Active room compensation for sound reinforcement using sound field separation techniques

This work investigates how the sound field created by a sound reinforcement system can be controlled at low frequencies. An indoor control method is proposed which actively absorbs the sound incident on a reflecting boundary using an array of secondary sources. The sound field is separated into incident and reflected components by a microphone array close to the secondary sources, enabling the minimization of reflected components by means of optimal signals for the secondary sources. The method is purely feed-forward and assumes constant room conditions. Three different sound field separation techniques for the modeling of the reflections are investigated based on plane wave decomposition, equivalent sources, and the Spatial Fourier transform. Simulations and an experimental validation are presented, showing that the control method performs similarly well at enhancing low frequency responses with the three sound separation techniques. Resonances in the entire room are reduced, although the microphone array and secondary sources are confined to a small region close to the reflecting wall. Unlike previous control methods based on the creation of a plane wave sound field, the investigated method works in arbitrary room geometries and primary source positions.

General information
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology, d&b audiotechnik GmbH
Contributors: Heuchel, F. M., Fernandez Grande, E., Agerkvist, F. T., Shabalina, E.
Pages: 1346-1354
Publication date: 2018
Peer-reviewed: Yes
A Quiet Zone System, Optimized For Large Outdoor Events, Based on Multichannel FxLMS ANC
As part of the bigger EU project MONICA (Horizon2020) a local quiet zone system is being developed. This system provides a zone of quiet close to loud outdoor concerts in order to support communications or minimize the noise exposure of staff. Because the noise sources are the loudspeakers of the venue’s PA system an ideal reference signal can be obtained from the sound engineer’s mixing console, which can be used to apply methods of feedforward active noise control. This paper presents a real time application of the multichannel filtered reference least mean square algorithm (MCFxLMS), shows how it has been designed, implemented and tested under laboratory conditions.

General information
State: Published
Organisations: Acoustic Technology, Department of Electrical Engineering
Contributors: Plewe, D., Agerkvist, F. T., Brunskog, J.
Publication date: 2018

A Stepped Acoustic Transmission Line Model of Interference Tubes for Microphones
This paper presents an extension of the standing-wave model of interference tubes for microphones by Ono et al. The original model accounts for three acoustic parameters: tube length, tube radius, and constant acoustic conductance per unit length. Our extension allows a varying conductance per unit length along the side wall. The assumptions behind the extended model and its ability to predict the frequency response of interference tubes are validated through simulations and by fitting the model parameters to frequency response measurements of a tube with varying conductance per unit length, using two different mountings. Results suggest that a tube with varying conductance per unit length is most effective at attenuating the off-axis sound if the conductance per unit length is decreased towards the tail end of the tube.

General information
State: Published
Organisations: Acoustic Technology, Department of Electrical Engineering, EBB-consult, Aalborg University
Contributors: Bigoni, F., Agerkvist, F. T., Bøgh, E.
Publication date: 2018

Fractional Derivative Loudspeaker Models for Nonlinear Suspensions and Voice Coils
Moving-coil loudspeakers exhibit a number of linear effects, such as viscoelastic suspension creep and lossy inductance of the voice coil, which complicate their frequency response. Nonlinear models of the loudspeaker must include these effects in order to make accurate predictions for nonlinear compensation algorithms. While viscoelasticity and lossy inductance have been modeled using a variety of methods in the frequency domain, the discrete time-domain description using fractional order derivatives is both accurate and easily incorporated into existing nonlinear models. The influence of the fractional order derivative is demonstrated using a fractional order oscillator, resulting in a response that closely resembles viscoelastic suspension creep in a loudspeaker or an increase in displacement toward low frequencies. A full bandwidth loudspeaker model with a fractional order viscoelastic suspension and a fractional order lossy voice coil was used to fit measurement data from two loudspeakers. Further simulations with a nonlinear position-dependent suspension and a nonlinear position-dependent inductance were conducted, and this revealed unexpected frequency components due to the infinite memory of fractional derivatives.

General information
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology
Contributors: King, A. W., Agerkvist, F. T.
Pages: 525-536
Publication date: 2018
On the Interdependence of Loudspeaker Motor Nonlinearities

Two of the main nonlinearities in the electrodynamic loudspeaker are the position dependence of the force factor, $B_l$, and the voice coil inductance, $L_e$. Since they both are determined by the geometry of the motor structure, they cannot be independent. This paper investigates this dependence both analytically and via FEM simulations. Under certain simplifying assumptions the force factor can be shown to be proportional to the spatial derivative of the inductance. Using FEM simulations the implications of this relation is illustrated for drivers with more realistic geometry and material parameters.

General information
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology
Contributors: Agerkvist, F. T., Heuchel, F. M.
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Publisher: Audio Engineering Society
Electronic versions:
AES_2018_On_the_interdependence_of_loudspeaker_motor_nonlinearities.pdf
Source: PublicationPreSubmission
Source-ID: 160053849
Research output: Research - peer-review › Article in proceedings – Annual report year: 2018

Sound field control for reduction of noise from outdoor concerts

We investigate sound field control based on the concept of sound zones for the mitigation of low frequency noise from outdoor concerts to the surrounding area by adding secondary loudspeakers to the existing primary sound system. The filters for the secondary loudspeakers are the result of an optimization problem that minimizes the total sound pressure level of both primary and secondary loudspeakers in a sensitive area and the impact of the secondary loudspeakers on the audience area of the concert. We report results from three different experiments with increasing complexity and scale. The sound field control system was reducing the sound pressure level in the dark zone on average by 10 dB below 1 kHz in a small scale experiment in anechoic conditions, by up to 14 dB in a controlled large scale open-air experiment and by up to 6 dB at a pilot test at a music festival.

General information
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology
Contributors: Heuchel, F. M., Caviedes Nozal, D., Agerkvist, F. T., Brunskog, J.
Number of pages: 8
Publication date: 2018

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Publisher: Audio Engineering Society
Electronic versions:
AES_2018_Sound_field_control_for_reduction_of_noise_from_outdoor_concerts_1_.pdf
Source: PublicationPreSubmission
Source-ID: 160053900
Research output: Research - peer-review › Article in proceedings – Annual report year: 2018

An adaptive, data driven sound field control strategy for outdoor concerts

One challenge of outdoor concerts is to ensure adequate levels for the audience while avoiding disturbance of the surroundings. We outline the initial concept of a sound field control (SFC) system for tackling this issue using sound-zoning. The system uses Bayesian inference to update a sound propagation model. We present a simulation in which SFC and propagation model work together.
Position Dependence of Fractional Derivative Models for Loudspeaker Voice Coils with Lossy Inductance

Commonly used models of moving-coil loudspeaker voice coils, which include effects from eddy current losses, are either inaccurate or contain an abundance of parameters, and are difficult to extend to the nonlinear domain. On the contrary, fractional derivative models accurately describe the frequency and position dependence of the lossy inductance, with meaningful connections to the underlying physics, while keeping the number of parameters low. These fractional derivatives are also compatible with state-space polynomial methods of modeling nonlinear behavior. It is shown that the fractional order derivative approaches a value of 1, corresponding to an ideal inductance, when the voice coil is completely outside the magnetic system. Finally, the developed model reveals details about the effect of conductive voice coil formers.

Accelerometer Based Motional Feedback Integrated in a 2 3/4” Loudspeaker

It is a well known fact that loudspeakers produce distortion when they are driven into large diaphragm displacements. Various methods exist to reduce distortion using forward compensation and feedback methods. Acceleration based motional feedback is one of these methods and was already thoroughly described in the 1960s showing good results at low frequencies. In spite of this, the technique has mainly been used for closed box subwoofers to a limited extent. In this paper, design and experimental results for a 2 3/4” acceleration based motional feedback loudspeaker are shown to extend this feedback method to a small full range loudspeaker. Furthermore, the audio quality from the system with feedback is discussed based on measurements of harmonic distortion, intermodulation distortion and subjective evaluation.
Do wavelet filters provide more accurate estimates of reverberation times at low frequencies

It has been amply demonstrated in the literature that it is not possible to measure acoustic decays without significant errors for low BT values (narrow filters and low reverberation times). Recently, it has been shown how the main source of distortion in the time envelope of the acoustic decay is the frequency dependent group delay of the common implementations of the 1/3 and 1/1 octave filters. Some authors report good results using wavelet filter banks as an alternative to the usual filters. In this paper, a critical review of the performance of wavelet filter banks is undertaken. A filter bank using the continuous wavelet transform (CTW) has been implemented using a Morlet mother function. Although in general, the wavelet filter bank performs better than the usual filters, the influence of decaying modes outside the filter bandwidth on the measurements has been detected, leading to a biased estimation of the reverberation time in the frequency band of interest.

General information
State: Published
Organisations: Department of Acoustic Technology, Department of Electrical Engineering, Acoustic Technology, Microflown AVISA BV, University of Vigo
Contributors: Sobreira Seoane, M. A., Pérez Cabo, D., Agerkvist, F. T.
Pages: 1088-1096
Publication date: 2016

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ISBN (Electronic): 978-3-939296-11-9
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Source: FindIt
Source-ID: 2349144994
Research output: Research - peer-review › Article in proceedings – Annual report year: 2016

Force Factor Modulation in Electro Dynamic Loudspeakers

The relationship between the non-linear phenomenon of 'reluctance force' and the position dependency of the voice coil inductance was established in 1949 by Cunningham, who called it 'magnetic attraction force'. This paper revisits Cunningham’s analysis and expands it into a generalised form that includes the frequency dependency and applies to coils with non-inductive (lossy) blocked impedance. The paper also demonstrates that Cunningham’s force can be explained physically as a modulation of the force factor which again is directly linked to modulation of the flux of the coil. A verification based on both experiments and simulations is presented along discussions of the impact of force factor modulation for various motor topologies. Finally, it is shown that the popular L2R2 coil impedance model does not correctly predict the force unless the new analysis is applied.

General information
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology, Purifi, PointSource Acoustics
Contributors: Risbo, L., Agerkvist, F. T., Tinggaard, C., Halvorsen, M., Putzeys, B.
Number of pages: 10
Publication date: 2016

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Publisher: Audio Engineering Society
Source: PublicationPreSubmission
Source-ID: 127713032
Research output: Research - peer-review › Article in proceedings – Annual report year: 2016
Modelling the Perceptual Components of Loudspeaker Distortion

While non-linear distortion in loudspeakers decreases audio quality, the perceptual consequences can vary substantially. This paper investigates the metric $R_{nonlin}$ [1] which was developed to predict subjective measurements of sound quality in nonlinear systems. The generalisability of the metric in a practical setting was explored across a range of different loudspeakers and signals. Overall, the correlation of $R_{nonlin}$ predictions with subjective ratings was poor. Based on further investigation, an additional normalization step is proposed, which substantially improves the ability of $R_{nonlin}$ to predict the perceptual consequences of non-linear distortion.

General information
State: Published
Organisations: Department of Electrical Engineering, Hearing Systems, Acoustic Technology, DELTA Microelectronics
Contributors: Olsen, S. L., Agerkvist, F. T., MacDonald, E., Stegenborg-Andersen, T., Volk, C. P.
Number of pages: 9
Publication date: 2016

Design and Evaluation of Accelerometer based Motional Feedback

The electro dynamic loudspeaker is often referred to as the weakest link in the audio chain due to low efficiency and high distortion levels at low frequencies and high diaphragm excursion. Compensating for loudspeaker non-linearities using feedback or feedforward methods can improve the distortion and enable radical design changes in the loudspeaker which can lead to efficiency improvements. In combination this has motivated a revisit of the accelerometer based motional feedback technique. Experimental results on a 8 inch subwoofer show that the total harmonic distortion can be significantly reduced at low frequencies and large displacements.

General information
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology, Electronics, Technical University of Denmark
Contributors: Schneider, H., Pranjic, E., Agerkvist, F. T., Knott, A., Andersen, M. A. E.
Number of pages: 7
Publication date: 2015

Flux Modulation in the Electrodynamic Loudspeaker

This paper discusses the effect of flux modulation in the electrodynamic loudspeaker with main focus on the effect on the force factor. A measurement setup to measure the AC flux modulation with static voice coil is explained and the measurements shows good consistency with FEA simulations. Measurements of the generated AC flux modulation shows, that eddy currents are the main source to magnetic losses in form of phase lag and amplitude changes. Use of a copper cap shows a decrease in flux modulation amplitude at the expense of increased power losses. Finally, simulations show that there is a high dependency between the generated AC flux modulation from the voice coil and the AC force factor change.

General information
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology, PointSource Acoustics
Contributors: Halvorsen, M., Tinggaard, C., Agerkvist, F. T.
Number of pages: 10
Publication date: 2015
Improvements in Elimination of Loudspeaker Distortion in Acoustic Measurements

This paper investigates the influence of nonlinear components that contaminate the linear response of acoustic transducers, and presents improved methods for eliminating the influence of nonlinearities in acoustic measurements. The method is evaluated with pure sinusoidal signals as well as swept sine signal and is tested on models of memoryless nonlinear systems as well as nonlinear loudspeakers. The method is shown to give a clear benefit over existing methods. Two techniques that improve the signal-to-noise ratio are demonstrated: the first uses more measurement levels than the number of orders to be separated, whereas the other one is based on standard Tikhonov regularization. Both methods are shown to significantly improve the signal-to-noise ratio.

General information
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology, Hearing Systems
Contributors: Agerkvist, F. T., Torras Rosell, A., McWalter, R. I.
Number of pages: 10
Publication date: 2015

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Research output: Research - peer-review › Article in proceedings – Annual report year: 2015

Improving the efficiency of deconvolution algorithms for sound source localization

The localization of sound sources with delay-and-sum (DAS) beamforming is limited by a poor spatial resolution - particularly at low frequencies. Various methods based on deconvolution are examined to improve the resolution of the beamforming map, which can be modeled by a convolution of the unknown acoustic source distribution and the beamformer's response to a point source, i.e., point-spread function. A significant limitation of deconvolution is, however, an additional computational effort compared to beamforming. In this paper, computationally efficient deconvolution algorithms are examined with computer simulations and experimental data. Specifically, the deconvolution problem is solved with a fast gradient projection method called Fast Iterative Shrinkage-Thresholding Algorithm (FISTA), and compared with a Fourier-based non-negative least squares algorithm. The results indicate that FISTA tends to provide an improved spatial resolution and is up to 30% faster and more robust to noise. In the spirit of reproducible research, the source code is available online.

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BFI (2019): BFI-level 2
Web of Science (2019): Indexed yes
BFI (2018): BFI-level 2
Web of Science (2018): Indexed yes
BFI (2017): BFI-level 2
Scopus rating (2017): CiteScore 1.77 SJR 0.695 SNIP 1.224
Web of Science (2017): Impact factor 1.605
Web of Science (2017): Indexed yes
Investigation of Current Driven Loudspeakers

Current driven loudspeakers have previously been investigated but the literature is limited and the advantages and disadvantages are yet to be fully identified. This paper makes use of a non-linear loudspeaker model to analyse loudspeakers with distinct non-linear characteristics under voltage and current drive. A multi tone test signal is used in the evaluation of the driving schemes since it resembles audio signals to a higher degree than the signals used in total harmonic distortion and intermodulation distortion test methods. It is found that current drive is superior over voltage drive in a 5” woofer where a copper ring in the pole piece has not been implemented to compensate for eddy currents. However the drive method seems to be irrelevant for a 5” woofer where the compliance, force factor as well as the voice coil inductance has been optimized for linearity.

Modeling of Lossy Inductance in Moving-Coil Loudspeakers

The electrical impedance of moving-coil loudspeakers is dominated by the lossy inductance in high frequency range. Using the equivalent electrical circuit method, a new model for the lossy inductance based on separate functions for the magnitude and phase of the impedance is presented. The electrical impedances of five loudspeakers were measured by Klippel LPM analyzer, and the model parameters were identified by fitting the measured impedance curve. The obtained accuracy was evaluated with respect to the simplicity of different models. The results show that, this new model agrees well with the measured electrical impedance, and gives an accurate prediction of the lossy inductance varying with frequencies, especially for the frequency dependent phase. Additionally, there are just three parameters in this new model, which gives simple and rapid parameter identification.
BFI (2018): BFI-level 2
Web of Science (2018): Indexed yes
BFI (2017): BFI-level 2
Scopus rating (2017): CiteScore 1.28 SJR 0.472 SNIP 0.982
Web of Science (2017): Impact factor 1.129
Web of Science (2017): Indexed yes
BFI (2016): BFI-level 2
Scopus rating (2016): CiteScore 1.12 SJR 0.487 SNIP 0.889
Web of Science (2016): Impact factor 1.119
Web of Science (2016): Indexed yes
BFI (2015): BFI-level 2
Scopus rating (2015): CiteScore 1.11 SJR 0.579 SNIP 1.097
Web of Science (2015): Impact factor 0.897
Web of Science (2015): Indexed yes
BFI (2014): BFI-level 2
Scopus rating (2014): CiteScore 0.89 SJR 0.581 SNIP 1.052
Web of Science (2014): Impact factor 0.783
Web of Science (2014): Indexed yes
BFI (2013): BFI-level 2
Scopus rating (2013): CiteScore 1.05 SJR 0.573 SNIP 1.574
Web of Science (2013): Impact factor 0.679
ISI indexed (2013): ISI indexed yes
Web of Science (2013): Indexed yes
BFI (2012): BFI-level 2
Scopus rating (2012): CiteScore 0.81 SJR 0.557 SNIP 0.986
Web of Science (2012): Impact factor 0.714
ISI indexed (2012): ISI indexed yes
Web of Science (2012): Indexed yes
BFI (2011): BFI-level 2
Scopus rating (2011): CiteScore 0.65 SJR 0.46 SNIP 0.913
Web of Science (2011): Impact factor 0.569
ISI indexed (2011): ISI indexed yes
Web of Science (2011): Indexed yes
BFI (2010): BFI-level 2
Scopus rating (2010): SJR 0.481 SNIP 0.916
Web of Science (2010): Impact factor 0.552
Web of Science (2010): Indexed yes
BFI (2009): BFI-level 1
Scopus rating (2009): SJR 0.536 SNIP 0.769
BFI (2008): BFI-level 1
Scopus rating (2008): SJR 0.596 SNIP 0.962
Web of Science (2008): Indexed yes
Scopus rating (2007): SJR 0.49 SNIP 0.926
Web of Science (2007): Indexed yes
Scopus rating (2006): SJR 0.288 SNIP 0.871
Scopus rating (2005): SJR 0.253 SNIP 0.844
Web of Science (2005): Indexed yes
Scopus rating (2004): SJR 0.24 SNIP 0.645
Web of Science (2004): Indexed yes
Scopus rating (2003): SJR 0.344 SNIP 0.859
Web of Science (2003): Indexed yes
Scopus rating (2002): SJR 0.817 SNIP 0.883
Web of Science (2002): Indexed yes
Reproduction of nearby sources by imposing true interaural differences on a sound field control approach

In anechoic conditions, the Interaural Level Difference (ILD) is the most significant auditory cue to judge the distance to a sound source located within 1 m of the listener's head. This is due to the unique characteristics of a point source in its near field, which result in exceptionally high, distance dependent ILDs. When reproducing the sound field of sources located near the head with line or circular arrays of loudspeakers, the reproduced ILDs are generally lower than expected, due to physical limitations. This study presents an approach that combines a sound field reproduction method, known as Pressure Matching (PM), and a binaural control technique. While PM aims at reproducing the incident sound field, the objective of the binaural control technique is to ensure a correct reproduction of interaural differences. The combination of these two approaches gives rise to the following features: (i) an accurate reproduction of ILDs is achieved at the head positions considered by the method, (ii) the ILD variations in the vicinity of those positions are smoothed, thus lowering the ILD error, and (iii) the true wavefront is preserved. Given the properties of the presented method, intended distance and directional perception is expected.
Shift of the Acoustic Center of a Closed-Box Loudspeaker in a Linear Array: Investigation Using the Beamforming Technique

The center of the spherical waves radiated from a loudspeaker is defined as its acoustic center. This study aims to investigate how the acoustic center of a closed-box loudspeaker is shifted when the loudspeaker is placed in a linear array. That is, the acoustic center of the loudspeaker is estimated when the loudspeaker is placed alone and then the loudspeaker is placed in a linear array composed of two or three identical loudspeakers. The acoustic center of each loudspeaker in the linear arrays is estimated with the other loudspeakers turned off and compared with that in the single
louderspeaker case. In order to estimate the acoustic center based on the wave fronts, a method is proposed that measures sound pressure around the loudspeaker with an array of microphones and uses the beamforming method for the reduction of the effect of the experimental errors. Experimental results show that the acoustic center is shifted differently depending on the relative position of the loudspeaker in the array. This implies that the performance of sound field control with a linear array of loudspeakers can be improved by taking the shift of the acoustic center into account.

**General information**

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Organisations: Department of Electrical Engineering, Acoustic Technology, Technical University of Denmark
Contributors: Chang, J., Jensen, J., Agerkvist, F. T.
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**Publication information**

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Ratings:
BFI (2019): BFI-level 1
Web of Science (2019): Indexed yes
BFI (2018): BFI-level 1
Web of Science (2018): Indexed yes
BFI (2017): BFI-level 1
Scopus rating (2017): SJR 0.265 SNIP 0.853
Web of Science (2017): Impact factor 0.774
Web of Science (2017): Indexed yes
BFI (2016): BFI-level 1
Scopus rating (2016): CiteScore 0.95 SJR 0.306 SNIP 0.934
Web of Science (2016): Impact factor 0.707
Web of Science (2016): Indexed yes
BFI (2015): BFI-level 1
Scopus rating (2015): CiteScore 1.11 SJR 0.408 SNIP 1.457
Web of Science (2015): Impact factor 0.856
Web of Science (2015): Indexed yes
BFI (2014): BFI-level 1
Scopus rating (2014): CiteScore 1.05 SJR 0.592 SNIP 1.399
Web of Science (2014): Impact factor 1.123
Web of Science (2014): Indexed yes
BFI (2013): BFI-level 1
Scopus rating (2013): CiteScore 1.35 SJR 0.441 SNIP 1.973
ISI indexed (2013): ISI indexed yes
BFI (2012): BFI-level 1
Scopus rating (2012): CiteScore 0.68 SJR 0.411 SNIP 1.15
Web of Science (2012): Impact factor 0.831
ISI indexed (2012): ISI indexed yes
BFI (2011): BFI-level 1
Scopus rating (2011): CiteScore 0.58 SJR 0.409 SNIP 1.039
Web of Science (2011): Impact factor 0.432
ISI indexed (2011): ISI indexed yes
Web of Science (2011): Indexed yes
BFI (2010): BFI-level 1
Scopus rating (2010): SJR 0.32 SNIP 1.101
Web of Science (2010): Impact factor 0.483
Web of Science (2010): Indexed yes
BFI (2009): BFI-level 1
State-Space Modelling of Loudspeakers using Fractional Derivatives

This work investigates the use of fractional order derivatives in modeling moving-coil loudspeakers. A fractional order state-space solution is developed, leading the way towards incorporating nonlinearities into a fractional order system. The method is used to calculate the response of a fractional harmonic oscillator, representing the mechanical part of a loudspeaker, showing the effect of the fractional derivative and its relationship to viscoelasticity. Finally, a loudspeaker model with a fractional order viscoelastic suspension and fractional order voice coil is fit to measurement data. It is shown that the identified parameters can be used in a linear fractional order state-space model to simulate the loudspeakers' time domain response.

Compensation of the flux modulation distortion using an additional coil in a loudspeaker unit

Flux modulation is one of the main causes of distortion in electrodynamic loudspeaker units. A new compensation technique that eliminates this type of non-linearity using an additional compensation coil in the speaker unit is presented. An equivalent circuit model of the device including the compensation coil is derived. The compensation technique consists on feeding the compensation coil and voice coil with filtered versions of the wanted audio signal. Simulations show that a significant reduction in flux modulation distortion can be achieved with this technique. A simple magnetic circuit has been constructed to test the method on a real device, and the measurements show the method works, also when eddy currents are present.
Compensation of the flux modulation distortion using an additional coil in a loudspeaker unit.
Flux modulation is one of the main causes of distortion in electrodynamic loudspeaker units. A new compensation technique that eliminates this type of non-linearity using an additional compensation coil in the speaker unit is presented. An equivalent circuit model of the device including the compensation coil is derived. The compensation technique consists on feeding the compensation coil and voice coil with filtered versions of the wanted audio signal. Simulations show that a significant reduction in flux modulation distortion can be achieved with this technique. A simple magnetic circuit has been constructed to test the method on a real device, and the measurements show the method works, also when eddy currents are present.

Eigenbeamforming array systems for sound source localization
Microphone array technology has been widely used for the localization of sound sources. In particular, beamforming is a well-established signal processing method that maps the position of acoustic sources by steering the array transducers toward different directions electronically. The present PhD study aims at enhancing the performance of uniform circular arrays, and to a lesser extent, spherical arrays, for two- and three-dimensional localization problems, respectively. These array geometries allow to perform eigenbeamforming, beamforming based on the decomposition of the sound field in a series of orthogonal functions. In this work, eigenbeamforming is particularly developed to improve the performance of circular arrays at low frequencies. Compared to conventional delay- and-sum beamforming, the proposed technique, named circular harmonics beamforming, provides a better resolution at the expense of being more vulnerable to noise. A simple way to further improve the array performance is to flush-mount the transducers on a rigid scatterer. For a circular array, an ideal solution is a rigid cylindrical scatterer of infinite length. Due to its impracticality, the use of a rigid spherical scatterer is recommended instead. A better visualization in the entire frequency range can be achieved with deconvolution methods, as they allow the recovery of the sound source distribution from a given beamformed map. Three efficient methods based on spectral procedures, originally conceived for planar-sparse arrays, are adapted to circular arrays. They rely on the fact that uniform circular arrays present an azimuthal response that is rather independent on the focusing direction. Finally, a method based on the combination of beamforming and acoustic holography is introduced for both circular and spherical arrays. This new approach, also expressible in terms of eigenbeamforming, extends the frequency range of operation of conventional delay-and-sum beamforming toward the low frequencies.
Eliminating transducer distortion in acoustic measurements
This paper investigates the influence of nonlinear components that contaminate the linear response of acoustic transducer, and presents a method for eliminating the influence of nonlinearities in acoustic measurements. The method is evaluated on simulated as well as experimental data, and is shown to perform well even in noisy conditions. The limitations of the Total Harmonic Distortion, THD, measure is discussed and a new distortion measure, Total Distortion Ratio, TDR, which more accurately describes the amount of nonlinear power in the measured signal, is proposed.

General information
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology, Hearing Systems
Contributors: Agerkvist, F. T., Torras Rosell, A., McWalter, R. I.
Pages: 824-833
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Publisher: Audio Engineering Society
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Research output: Research - peer-review › Article in proceedings – Annual report year: 2014

Enhancing the beamforming map of spherical arrays at low frequencies using acoustic holography
Recent studies have shown that the localization of acoustic sources based on circular arrays can be improved at low frequencies by combining beamforming with acoustic holography. This paper extends this technique to the three dimensional case by making use of spherical arrays. The pressure captured by a rigid spherical array under free-field conditions is used to compute the expected pressure on a virtual and larger sphere by means of acoustic holography. Beamforming is then applied with the pressure predicted at the virtual array. Since the virtual array has a larger radius compared to the one of the physical array, the low frequencies (the ones with larger wavelength) are better captured by the virtual array, and therefore, the performance of the resulting beamforming system is expected to improve at these frequencies. The proposed method is examined with simulations based on delay-and-sum beamforming. In addition, the principle is validated with experiments.

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Organisations: Department of Electrical Engineering, Acoustic Technology
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Electronic versions: BeBeC2014-03.pdf
Source: dtu
Source-ID: u::10656
Research output: Research - peer-review › Article in proceedings – Annual report year: 2014

Model Based Beamforming and Bayesian Inversion Signal Processing Methods for Seismic Localization of Underground Source
This PhD study examines the use of seismic technology for the problem of detecting underground facilities, whereby a seismic source such as a sledgehammer is used to generate seismic waves through the ground, sensed by an array of seismic sensors on the ground surface, and recorded by the digital device. The concept is similar to the techniques used in exploration seismology, in which explosions (that occur at or below the surface) or vibration wave-fronts generated at the surface reflect and refract off structures at the ground depth, so as to generate the ground profile of the elastic material properties such as the elastic wave speeds and soil densities. One processing method is casting the estimation problem into an inverse problem to solve for the unknown material parameters. The forward model for the seismic signals used in
the literatures include ray tracing methods that consider only the first arrivals of the reflected compressional P-waves from the subsurface structures, or 3D elastic wave models that model all the seismic wave components. The ray tracing forward model formulation is linear, whereas the full 3D elastic wave model leads to a nonlinear inversion problem.

In this PhD study, both the linear and nonlinear inverse problems are investigated, in order to solve the problem to locate the position of an underground tunnel. One practical limitation of geophysics inversion problem is the high dimension of the unknown parameter space, such as the elastic wave speeds, soil density values of the discretized ground medium, which leads to time-consuming computations and instability behaviour of the inversion process. In addition, the geophysics inverse problem is generally ill-posed due to non-exact forward model that introduces errors. The Bayesian inversion method through the probability density function permits the incorporation of a priori information about the parameters, and also allow for incorporation of theoretical errors. This opens up the possibilities of application of inverse paradigm in the real-world geophysics inversion problems.

In this PhD study, the Bayesian inversion paradigm for the tunnel localization problem was investigated. A formulation of the mathematical framework of the inverse problem to solve the specific tunnel localization problem defined in the PhD study has been proposed. On this basis, two optimization algorithms, namely the Monte Carlo Metropolis Hasting and Simulated Annealing have been studied, and a new reduced modelling scheme to reduce the dimension of the ground material elastic parameter space has been proposed. Also, the linear ray tracing and nonlinear 3D elastic wave models have been examined using the Bayesian inversion algorithms and conventional source localization beamforming algorithms. Additionally, an experiment validation of the inversion framework is performed through conducting seismic measurements at an underground tunnel site using an array of geophones deployed on the ground surface and using a surface seismic source.

The examples show with the field data, inversion for localization is most advantageous when the forward model completely describe all the elastic wave components as is the case of the FDTD 3D elastic model. The simulation results of the inversion of the soil density values show that both the global optimization method, i.e., Monte Carlo Metropolis Hasting algorithm and Simulated Annealing, are able to provide fairly good estimates which agree with the investigations in the literatures that focus only on geo-inversion of the elastic medium. The results of Monte Carlo Metropolis Hasting inversion to solve the source localization problem, i.e., invert for source depth and source range, display large fluctuations in the range and depth samples generated. However the point MAP estimates derived from 5000 runs of the Metropolis Hasting method are relatively close to the true values. The results of the Simulated Annealing using an initial guess as the MAP estimate calculated from a small number of runs of the Monte Carlo Metropolis Hasting algorithm (in the simulation, we use 50 runs), is able to improve the accuracy of the range and depth estimate of the source. The field results of the joint inversion of material elastic parameters and tunnel location show an agreement with the simulated results. The PDF curves of range and depth derived from Monte Carlo Metropolis Hasting samples shows multi-modal distribution behaviour, which made the mean estimate not a suitable parameter for processing the Monte Carlo samples. The MAP estimates derived from both the Monte Carlo Metropolis Hasting and Simulated Annealing methods however match well against the location of the underground tunnel. These results reflect that the point MAP estimate, in agreement with the simulation results, provides a more accurate representation for the location parameters exhibiting multi-modal distribution behaviour.

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New measurements techniques: Optical methods for characterizing sound fields
Acoustic measurements are traditionally based on transducers, and in particular, the most advanced measurement techniques are nowadays based on transducer arrays. This poses a fundamental problem, namely the influence of the transducer itself on the actual properties of sound when the transducer is immersed into the sound field. Typically, this influence is assumed to be negligible when the size of the transducer is small compared to the wavelength of the sound wave, or is rendered negligible by using a transducer-based correction that depends on the frequency. Either solution introduces additional uncertainties to the measurement process. Optical techniques may help overcoming this problem because the sensing element is not a bulky instrument, but a beam of light that does not change the properties of sound. Optical methods are thus non-invasive and can thereby enhance the current state of the art in the measurement of sound. The present PhD study primarily examines the use of the acousto-optic effect, that is, the interaction between sound and light, as a means to characterize acoustic fields. The acousto-optic measuring principle does not provide a direct measure of the pressure, but the integral of the pressure encountered by the ‘sensing’ light when traveling through the acoustic field. Far from being a limitation, this integral principle is exploited for sound field visualization using tomography. The most innovative contribution of this PhD project is the applicability of the acousto-optic measuring principle to acoustic
holography and beamforming. On the one hand, a new method called near-field acousto-optic holography (NAOH) has been proposed and makes it possible to predict properties of sound at planes different from the measuring one. In comparison with conventional near-field acoustic holography (NAH), the suggested holographic method features novel spectral properties in the wavenumber domain. On the other hand, an acousto-optic beamformer has been designed and validated experimentally for the localization of sound sources located in the far field. In this case, a laser beam is interpreted as a line array of microphones with infinite resolution, which makes the proposed acousto-optic beamformer immune to spatial aliasing.

In addition, the present PhD study investigates the applicability of photon correlation spectroscopy as a primary method for microphone calibration under free-field conditions. Various signal processing refinements are proposed to improve the accuracy of this measurement technique.

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Nonlinear Distortion Mechanisms and Efficiency of Balanced-Armature Loudspeakers
Nonlinear distortion added by the loudspeaker (often referred to as a receiver) in a hearing aid reduces the signal-to-noise ratio in the acoustic output and may degrade the user’s ability to understand speech. The balanced-armature-type loudspeakers predominantly used in hearing aids are inherently nonlinear devices, since any displacement of the loudspeaker diaphragm inevitably changes the magnetic and electrical characteristics of the loudspeaker. Additionally, for the balanced-armature loudspeaker the signal has to be transmitted through the magnetic domain (as a magnetic B-field) and the linearity of the magnetic material is therefore of great importance. This thesis describes the inherent nonlinear parameters of the balanced-armature loudspeaker and demonstrates how the nonlinearity of these parameters may be reduced by design. A simple technique for incorporating magnetic leakage effects is introduced and it is shown how the leakage affects the linearity of the loudspeaker. Magnetic hysteresis, saturation, and eddy current losses and how these effects might affect the performance of the loudspeaker are also discussed. FEM simulation software is used to investigate magnetic effects and to validate simpler equivalent circuit models. A large scale model of a balanced-armature loudspeaker has been developed and its inherent nonlinear parameters have been measured and compared to the theoretically predicted values. A measurement setup for determining the magnetic properties of soft magnetic materials has also been developed, since it is of great importance to understand what kind of linear and nonlinear transformations the magnetic materials impose on the signal. In hearing aid applications the power efficiency of the loudspeaker is important because every reduction in power consumption will help prolong battery life and thereby reduce the frequency of necessary service checks. A great deal of the power consumed in a hearing aid goes into the amplifier that drives the loudspeaker. If the efficiency of the balanced-armature loudspeaker can be improved, the operation time of the hearing aid may be extended or the size of the hearing aid could be reduced using a smaller battery, or new features and more advanced algorithms could be embedded without compromising the operation time of the hearing aid. A new loudspeaker efficiency performance metric is proposed and it is shown how the balanced-armature loudspeaker may be optimized in terms of this. The maximum level of the acoustic output of a balanced-armature loudspeaker is an important parameter since these miniature loudspeakers sometimes need to be capable of compensating for substantial hearing losses. It is demonstrated that magnetic saturation of the loudspeaker armature is likely to be the most significant cause of compression in the balanced-armature loudspeaker. It is furthermore shown which conditions should be fulfilled in order to reduce the risk of armature saturation and thereby increase the maximum output and reduce distortion.

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Arrangements of a pair of loudspeakers for sound field control with double-layer arrays
Recent studies have attempted to control sound fields, and also to reduce room reflections with a circular or spherical array of loudspeakers. One of the attempts was to suppress sound waves propagating to the walls outside the array with a circular double-layer array of loudspeakers. The double-layer array represents a set of a monopole and a dipole in the Kirchhoff-Helmholtz integral equation, and thus the distance between these layers should be short compared with the wavelength. In practice, however, this condition is occasionally hard to satisfy because of the sizes of loudspeaker cabinets. In order to solve this problem, this study aims to examine several arrangements of a pair of loudspeakers that has a short distance between the acoustic centres. The effect of diffraction of sound waves due to the enclosure of another loudspeaker is investigated in simulations in terms of the position of the acoustic centre. As a result, it is shown that a loudspeaker has an approximately omni-directional radiation pattern at low frequencies in spite of the other loudspeaker cabinet, but the acoustic centre is shifted to the opposite direction of the cabinets.
A comparison of two strategies for generating sound zones in a room.
For some purposes it may be of interest to generate sound zones with different acoustic properties in a room. This paper compares two strategies for generating such zones. One method is based on 'contrast optimisation': the idea is to maximise the ratio of the potential energy in a 'bright' (ensonified) zone to the potential energy in a 'dark' (quiet) zone with a given source configuration. An alternative method based on 'sound field synthesis' has the more ambitious goal to control the sound field in the bright zone in detail, for example, to generate a plane wave that propagates in a certain direction. The two methods are analysed theoretically and examined through simulations and experimentally.

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Adaptive Feedback Cancellation With Band-Limited LPC Vocoder in Digital Hearing Aids
Feedback oscillation is one of the major issues with hearing aids. An effective way of feedback suppression is adaptive feedback cancellation, which uses an adaptive filter to estimate the feedback path. However, when the external input signal is correlated with the receiver input signal, the estimate of the feedback path is biased. This so-called “bias problem” results in a large modeling error and a cancellation of the desired signal. This paper proposes a band-limited linear predictive coding based approach to reduce the bias. The idea is to replace the hearing-aid output with a synthesized signal, which sounds perceptually the same as or similar to the original signal but is statistically uncorrelated with the external input signal at high frequencies where feedback oscillation usually occurs. Simulation results show that the proposed algorithm can effectively reduce the bias and the misalignment between the real and the estimated feedback path. When combined with filtered-X adaptation in the feedback canceller, this approach reduces the misalignment even further.

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Scopus rating (2010): SJR 1.127 SNIP 2.502
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Extracting the invariant model from the feedback paths of digital hearing aids
Feedback whistling is a severe problem with hearing aids. A typical acoustical feedback path represents a wave
propagation path from the receiver to the microphone and includes many complicated effects among which some are
invariant or nearly invariant for all users and in all acoustical environments given a specific type of hearing aids. Based on
this observation, a feedback path model that consists of an invariant model and a variant model is proposed. A common-
acoustical-pole and zero model-based approach and an iterative least-square search-based approach are used to extract
the invariant model from a set of impulse responses of the feedback paths. A hybrid approach combining the two methods
is also proposed. The general properties of the three methods are studied using artificial datasets, and the methods are
cross-validated using the measured feedback paths. The results show that the proposed hybrid method gives the best
overall performance, and the extracted invariant model is effective in modeling the feedback path.
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Generation of sound zones in 2.5 dimensions
A method for generating sound zones with different acoustic properties in a room is presented. The method is an extension of the two-dimensional multi-zone sound field synthesis technique recently developed by Wu and Abhayapala; the goal is, for example, to generate a plane wave that propagates in a certain direction within a certain region of a room and at the same time suppress sound in another region. The method is examined through simulations and experiments. For comparison a simpler method based on the idea of maximising the ratio of the potential acoustic energy in an ensonified zone to the potential acoustic energy in a quiet zone is also examined.

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Nonlinear time-domain modeling of balanced-armature receivers
Nonlinear distortion added by the loudspeaker in a hearing aid lowers the signal-to-noise ratio and may degrade the hearing aid user's ability to understand speech. The balanced-armature-type loudspeakers, predominantly used in hearing aids, are inherently nonlinear devices, as any displacement of the loudspeaker diaphragm inevitably changes the magnetic and electrical characteristics of the loudspeaker. A numerical time-domain model capable of describing these nonlinearities is presented. By simulation it is demonstrated how the output distortion could potentially be reduced significantly through careful design of the mechanical properties of the armature.
Non linear viscoelastic models

Viscoelastic effects are often present in loudspeaker suspensions, this can be seen in the displacement transfer function which often shows a frequency dependent value below the resonance frequency. In this paper nonlinear versions of the standard linear solid model (SLS) are investigated. The simulations show that the nonlinear version of the Maxwell SLS model can result in a time dependent small signal stiffness while the Kelvin Voight version does not.

The effect of vocal tract impedance on the vocal folds

The importance of the interaction between the acoustic impedance of the vocal tract with the flow across the vocal cords is well established. In this paper we are investigating the changes in vocal tract impedance when using the different modes of phonation according to Sadolin [1], going from the soft levels of the Neutral mode to the high levels of the fully 'metallic' Edge mode. The acoustic impedance of vocal tract as seen from the mouth opening is measured via a microphone placed close to the mouth when exciting the system with a volume velocity source [2]. At the same time a Laryngograph frontend is used to measure the electrotototograph signal which reflects the opening and closing pattern of the vocal folds. The measurements were carried out for all four modes (Neutral, Curbing, Overdrive and Edge) for the vowel [a] in three different pitches: C3 (131 Hz), G3 (196 Hz) and C4 (262 Hz). The results show changes in the resonance frequencies of the vocal tract with increasing pitch, whereas the changes between the modes are less clear due to the measurement signal being weak in comparison to the louder modes, especially at high pitches. The electrototograph shows a very different waveform for the Neutral mode compared to the other, so-called metallic modes. The differences in waveform between Curbing, Overdrive and Edge modes are minor. However, the spectrum of the Overdrive mode shows stronger 2nd harmonic and weaker 4th and 6th harmonic compared to Curbing and Edge. Finally the Overdrive mode, which is the mode that is most limited in pitch range, was tested at its pitch limit C5 (523 Hz) under normal conditions and when the singer has inhaled Helium. When inhaling Helium the acoustic impedance of the vocal tract is reduced in magnitude and the resonances are scaled upwards in frequency due to different density and speed of sound in Helium. The electrototograph shows a change in waveform when the singer inhales helium. The percentage of the glottal cycle when the vocal cords are open, the so-called open quotient, increases from 40 to 55%. When inhaling helium the male singer was able reach Eb5, a minor third over the normal limit for males, this seems to indicate that the vocal tract impedance is at least partially responsible for the pitch limit in ‘Overdrive’.
Circular Loudspeaker Array with Controllable Directivity

Specific directivity patterns for circular arrays of loudspeakers can be achieved by utilizing the concept of phase-modes, which expands the directivity pattern into a series of circular harmonics. This paper investigates the applicability of this concept applied on a loudspeaker array on a cylindrical baffle, with a desired directivity pattern, which is specified in the frequency interval of 400–5000 Hz. From the specified frequency independent directivity pattern filter transfer functions for each loudspeaker are determined. The sensitivity of various parameters, related to practical implementation, is also investigated by introducing filter errors for each array element. Measurements on a small scale model using 2 inch drivers are compared with simulations, showing good agreement between experimental and predicted results.

Frequency Dependence of Damping and Compliance in Loudspeaker Suspensions

A loudspeakers suspension is commonly not, even by small signals, to be regarded as a simple spring following Hooke's law – as otherwise presumed by traditional theory. Different types of polymers (rubber or plastic, either vulcanized, foamed or TPE) used for surrounds - besides impregnated textiles used for spiders - have more or less visco-elastic properties; best known is the "creep" effect. This phenomenon in itself is normally of little interest in the audio frequency range. It is mainly a DC phenomenon. As such it manifests itself when a static (DC) force probes the speaker voice coil. At first, it moves fast very close to the expected position, but then it "creeps" slowly a little further (see [1, Fig.1]) A widely accepted model giving account for the visco-elastic effects in loudspeaker surrounds is the so called LOG-model evaluated by M.H. Knudsen and J. G. Jensen [1]. It is an empirical model mathematically describing the effects of visco-elasticity in loudspeaker suspensions. The evaluation is to a high degree based on test loudspeakers with rubber surrounds with a high content of plasticizer combining high compliance and high damping. This is very effective to reduce rim resonances, but less used in high quality loudspeakers today – where "Low Loss Rubber Surround" is currently seen as a marketing feature, as it is expected to have positive impact on sound quality. The plasticized type of surround shows significant creep, followed by compliance and damping increasing towards lower frequencies. The LOG-model is found to give good agreement with measurements, also for loudspeakers with low loss surrounds. However, it is not supported by a theory explaining visco-elastic properties in a physical way. Surrounds today are mostly made from SBR rubber for which - with the additives normally used to adjust stiffness and damping - neither frequency dependency of compliance nor creep are significant problems. Despite this, experience shows that frequency dependent mechanical damping nevertheless might be present. In this paper some modifications to the LOG-model are proposed, and a suggestion is given to explain the phenomenon physically. Frequency dependency of damping - despite lack of other visco-elastic characteristics - is explained as a possible effect of mechanical hysteresis. To derive reliable parameters a new method named "Advanced Added Mass" is evaluated and a new "Hysteresis-model" proposed. Different 2 equivalent circuit models for frequency dependency of compliance and damping are tested in order to find the best usable tool for the loudspeaker engineer.
Modelling Viscoelasticity of Loudspeaker Suspensions using Retardation Spectra

It is well known that, due to viscoelastic effects in the suspension, the displacement of the loudspeaker increases with decreasing frequency below the resonance. Present creep models are either not precise enough or purely empirical and not derived from the basis of physics. In this investigation, the viscoelastic retardation spectrum, which provides a more fundamental description of the suspension viscoelasticity, is first used to explain the accuracy of the empirical LOG creep model (Knudsen et al.). Then, two extensions to the LOG model are proposed which include the low and high frequency limit of the compliance, not accounted for in the original LOG model. The new creep models are verified by measurements on two 5.5 loudspeakers with different surrounds.

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Noise injection for feedback cancellation with linear prediction

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Using a reflection model for modeling the dynamic feedback path of digital hearing aids

Feedback whistling is one of the severe problems with hearing aids, especially in dynamic situations when the users hug, pick up a telephone, etc. This paper investigates the properties of the dynamic feedback paths of digital hearing aids and proposes a model based on a reflection assumption. The model is compared with two existing models: a direct model and an initialization model, using the measured dynamic feedback paths. The comparison shows that the proposed approach is able to model the dynamic feedback paths more efficiently and accurately in terms of mean-square error and maximum stable gain. The method is also extended to dual-microphone hearing aids to assess the possibility of relating the two dynamic feedback paths through the reflection model. However, it is found that in a complicated acoustic environment, the relation between the two feedback paths can be very intricate and difficult to exploit to yield better modeling of the dynamic feedback paths. (C) 2010 Acoustical Society of America. [DOI: 10.1121/1.3290989]

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A new approach for modelling the dynamic feedback path of digital hearing aids

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Concha headphones and their coupling to the ear
The purpose of the study is to obtain a better understanding of concha headphone. Concha headphones are the small types of earpiece that are placed in the concha. They are not sealed to the ear and therefore, there is a leak between the earpiece and the ear. This leak is the reason why there is a significant lack of bass when using such headphones. This

Volterra Series Based Distortion Effect
A large part of the characteristic sound of the electric guitar comes from nonlinearities in the signal path. Such nonlinearities may come from the input- or output-stage of the amplifier, which is often equipped with vacuum tubes or a dedicated distortion pedal. In this paper the Volterra series expansion for nonlinear systems is investigated with respect to generating good distortion. The Volterra series allows for unlimited adjustment of the level and frequency dependency of each distortion component. Subjectively relevant ways of linking the different orders are discussed.

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Contributors: Agerkvist, F. T.
Number of pages: 7
Publication date: 2010

Host publication information
Title of host publication: Proceedings of the 129. Convention of the Audio Engineering Society
Source: orbit
Source-ID: 259658
Research output: Research - peer-review › Journal article – Annual report year: 2010

Bibliographical note
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Source: orbit
Source-ID: 266841
Research output: Research › Article in proceedings – Annual report year: 2010
paper investigates the coupling between the headphone and the ear, by means of measurement in artificial ears and models. The influence of the back volume is taken into account.

**General information**
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology
Contributors: Blanchard, L. J. K. O., Agerkvist, F. T.
Publication date: 2009

**Host publication information**
Title of host publication: Audio Engineering Society Convention Papers: Papers 7651-7819
ISBN (Print): 978-0-937803-67-7
Source: orbit
Source-ID: 251393
Research output: Research - peer-review › Article in proceedings – Annual report year: 2009

**Time varying behavior of the loudspeaker suspension: Displacement level dependency**

**General information**
State: Published
Organisations: Acoustic Technology, Department of Electrical Engineering, Aalborg University
Contributors: Agerkvist, F. T., Pedersen, B. R.
Publication date: 2009

**Host publication information**
Title of host publication: Audio Engineering Society Convention Papers
Volume: 127th
ISBN (Print): 978-0-937803-71-4
Source: orbit
Source-ID: 250407
Research output: Research - peer-review › Article in proceedings – Annual report year: 2009

**A Study of the Creep Effect in Loudspeaker Suspension**
This paper investigates the creep effect, the visco elastic behaviour of loudspeaker suspension parts, which can be observed as an increase in displacement far below the resonance frequency. The creep effect means that the suspension cannot be modelled as a simple spring. The need for an accurate creep model is even larger as the validity of loudspeaker models are now sought extended far into the nonlinear domain of the loudspeaker. Different creep models are investigated and implemented both in simple lumped parameter models as well as time domain non-linear models, the simulation results are compared with a series of measurements on three version of the same loudspeaker with different thickness and rubber type used in the surround.

**General information**
State: Published
Organisations: Acoustic Technology, Department of Electrical Engineering, Tymphany A/S
Contributors: Agerkvist, F. T., Thorborg, K., Tinggaard, C.
Number of pages: 10
Publication date: 2008

**Host publication information**
Title of host publication: Proceedings of the 125th Audio Engineering Society
Source: orbit
Source-ID: 222383
Research output: Research - peer-review › Article in proceedings – Annual report year: 2008

**Non-linear Loudspeaker Unit Modelling**
Simulations of a 6½-inch loudspeaker unit are performed and compared with a displacement measurement. The non-linear loudspeaker model is based on the major nonlinear functions and expanded with time-varying suspension behaviour and flux modulation. The results are presented with FFT plots of three frequencies and different displacement levels. The model errors are discussed and analysed including a test with loudspeaker unit where the diaphragm is removed.

**General information**
State: Published
Organisations: Acoustic Technology, Department of Electrical Engineering, Aalborg University
Contributors: Pedersen, B. R., Agerkvist, F. T.
Time Variance of the Suspension Nonlinearity
It is well known that the resonance frequency of a loudspeaker depends on how it is driven before and during the measurement. Measurement done right after exposing it to high levels of electrical power and/or excursion giver lower values than what can be measured when the speaker is cold. This paper investigates the changes in compliance the driving signal can cause, this includes low level short duration measurements of the resonance frequency as well as high power long duration measurements of the non-linearity of the suspension. It is found that at low levels the suspension softens but recovers quickly. The the high power and long term measurements affect the non-linearity of the speaker, by increasing the compliance value for all values of displacement. This level dependency is validated with distortion measurements and it is demonstrated how improved accuracy of the non-linear model can be obtained by including the level dependency.

Modelling Loudspeaker Non-Linearities
This paper investigates different techniques for modelling the non-linear parameters of the electrodynamic loudspeaker. The methods are tested not only for their accuracy within the range of original data, but also for the ability to work reasonable outside that range, and it is demonstrated that polynomial expansions are rather poor at this, whereas an inverse polynomial expansion or localized fitting functions such as the gaussian are better suited for modelling the BI-factor and compliance. For the inductance the sigmoid function is shown to give very good results. Finally the time varying property of the suspension is studied and it demonstrated that significant part of the variation can be predicted from the dissipated power.

Monaural separation of dependent audio sources based on a generalized Wiener filter
This paper presents a two-stage approach for single-channel separation of dependent audio sources. The proposed algorithm is developed in the Bayesian framework and designed for general audio signals. In the first stage of the algorithm, the joint distribution of discrete Fourier transform (DFT) coefficients of the dependent sources is modeled by complex Gaussian mixture models in the frequency domain from samples of individual sources to capture the properties of the sources and their correlation. During the second stage, the mixture is separated through a generalized Wiener filter, which takes correlation term and local stationarity into account. The performance of the algorithm is tested on real audio signals. The results show that the proposed algorithm works very well when the dependent sources have comparable variances and linear correlation.
Time Varying Behavior of the Loudspeaker Suspension
The suspension part of the electrodynamic loudspeaker is often modelled as a simple linear spring with viscous damping, however the dynamic behaviour of the suspension is much more complicated than predicted by such a simple model. At higher levels the compliance becomes non-linear and often changes during excitation at high levels. This paper investigates how the compliance of the suspension depends on the excitation, i.e. level and frequency content. The measurements are compared with other known measurement methods of the suspension.

Efficient Non Linear Loudspeakers
Loudspeakers have traditionally been designed to be as linear as possible. However, as techniques for compensating non-linearities are emerging, it becomes possible to use other design criteria. This paper present and examines a new idea for improving the efficiency of loudspeakers at high levels by changing the voice coil layout. This deliberate non-linear design has the benefit that a smaller amplifier can be used, which has the benefit of reducing system cost as well as reducing power consumption.
Development of a Sound Quality Evaluation System
This paper describes the development of the first version of the Sound Quality Evaluation System. The purpose of the system is to predict the subjective sound quality of home theater systems from objective measurements. 16 home theater systems were measured in an anechoic room. Several metrics expected to correlate with the subjective quality were proposed and tested. A model for the sound quality was created by mapping the subjective evaluations of the Home Theater Systems with the metrics calculated for each system. Correlation between subjective listening test and the prediction is present.

General information
State: Published
Organisations: Department of Electrical Engineering
Contributors: Kvist, P., Thomsen, C., Lee, S., Park, J., Agerkvist, F. T.
Publication date: 2004

Subjective test of class D amplifiers without output filter
This paper presents the results of subjective listening tests designed to determine whether the output filter on class D amplifiers used in active loudspeakers can be omitted without audible errors occurring. The frequency range of the amplifiers was limited to 0-3 kHz corresponding to a woofer or midrange unit. A listening panel of 7 persons were used in the test which showed that no audible errors could be detected.

General information
State: Published
Organisations: Department of Electrical Engineering
Contributors: Agerkvist, F. T., Fenger, L. M.
Publication date: 2004

Measuring the complexity of the ocean floor

General information
State: Published
Organisations: Department of Electrical Engineering, Acoustic Technology
Contributors: Sams, T., Stage, B., Agerkvist, F. T., Christensen, T.
Pages: 576-583
Publication date: 2000
A Study of Simple Diffraction Models

General information
State: Published
Organisations: Department of Acoustic Technology
Contributors: Agerkvist, F.
Number of pages: 12
Publication date: 1997

Publication information
Original language: English
Source: orbit
Source-ID: 167719
Research output: Research - peer-review › Report – Annual report year: 1997

A Study of Simple Diffraction Models
In this paper two simple methods for cabinet edge diffraction are examined. Calculations with both models are compared with more sophisticated theoretical models and with measured data. The parameters involved are studied and their importance for normal loudspeaker box designs is examined.

General information
State: Published
Organisations: Department of Acoustic Technology
Contributors: Agerkvist, F.
Publication date: 1997
Peer-reviewed: Yes
Event: Paper presented at 102nd Audio Engineering Society Convention, Munich, Germany.
Source: orbit
Source-ID: 167723
Research output: Research - peer-review › Paper – Annual report year: 1997

A Time-Frequency Auditory Model Using Wavelet Packets

General information
State: Published
Organisations: Department of Acoustic Technology
Contributors: Agerkvist, F.
Pages: 37-50
Publication date: 1996
Peer-reviewed: Yes

Publication information
Journal: Journal of The Audio Engineering Society
Volume: 44
Issue number: 1/2
ISSN (Print): 1549-4950
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Web of Science (2019): Indexed yes
BFI (2018): BFI-level 1
Web of Science (2018): Indexed yes
BFI (2017): BFI-level 1
Scopus rating (2017): SJR 0.265 SNIP 0.853
Web of Science (2017): Impact factor 0.774
Web of Science (2017): Indexed yes
BFI (2016): BFI-level 1
Scopus rating (2016): CiteScore 0.95 SJR 0.306 SNIP 0.934
Web of Science (2016): Impact factor 0.707
Web of Science (2016): Indexed yes
BFI (2015): BFI-level 1
Scopus rating (2015): CiteScore 1.11 SJR 0.408 SNIP 1.457
Time-frequency analysis with temporal and spectral resolution as the human auditory system

The human perception of sound is a suitable area for the application of a simultaneous time-frequency analysis, since the ear is selective in both domains. A perfect reconstruction filter bank with bandwidths approximating the critical bands is presented. The orthogonality of the filter makes it possible to examine the masking effect with realistic signals. The tree structure of the filter bank makes it difficult to obtain well-attenuated stop-bands. The use of filters of different length solves this problem

General information
State: Published
Organisations: Department of Acoustic Technology
Contributors: Agerkvist, F. T.
Pages: 425-428
Publication date: 1992

Host publication information
Title of host publication: Proceedings of the IEEE-SP International Symposium of Time-Frequency and Time-Scale Analysis
Bibliographical note
Copyright: 1992 IEEE. Personal use of this material is permitted. However, permission to reprint/republish this material for advertising or promotional purposes or for creating new collective works for resale or redistribution to servers or lists, or to reuse any copyrighted component of this work in other works must be obtained from the IEEE

Projects:

**Sound Field Analysis and Microphone Array in Rooms**
Hahmann, M., PhD Student, Department of Electrical Engineering
Fernandez Grande, E., Main Supervisor, Department of Electrical Engineering
Agerkvist, F. T., Supervisor, Department of Electrical Engineering
01/02/2019 → 31/01/2022
Project: PhD

**Acoustic Array Processing and Sound Field Analysis in Rooms**
Verburg Riezu, S. A., PhD Student, Department of Electrical Engineering
Fernandez Grande, E., Main Supervisor, Department of Electrical Engineering
Agerkvist, F. T., Supervisor, Department of Electrical Engineering
Eksternt finansieret virksomhed
01/07/2018 → 30/06/2021
Award relations: Acoustic Array Processing and Sound Field Analysis in Rooms
Project: PhD

**Optimization of loudspeakers using material and shape optimization**
Nielsen, D. G., PhD Student, Department of Electrical Engineering
Agerkvist, F. T., Main Supervisor, Department of Electrical Engineering
Jensen, J. S., Supervisor, Department of Mechanical Engineering
Institut stipendie (DTU)
01/03/2018 → 28/02/2021
Award relations: Optimization of loudspeakers using material and shape optimization
Project: PhD

**Real time sound field control for outdoor concerts - silent zones, adaptation and objective-subjective performance**
Plewe, D., PhD Student, Department of Electrical Engineering
Agerkvist, F. T., Main Supervisor, Department of Electrical Engineering
Brunskog, J., Supervisor, Department of Electrical Engineering
Fernandez Grande, E., Supervisor, Department of Electrical Engineering
Samfinansieret - Andet
01/05/2017 → 14/05/2020
Award relations: Real time sound field control for outdoor concerts - silent zones, adaptation and objective-subjective performance
Project: PhD

**Outdoor Sound Propagation and Monitoring for Sound Field Control Applications**
Caviedes Nozal, D., PhD Student, Department of Electrical Engineering
Brunskog, J., Main Supervisor, Department of Electrical Engineering
Agerkvist, F. T., Supervisor, Department of Electrical Engineering
Fernandez Grande, E., Supervisor, Department of Electrical Engineering
Samfinansieret - Andet
15/12/2016 → 14/12/2019
Award relations: Outdoor Sound Propagation and Monitoring for Sound Field Control Applications
**Sound field control for outdoor concerts**
Heuchel, F. M., PhD Student, Department of Electrical Engineering
Agerkvist, F. T., Main Supervisor, Department of Electrical Engineering
Brunskog, J., Supervisor, Department of Electrical Engineering
Fernandez Grande, E., Supervisor, Department of Electrical Engineering
Samfinansieret - Andet
15/12/2016 → 14/12/2019
Award relations: Sound field control for outdoor concerts
Project: PhD

**Minimering af vibro-akustik i tilbagekobling (feedback) i høreapparater**
Friis, L., PhD Student, Department of Electrical Engineering
Ohlrich, M., Main Supervisor, Department of Electrical Engineering
Jacobsen, F., Supervisor, Department of Electrical Engineering
Jensen, L. B., Supervisor
Agerkvist, F. T., Examiner, Department of Electrical Engineering
Carcaterra, A., Examiner
ErhvervsPhD-ordningen VTU
01/07/2005 → 20/05/2009
Award relations: Minimering af vibro-akustik i tilbagekobling (feedback) i høreapparater
Project: PhD

**Akustisk Identifikation af Søminer**
Wendelboe, G., PhD Student, Department of Electrical Engineering
Sørensen, H. B. D., Main Supervisor, Department of Electrical Engineering
Damsgaard, B., Supervisor
Jacobsen, F., Supervisor, Department of Electrical Engineering
Agerkvist, F. T., Examiner, Department of Electrical Engineering
Juhl, P. M., Examiner, Department of Acoustic Technology
Lyons, A. P., Examiner
Anden sektorministeriel finans
01/03/2002 → 30/03/2007
Award relations: Akustisk Identifikation af Søminer
Project: PhD

**Nonlinear fractional order derivative models of components and materials in hearing aids and transducers**
King, A. W., PhD Student, Department of Electrical Engineering
Agerkvist, F. T., Main Supervisor, Department of Electrical Engineering
Brunskog, J., Supervisor, Department of Electrical Engineering
Jensen, J. S., Supervisor, Department of Mechanical Engineering
Forskningsrådsfinansiering
01/05/2016 → 30/04/2019
Award relations: Nonlinear fractional order derivative models of components and materials in hearing aids and transducers
Project: PhD

**Quantification of diffuseness chambers for sound absorption measurements**
Nolan, M., PhD Student, Department of Electrical Engineering
Brunskog, J., Main Supervisor, Department of Electrical Engineering
Jeong, C., Supervisor, Department of Electrical Engineering
Agerkvist, F. T., Examiner, Department of Electrical Engineering
Nilsson, E., Examiner
Svensson, U. P., Examiner
Samfinansieret - Andet
01/08/2015 → 23/01/2019
Award relations: Quantification of diffuseness chambers for sound absorption measurements
Project: PhD

**Nonlinear balanced armature receivers**
Jensen, J., PhD Student, Department of Electrical Engineering
Agerkvist, F. T., Main Supervisor, Department of Electrical Engineering
Harte, J., Supervisor, Department of Electrical Engineering
Brunskog, J., Examiner, Department of Electrical Engineering
Bard, D., Examiner
Klippel, W., Examiner
Bard, D., Examiner
Klippel, W., Examiner
ErhvervsPhD-ordningen VTU
01/07/2010 → 15/11/2014
Award relations: Nonlinear balanced armature receivers
Project: PhD

New measurement techniques: Optical methods for characterizing sound fields
Torras Rosell, A., PhD Student, Department of Electrical Engineering
Agerkvist, F. T., Main Supervisor, Department of Electrical Engineering
Barrera Figueroa, S., Supervisor, Department of Electrical Engineering
Jeong, C., Examiner, Department of Electrical Engineering
Gazengel, B., Examiner
Humphrey, V. F., Examiner
ErhvervsPhD-ordningen VTU
01/04/2010 → 19/03/2014
Award relations: New measurement techniques: Optical methods for characterizing sound fields
Project: PhD

Speech intelligibility enhancement using modern envelope and phase manipulations
Decorsière, R. J. B., PhD Student, Department of Electrical Engineering
Dau, T., Main Supervisor, Department of Electrical Engineering
MacDonald, E., Supervisor, Department of Health Technology
Søndergaard, P. L., Supervisor, Department of Electrical Engineering
Agerkvist, F. T., Examiner, Department of Electrical Engineering
Ghitza, O., Examiner
Stone, M. A., Examiner
Eksternt finansieret virksomhed
15/02/2010 → 19/11/2013
Award relations: Speech intelligibility enhancement using modern envelope and phase manipulations
Project: PhD

Samtidig tids- og frekvensanalyse
Agerkvist, F. T., PhD Student, Department of Electrical Engineering
Jacobsen, F., Main Supervisor, Department of Electrical Engineering
Elberling, C., Examiner
DTU-stipendium
01/02/1992 → 02/08/1995
Award relations: Samtidig tids- og frekvensanalyse
Project: PhD

New Strategies for Feedback Suppression in Hearing Instruments
Guilin, M., PhD Student, Department of Electrical Engineering
Jacobsen, F., Main Supervisor, Department of Electrical Engineering
Agerkvist, F. T., Supervisor, Department of Electrical Engineering
Gran, F., Supervisor, Department of Electrical Engineering
Buchholz, J., Examiner, Department of Electrical Engineering
Kates, J. M., Examiner
Rubak, P., Examiner
ErhvervsPhD-ordningen VTU
01/05/2007 → 01/09/2010
Award relations: New Strategies for Feedback Suppression in Hearing Instruments
Project: PhD

Topology optimization for medium- to high-frequency applications
Christiansen, R. E., PhD Student, Department of Mechanical Engineering
Sigmund, O., Main Supervisor, Department of Mechanical Engineering
Jensen, J. S., Supervisor, Department of Mechanical Engineering
Lazarov, B. S., Supervisor, Department of Mechanical Engineering
Agerkvist, F. T., Examiner, Department of Electrical Engineering
Berggren, M., Examiner
Schevenels, M., Examiner
Berggren, M., Examiner
Institut, samfinansiering
15/06/2013 → 30/09/2016
Award relations: Topology optimization for medium- to high-frequency applications
Project: PhD

Deconvolution of seismic transient: A model-based signal processing approach
Oh, G. L., PhD Student, Department of Electrical Engineering
Brunskog, J., Main Supervisor, Department of Electrical Engineering
Agerkvist, F. T., Supervisor, Department of Electrical Engineering
Jensen, J. S., Examiner, Department of Mechanical Engineering
Mosegaard, K., Examiner, Center for Energy Resources Engineering
Rydén, N., Examiner
Rydén, N., Examiner
Ansat eksternt
15/03/2011 → 30/09/2014
Award relations: Deconvolution of seismic transient: A model-based signal processing approach
Project: PhD

Computational methods for detection and imaging of oil in sea water
Xenaki, A., PhD Student, Department of Electrical Engineering
Knudsen, K., Main Supervisor, Department of Applied Mathematics and Computer Science
Brunskog, J., Supervisor, Department of Electrical Engineering
Mosegaard, K., Supervisor, Center for Energy Resources Engineering
Agerkvist, F. T., Examiner, Department of Electrical Engineering
Edelmann, G. F., Examiner
Nordborg, A., Examiner
Edelmann, G. F., Examiner
Nordborg, A., Examiner
1/3 FUU, 1/3 inst 1/3 Andet
01/07/2011 → 13/08/2015
Award relations: Computational methods for detection and imaging of oil in sea water
Project: PhD

Acoustic array methods for identification of noise sources in vehicles
Tiana Roig, E., PhD Student, Department of Electrical Engineering
Jeong, C., Main Supervisor, Department of Electrical Engineering
Jacobsen, F., Supervisor, Department of Electrical Engineering
Juhl, P. M., Examiner
Rafaely, B., Examiner
Song, W., Examiner
Institut stipendie (DTU)
01/10/2010 → 26/01/2015
Award relations: Acoustic array methods for identification of noise sources in vehicles
Project: PhD

Loudspeaker cabinet edge diffraction
Not only the loudspeaker units themselves, but also the shape of the loudspeaker cabinet determine the frequency response of a loudspeaker system. In this project models for predicting the influence of the cabinet diffraction are investigated, with the aim of improving the current models.
Agerkvist, F. T., Project Manager, Department of Acoustic Technology
01/10/1994 → 31/01/1997
Project: Research