Measuring the impulse response of microphones using noise: a feasibility study

Dr. Hans R.E. van Maanen (Temporal Coherence B.V., Netherlands)

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Abstract

The measurement of the impulse response of microphones is crucial to understand their temporal properties. Little information is disclosed by manufacturers, claiming this is hard to measure due to the requirement of specific hardware and the risk of non-linear response of the microphone. Both objections could be circumvented by using Gaussian distributed white noise as exciting signal and the cross correlation technique. As the use of a loudspeaker is required, a major stumbling block is the requirement of deconvolution of the obtained impulse response. This study shows that the deconvolution is feasible in case of a measured impulse response of a loudspeaker and the impulse response of a modelled (and thus well known) microphone, even if the impulse response of the loudspeaker is significantly wider than the impulse response of the microphone.

Abstract of part 2

The second part of the feasibility study includes a Monte Carlo simulation of the actual measurement procedure, including the exciting Gaussian distributed white noise signal. The exciting signal has been verified to fulfill the requirements for this application. The main conclusion is that the technique is feasible for the determination of the microphone impulse response, but the results show some slight imperfections, which can be reduced by using the same noise signal for the determination of the impulse response of the loudspeaker.

The major conclusion of both parts is that the measurement of the microphone impulse response using noise is feasible without special hardware and that there is no reason for manufacturers to keep this information away from users.

1. Introduction

Sound, including music, is recorded using microphones, so it is an essential link in the chain. But the question arises how well these perform *perceptually*. Of course, the frequency response should cover the range from 20 Hz to 20 kHz, but is this response the only parameter which determines the perceived quality? For a long time, the frequency response in the audio band has been regarded as a major parameter, another one being the harmonic distortion. However, the experiences with CD reconstruction filters and high resolution digital formats have revealed that the perceived quality also depends on the properties of the anti-aliasing and reconstruction filtering: the response *above* 20 kHz is important for the perceived quality (ref. 1).

As the microphone also acts as a (lowpass) filter, it is clear that other properties are of importance too, including its response above 20 kHz. Although response measurements above 20 kHz can be made, this is usually limited to the modulus of the response, the phase response is either not measured or not reported in public domain. The combined modulus and phase responses determine the *temporal* response of the microphone and there are clear indications that the temporal response is of importance for the perceived quality (ref. 2). However, it is not common practice to publish the impulse response of microphones in the datasheets, provided by the manufacturers. Regretfully, as an impulse response covers both the frequency and the phase responses of the device. This kind of information would be very useful for the application and comparison of microphones.

Whether difficulties to measure the impulse response or the revelation of the non-ideal properties of the microphone lie at the basis of this reluctance remains unclear, Yet, some manufacturers do provide impulse responses (ref. 3). In this paper we will describe an alternative way to measure the impulse response of microphones using noise. By this approach, some stumbling blocks, which occur with the more common approach using sparks, can be circumvented. In sec. 2, the basic technique is described, including the mathematics. In sec. 3 a simulation and its results will be presented, followed by a discussion on the feasibility of this technique for use in practical situations. In sec. 4, conclusions will be reported and suggestions for future work listed.

2. The basic technique and the related mathematics

2.1 The problems with the spark technique

It is logical to measure an impulse response with an impulse. But it is not easy to create a sound, sufficiently resembling an impulse, which can be used for this purpose. The currently best option is to use the sound, generated by a spark discharge. However, a close look at its properties reveals that it is not perfect in frequency domain (ref. 4) and thus neither in time domain. Also, no two sparks are completely identical.

And there is another problem: the spark generates a high sound pressure level, so how can be certified that the microphone is still operating in its linear range? This problem could be tackled by using a smaller, less powerful, spark, but that will decrease the signal-to-noise ratio (SNR). The SNR can be improved by ‘conditional averaging’ (ref. 5), but this requires a complete absence of ambiguity in the sound, generated by all the individual sparks. This is, probably, asking too much, leading to a ‘smear’ of the averaged impulse response, an undesirable phenomenon, exactly the opposite of what needs to be achieved.

It thus shows that the ‘spark’ technique has its limitations and stumbling blocks, which hamper its application. Therefore, a different technique to measure the impulse response would be attractive, especially when this could be realized without the use of very specialized equipment.

2.2 The use of white noise for the measurement of impulse responses

From stochastic signal analysis, it is known that there is a one-to-one relation between the transfer function of a system and its response to Gaussian distributed white noise (ref. 5). Without repeating the theory, some of its theorems will be stated without proof. The ones, which are needed in this paper are:

- The autocorrelation function of white noise is a delta function at $\tau = 0$ (albeit with an amplitude of 1 (one)).
- The cross correlation function between the input Gaussian distributed white noise signal and the response of the system equals the impulse response of the system (including its time delay).

From Fourier theory, it is known that multiplying complex valued frequency responses of systems equals convolution of the impulse responses in time domain (refs. 6, 7).

This basically opens the door to measure the impulse response of a system using Gaussian distributed white noise. This has the advantage that it can be certified that the system is never driven into non-linear responses. Any level of accuracy / uncertainty can be reached by using an averaging time as long as needed.

2.3 Basic set-up for the measurement of the impulse response using white noise

The problem with this approach when applied to microphones is, of course, the generation of the white noise sound. It is no real problem to generate a Gaussian distributed noise signal of the required bandwidth and any desired length with computers (ref. 8). Amplifiers, which are able to deliver the noise signal with sufficient power over a wide frequency range to a loudspeaker, are no problem either. But to create the related sound field, a loudspeaker, which has both a very wide frequency response and an excellent impulse response itself, is required. This is not possible with the current 'state of the art' of loudspeakers. However, there is a way around this problem.

The loudspeaker can be regarded as a band-pass filter and its properties are completely determined by its impulse response. Using an excellent wide-band measurement microphone (which are available, e.g. ref. 9) the impulse response of the loudspeaker, used to generate the sound field, can be measured. It will be labelled a $f_{ls}(t)$. The complex transfer function of the loudspeaker can be determined by Fourier Transformation of the impulse response. It will be denoted as $F_{ls}(\omega)$. The microphone under test (MuT) is also a bandpass filter and its complex transfer function will be denoted as $G_{mp}(\omega)$ and its impulse response $g_{mp}(t)$. The overall response of the combination of the loudspeaker and the microphone, denoted as $H_{lm}(\omega)$, is, of course, the product of $F_{ls}(\omega)$ and $G_{mp}(\omega)$. In time domain, this equals the convolution of $f_{ls}(t)$ and $g_{mp}(t)$ and this will be denoted as $h_{lm}(t)$. When the measurement microphone is replaced by the MuT, the measured impulse response equals $h_{lm}(t)$ because the noise is filtered both by the loudspeaker and the MuT. Using the independently measured $f_{ls}(t)$, $g_{mp}(t)$ can be determined by deconvolution of $h_{lm}(t)$ and $f_{ls}(t)$.

Deconvolution is -in general- a bit tricky. So in this study the feasibility to obtain the impulse response of a simulated microphone using the measured impulse response of a tweeter has been investigated. The tweeter impulse response has been measured by Geoff Hill (Hill Acoustics) in the UK and it is shown in fig. 1.

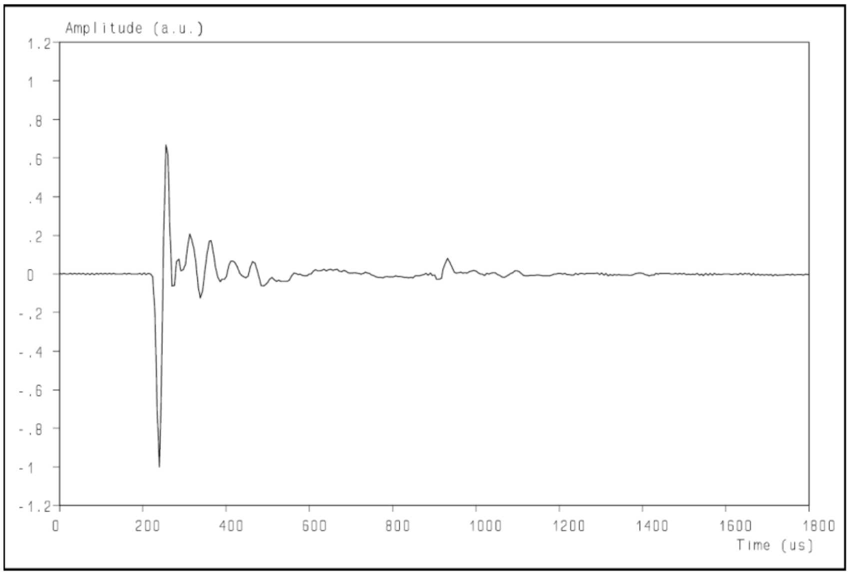

Figure 1: *The measured impulse response of the tweeter, used in this study (courtesy of Hill Acoustics). The small peak at approx. 950 µs is probably caused by a damped reflection in the tetrahedral anechoic chamber.*

The properties of the simulated microphone are illustrated in fig. 2 (frequency response) and fig. 3 (impulse response). The properties of the microphone are based on a general approach: the frequency response tends to decrease at higher frequencies in the audio band. To increase the response in the upper part of the audio band, a slight resonance has been added. This resonance can be discerned in the frequency response, but it is far more clear in the impulse response.

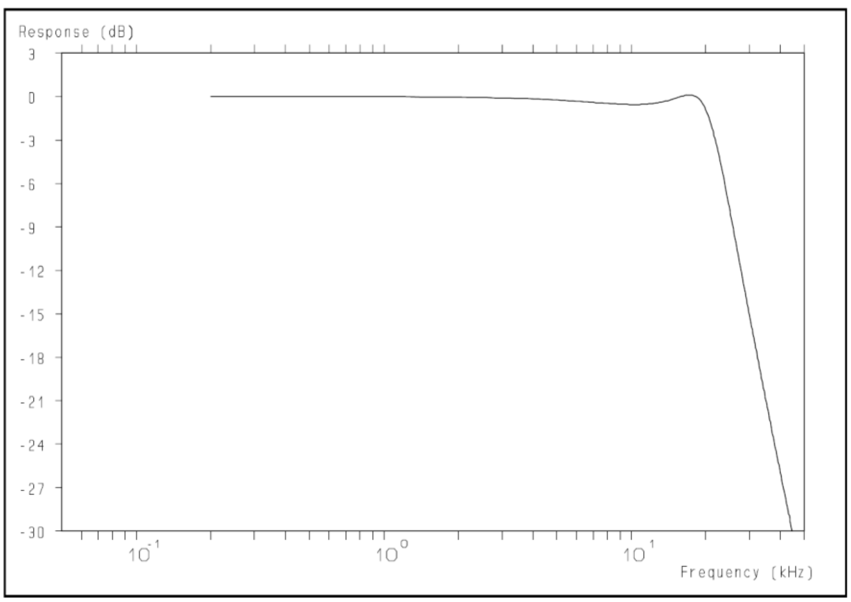

Figure 2: *The frequency response of the simulated microphone.*

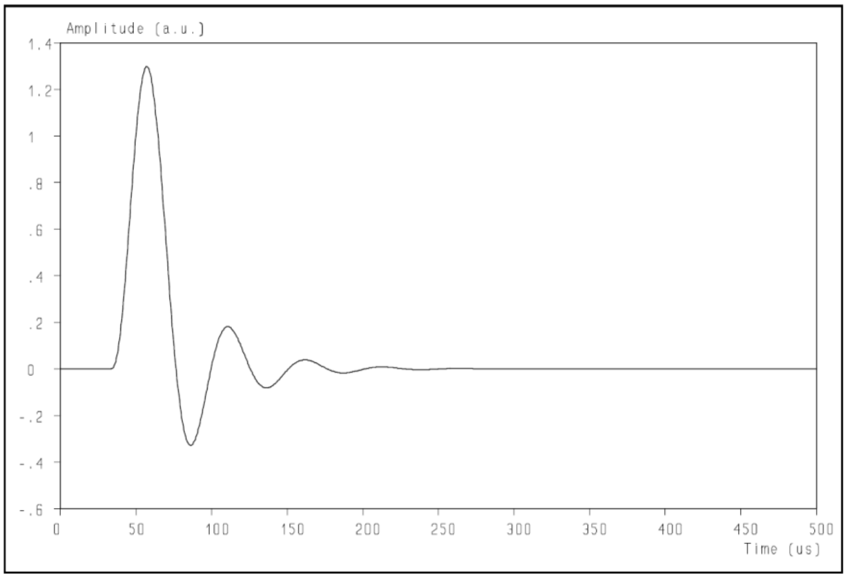

Figure 3: *The impulse response of the simulated microphone as derived by Fourier Transformation of its (complex) frequency response.*

The calculation of the convolution of the impulse responses of the loudspeaker and the microphone is straightforward and the result is shown in fig. 4: the signal in time domain as it would come out of the MuT when the impulse would be delivered to the loudspeaker directly (and ignoring non-linearities). In this case, it can also be obtained by cross correlating the input noise signal and the output signal of the MuT. The question is now whether it is possible to retrieve the properties of the MuT from the impulse response of the loudspeaker and the signal of fig. 4.

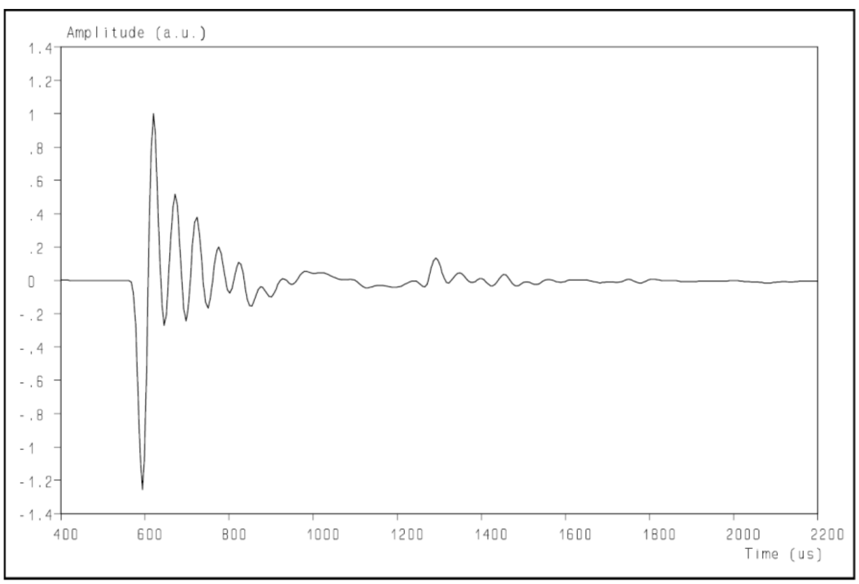

Figure 4: *The convolution of the impulse responses of the (measured) tweeter and the simulated microphone.*

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3. Results and discussion

The deconvolution process includes the following steps:

- Fourier transformation of $f_{ls}(t)$ to obtain $F_{ls}(\omega)$.
- Fourier transformation of $h_{lm}(t)$, which will yield $H_{lm}(\omega)$, the product of $F_{ls}(\omega)$ and $G_{mp}(\omega)$.
- Dividing $H_{lm}(\omega)$ by $F_{ls}(\omega)$, yielding $G_{mp}(\omega)$.
- Use $G_{mp}(\omega)$ to determine the frequency response of the simulated microphone.
- Inverse Fourier Transformation of $G_{mp}(\omega)$ to determine the impulse response of the simulated microphone.

The result for the frequency response is shown in fig. 5. The frequency response as used for the modelled microphone is shown in fig. 2 and the comparison shows a very good agreement.

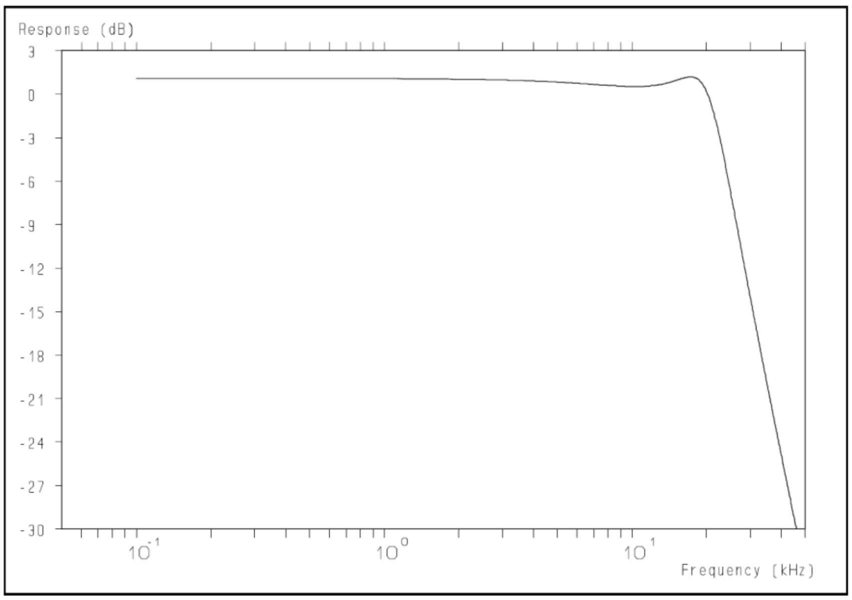

Figure 5: *The frequency response of the simulated microphone as determined from the convoluted impulse response of fig. 4. Compare with fig. 2.*

The impulse response of the microphone, using the deconvolution algorithm, is shown in fig. 6. Compared with the impulse response, directly derived from the simulated microphone as shown in fig. 3, there are only minor differences, mainly before the onset of the impulse.

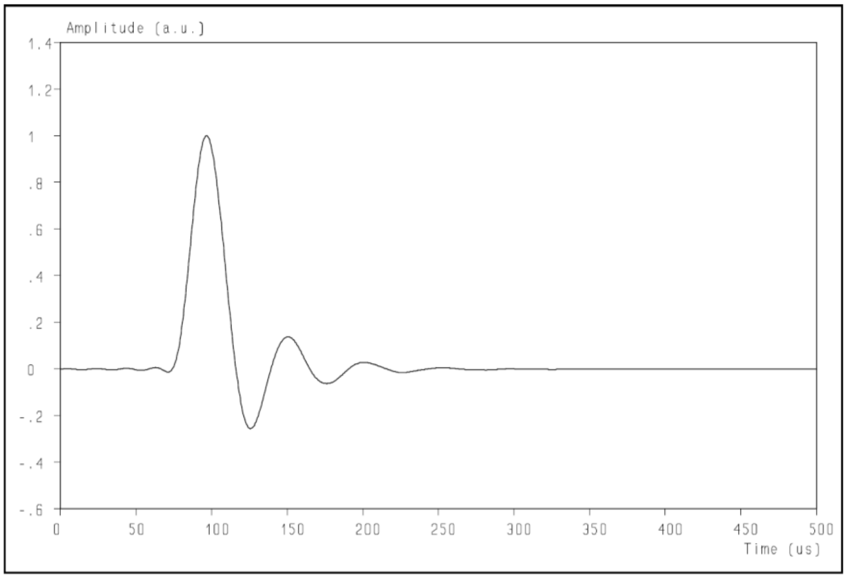

Figure 6: *The impulse response of the simulated microphone as retrieved from the convoluted impulse response of fig. 4. Compare with fig. 3.*

The slight oscillation before the onset of the impulse might be caused by aliasing effects and / or truncation errors, but it is not unlikely that a further development of the algorithm can improve the result. This is an interesting subject for further work on this technique.

The deconvolution did yield the frequency response curve very accurately. The impulse response also came very close to the actual response. As the impulse response of a real tweeter has been used, this major hurdle can be tackled. Even though the impulse response of the tweeter is wider than the impulse response of the simulated microphone, the algorithm reproduced it well. Yet, it is obvious that the shorter the impulse response of the loudspeaker is, the better the impulse response of the MUT can be retrieved. Which is why it is recommended to use a tweeter which has excellent temporal properties (ref. 2), but it will still be inferior, compared to the white noise signal applied.

A question is how well the impulse response of the loudspeaker will reproduce in time: how much will it have changed between the two successive tests? In this investigation, the same loudspeaker impulse response has been used to convolve the microphone impulse response with as is used in the deconvolution process. However, in reality the actual loudspeaker impulse response is used and is it any different from the stored impulse response, measured previously? This question should be addressed in future work.

If the loudspeaker impulse response is changing noteworthy in time, how sensitive will the derived microphone impulse response be for such changes? Such a sensitivity analysis could also be an interesting subject for future work too.

The third issue is whether it is possible to measure the loudspeaker impulse response simultaneously with the MuT test by using the measurement microphone at the same time. Do the microphone housings influence the responses of the two devices? Is the sound field at both positions sufficiently identical to make this a viable option? This also needs to be investigated further, most likely by experiments.

4. Conclusions and future work

This feasibility study shows that it is possible to determine the impulse response of a microphone using Gaussian distributed white noise and a real loudspeaker, in this case a tweeter. The data processing is sufficiently accurate, even if the impulse response of the loudspeaker is wider than the impulse response of the microphone. The use of Gaussian distributed white noise is very attractive as it does not require specialized and / or expensive equipment and it can be certified that the microphone under test will never be driven outside its linear range.

Future work should include i) experiments on the stability / reproducibility of the loudspeaker impulse response, ii) a sensitivity analysis for (small) changes in the loudspeaker impulse response and iii) the ability to determine the loudspeaker impulse response simultaneously with the test of the microphone under test.

The technique to use Gaussian distributed white noise seems a very promising approach, which needs further development. It would enable the determination of microphone impulse responses with relatively simple equipment and at low costs. As the impulse response is important for the perceived quality of the recorded sound, this would be an important extension of the microphone data, available to users.

References

1. Helen M. Jackson, Michael D. Capp, and J. Robert Stuart, “The audibility of typical digital audio filters in a high-fidelity playback system”, Audio Engineering Society, 137th Convention, Los Angeles, (2014), Paper 9174
2. “Perception of Temporal Response and Resolution in Time Domain”, Workshop #3, AES convention Berlin (Germany), 2017
3. https://earthworksaudio.com/products/microphones/measurement-series/m50/
4. Alex Khenkin, “How Earthworks Measures Microphones”, http://recordinghacks.com/pdf/earthworks/how-earthworks-measures-mics.pdf
5. J.S. Bendat and A.G. Piersol, “Random Data, Analysis and Measurement Procedures”, John Wiley & Sons, New York (1986)
6. A. Papoulis, “The Fourier Integral and its Applications”, McGraw-Hill Book Company, New York (1962)
7. A. Papoulis, “Signal Analysis, McGraw-Hill Book Company, New York (1984)
8. H.R.E. van Maanen, “Retrieval of Turbulence and Turbulence Properties from randomly sampled Laser-Doppler Anemometry data with noise”, (chapter 2), Ph.D. Thesis, Delft University of Technology (Delft, Netherlands), 1999
9. https://www.bksv.com/en/transducers/acoustic/microphones/microphone-cartridges/4939

Measuring the impulse response of microphones using noise: a feasibility study
Part 2

Dr. Hans R.E. van Maanen (Temporal Coherence B.V., Netherlands)

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1. Introduction

In the first part of this study (ref. 1), the feasibility of the measurement of the impulse response of a microphone using a real loudspeaker has been investigated. The conclusion was that this is feasible under the conditions, stated in the paper. The focus was to look at the essential step: the deconvolution of the convolved impulse response of loudspeaker and microphone, using the measured impulse response of the loudspeaker. As deconvolution can lead to inaccurate results when e.g. the impulse response of the loudspeaker is wider than the impulse response of the microphone, this step is not trivial. The above conclusion was reassuring as the results proved to be very good, both for the frequency response and the impulse response. Yet, a major step in the whole procedure had not been included in the study as reported in part 1: the noise itself.

In this second part of the study, the measurement procedure with Gaussian distributed white noise has been included in the simulation. In sec. 2, the Monte Carlo procedure will be elucidated, in sec. 3, the results will be described and discussed. In sec. 4, conclusions and future work will be presented.

2. The extension of the simulation with Gaussian distributed white noise

The technique under study is to excite the loudspeaker with Gaussian distributed white noise and to record the sound from the loudspeaker with the microphone under test (MuT). The signal from the microphone is subsequently stored to be cross correlated with the exciting noise signal. For the limiting case of the measurement time going to infinity, the cross correlation function is identical to the convolution of the impulse responses of the loudspeaker and the MuT. The basics of this theory have been presented (without proof) in part 1 and the theory itself can be found in literature (e.g. ref. 2).

In this part of the study, the procedure has been extended with a Monte Carlo simulation of the actual measurement procedure. First of all, a computer file, consisting of 5.2 seconds of Gaussian distributed white noise has been generated with a sampling frequency of 192 kHz. The same sampling frequency has been chosen as for the loudspeaker impulse measurement and the record entails 1 million samples. The procedure for its generation can be found in ref. 3; the basics are provided in Appendix 1.

N.B. It should be noted that this signal extends up to 96 kHz as white noise when directly used as input to a D/A converter, which could introduce further limitations, due to its reconstruction filter. The noise signal, as stored in the file, has been analyzed to verify whether it fulfills the requirements for this simulation. This analysis is reported in Appendix 2.

This noise signal is subsequently convoluted (in time domain) with the measured impulse response of the tweeter (see fig. 1 of part 1), resulting in the temporal response of the loudspeaker to such a signal. We will refer to the sound, coming from the loudspeaker, as the 'loudspeaker filtered noise'.

The loudspeaker filtered noise can subsequently be convoluted (in time domain) with the impulse response of the microphone. This results in the temporal output signal of the microphone as it would be recorded when the sound field, reaching the microphone, is the loudspeaker filtered noise. We will refer to this signal as the 'microphone output signal'.

The microphone output signal can now be cross correlated with the Gaussian distributed white noise which has excited the loudspeaker. In the ideal case with an infinite averaging time, the cross correlation function would be identical to the convolution of the loudspeaker impulse response and the microphone impulse response. This has been reported in part 1 and for the ease of comparison, the direct convolution result is shown again in fig. 1 below.

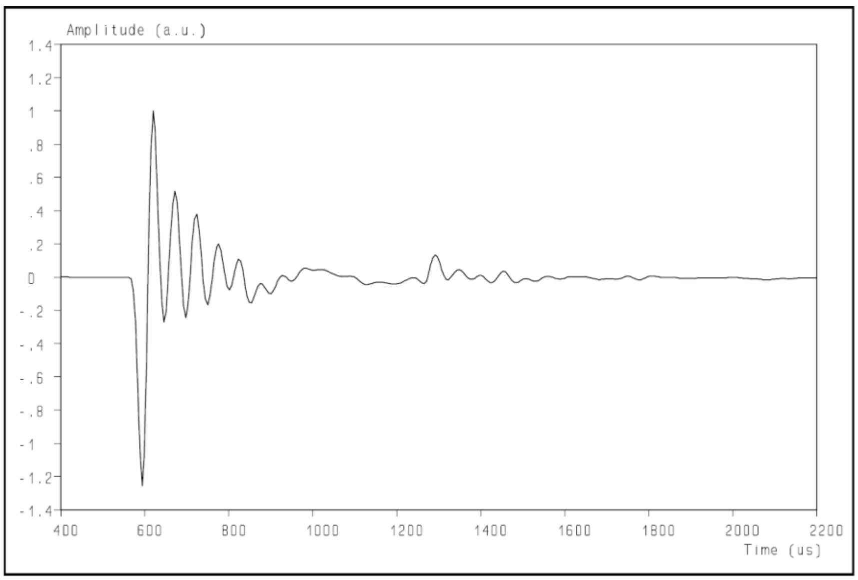

Figure 1: *The impulse response of the loudspeaker + simulated microphone, obtained by direct convolution of the measured loudspeaker impulse response and the impulse response of the microphone. For details see figs. 1 and 3 of part 1.*

The cross correlation function is shown in fig. 2 and as can be seen, the differences between figs. 1 and 2 are invisible to the unaided eye. But this does not yet mean that the results, derived for the properties of the microphone are (almost) identical when the cross correlation function is used in the deconvolution process. This is an essential requirement for the practical application of the technique. Yet, as a preliminary conclusion, it can be stated that the averaging time of 5 seconds and 1 million samples is sufficient to approach the theoretical result with a close resemblance.

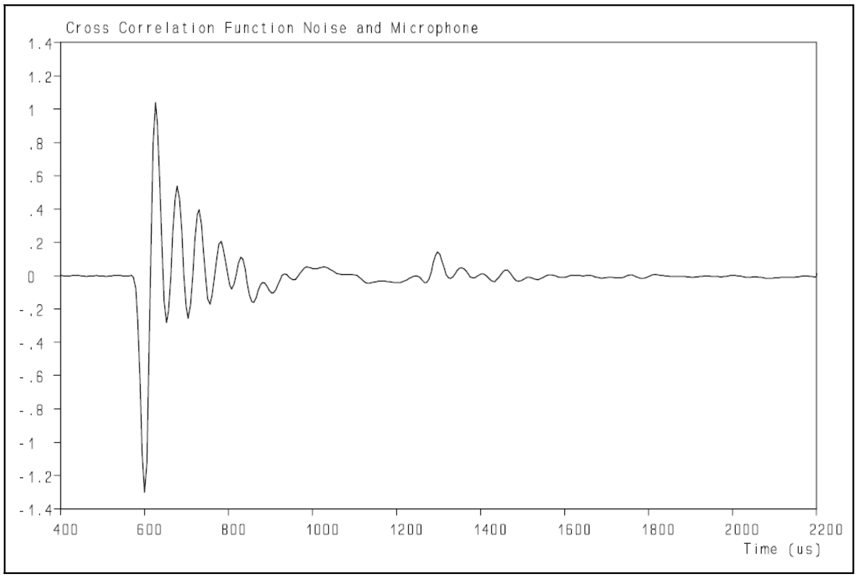

Figure 2: *The cross correlation function between the exciting Gaussian distributed white noise and the microphone output signal, converging to the impulse response of the loudspeaker + simulated microphone. Compare with fig. 1.*

3. Results and discussion

The cross correlation function of fig. 2 can be used as input for the deconvolution algorithm instead of the impulse response of fig. 1. The output of the deconvolution algorithm consists of the frequency response curve and the impulse response. These are shown in figs. 3 and 4.

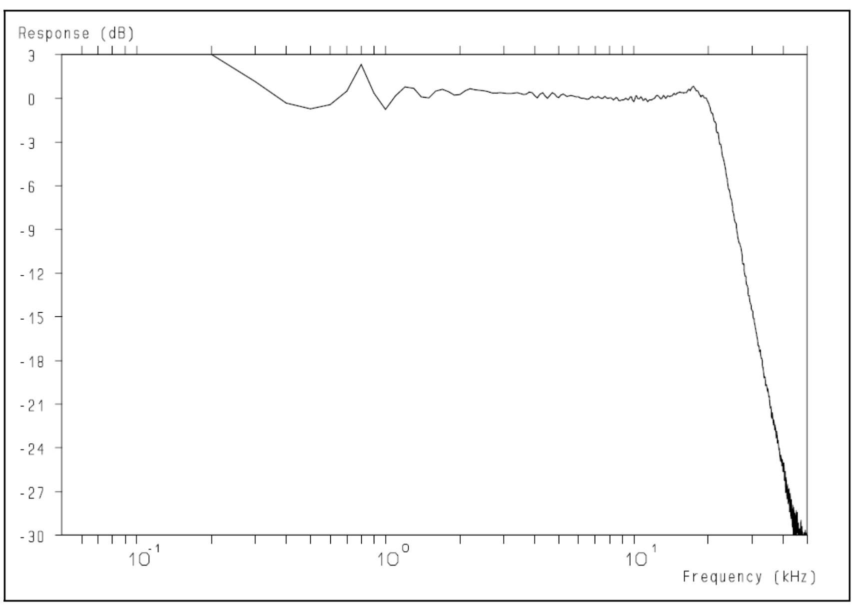

Figure 3: *The modulus of the frequency response of the simulated microphone as derived by deconvolution of the cross correlation function of fig. 2, using the measured impulse response of the loudspeaker as shown in fig. 1 of part 1. Compare with figs. 2 and 5 of part 1.*

The frequency response curve is obviously more ‘wiggly’ than the input curve, which is shown in fig. 2 of part 1 and the retrieved frequency response curve, using the measured tweeter response, shown in fig 5 of part 1. Yet, the overall properties can still be revealed.

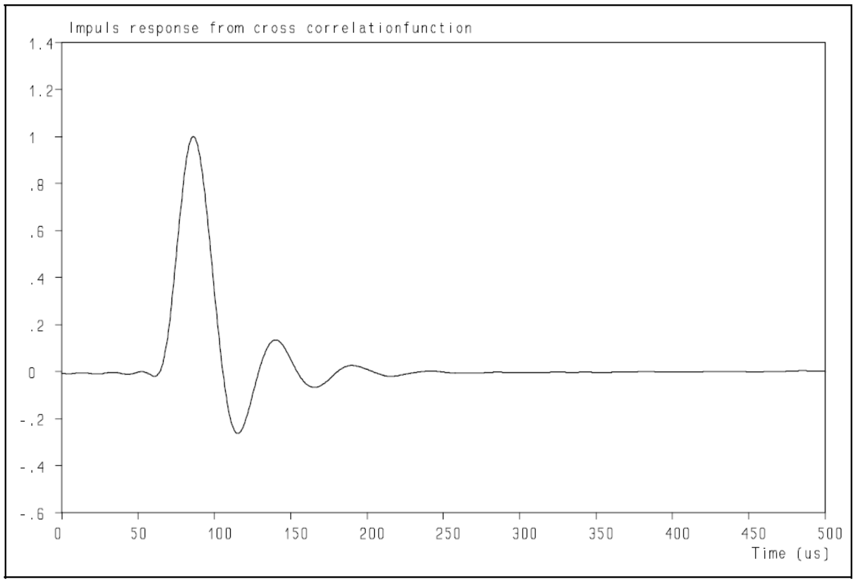

Figure 4: *The impulse response of the simulated microphone as derived by deconvolution of the cross correlation function of fig. 2, using the measured impulse response of the loudspeaker as shown in fig. 1 of part 1. Compare with figs. 3 and 6 of part 1.*

The impulse response though, is barely indistinguishable from the impulse responses as shown in figs. 3 and 6 of part 1. A very detailed comparison shows that the impulse response, derived from the cross correlation function, has a very small offset just in advance of the impulse and shows a very slight wiggle in the tail, these imperfections are neither present in fig. 6 of part 1. Overall, the retrieval of the impulse response is excellent and the degradation, compared to the response as shown in fig. 6 of part 1, can be ignored in most practical applications.

As shown in fig. 4, the variance increases at lower frequencies. This can, qualitatively, be understood by the decreasing statistics for lower frequencies: the number of cycles in the 5 sec. signal is less at lower frequencies. Furthermore, the signal strength decreases with lower frequencies, due to the properties of the tweeter as can be seen from fig. 5 below.

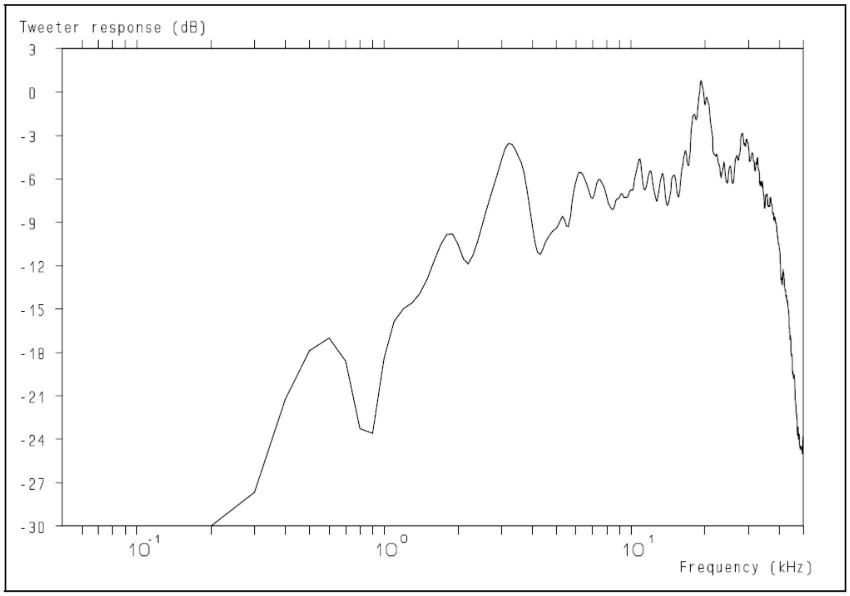

Figure 5: *The modulus of the tweeter response as a function of frequency, as derived from the measured impulse response by Fourier Transformation. Note the decrease in response below 300 Hz.*

Fig. 5 shows the modulus of the tweeter response as a function of frequency, derived from the measured loudspeaker impulse response. Note that the technique under study focusses on the short duration of the impulse response and not on an optimal determination of the frequency response curve. Other techniques are far better suited for that purpose.

A part of the slight imperfections could be attributed to the statistical approach which is inherent to the use of Gaussian distributed white noise for the determination of the impulse response. It is possible to determine the impulse response of the loudspeaker by cross correlation of the loudspeaker filtered noise and the exciting Gaussian distributed white noise. When this is done, it is not unreasonable to assume that the statistical variability in it will be similar to that in the microphone output signal. Using this loudspeaker impulse response, instead of the independently measured one, could result in less imperfections for the microphone results. To verify this assumption, the loudspeaker filtered noise has been cross correlated with the Gaussian distributed white noise input signal, as shown in fig. 6. Subsequently, this result has then been used for the deconvolution algorithm and the results are presented in figs. 7 and 8.

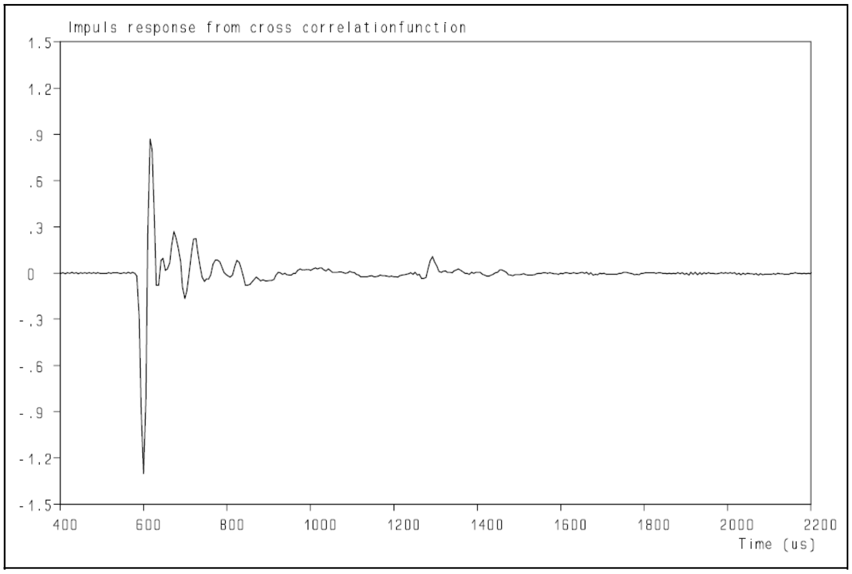

Figure 6: *The cross correlation function between the exciting Gaussian distributed white noise and the loudspeaker filtered noise, converging to the impulse response of the loudspeaker. Compare with fig. 1 of part 1.*

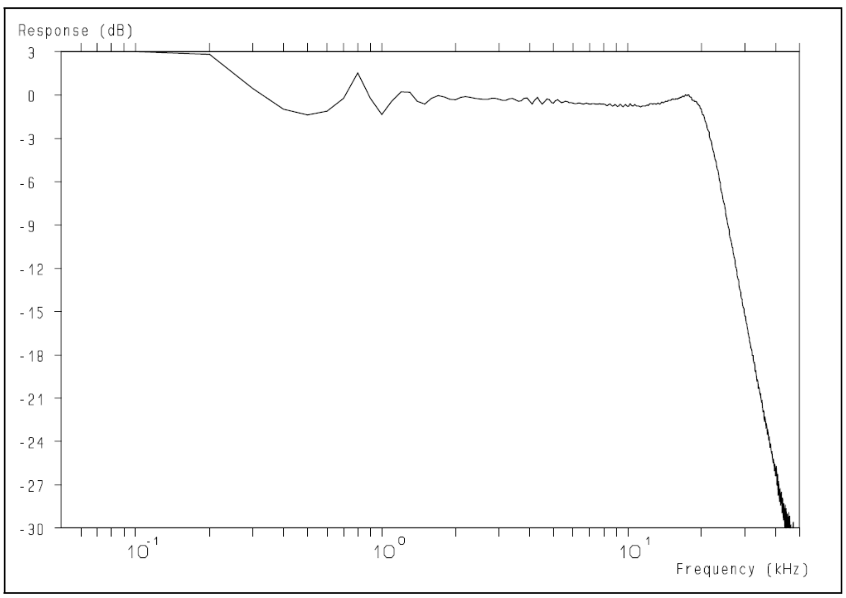

Figure 7: *The modulus of the frequency response of the simulated microphone as derived by deconvolution of the cross correlation function of loudspeaker + microphone (see fig. 2). Instead of the independently measured impulse response (see fig. 1 of part 1), the cross correlation of the loudspeaker filtered noise and the Gaussian distributed white noise (see fig. 6) has been used. Compare with fig. 3.*

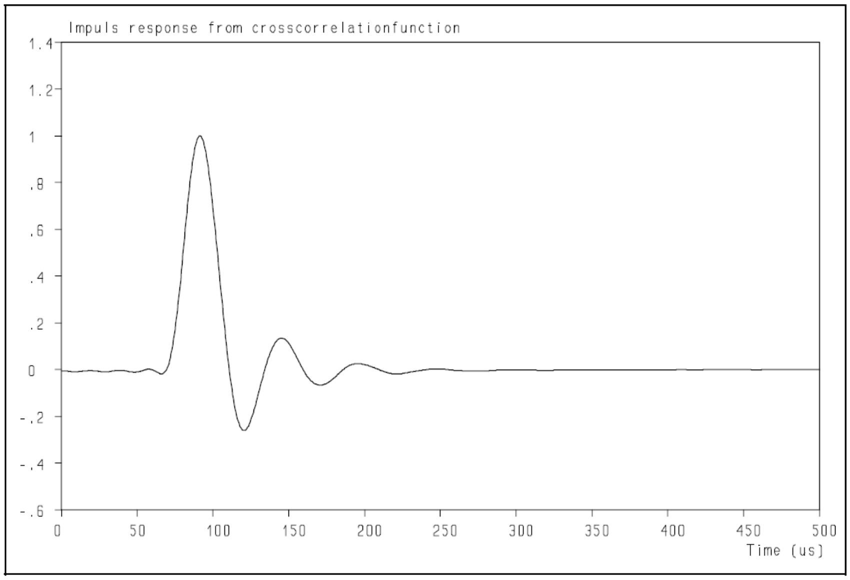

Figure 8: *The impulse response of the simulated microphone as derived by deconvolution of the cross correlation function of loudspeaker + microphone (see fig. 2). Instead of the independently measured impulse response (see fig. 1 of part 1), the cross correlation of the loudspeaker filtered noise and the Gaussian distributed white noise (see fig. 6) has been used. Compare with fig. 4.*

These results, presented in fig. 7 and 8, do show a decrease in the variability when compared with figs. 3 and 4. By enlarging the vertical scale by a factor of 25, the differences between the two results become more clear, as is shown in figs. 9 and 10. So at least a part of the variability can be attributed to the statistical nature of the measurement technique. Another advantage is, of course, that the impulse response of the loudspeaker is determined ‘on the fly’. So there is no need to verify whether the loudspeaker properties have changed between the initial measurement and the measurement of the MuT.

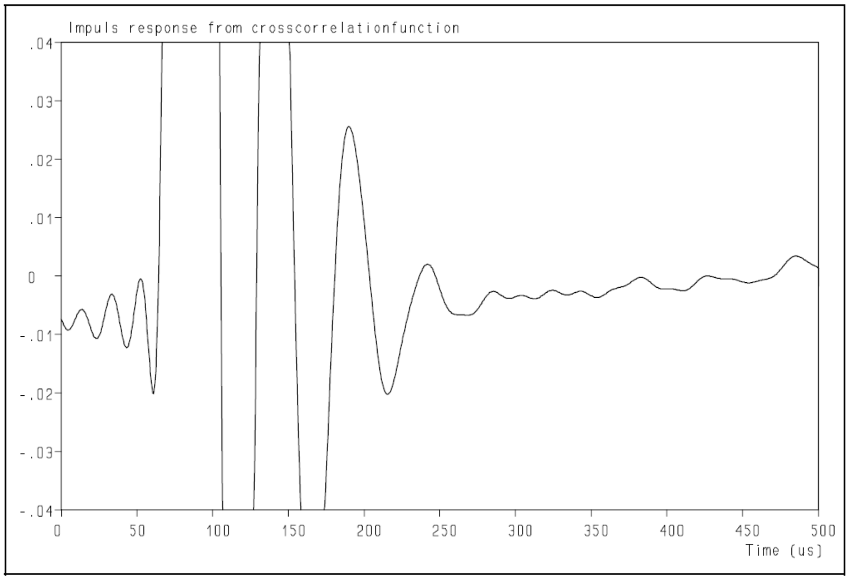

Figure 9: *The imperfections of the impulse response of fig. 4 are shown more clearly by extending the vertical scale by a factor of 25. Note the offset before the onset of the impulse response and the variability of the response in the tail.*

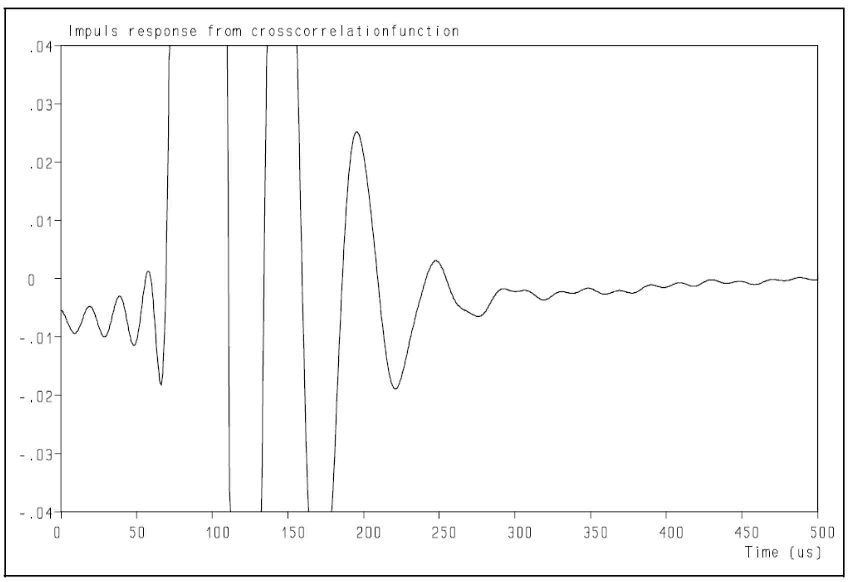

Figure 10: *The imperfections of the impulse response of fig. 8 are shown more clearly by extending the vertical scale by a factor of 25. Note that both the offset before the onset of the impulse response and the variability in the tail are reduced, compared to fig. 9.*

However, a disadvantage is that the measurement microphone and the MuT share the same space, so the housings of both microphones could influence each other. This problem can be circumvented in the following way: as the Gaussian distributed white noise signal is coming from a computer file, the noise signal can be repeated without changes. So the measurements of the loudspeaker + MuT and the loudspeaker alone could be performed shortly after each other, yet using sound fields with identical properties. This prevents that the microphone housings could influence one another and the membranes of the MuT and the measurement microphone can be positioned at the same place. Also, the actual loudspeaker properties are determined, so one does not have to rely on the loudspeaker measurements of a while ago.

4. Conclusions and future work

The extension of the feasibility study has revealed that the use of Gaussian distributed white noise for the determination of microphone impulse responses is a realistic option, even with noise signals as short as 5 seconds.

The cross correlation function, obtained in this way, is very close to the direct convolution of the loudspeaker and microphone impulse responses.

The deconvolution of the cross correlation function, using a previously measured loudspeaker impulse response, shows some slight imperfections, due to the statistical nature of the measurement technique. This is most clear when looking at the lower frequencies of the frequency response curve. This technique is not optimized for the measurement of the frequency response curve, but it focusses on the short time interval of the impulse response.

The determination of the microphone impulse response from the cross correlation function is, using a previously measured loudspeaker impulse response as input for the deconvolution process, sufficiently accurate for most practical applications.

The loudspeaker impulse response can also be determined by cross correlating the loudspeaker filtered noise with the Gaussian distributed white noise. In that case, the same noise file is used as for the microphone impulse response. Using this loudspeaker impulse response instead of the independently measured one in the deconvolution, the statistical fluctuations in both inputs partially cancel. This will give a better result as has been proven by detailed analysis of the results of the Monte Carlo simulation.

Simultaneous measurement, however, carries the risk of mutual influences on the measurement microphone and the MuT by their housings. By using two separate measurements, which use the same Gaussian distributed white noise computer file, this problem can be circumvented. This also offers the option to position the membranes of the MuT and the measurement microphones at the same place. The small price to pay is an additional measurement. Another advantage is that the loudspeaker properties are determined under almost the same conditions and at almost the same time. Therefore, its properties cannot have changed worth mentioning between the two measurements.

By using longer noise signals, of course, the statistical fluctuations can be reduced to any desired level, at the expense of longer measurement times and data processing requirements. The statistical fluctuations reduce with the square root of the measurement time, thus rapidly increasing the data processing requirements.

This feasibility study has shown that the measurement of microphone impulse responses can very well be done using Gaussian distributed white noise. By using this technique, it can be ascertained that the microphone will operate in its linear range and no specific hardware is required. So there is no reason to keep this information from the end users and it should therefore be demanded by (potential) end users to be provided by the manufacturers.

The next steps include a real measurement in an anechoic chamber, similar to the one, used to measure the loudspeaker impulse response. The most interesting is, of course, to use a microphone with a known impulse response to compare the results. Another aspect is the development / optimization of the software for data processing. Although the software, used in this feasibility study, is, mathematically speaking, correct, it is not optimized for regular application.

From a perceptional point of view, it would be very interesting to study if there is a relation between the perceived quality of microphones and their impulse responses. As different CD reconstruction filters with different impulse responses give different perceived quality assessments, there could be also differences between microphones, related to their temporal properties.

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References

1. Dr. Hans R.E. van Maanen, "Measuring the impulse response of microphones using noise: a feasibility study", https://www.temporalcoherence.nl/cms/images/docs/MikeImpulse.pdf
2. J.S. Bendat and A.G. Piersol, "Random Data, Analysis and Measurement Procedures", John Wiley & Sons, New York (1986)
3. H.R.E. van Maanen, "Retrieval of Turbulence and Turbulence Properties from randomly sampled Laser-Doppler Anemometry data with noise", (chapter 2), Ph.D. Thesis, Delft University of Technology (Delft, Netherlands), 1999

APPENDIX 1

Generation of the simulated Gaussian distributed white noise

The file, containing 1 million samples of Gaussian distributed white noise is generated using the Random Number Generator (RND) of the computer. However, the RND generator provides numbers with a rectangular distribution between 0 and 1. This distribution has to be converted into a Gaussian distribution, which is done in the following way:

- Call the RND generator, it returns a value between 0 and 1
- Subtract 0.5 from its value, so it will be in the range between -0.5 and +0.5
- Call again the RND generator, it will return another value between 0 and 1
- Subtract 0.5 from this value too, so it will be a value between -0.5 and +0.5 and add this to the previous value
- Repeat the above procedure until 21 values in the range of -0.5 to +0.5 have been summed

Statistics learn that numbers, obtained in this way, will have a Gaussian distribution, but the related variance will differ from 1 (one). This can easily be solved by normalization. The correct value for normalization is the square root of 1.75 because of the number of summations (21) and the amplitude of the rectangular distribution (0.5).

Each individual sample in the file is generated by the following instructions:

```
Sigma0 = SQR(1.75)          Normalization constant
A = RND - 0.5               Call RND generator and subtract 0.5
FOR L = 1 TO 20             Repeat another 20 times
   A = A + RND - .5         Sum all these values
NEXT L
A = A / Sigma0              Normalize
STORE A                     Store the sample in the file
```

This procedure is repeated 1 million times and the file thus includes 1 million samples of Gaussian distributed white noise with a variance of 1 (one), which can be used for the Monte Carlo simulation. The sampling frequency will be equal to the playback frequency used by the D/A converter. A more detailed description of the generation procedure can be found in ref. 3.

APPENDIX 2

Properties of the simulated Gaussian distributed white noise

The results and trustworthiness of the Monte Carlo simulation rely on the correctness of the properties of the Gaussian distributed white noise input, as the theory is based on the assumption that the input noise signal fulfills all the requirements, imposed on it. The three major requirements are:

- The probability distribution of the noise is Gaussian.
- The noise signal should be completely random, it should have no 'memory', in other words, the correlation between any two consecutive samples should be zero.
- The requirement of randomness also means that the correlation between *any* pair of samples should be zero.

To verify whether the data in the noise input file fulfill the above mentioned requirements, its properties have been verified by common signal analysis techniques. In fig. A-1, the non-normalized probability distribution is shown (wiggly line), together with the theoretical Gaussian distribution curve (drawn line). (**N.B.** Non-normalized means that the integral of the curve is not equal to 1, so it is not a probability *density* distribution, but, for the sake of simplicity, it just has been scaled to 1 (one) at its maximum.)

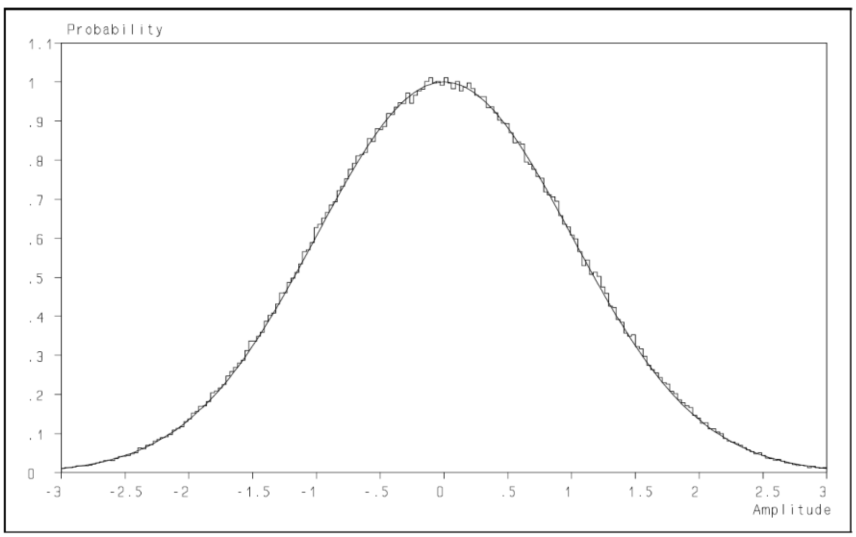

Figure A-1: *The non-normalized probability function of the noise, as used in the Monte Carlo simulation (wiggly line) and the theoretical Gaussian distribution function (drawn line).*

N.B. The probability function of the noise is wiggly, because the calculation of the distribution uses 'bins' (in this case 200) of a certain width and each sample is assigned as an element to a bin, depending on its value. Therefore, the number of elements in each bin is limited and as a result, this number will show a certain statistical fluctuation, leading to the 'wiggly' character.

It is obvious that the probability distribution follows the theoretical Gaussian distribution very well, in full agreement with the results of ref. 3 and thus fulfilling the first requirement.

The noise should be completely random, so any two samples from the noise file should be completely independent. In other words, there should be no relation between the two, a sample should not be related to *any* other sample. This can be verified by the calculation of the autocorrelation function, which should be 1 (one) for zero time shift (because a sample should fully correlate with itself) and 0 (zero) for all other time shifts (in units of the inverse of the sampling frequency).

The autocorrelation function is shown in fig. A-2 and behaves accordingly. In order to get a more detailed view of the correlation coefficients with time shifts larger than 0, the value at zero time shift has been put to zero and the vertical scale expanded by a factor of 20. The result is presented in fig. A-3. As can be seen, the correlation coefficients are very close to zero, with some remaining statistical fluctuations, as is to be expected with a limited number of samples (even if this number is a million).

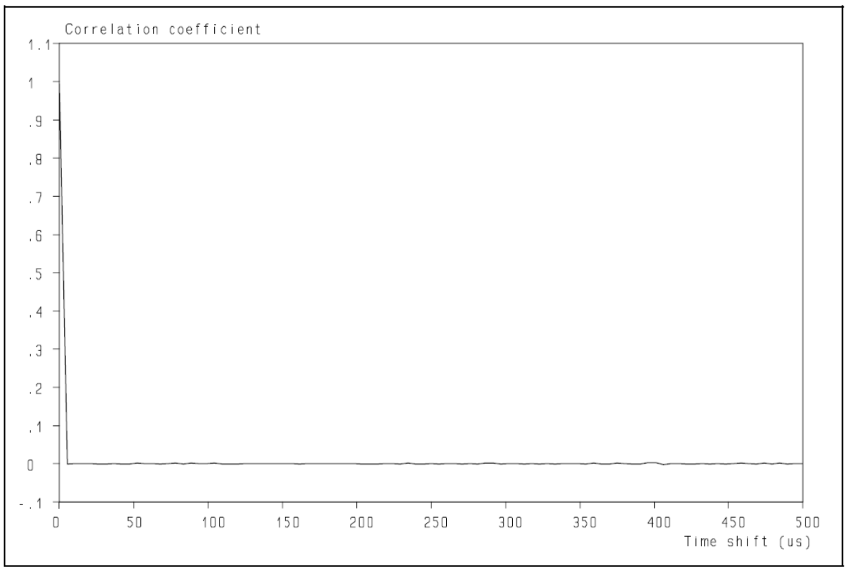

Figure A-2: *The autocorrelation function of the noise as used in the Monte Carlo simulation.*

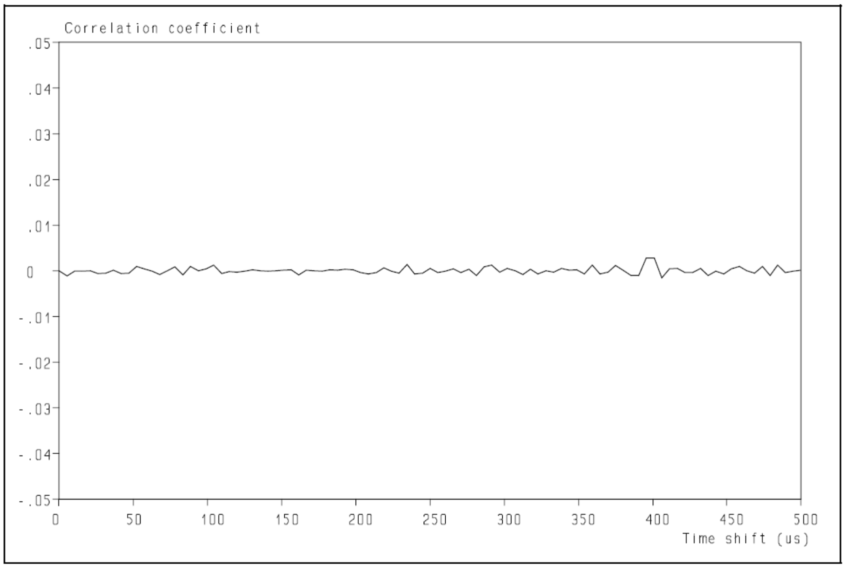

Figure A-3: *The autocorrelation function of the noise as used in the Monte Carlo simulation with an extended vertical scale. For details: see text.*

So it can be concluded that the noise in the file, used for the Monte Carlo simulation, also fulfills the second and third requirement. The noise is therefore suited for the simulation and the conclusions from the simulation can be qualified as trustworthy.