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Precision measurement and correction of microphone distortion

Prof R A Belcher and Prof J A Chambers, Cardiff University

1. Background and Context

Precision 'metrology-class' or measurement microphones are key components in acoustic measurement and recording instruments. These microphones are difficult to manufacture and are assumed to have very low non-linear distortion. Their high cost precludes their use in consumer and multi-sensor applications where low distortion would certainly be of benefit. MEMS (Micro-Electro-Mechanical System) microphones can be less expensive to manufacture and are potentially able to produce metrology class results. Unfortunately, low cost MEMS microphones are limited by non-linear distortion. Manufacturers and standards laboratories test the non-linearity of a microphone by using acoustic resonators to reduce the harmonics generated by a loudspeaker. Non-linearity introduces an error in the measured value of sound pressure level (SPL) as total harmonic distortion (THD) increases with SPL. The maximum acceptable THD is set by standards at 3%. Published theory indicates mainly second order non-linear distortion in measurement microphones at low audio frequencies [1]. Although the predicted lowest THD is much less than 0.25%, direct measurements have been unable to confirm this owing to the difficulty of generating pure acoustic test signals [1].

When small and large amplitude signals are present, non-linearity produces intermodulation distortion (IMD) products. Their presence is an important issue in consumer and multi-sensor applications as they impose a limit on separating small and large signals. The difficulty of generating acoustic sine waves with sufficient purity has therefore limited researchers' ability to verify theoretical THD figures and the non-linear model for microphone distortion. In principle, digital signal processing could correct for THD in microphones but the problems of precision acoustic measurement and accurate modelling has resulted in limited success.

A recent patent application by Prof Belcher [2] described a method that could facilitate increased precision in the measurement of microphone distortion. This method could enable the validation of a precision mathematical model for the distortion. In principle, compensation or correction could be achieved by inverting the model. This approach could be applied to low cost MEMS microphones [3] to obtain increased linearity and is one potential outcome of the project.

The 12 month programme of work summarised in this report is a 'proof of concept' project. The central aim of this endeavour was to discover whether the proposed new test method could facilitate low cost precision measurement and compensation of microphone non-linearity. In order to establish this it was necessary to develop and test a theoretical model of the signal processing steps and approximations involved in implementing the system described in [2]. The highly experimental nature of this project resulted in the necessity of most effort being applied to testing theoretical predictions and refining the theory in the light of experimental results. This revealed unexpected theoretical and practical factors that limited the precision of measurement and therefore achieved one of the goals. It also showed that correction by inverting the model was problematic as only the output signal was available. A new method was therefore devised that enabled precision measurement of the quadratic coefficient of distortion. A correction algorithm was also invented that provided 'ideal' correction using only the output signal. The work leading to these advances will be outlined in the next section.

2. Key advances and supporting methodology

Initial tests of a selection of microphones were made in the Intelligent Office within the Cardiff Centre of Digital Signal Processing (DSP). Test signal sequences were designed to be one second bursts of signal followed by silence. This allowed the measurement of reverberation time, intermodulation distortion (IMD) and noise. IMD may have been influenced by the reverberation time of the room but unfortunately this proved impossible due to the high background noise. A PhD Student, Yonggang Zhang, assisted with this investigation by simulating the reverberation time as part of an EPSRC project [4], [5-7]. The simulated reverberation time agreed with that measured using the test signals. In principle his work could be used to simulate a noise free reverberant room in a future project but there are other practical factors to take into account, such as acoustic cross-talk, before this could be used for non-linearity simulations. Measurements were made using the 'double comb filter' or DCF method. This measurement result was much less affected by the low frequency room noise as it was below the frequency range of the test signal and could be removed by filtering. The recordings are included in the database for reference. However, as IMD could not be measured a meaningful comparison of both methods could not be made. This part of the project could not be taken any further as a low noise listening room was not available.

Microphone test recordings were successfully made using the anechoic room at NPL. The use of this facility was offered by Richard Barham of NPL during his introductory visit to Cardiff. The database of recordings made at NPL was used in investigations in this project. Recordings in a real environment were necessary as these would reveal practical limitations that may not be included in a simulation.

Initially, mathematical models for condenser and electro-dynamic microphones were investigated and samples of these tested. A literature search had revealed that no comprehensive models for the non-linearity errors of either dynamic or condenser microphones had been produced. Discussions with NPL and QinetiQ confirmed this conclusion. To assist with this investigation, NPL arranged for Prof Belcher to meet with two international experts in microphone design: Prof Knud Rasmussen senior lecturer at the Institute of Acoustic Technology, Technical University of Denmark (DTU) and Mr Erling Frederiksen department manager for microphone calibration Bruel &Kjaer (B&K). The discussion covered distortion mechanisms and models for most types of microphones and included the advice that electro-dynamic microphones are very challenging to model as the distortion varies with frequency. Fortunately, these are of less importance as they are being replaced by low cost electret (these are condenser or capacitor types) microphones. The main conclusion of this meeting was that the project should next focus on testing and modelling condenser microphones. Below resonance the distortion for measurement microphones was expected to be frequency independent and only second order. Measurement microphones and MEMS microphones are both condenser types. This justified starting with a quadratic, memory-less model for simulations. QinetiQ had overall computer simulations of their MEMS microphone designs and were investigating the problem of isolating and modelling the non-linearity aspects of their design. These issues were discussed in collaboration meetings. At a meeting close to the end of this project, their work had produced a model showing a mechanism that could produce small third order non-linearity. Samples of their MEMS microphones were provided for testing at NPL. These tests had shown that while these devices had a quadratic term, a small cubic term was also present. This provided support for the QinetiQ improved model. Further work would be required in a future project to determine the magnitude and significance of this particular cubic term in a compensation algorithm. Theory developed in this project enabled the value of the quadratic and cubic coefficients in a model of the non-linearity to be calculated from IMD. THD can then be calculated by applying a sinewave as the input to this model. This approach had not been reported in the literature. As it is one of the study topics of an AES standards working group, the planned publication revealing this approach will be of interest to the audio community.

During the NPL tests very low frequency signals (infra sound) were recorded *only* when measurement microphones were tested. A 15 Hz analogue high pass filter removed most of this noise but the filter introduced extra non-linearity and so could have prevented a true measure of the microphone distortion. Measurement microphones were tested with and without this in place but noise below 100Hz limited the accuracy of measurement of IMD. This was a problem that could not be resolved easily at NPL. It could have been due either to electrical problems with the microphones or to acoustic sources. If the infra noise levels were significantly different at another location this would tend to eliminate the possibility of electrical problems with the microphone as a source of the noise. The possibility of making tests on measurement microphones at other locations was therefore investigated.

Mr Fredreiksen of B&K was supplied with the THD test results estimated from IMD measurements. He commented that the THD results for the B&K microphones were higher than he expected in some cases. As a result of further discussions, B&K offered the use of their test facility and microphones in Denmark to assist in resolving this problem. They were also interested in obtaining a better understanding of the test methods. Prof Belcher accepted this offer and took with him a complete Cardiff test system. The tests were undertaken in B&K's anechoic chamber using the same models of microphone as used at NPL. These tests revealed the presence of a small amplitude very low frequency noise. This was much smaller in amplitude than at NPL so did not require a high pass filter. B&K explained that their measurement microphones have a sub Hz response so can detect very low frequency noise. This effect can be produced by changes in air pressure caused by wind or by doors opening and closing. Anechoic chambers are not air-tight as ducts are needed for cables. This appears to be a reasonable explanation for the source of infrasound noise. NPL stated that their chamber had less than 0dBA noise below 1Hz. 'A' weighting introduces more than 60 dB attenuation below 1 Hz compared to un-weighted measurements. It is therefore possible for a 0 dBA SPL noise level to be measured when the unweighted infra noise is 60 dB SPL. To put this into perspective, if -100dB second order IMD products are predicted with a measurement microphone this is equivalent to 10 dBSPL so measurement precision is strongly influenced by acoustic noise levels. For reference purposes, IMD testing was also undertaken using B&K test equipment and this produced an unexpected result. It indicated that there was an unknown factor in the Cardiff test system that produced a frequency dependent IMD measurement and limited IMD. This frequency dependent effect did not occur in the B&K test method. The possibility of a problem had not been indicated when the system was configured and evaluated using standardised test methods. Fortunately, this problem was not significant for the MEMS and other microphones as they produced IMD results at least 20 dB higher than this limit. Discussions took place at B&K regarding the theoretical and practical issues relevant to modelling measurement microphones and this collaboration has provided a solid basis for future work.

On return from Denmark, in the final weeks of the project, a test fixture was constructed and measurements made to investigate the possibility of the 24 bit ADC-DAC having become faulty. The problem was finally identified as slew rate dependent nonlinear cross talk in the stereo digital to analogue converter. This was a completely unexpected problem as it is not covered by test standards. Three papers were prepared describing these results and submitted to the IMTC 2007 Conference [8], AES 2006 International Conference [9] and AES 2006 European Convention [10]. A replacement 24 bit ADC-DAC was purchased and tested and this did not have this fault. It provided an IMD threshold of better than -100 dB in the same test configuration. Simulations based on FFT analysis predicted a figure of -150 dB. Tests showed that the -100dB limit was due to random noise produced by the amplifiers and data converters. This IMD threshold could be improved with averaging or improved circuits.

Prior to the visit to B&K, research and collaboration had so far not provided an accurate model for sources of non-linearity higher than second order. This may have been partly because higher order terms were not considered an important issue for SPL measurement and partly because they were difficult to measure. Prof Belcher had previously undertaken a literature search in journal and patent publications for mathematical models of microphone distortion and little was found. He is a member of various international standards committees concerned with audio measurements and through the Audio Engineering Society SC04-04 committee became aware of interest in methods of testing microphone pre-amplifiers. The committee members revealed that 'difference frequency' tests were used by studio microphone manufacturers at the design stage. This test was not used by manufacturers of measurement microphones as they only required THD figures. This 'difference frequency' test had been reported in an AES convention pre-print [8], by employees of the Neumann company in Germany.

A previous search of AES Journals had not revealed this publication. Pre-prints do not necessarily become Journal publications and are not fully abstracted so relevant ones may not always be found by searching titles. An in depth search of the AES electronic archive was then undertaken using many variations of search words and related articles were read. This revealed a paper by Vakhitov on mathematical modelling of microphone distortion [12]. It claimed to predict measured distortion with good accuracy and covers many practical issues that must be included in the model. It stated that third order distortion is due to non-ideal flexibility of the sensing capacitive membrane. Second order distortion is claimed to be due to stray capacitance and this particular claim is supported in many publications. E-mail communication with Mr Vakhitov has been informative and he has supplied a copy of part of his Monograph in Russian. This was translated informally in Cardiff and revealed a more in depth treatment of sources of non-linearity than found previously.

In summary, the key advances are:

(1) Mathematical prediction of the coefficient of quadratic distortion from measurements of the non-linear distortion power generated by the Double Comb Filter (DCF) method.

(2) Extension of the 1945 seminal work by Brockbank and Wass [13] on calculating multi-tone distortion power to sampled and quantised systems.

(3) Analysis of the systematic error sources in a practical implementation of the DCF test method. This provides a basis for specifying the minimum stop band attenuation required for source and measurement filters.

(4) A multi-rate filter was designed that enabled the analysis time on the PC to be reduced from 15 minutes to 5 minutes.

(5) Collaboration between Prof Chambers, Prof Belcher and Dr Lambotharan resulted in the invention of a quadratic modelling and correction method that applied to large scale signals and did not need access to the input signal to determine the coefficient values [14]. This invention provided perfect correction in the case of a purely quadratic model. In comparison the 'known art' of model inversion provided limited compensation.

(6) A mathematical investigation of modelling the amplitude probability density function (APDF) of the DCF test signal by using a limited set of basis functions has shown that a Fourier series with 10 coefficients provides the best fit. A paper on this has been accepted for the IMA conference in December [15]. This has revealed a new area of research: is it possible to find a set of shift register clock frequencies and generating polynomials for each maximum length sequence that minimises the residual error in the APDF fit?

(7) The discovery of how even and odd order terms relate to the APDF [15]. Previous published work on measuring non-linearity using a noise-like test signal has assumed that the test signal has a Normal distribution [16]. As the accuracy of this assumption is low for realistic test durations it provides an estimate only of the integral non-linearity.

(8) Analysis of the DCF non-linear distortion error signal in the time, frequency and APDF domains to assess its suitability for adaptive correction. This has shown that

- In the time domain, the crest factor increases with the order of non-linearity so may be suitable for use in adaptive correction of higher order terms.

- In the frequency domain, aliasing of out of band distortion products can mask the changes in power spectral density (PSD) predicted by Brockbank and Wass. Oversampling at 192 kHz can, however, be used to show the predicted PSD changes for quadratic and cubic distortion. It may be possible to use PSD as an error signal to find low order terms in adaptive correction.

- The symmetry of the APDF is a basis for estimating the order of the non-linearity.

Perhaps the most important overall conclusion is that the factors that limit the precision measurement of the DCF test method [2] have been identified and the method fully validated. Its ultimate precision is limited by aliasing effects in digital filters and by cross talk between audio sources. Taking all these factors into account, low cost audio equipment provides sufficient performance to test the best available microphones. The DCF test is superior to IMD tests as it is influenced much less by low frequency or line spectrum interference. It is also in principle less affected by variations in the amplitude frequency response and reverberation time of the room but an experimental investigation of this was beyond the scope of this short project. It has also been demonstrated that much superior correction of quadratic distortion is possible than is available by model inversion and that both quadratic and cubic coefficients can be found from DCF results.

3. Project plan review

The original project plan was followed with the following minor changes.

(1) It did not prove possible to present papers at conferences held during the term of the project. This was due entirely to the delay introduced by the decision to apply for a patent [14]. However, six conference papers, one journal and one patent application were submitted in the final stage of the project, which exceeds the planned one conference and one journal publication.

(2) It was concluded at an early stage that an investigation of adaptive correction required a real time filter as the computer storage requirements for off-line drift analysis was impractical. The real time filter design and implementation was beyond the scope of a 'proof of concept' activity.

(3) Some of the time allocated to an investigation of adaptive correction algorithms was instead allocated to an originally unplanned investigation of two tone intermodulation measurement for the following reasons:

In order to compare the DCF method with the best available conventional test method, some time was spent examining the possibility of using sine wave tests. It occurred to Prof. Belcher that by splitting the IMD test into two acoustic sources a limited measurement of microphone IMD could be made by examining the spectrum of the microphone output signal. It was felt that the performance of the DCF method in this application should now be evaluated against a two tone IMD test as this would be the most demanding and realistic comparison. The additional analysis and experimental investigations required to evaluate the IMD test resulted in less time being available for investigations of adaptive correction methods planned for later in this project. However, this change in direction was considered essential in a 'proof of concept' research project.

The use of IMD testing to measure microphone non-linearity coefficients was not found in a search of abstracted publications nor in previous discussions with measurement microphone design experts so was initially believed to be novel and perhaps suitable for a patent application. As already mentioned in Section 2 of this report, meetings and e-mail discussions in the very last part of the project revealed that this IMD test was used by studio microphone manufacturers for design purposes, but known as a 'difference frequency' test. One of the reasons put forward for the limited use of IMD was that microphone standards specified THD not IMD. This revelation fully justified the earlier decision to change emphasis in the project. A comparison with the DCF method has now been made against the best method used by studio microphone manufacturers. From this, two interesting conclusions were drawn. First, THD was of more importance commercially than IMD, which may indicate why this IMD test method was not used by manufacturers of measurement microphones. Second, the very limited availability of publications on this subject may explain why engineers in the microphone applications area failed to realise that THD could be calculated from IMD.

4. Research impact and benefits to society

Developments in signal processing and new areas of study resulting from this project will be of benefit to both academic and industrial organisations. For example, the low cost nature of DSP corrected MEMS microphones could lead to improved dynamic range for consumer audio and telephony units. This is a particular problem for handsets used for 3G video calling as the handset is much further away from the face than conventional handsets. It could also enable more intelligible hands-free communication in noisy environments. The low cost per precision audio sensor could also make it economic to use multi-sensor networks. Some examples include the monitoring of noise pollution, multi sensor fault detection in rotating machines and active noise reduction in aircraft and motor vehicles.

5. Explanation of expenditure

A notebook PC was purchased to enable the audio test unit to have the lowest acoustic and electrical noise. When the project was first planned, the performance required of a 24 bit ADC/DAC was available only from a PCI based audio card, necessitating a PCI to PCMCIA adapter. However, at the start of the project it was found that the performance of PCMCIA based 24 bit audio systems had become equal to PCI versions so the adapter was not needed. This produced a cost saving and enabled the purchase of a second 24 bit audio system at a later date when the performance of the original audio unit became suspect. Travel was undertaken to QinetiQ, to discuss modelling, and to NPL to make measurements of microphones. The anechoic measurements at B&K Denmark were originally unplanned and required the Cardiff test system to be completely transportable, within the weight limit and size of checked luggage. This necessitated the purchase of a set of portable stands and GENELEC active loudspeakers from funds originally allocated for travel and consumables. As Prof Belcher was employed part time, a small amount of the cost saving in travel and consumables was used to pay for his extra time required to undertake the work in Denmark. Prof Chambers attended a conference on digital communications in order to explore the possibility of using the findings of this project in the microphones within the nodes of wireless sensor networks, and in the handsets of mobile systems. Discussions with international researchers proved positive and this matter is the subject of the ongoing work with QinetiQ as part of the Cardiff-QinetiQ partnership.

6. Further research or dissemination activities

It is possible that if the APDF of combined odd and even order terms can be separated into odd and even symmetry APDFs then these could be used to estimate the coefficient values and number and order of terms present.
It is also proposed that the APDF of the original test signal is stored as a set of ‘undistorted’ reference data. By comparing a measured APDF with this reference, the difference can be used to calculate the integral non-linearity error function. A measurement of the microphone distortion made this way would be accurate and quick as only one period of this test signal is required. However, the amount of data storage required for the reference is a disadvantage of this approach. If, instead, the APDF of the test signal could be modelled by a set of basis functions then it would remove the need to store the actual APDF. It should be possible to fit Tchebyschev polynomials to this function [16] and thus obtain accurate coefficient values to model the microphone non-linearity.

Further acoustic tests and simulations should be undertaken to show if the non-linear distortion of a measurement microphone can be determined with sufficient accuracy to validate its theoretical model and also whether this can be determined in the presence of reverberation.

Publications are planned to disseminate the results of the microphone distortion correction scheme.

International collaboration with Denmark and Russia has provided links that will enable more in depth modelling of microphone distortion to be undertaken in a future project.

Clearly a hardware or DSP based filter would be an advantage for future work as it would enable adaptive correction to be investigated.

Due to collaboration with QinetiQ in the area of measurement and modelling of MEMS microphones, Cardiff has been awarded a sub-contract as part of the £3.2M three year MOD funded ‘MEMS Applications for Defence’ programme.

7. Reports from project partners

Letters From both NPL and QinetiQ describing their contributions and their views of the benefits of collaboration are provided as separate documents. Extracts are provided below:

From NPL:
"The EPSRC project was therefore able to make a direct and valuable contribution to our research by providing a means of determining the distortion characteristics of the MEMS microphones, independent of the sound source used to create the high sound pressure level."

From QinetiQ:
"Interaction with Professors Chambers and Belcher provided technical staff within our Microsystems group with a useful fundamental insight into the innovative non-linear measurement techniques"

A letter of support from B&K has been requested and offered by email but it was not received at the time of submission of this report.

8. References

[1] Erling Frederiksen, "System for Measurement of Microphone Distortion and Linearity from Medium to Very High Level", Brüel&Kjær Technical Review no1 2002.

[2] R A Belcher "Measuring non-linear distortion in transducers" UK Patent GB2426404A . Published 22 November 2006.

[3] K M Brunson et al "MEMs Microphone Technology", IEE Seminar on MEMs Sensor Technology, IEE London, 25 April 2005.

[4] J Chambers , EPSRC project GR/577547, "Room Acoustics parameters from music".

[5] Y. Zhang, J.A. Chambers, S. Sanei, P. Kendrick, and T.J. Cox, "A new variable tap-length LMS algorithm to model an exponential decay impulse response", accepted for IEEE Signal Processing Letters, August 2006.

[6] Y. Zhang, J.A. Chambers, P. Kendrick, T. Cox, and F. Li, "Acoustic Parameter Extraction from Occupied Rooms Utilizing Blind Source Separation"KES, Brighton, UK, 06.

[7] Y. Zhang, J.A. Chambers, P. Kendrick, T. Cox, and F. Li, "Blind Estimation of Reverberation Time in Occupied Rooms"EUSIPCO, 06, Florence, Italy, 06.

[8] R A Belcher and J Chambers, "Data Converter Dynamic tests with wide band signals", paper submitted to IEEE IMTC 2007 International Conference, Warsaw, March 2007.

[9] R A Belcher and J Chambers, "Precision measurement of ADC effective number of bits using multi-tones" Paper submitted to AES International Conference, London, June 2007.

[10] R A Belcher and J Chambers "Non-linear cross talk in personal computer based audio systems", paper submitted to AES International Convention, Vienna, April 2007.

[11] M Hibbing and H-J Griese, "New investigations on linearity problems of capacitive transducers" AES 68th Convention, Hamburg March 1981, Preprint 1752

[12] S Vakhitov, "Non-linear model of Condenser microphone capsule", AES 21st International Conference, St Petersburg, June 2002 Pre-print paper 2 pages.

[13] R A Brockbank and C A A Wass, "Non-linear distortion in transmission systems", Journal of IEE, Vol 92, Part 3, No 17, pp 45-56,March 1945

[14] R A Belcher, J Chambers and S Lambotharan , " Non-linear signal processing" , UK patent application GB 0619320.5, filed 30 September 2006.

[15] R A Belcher and J Chambers, "Integral Non-linearity error measurement through statistical analysis of a multi-tone test signal", paper accepted for IMA 7th Mathematics in Signal processing Conference, December 2007.

[16] N. Csizmadia and A J E M Janssen,"INL and DNL Estimation based on Noise for ADC test" IEEE T. Inst. and Meast. Vol. 53(5), pp1391-1395, October 2004.