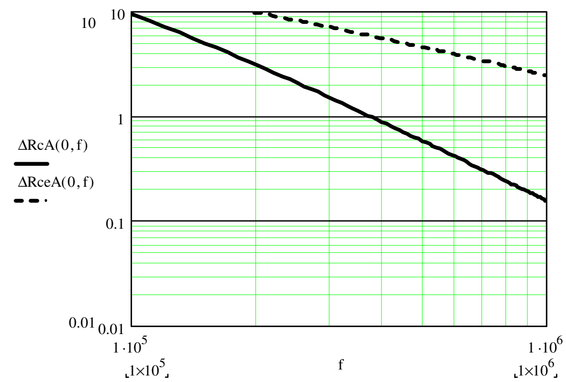

Figure 12 NADD PWM. Relative loss factor vs. frequency at M=0 (worst-case) for the contributions Rc and Rce (NADD)

5.2. Intelligent modulation

In order to reduce the effects of high frequency losses, an alternative modulation technique proves to have significant advantages, and fundamentally overcomes the HF transducer loss problem. In [1] a range of pulse modulation methods were analyzed, including three-level modulation techniques as NBDD PWM. The double Fourier series for NBDD PWM is [1,7]

$$V_{O,NBDD}(t) = MV_P \cos(y)$$
$$- 4V_P \sum_{m=1}^{\infty} \sum_{n=\pm 1}^{\infty} \frac{J_n(m\pi \frac{M}{2})}{m\pi} \sin(\frac{(m+n)\pi}{2}) \sin(\frac{n\pi}{2}) \sin((mx+ny) - \frac{n\pi}{2})$$

The time and frequency domain characteristics are shown in Figure 13. The elimination all components around odd harmonics of the carrier, combined with the proportional relationship with modulation index is advantageous in relation to reducing the losses from the carrier components. For NBDD, assuming that the main contribution is from the components (m=2, n=+/-1), we get the following approximate expressions:

$$PRc(\omega_c, M) = \frac{1}{2 \cdot Rc} \cdot 2 \cdot \left(4V_P \frac{J_1(2\pi \frac{M}{2})}{2\pi}\right)^2 \cdot \left|\frac{R_c}{R_c + (Rce(\omega_c) \| j\omega_c L_c)}\right|^2$$

And:

$$PRce(\omega_c, M) = \frac{1}{2Rce(\omega_c)} \cdot 2 \cdot \left(4V_P \frac{J_1(2\pi \frac{M}{2})}{2\pi}\right)^2 \cdot \left|\frac{Rce(\omega_c) \| j\omega_c L_c}{R_c + (Rce(\omega_c) \| j\omega_c L_c)}\right|^2$$

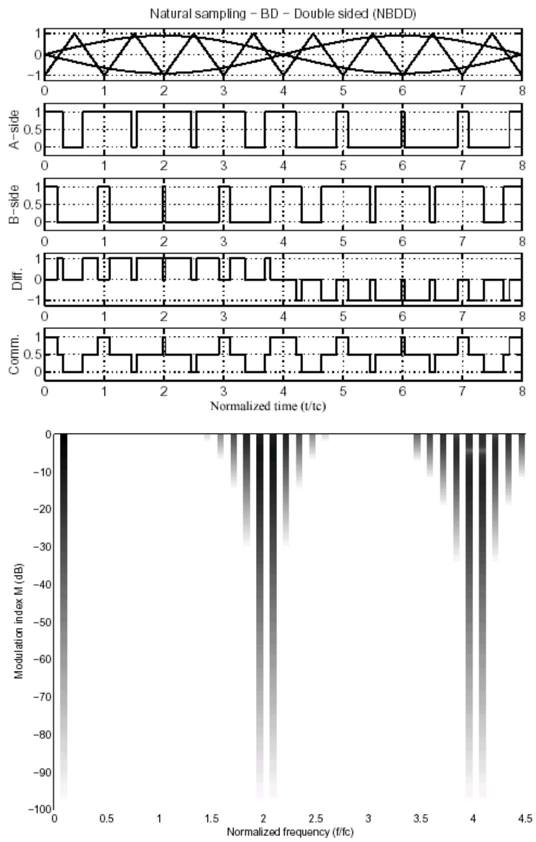

Figure 13 Time and frequency domain characteristics for NBDD PWM (NBDD)

The carrier related losses in the coil are reduced, since Bessel functions of 1st order converge to zero at zero modulation index. Furthermore, the output switching frequency is effectively doubled.

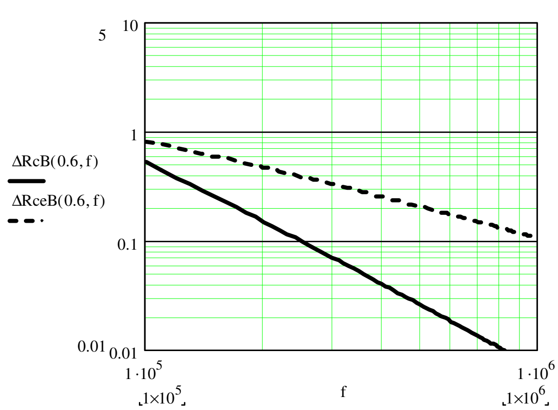

Figure 14 Relative loss factor vs. frequency at M=0.6 (worst-case) for the contributions Rc and Rce (NBDD)

The resulting relative loss factors for the case example are illustrated in Figure 14, with a modulation index M = 0.6, corresponding to the worst case situation. Clearly, the result is radically different and the loss from both Rc and Rce residing from the high frequency carrier will be neglible, with the given assumptions.

The relationship between relative loss factor and modulation depth is illustrated in Figure 15, investigated with a typical carrier frequency of 400 kHz. By selection of the NBDD type modulation scheme or at scheme with similar characteristics, we can eliminate the contributions from carrier components as a very important conclusion and foundation for the Active Transducer architecture.

Equally important, the total system efficiency improvement can now be estimated using output stage models as in [1]. By elimination of the 4 contributions from the inductor, the idle consumption can be approximately halved and the efficiency at maximal power improved by generally 2-4% [1]. Moreover the elimination of energy consuming Zoebel networks improves efficiency at higher output frequencies with a larger factor.

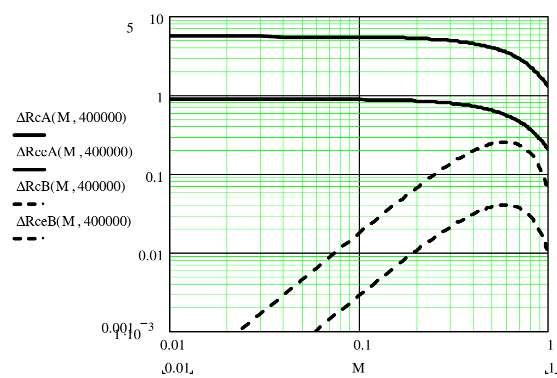

Figure 15 Relative loss factor vs. M for both NADD and NBDD cases (fc = 400kHz)

The remaining contributor to power dissipation will be conduction losses and switching losses from the output stage [1].

The residual heat from the AT core is comfortably handled by the magnet structure of the transducer section. The contribution from the AT core, will generally more than an order of magnitude lower than the contributions from the transducer itself, with a well designed 95-96% efficiency AT core.

A more detailed total system efficiency model is under development, including all contributions in the AT core and the transducer itself. However, based on this initial investigation, it can be concluded that pulse modulating transducers is viable with NBDD type modulation methods, with improvements in total system efficiency as an important benefit.

6. MODULATION & CONTROL

Despite the advantages of the NBDD scheme, general PWM has proven is disadvantages in PMA design and some of the same problems will convert to the AT architecture. In [12] the COM modulator structure was introduced as a advantageous modulator & control system structure with the basic advantages:

- Excellent linearity
- Inherent stability
- Control system bandwidth equal to the carrier frequency.
- Infinite power supply rejection ratio.

It is desirable to adapt these advantages to the Active Transducer, however the COM principle synthesizes a classical 2-level output it its conventional form. Subsequently, the basic principle has been further developed specifically for the AT core – in order to synthesize the needed NBDD or NBDD like 3-level output.

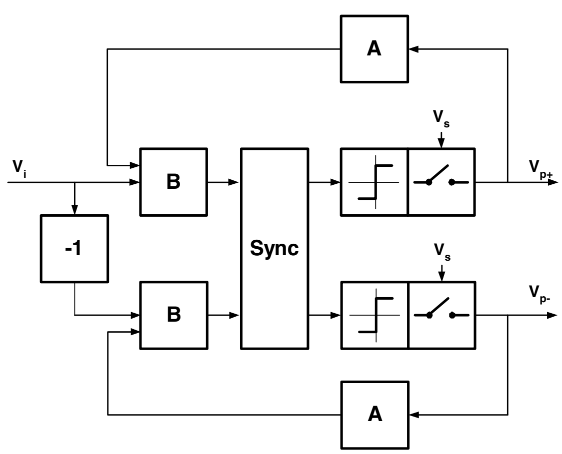

Figure 16 SCOM architecture

A block diagram of the principle which is called the Synchronized Controlled Oscillation Modulator (SCOM) is shown in Figure 16. The novel topology consists of 2 COM modulators, which are synthesized with the same parameters, however a synchronization element is introduced between the two modulators in the forward path. The input signal is *inverted* to one COM modulator hence generating a situation where the oscillation is in phase – and the audio signal is out of phase in each COM legs. The result is a "pseudo BD" modulation, exactly as desired for the AT core.

The principle is simulated in Figure 17, in a 250W case example. Notice the characteristic sinusoidal COM oscillation method, where the modulating carrier signal is phase, whereas the LF audio signal is out of phase. The differential output is clearly double sided 3-level in nature, with two pulses per switching period.

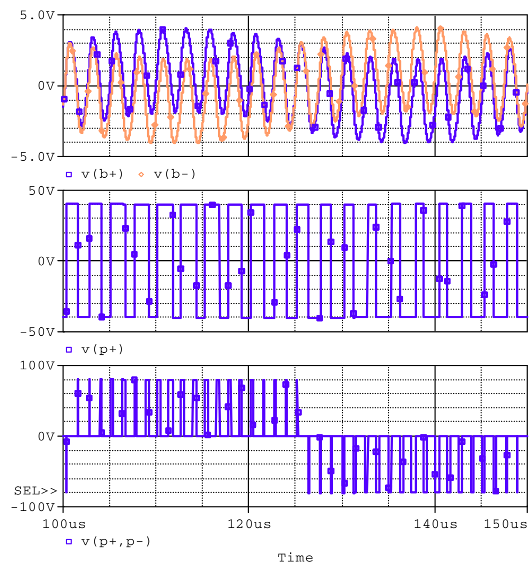

Figure 17 SCOM – essential signals

Figure 18 illustrates simulated characteristics of a reference design. The elimination of the output filter enables a high bandwidth design, in this case –3dB bandwidth is approximately 200kHz.

Another interesting characteristic is the output impedance characteristics which is now defined by the filterless SCOM feedback AT core design. Output impedance in the proposed design is thus less than 2 mΩ @ 20kHz.

The SCOM algorithm inherits the theoretically infinite power supply rejection ratio [12], which significantly reduces the requirements for power supply design using Active Transducer components.

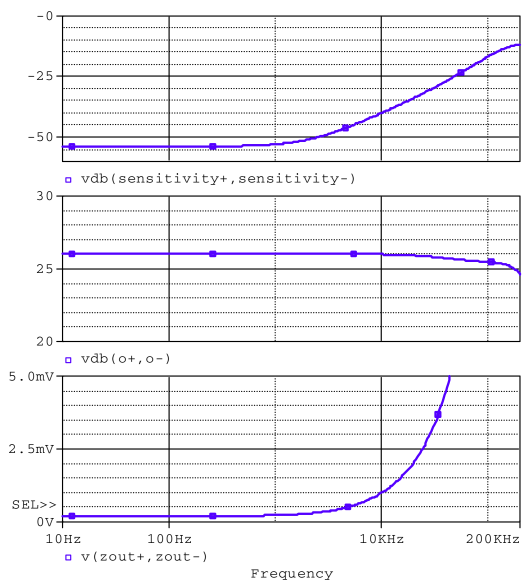

Figure 18 Active Transducer core design based on SCOM algorithm. Case example simulation of sensitivity function (top), frequency response (mid) and output impedance (bottom, 1mV=1mΩ)

7. Intelligent supplies (PAWM)

Presently, we have presumed that the DCAT is powered by a DC supply. The COM principle, although of the width modulating type, operates fundamentally different from PMA in that the transfer function is *independent* upon the power supply variable and exclusively determined by the feedback path gain A [12].

Further advantages can be obtained by implementing intelligent control on the DCAT power supply. We propose a new control method, called Pulse Amplitude Width Modulation (PAWM), illustrated in Figure 19. The power supply variable is controlled by the signal, as:

$$V_S = K \cdot |\max(v_i)| + \Delta \quad i \in \{1; N\}$$

The output stage modulation characteristics effectively change from a width modulating type to an amplitude *and* width modulating type, hence the PAWM designation. A PAWM simulation is shown in Figure 20. The modulation scheme is actually a new amplifier class (PAWM), where the output will be a hybrid between AD and BD mode. The differential benefits are achieved without the common-mode drawbacks of BD modulation [1].

Since power dissipation is strongly related to the supply voltage, either proportional to the supply voltage (switching losses) or proportional to the squared supply voltage (idle losses related to the FET output parasitic capacitance), PAWM significantly improves overall efficiency.

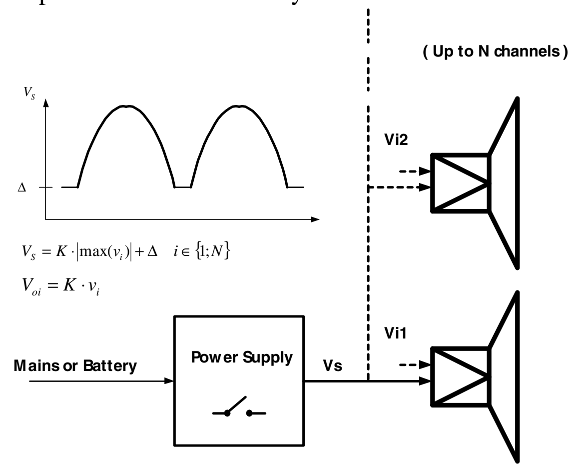

Figure 19 Intelligent Power Supply Control implementing Pulse Width Amplitude Modulation (PAWM) in the AT audio architecture

EMI is affected in a positive way, in that the ripple current and shoot-through/diode related noise is reduced, equally approximately proportional to the power supply voltage *Vp*.

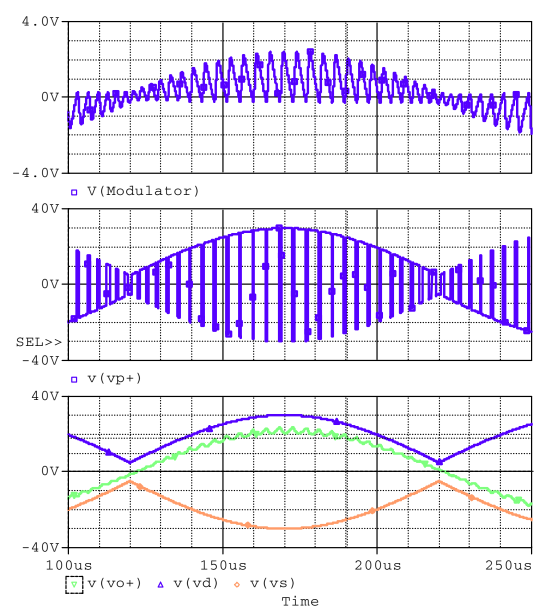

Figure 20 Simulation of PAWM control. Top is modulating signal, Mid – Output, Bottom – Supply rails with demodulated output

A further advantage is, that the PAWM method allows for simpler implementation of the output stage in terms of a two-transistor single ended (SE) output stage operating from dual supplies.

Hence PAWM in combination with a 2-transistor SE output stage, is an alternative "pseudo" BD method suitable for AT core implementation, having further positive effects on EMC and efficiency.

8. AT CORE POWER STAGE DESIGN

The power conversion stage studied so far has been the H-bridge, allowing for intelligent pseudo 3-level waveforms to be synthesized. In the AT architecture, the advantage of dedication should be utilized to the full extent, i.e. utilizing that crest factors and power requirements are different in each frequency bands.

Advantages can be gained by obviating the conventional thinking of 4, 8 Ω loads. Optimum loads should be defined by integrated Active transducer topology, by total optimization of efficiency and performance. Presently, a total AT efficiency model is under development and we plan to present the model in a future paper.

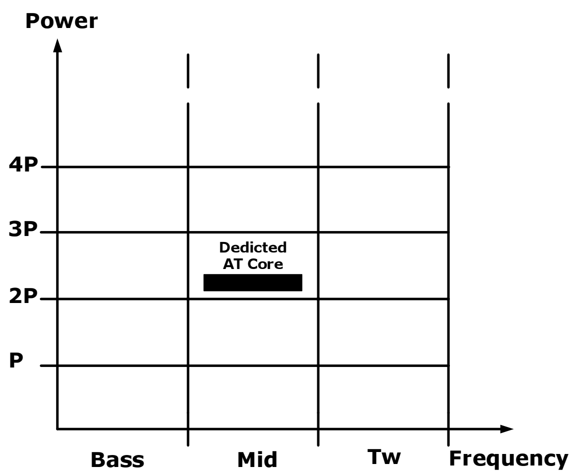

Figure 21 Dedicating power stages in AT core for power range, crest factor and target bandwidth.

The AT can advantageously be dedicated to the target bandwidth, e.g. divided by bass, midrange and tweeter electro dynamic transducer. *Lc* varies significantly with frequency band, in the range of more than one magnitude order. This should be utilized to optimize for lowest total power dissipation, e.g. using dedicated carrier frequencies for each band.

Also, the crest factors are very different in each frequency band, hence the continuous power can be significantly de-rated in midrange and especially tweeter band, simplifying the AT core for these particular bands.

An implementation matrix is proposed in Figure 21. By the definition of 3 dedicated frequency bands, and by quantizing the power range in factors of 2, all applications and transducer can be covered by a limited number of AT cores.

8.1. EMC considerations

EMC has proven to be a vital limitation in PMA designs, and one of the major obstacles preventing commercial use. As mentioned above, one of the major contributions of the Active Transducer is the potential benefits that can be realized over traditional PMA topologies. The major EMI sources for the conventional PMA are conducted noise coupled to the mains residing from the PMA output stage, and radiated noise from output stage, filter and speaker cables. The AT architecture reduces the sources of emission by:

- Elimination of the output filter as contributor
- Minimizing radiation loops consisting of output stage, filter, cables and transducers.
- Minimizing actual sources (di/dt primarily) by intelligently modulated switching directly on a dedicated and known transducer impedance.

Typical inductors values in PMA system are 20uH – 40uH. Assuming that the bass driver in Figure 10 is driving directly by an integrated AT core, ripple current amplitude in the load and mains will be reduced by the ratio of the inductors, or by a factor of 20-50 times.

9. IMPLEMENTATION CONSIDERATIONS

New topologies and principles may be appealing in theory but worthless if not practical for the world to enjoy as actual products.

Integration of the AT electronic core into the actual transducer structure is indeed mentally a big hurdle, with the conventional breakup of amplifier and transducer for the last century. However, the simplifications of the electronic architecture, especially the elimination of output LC filters and a dedicated power conversion, allows for a very compact implementation even in higher power AT systems in the kW area.

Most of the AT core can be implemented into silicon, and assembled on a aluminum substrate or ceramic substrate yielding a extremely compact package of less that 1/4 cubic inch pr. 100W, with the elimination of voluminous components as the LC filter and heat sinking.

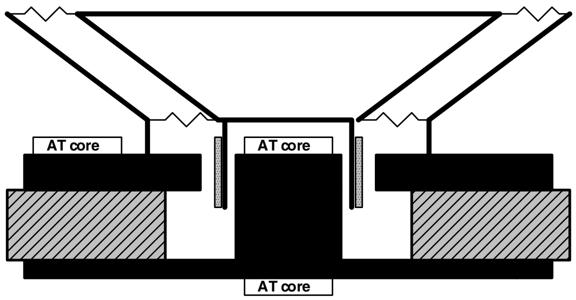

Figure 22 Implementation of the AT core

Substrate technology, e.g. IMST [17] is preferable in medium and high power applications due to the thermal characteristics, EMI characteristics and low cost. At lower power, small monolithic power packages can be utilized. The resulting package is sealed to the magnet structure – were more options are viable as illustrated in Figure 22.

10. BENCHMARKING

Consider the implementation of the 250W/8Ω case example. The power rating refers to conventional thinking, in the AT architecture, target SPL would be relevant target parameter. A benchmarking to a conventional PMA design, the ICEpower500A [18] is carried out, in order to put the potential advantages of the AT component into perspective. A picture of the electronics sections for the 250W module and the equivalent 250W AT core is shown in Figure 23.

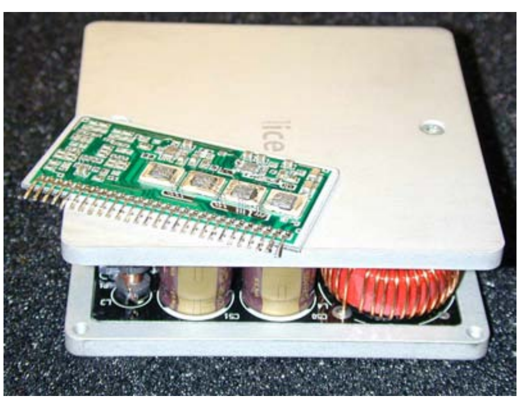

Figure 23 Comparing 250W/8Ω AT core (Substrate on top) with 250W/8Ω PMA module (ICEpower500A size 4”x 4” x 1”)

Based on the simulation results for the SCOM system above, the following results are achieved.

Parameter	250W PMA	250W AT core
THD+N all f, Po	<0.05%	<0.05%
Power Bandwidth	20kHz	> 100kHz
SR	4V/us	> 20V/us
PSRR	60-80dB	> 80dB
Zo @ 20kHz	200 mΩ	< 2mΩ
Idle loss	4W	1.5W
Efficiency@250W	93%	96%
Volume	16 cubic inch	0.5 cubic inch
Relative BOM	100%	30%

Advantages can be found on most parameters. Whereas the audio performance benefits may be of academic nature with the high reference performance of the 250W PMA system in mind, the benefits in efficiency, power density and system cost are substantial.

11. AUDIO SYSTEM ARCHITECTURES

The AT concept fundamentally changes audio system design, and new audio architectures emerge. Figure 24 illustrates a 2-way speaker design based on dedicated Active Transducers for the bass and the tweeter region. The DC supply feeds both Active Transducers and a signal interface implements cross-over’s (by analog or digital means).

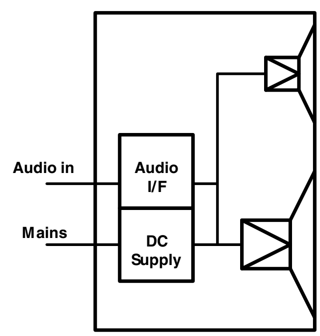

Figure 24 Active speaker based on dedicated Active Transducers (DCAT)

Compared to conventional active speaker design, the design of the DCAT based speaker is much simplified and enjoys the efficiency and performance benefits of the Active Transducer.

In Figure 25 a new multi-channel architecture is proposed, where the DC supply for the DCAT is in the subwoofer. Compared to present passive and active multi-channel architectures, significant advantages are gained in performance and system complexity.

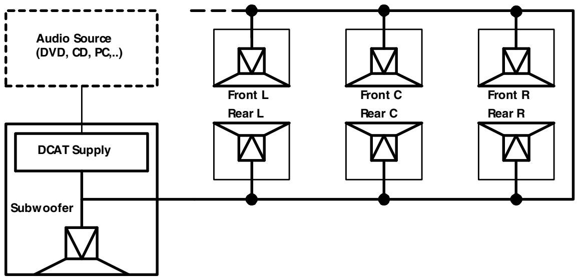

Figure 25 Multi-channel architecture based on Active transducers

In the automotive environment, where the DC bus is present, DCAT's can be connected directly to the battery supply already wired throughout the car, hence eliminating any power conversion in the compact main unit or in a separate amplifier.

12. FURTHER RESEARCH

The Active Transducer paradigm represents an ultimate challenge by its wide mix of scientific fields, that are generally considered complementary as electro acoustics, power electronics, modulation techniques, RF/EMC and more.

While the general framework has been presented in this paper, further research and development is needed before any commercialization can take place.

Research is going on in co-operation with the Technical University of Denmark (DTU) and covers the following primary fields of interest:

- Developing a AT system efficiency model (mains to acoustic power)
- Transducer optimization / dedication for the AT concept.
- Transducer compensation algorithms for integration into the AT core (linearization, equalization, …)
- Digital interface solutions enabling true Digital Active Transducers (DATs)
- Single ended AT core topologies for further simplification.
- Intelligent supply control (as the PAWM method).
- ACAT implementation based on direct AC-AC conversion in the AT core.
- Optimal mechanical integration methods of the AT core.
- EMC characteristics of Active Transducers
- Miniaturization of the AT core electronics into silicon and substrate technology

Future papers on the result of this research are planned.

13. CONCLUSIONS

A holistic but far from complete introduction to a new audio power conversion topology and architecture, the Active Transducer, was given. The variants DC Active Transducer (DCAT) and the AC Active Transducer (ACAT) were introduced, and the DC Active Transducer was subjected to a more through introduction and analysis.

The primary objectives and motivation for the Active Transducer was outlined to be efficiency and performance improvements, combined with a remarkable reduction in overall topological complexity.

Permanent elimination of LC filtering is critical to reach the objectives of the Active Transducer. Pulse modulated drive of electro dynamic transducers was addressed and it was shown that intelligent pulse modulation methods are needed in order for the carrier generated high frequency losses to be neglible. Two methods were proposed, the SCOM algorithm for fixed DC supply and the PAWM method enable simple single ended output stages for further reduced system complexity.

Performance, efficiency and topological complexity were analyzed and a direct benchmarking with the closest competitor – the PMA + transducer combination, was given. On of the most powerful features of the AT topology and AT architecture will probably be the quite dramatic reductions in system complexity reducing system cost and volume and the overall simplified audio power conversion architecture, which should appeal to general audio designer, whether for consumer, professional, car or any other audio field.

Overall, presents results are very encouraging, and illustrates the potential to revolutionize audio system design, if the remaining technical difficulties and market conservatism can be overcome.

14. ACKNOWLEDGMENT

The research is carried out in a co-operation with the Technical University of Denmark (DTU), and Danish Sound Technology (DST). We would like to thank professor Michael A.E. Andersen for our fruitfull co-operation within audio power coversion research.

15. PATENT NOTE

The Active Transducer topology, including the mentioned algorithms and methods for power conversion, and the AT audio architectures presented in this paper is protected by a range of pending patents.

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