Transients in Microphones: Pop and Impulse

Martin Schneider

Georg Neumann GmbH

Ollenhauerstr. 98, D-13403 Berlin

Germany

Abstract

Signals outside the human auditory range nonetheless have some impact on audio recording. Pop and impulse measurements analyze the sub-audio and ultrasonic regions. While pop measurements can be directly understood and correlated with known subjective impressions, impulse measurements still leave some questions open. Data from both measurement types is presented and discussed quantitatively, and regarding their relevance for the audio engineer. Their dependency on microphone construction aspects is explained.

1. INTRODUCTION

The technical data and information published on microphones generally comprises many, if not all, aspects concerning the human auditory range, i.e. assuming that all sound signals of concern that are to be transmitted or recorded be localized in the spectrum between 20 Hz and 20 kHz.

That this is not always the case and that subsonic and ultrasonic phenomena have to be taken into account becomes obvious when one analyzes pop disturbances. These very low frequent air flows can produce diaphragm excursions in the range of or exceeding the maximum permissible sound pressure levels. If not mechanically overloading a microphone, the resulting output signal might still cause overload situations in the following recording chain, producing distortion in the audible frequency range. A closer look at pop sensitivities and protective measures thus seems appropriate.

On the opposite side of the frequency spectrum, impulse measurements provide an insight into the behaviour of a microphone diaphragm in the sonic and ultrasonic range. While most measurements so far have been performed purely in the time domain, with modern lab devices a frequency domain approach becomes feasible. With the possible advent of extended frequency range digital devices, see the 96 kHz discussion, microphone sensitivity above 20 kHz should be of interest as well. Furthermore impulse measurements provide information of importance to the microphone designer.

2. POP

The problem of shielding microphones from the influence of external air-flows, be it wind or pop, has been omnipresent for a long time. With the advent of pressure-gradient microphones, first bi-directional and later uni-directional, in the 20's to 30's the problems induced by wind and pop became obvious. Very large microphone output signals due to air movements are registered that
can seriously affect the quality of a recording up to the point of making it an impossibility.

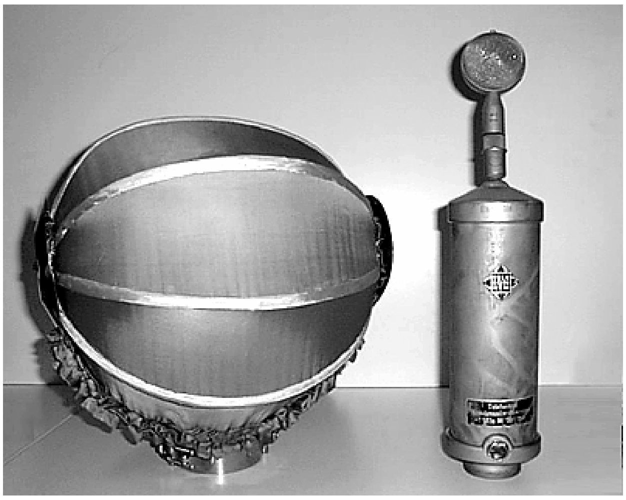

Fig. 1. CMV 3 microphone with wind shield (1930’s)

At least since the 30's the ancestors of current wind shields („zeppelins“) were thus used (fig. 1) to improve sonic quality in adverse circumstances.

These circumstances may not be perceived in their full extent by the human ear due to the fact that the ear functions as a pressure transducer. Only the field engineer working outside will fully appreciate the devastating effect of a microphone membrane fluttering in the wind.

A similar effect arises when speech is close-miked. Especially with plosive sounds, "t" and "p", an over-pressure is suddenly released when opening the lips, producing an air-flow resembling a small explosion, or a step function. In the worst case the air-flow is directly aimed at the microphone's diaphragm, causing a massive excursion of the diaphragm from its stationary position. In condenser microphones this excursion from the stationary position might only be a few microns at the most. Still, this can equal a displacement otherwise only produced by sound levels in the region of 140 dB or more.

To attenuate the adverse effects of plosives, some protection, be it of internal constructive nature or of external shielding, has to be used. To be able to analyze and quantify improvements a specification for pop measurements has been included in the new IEC 60268-4 standard on microphones [1]. There, a setup is proposed, to be evaluated still, that has been in use with some manufacturers since the 60's [2,3]. A single, low-pass filtered step function is applied to a woofer loudspeaker, thereby producing air motion resembling those found with plosives [4]. This setup’s drawbacks lie in the difficulty of producing similar stimuli, and in the comparability between different installations.

A second proposal for pop measurements is included in the standard [1], for discussion, specifying a simpler setup (fig. 2). In this second setup a function generator feeds a loudspeaker with a periodic sine signal. The loudspeaker in an enclosed chamber is driven at the frequency of 5 Hz, producing a sound pressure level of 140 dB inside the enclosure. Air motion is allowed to escape through 9 holes situated in the plate covering the loudspeaker. The microphone is positioned on-axis in front of the holes at a specified distance.

The quasi-stationary microphone output can be registerd with a standard level meter. For comparability all results were normalized re. free-field sensitivity at 1 kHz. Standard measurement distance was 10 cm. An „equivalent pop level“ [EPL] was thus calculated:

EPL = 20 * $\log_{10}$ (V_{pop} / V_{sens}) + 94 [dB SPL] with

EPL : *equivalent pop level*,
V_{pop} : *voltage measured at the microphone output*,
V_{sens} : *mic. output voltage at 1 kHz (see free-field sensitivity re. 1 Pa = 94 dB SPL).*

EPL was calculated with values of V_{pop} A-weighted (RMS) **and** unweighted (IEC 268-1,quasi-peak).

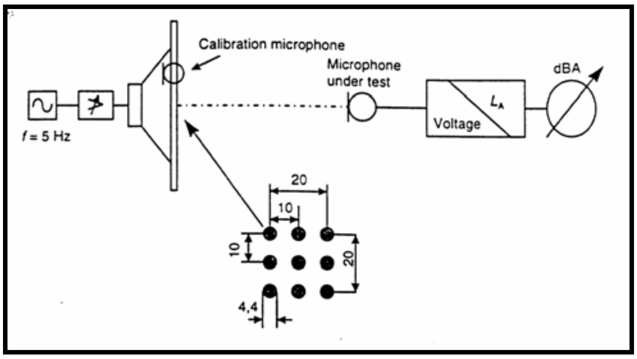

Fig. 2. Pop measurement setup [1, Annex B]

In investigations made by the IRT [5,6] it was found that the 5 Hz periodic sine-tone produces air movement strongly resembling those created by the plosives in speech. In listening tests the correlation between measured data and subjective annoyance of pop responses was found to be good. The exact positioning of the micro phone furthermore seems to be less critical than in the setup first mentioned. More details on the second setup, also used to obtain the following results, are given in [7].

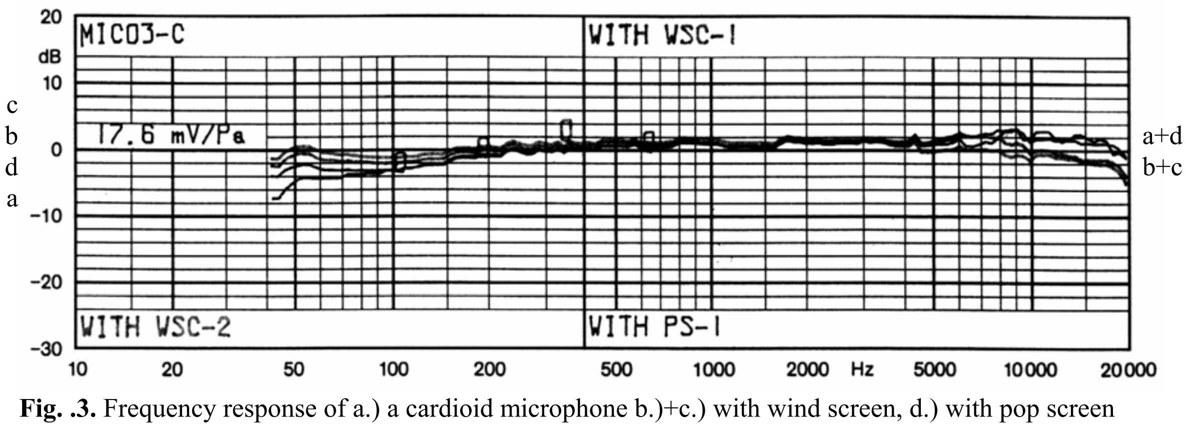

Fig. .3. Frequency response of a.) a cardioid microphone b.)+c.) with wind screen, d.) with pop screen

One aspect that has not been fully investigated yet, and is left open to discussions in IEC 60268-4 as well, remains how the differing near-field sensitivities of microphones should be taken into account by the pop measurement. As mentioned, the following data has been obtained by just normalizing with respect to different free-field sensitivities.

2.2 Transducer Principle

As already mentioned, pressure transducers are rather insensitive to pop. This is underlined by decades long practice of ENG engineers using omnidirectional microphones outdoors, making up for the non-directionality of the microphone by using it extremely close to the mouth. Pressure transducers do not have any proximity effect, can have a flat frequency response theoretically down to zero Hz, and with their diaphragms under very high mechanical tension do not react overtly to pop.

On the other side of the microphone spectrum, the low-frequency response of the pressure-gradient microphone depends to a large extent on the ability of the diaphragm to react to even the slightest differences in sound pressure between the front and the back of the diaphragm. With a diaphragm as loosely tensioned as possible, the membrane displacement can be very large, when confronted with an explosive air-flow as pop.

Microphone	Pattern	EPL unweighted	EPL A-weighted
Mic00	Omni	102 dB	72 dB-A
Mic03	Cardioid	127 dB	92 dB-A
Mic04	Hyper-Card.	129 dB	97 dB-A

Table 1. Equivalent pop level of small, cylindrical condenser microphones.

Table 1 exemplifies the dependency on the transducer principle by showing the pop sensitivities of microphones with identical proportions but different transducer principles. As the diaphragms are situated directly at the front end, shielded just by a thin metal gauze, the measured EPLs are rather high. Of course, in a pop-prone recording situation these microphones would be equipped with some wind screen or pop shield. Their helpful effect will be quantified further below.

2.3 Capsule Construction

For other microphones (see Table 2), of the large diaphragm type, the same behaviour applies, with the difference that all the equivalent pop levels are much lower. As the capsule is situated in a wire-mesh cage of some 5 cm diameter, the effectiveness of this construction in attenuating pop can be clearly estimated.

Still some exceptions do exist, as shown in Table 3. This other large-diaphragm capsule yields pop sensitivities almost independent of the chosen polar pattern, actually with a slight rise in EPL towards omnidirectional. This might be due to the diaphragm not being centrally supported as in the classic capsule design of Table 2.

Microphone	Pattern	EPL unweighted	EPL A-weighted
Mic10_O	Omni	75 dB	50 dB-A
Mic10_C	Cardioid	89 dB	63 dB-A
Mic10_8	Figure-8	93 dB	66 dB-A

Table 2. Equivalent pop level of a large diaphragm multi-pattern condenser microphone.

Microphone	Pattern	EPL unweighted	EPL A-weighted
Mic20_O	Omni	98 dB	64 dB-A
Mic20_C	Cardioid	95 dB	66 dB-A
Mic20_8	Figure-8	97 dB	71 dB-A

Table 3. Equivalent pop level of a large diaphragm multi-pattern condenser microphone.

It might be mentioned that the microphone in Table 2 is most widely used as a broadcasting and studio vocalist microphone. Perhaps the combination of sound and good pop protection had a helpful influence.

2.4 Cage Construction

That size and mesh layers of the cage construction around a capsule are most important parameters in the design of a microphone can be seen in Table 4. The same cardioid is set into different cage constructions, with 2 or 3 layers, small to large size.

Size	Mesh layers	EPL unweighted	EPL A-weighted
Small	2	109 dB	82 dB-A
Small	3	102 dB	74 dB-A
Medium	3	102 dB	71 dB-A
Large	3	95 dB	66 dB-A

Table 4. Cardioid capsule in diverse cage constructions.

The effects on the frequency response of the microphones are close to negligible, with some 0.5 dB difference in the region above 7 kHz.

2.5 Vocalist microphones

Of all microphone types in general use, it must be the dedicated vocalist microphones that show the largest variety of constructions. In general, a combination of different meshes and gauzes, together with the omnipresent foam is used. As these microphones are almost always in the near-field, at distances of between zero and, say, 20 cm, manufacturers of these microphones have always been interested in measurement methods for pop sensitivity [2, 8, 9]. Still this had so far not led to a standardized measuring method.

It might be an ideological debate if the use of foam or gauze is to be recommended. Some of the pros and cons are listed here:

Foam screens

- attenuate the high frequency response (fig. 3),
- can have an effect in distorting the polar pattern, and/or raising the low frequencies (fig.3 and [10]),
- provide simple protection against soiling of the diaphragm,
- are a low cost solution.

Wire mesh and gauze networks

- have very limited influence on frequency response and polar pattern (see 2.4 and [10,11]),
- have to be carefully designed in their combination of diverse mesh sizes and distances [11],
- are more expensive in production.

In proper constructions of both types, the main parts can be disassembled for cleaning.

In contrast to other microphones which are mostly conceived for an extended frequency range, vocalist microphones are often just optimized to the vocal frequency range. This can imply wanted coloration of the mid to high frequency response, plus a bass roll-off designed to compensate to some extent the proximity effect inherent in pressure gradient transducers.

With the very diverse frequency responses vocalist microphones can have, a full comparison between types seems difficult, taking all aspects into account. Table 5 shows EPLs of some vocalist microphones. Clearly, well-constructed mesh networks without any use of foam can be as protective as the typical foam used in many vocalist microphones.

Yet, the above measurements have to be relativated slightly, as they were made at 10 cm distance. In „lips to the mic“ situations, the wire mesh-only construction can lose some of its efficiency as the plosives are articulated directly at the first mesh, which is thus bypassed. With foamless constructions it is thus recommendable to keep at least a slight distance from the microphone.

Micro-phone	Pattern	Construc-tion	EPL unweight.	EPL A-wgt.
Mic51	Hyp.Card.	Wire mesh	81 dB	62 dB-A
Mic52	Card.	Wire mesh	dB	58 dB-A
Mic53	Card.	Mesh/foam	85 dB	64 dB-A
Mic54	Card.	Mesh/foam	88 dB	66 dB-A
Mic55	Card.	Mesh/foam	91 dB	73 dB-A

Table 5. Equivalent pop level of vocalist microphones.

2.6 Wind and Pop Screens

The same recommendation, i.e. to keep a distance applies to the effective use of pop screens. Although they find wide-spread use, their beneficial effect seems to be not universally known. Just as no ENG sound engineer would venture outdoors with his shotgun microphone or his figure-8 for MS-recordings, without adequate protective devices as windshields and windjammers, any studio should be provided with a number of pop screens for vocal recordings.

As shown in the following, the simple pop screen system of two stretched layers of very fine gauze provides the best and acoustically most neutral pop protection. With high quality microphones pop screens furthermore ensure that all sorts of extraneous „objects“, be it breath, spittle or food particles, are kept away from the diaphragms, thus keeping servicing costs low .

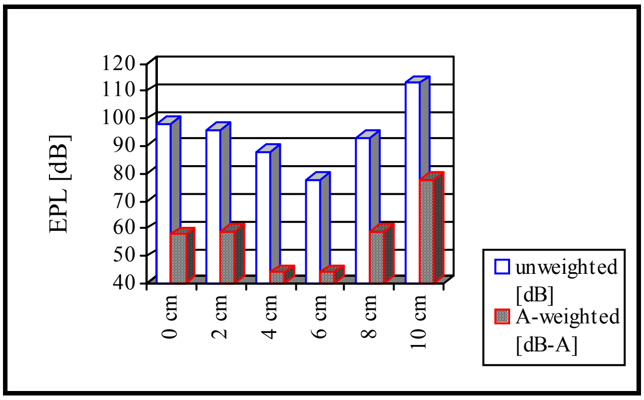

Fig. 4. Equivalent pop level with pop screen at different positions

The influence of the positioning of the pop screen on the microphone pop sensitivity can be seen in fig. 4. Zero cm distance signifies pop screen directly at the pop source („mouth“), 10 cm is with the pop screen touching the microphone cage. Optimal positioning is with the pop screen halfway between stimulus source and microphone body. This can be easily verified empirically.

The protective effect of different pop screens and wind screens is shown in fig. 5, where a small microphone with high EPL was chosen as test object. As can be seen, a small wind screen offers some protection (20 dB approx.), a large foam ball of 9 cm diameter provides 10 dB additional attenuation. Some extra 14 dB of pop attenuation arise when one switches over to pop screens, blocking off large portions of the low frequency content of the pop stimulus.

A typical wind shield was included in the test. It proved to be very adequate as well, but not as efficient as the pop screens. This should be directly traceable to the cage construction of the wind shield which produced some whistling noises, and the fact that the wind shield had just one layer of gauze. Still, for these very high „wind levels“ as they are produced by the pop apparatus, the wind shield proves to be quite efficient.

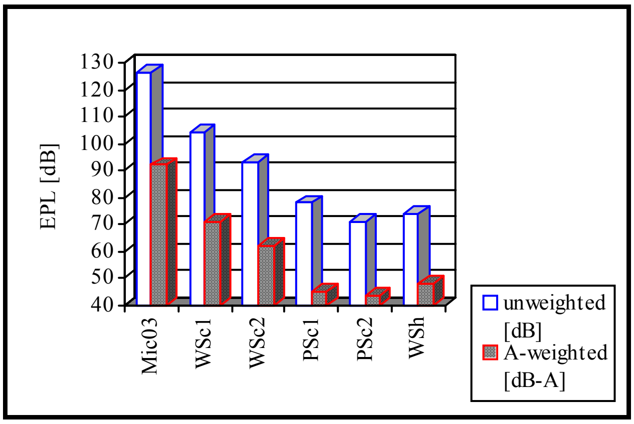

Fig. 5. Pop and wind screen efficiency

2.7 Overload

Modern condenser microphones are capable of producing very high output voltages when subject to sound pressure levels close to the microphone's specified maximum. Before the advent of transformerless microphones a typical maximum output level was 300 mV = -8 dBu. Nowadays maximum values of 2.5 to 5 volts (= +10 to +16 dBu) are standard.

The maximum sound pressure levels for which these voltages can be produced rarely arise with „normal" recording situations. A close-miked trumpet or a kick drum would be the only instruments that come to mind. On the other hand, a pop signal level, when the pop stimulus hits a diaphragm directly, of 140 dB is not uncommon. Most microphones will have some constructive protection, such as gauze, or foam, while standard small cylindrical microphones would be used with a wind or pop screen. Still, the above results have shown that impressive low frequency signals can be transmitted by a microphone.

As the pop response consists mostly of low frequent and actually subsonic components, the presence of some sort of low-cut filtering is helpful, as modern transformerless circuitry can easily have a lower corner frequency of, say, 3 Hz. Any very low frequent signal could thus reach the console or other recording device input.

Three possibilities can arise:

- The first input stage is capable of handling very high levels, and the gain is set as to provide enough headroom for any non-anticipated signal. Then a subsequent low-cut filter might be able to attenuate some parts of the low frequent disturbance.
- Headroom is not high enough, so the input stage will clip. Alternatively, the maximum input level is too low, leaving the unwanted alternative of using e.g. line level inputs.
- The basic specification of the input in the audio range is good enough, just that for subsonic frequencies distortion levels rise unproportionately. This can happen with under-dimensioned transformers [16].

Other low frequent disturbances than pop are air conditioners, traffic noise and structure-borne noise in general. Although with the increased use of pressure transducers, theoretically transmitting sound down to zero Hz, there has been some interest in electrical corner frequencies „as low as possible", in most cases a low-cut filter inside the microphone is to be strictly recommended.

2.8 Summary

The expected dependency of pop sensitivity on the transducer principle was seen in tables 1 to 3. Pop sensitivity in general increases from the pure pressure transducer (omni-directional) to the pure pressure-gradient transducer (figure-8).

A further ranking can be seen, with some overlap, from highest to lowest pop sensitivity:

1. miniature, cylindrical microphones,
2. large diaphragm microphones, „free" diaphragm
3. large diaphragm microphones, centrally supported diaphragm
4. dedicated vocalist microphones

Small, cylindrical microphones have the highest sensitivity to pop as their construction in general provides no further protection than a thin metal gauze, and the diaphragm is thus freely accessible for the pop stimulus.

Large diaphragm microphones have the advantage that the capsule is situated in a wire mesh cage, acting like a wind shield. Protection increases with size of the cage, and careful selection of multiple, different meshes. Here a compromise has to be found between an acoustically ideal open construction, eliminating reflections inside the cage, and an ideally protective construction.

Dedicated vocalist microphones, as to be expected, show the least influence to pop stimuli. It has to be pointed out however, that the usual foam does not necessarily provide the best pop attenuation. A well-constructed network of different meshes provides suitable protection as well, without affecting the acoustical accessibility of the capsule for the intended audio signal.

Some issues regarding the use of weighting functions and the inclusion of the frequency response into the measured data are still open to discussion. With the high SPLs, at low frequencies, usually associated with pop disturbances, the use of the unweighted data seems straightforward. Still, for some applications, e.g. an announcer's microphone at 50 cm distance, the relative pop level might be small enough as to allow the use of the A-weighting curve, which is just intended for low level signals. As more measurements will be made, experiences will hopefully lead to a standard recommendation taking all aspects into account.

3. IMPULSE

Microphone impulse response measurements as they are performed today have been known since at least 1939. In [12] Weber, well-known from the Braunmühl-Weber patent of multi-pattern microphone capsules, analyzed the spectral distribution of two impulsive sources: pistol shots and electrical spark discharges. Here we have a similarity to the two pop setups mentioned in chapter 2: pistol shots provide single impulses, while spark discharges can be reproduced periodically in a setup shown in fig. 6.

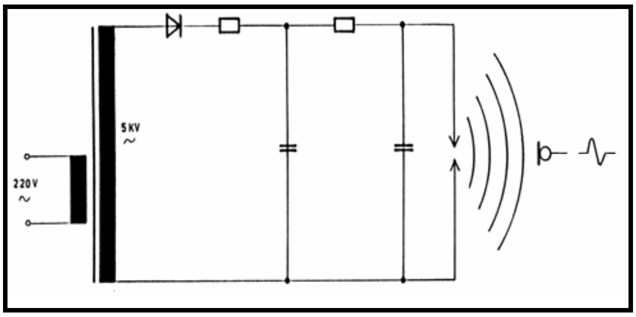

Fig 6. Impulse measurement setup [13]

According to theory the impulse response of a linear time-invariant system describes the system fully. The Fourier transform of the impulse response yields the complex spectrum, i.e. the frequency and phase response, which contain the exact same information as the impulse response, just represented in the frequency domain.

The above argument is based on three assumptions:

- that the impulse be an ideal delta (or Dirac) impulse,
- that the system be linear,
- that the system be time-invariant.

With the system under test being a microphone, all three assumptions are hard to fulfil. Time-invariance may be assumed, but linearity definitely ends e.g. with very large displacements of the diaphragm. Lastly, an ideal acoustic impulse cannot exist. Weber has shown in [12] and Cremer refined in [14] that the shape of an acoustic impulse as in a spark discharge between two electrodes will have the appearance of a heavily damped period close to the aperiodic boundary. As in a miniature explosion, a short positive overpressure peak will be followed by a more extended time span of underpressure, when the suddenly expanded air flows back into the centre of the „explosion“.

The spectrum of such an impulse is determined by a flat spectral maximum, depending on the energy of the discharge, with the two slopes towards higher and lower frequencies falling off at 6 dB per octave. Axes in fig. 7 are with arbitrary normalization.

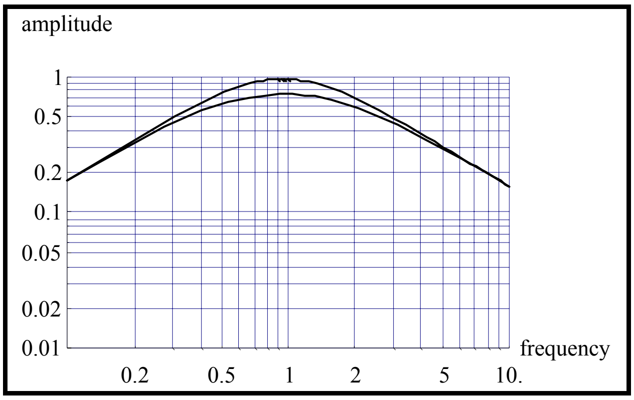

Fig 7. Spectrum of a spark discharge according to [12] (lower curve) and [14] (upper curve).

Still, a spark discharge produces a wide-band spectrum especially helpful for the analysis of the high frequency response of microphones, i.e. their behaviour when confronted with transient signals of very short duration. In a time where the audio industry is discussing the possible adoption of 96 kHz sampling rate as a future standard, the responses of microphones above 20 kHz will have to be looked at. Even though the main aim of an elevated bit rate is the simplification of A/D conversion circuits and the corresponding steep anti-aliasing filters one should know if the microphones will deliver any signals, if there should be any of relevant level, in the range from 20 to 48 kHz.

3.1 Time Domain

As spectral analysis of impulse responses necessitates either Fourier transform devices or expensive filter banks, as in [12], most publications have investigated the time domain, i.e. the actual impulse response. By comparing the response of a microphone to its theoretical ideal some relevant information can be obtained.

The typical data provided in the technical manuals of microphones includes a frequency response curve. As seems to be the case, microphones with similar frequency responses can subjectively sound different. Seen from a signal processing point of view, a microphone acts like a band-pass filter. Different implementations of filters with the same frequency response can differ in their phase response, and conclusively in their impulse response. As phase response measurements of microphones can be especially difficult, the impulse measurement method provides a feasible alternative.

With this method, hidden aspects of microphone construction can be discovered that are not directly apparent, e.g. resonance effects and filtering.

Fig. 8 shows the impulse responses of three different omnidirectional microphones, all of similar small capsule dimensions.

In fig. 8a. a standard diffuse-field equalized omni-directional is shown. It has a very wide-band impedance converter and contains no filter stages. The impulse response is very short and only minimal amounts of overlaid, very high frequency „ringing“ appear. The microphone shows no resonance effects.

Fig 8b. is a microphone with a highly undamped diaphragm. The frequency response would be similar to the spectrum of the spark discharge in fig. 7, i.e. falling off steeply towards the limits of the frequency range, were it not for the filter stage which smoothes out the frequency response. The resulting frequency curve is very flat, but the impulse response shows some hundreds of microseconds ringing with a „beat“ frequency of 22 kHz approx. Whether this type of „smooth“ ringing is noticeable, is open to discussion. It can be argued that such a microphone subjected to transient sound signals might be kept in a state of „constant ringing“. Still, no evidence in either direction has been published.

What definitely can be heard is the devastating effect of an impulse response as shown in fig. 8c. Although intended as an omnidirectional microphone, which is rather simple to design for a good frequency response, the microphone shown here has three discrete +6 dB peaks at 3 kHz, 6.5 kHz and 15 kHz approx. The extended, irregular ringing points to coupled resonators in the capsule housing, as was here the case.

Further examples of impulse responses are found in [15]. There some remarks are made also concerning the „ideal“ impulse response. The length of the first half-wave of the impulse response correlates with the upper frequency bound of the microphone. A wider half-wave points to a lower maximum frequency, a very short first half-wave can be achieved with pressure transducers as the diaphragm is under high mechanical tension and the resonance frequency is correspondingly high.

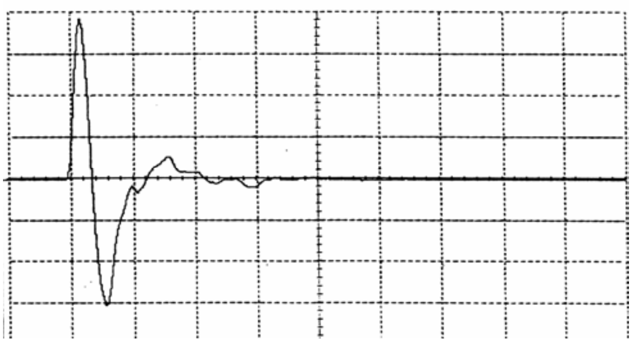

Fig 8a. Impulse response of a small omnidirectional microphone (50 µs per division)

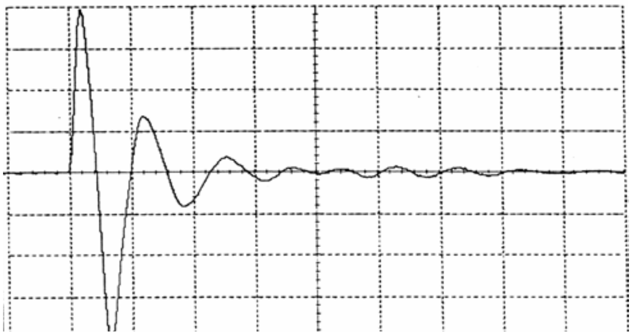

Fig 8b. Impulse response of an omnidirectional microphone with filtering (50 µs per division)

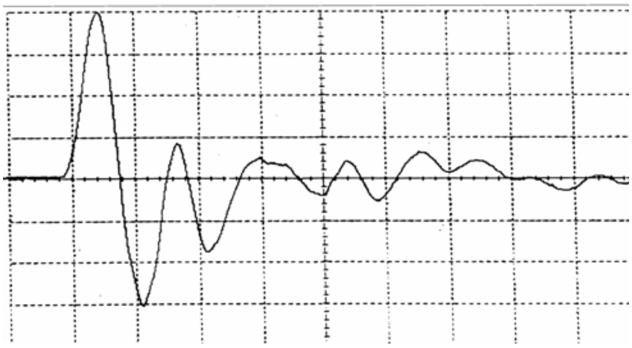

Fig 8c. Impulse response of an omnidirectional microphone with resonances (50 µs per division)

It is further argued that a smooth transition at the upper corner frequency should be aspired to. The frequency response should have a gentle slope above the wanted upper corner frequency, leading to a damped wave of a certain length. A steep cut-off in the high frequencies can produce a short ringing phase, but of possibly relatively high amplitude.

3.2 Frequency Domain

Less data has been published concerning the transformation of the impulse response into the frequency domain. With digital oscilloscopes having become common tools this has become a realistic approach.

Although the procedure appears to be quite simple, it still poses many problems that have yet to be looked into, so the following is to be seen as part of a „work in progress“.

The first aspect is that the frequency content of the actual spark being used has to be determined. As this can only be done with another (measurement) microphone, questions arise whether one is actually measuring the spark or rather the measurement microphone itself.

Fig. 9 shows the calculated spectrum of the impulse response of a 1/2" measurement microphone of the free-field equalized type. The upper corner frequency of the microphone according to its specifications lies in the range of 40 kHz. Up till that bound the calculated Fourier transform corresponds to some extent with the theoretical spectrum of fig. 7. Beyond 40 kHz the microphone's response begins to fall off, showing some peaks and dips in the measured spectrum. These cannot be directly traced to any reflection in the apparatus or any characteristic of the spark so should be part of the microphone's characteristics.

The calculated impulse spectrum of a standard omnidirectional studio microphone (fig. 10) shows some correspondences, with an additional 8 kHz peak due to pressure congestion (seen also in the frequency curve), and a not so extended frequency range, with the higher peaks positioned differently, but in a similar pattern that will have to be analyzed yet.

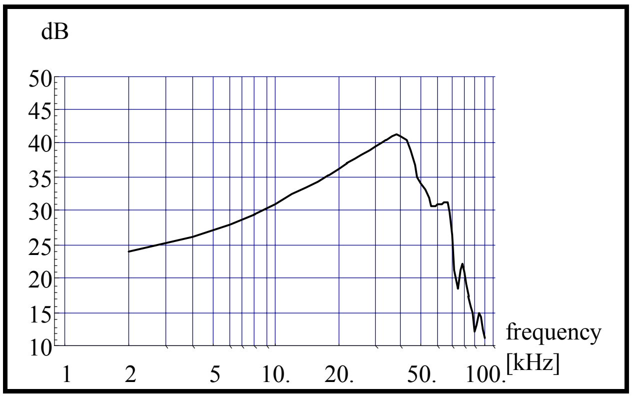

Fig 9. Calculated spectrum of the impulse response of a measurement microphone

Further tests will have to be made with other, smaller measurement microphones in order to establish the spectrum of the impulse. With the impulse spectrum determined, the calculated spectra can be normalized, yielding the wide-band frequency response of the microphone under test, measured with impulses.

With a method of measuring frequency responses as a possible outcome of the investigations under way, the impulse method will not be able to replace current methods. Most of the impulse energy is localized in one spectral peak, yielding only limited energy in the lower frequency bands. As the data acquisition hardware at high sample rates has only limited accuracy, low frequency data will always have limited reliability.

For accuracy in the mid to lower audio frequencies the measurement time span will have to be expanded as well. In the current setup, the free time span until the arrival of the first reflections from the housing is only 500µs, thus producing relevant data only in 2 kHz intervals. The setup will have to be rearranged to anechoic surroundings.

One of the interesting points will nevertheless be if an exact correspondence can be found between the frequency response measured the „classical“ way, with sine tones or sweeps, and measured via impulses. Any non-correspondence would point to limits in the assumption of microphones as linear, time-invariant systems.

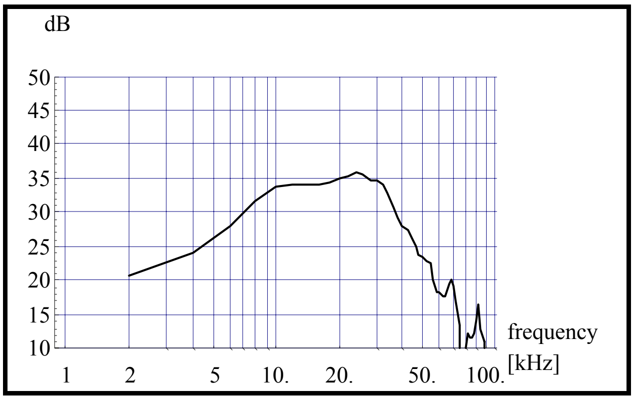

Fig 10. Calculated spectrum of the impulse response of an omnidirectional studio microphone

4 CONCLUSIONS

Measurements at the boundaries of the standard audio frequency range have been discussed. Pop measurements are being standardized at the moment, and pop can be seen as a rather well understood phenomenon. With appropriate microphone designs and recording technique, pop should not pose a major problem anymore. Possibly, a value specifying pop sensitivity will be included in the future in standard manufacturer's specifications.

Impulse measurements, on the other hand, are yet to be fully investigated. Some aspects of microphone characteristics not yet specified in data sheets can be traced with the time domain approach. Whether the frequency domain will produce further insights remains yet to be seen.

Bibliography

[1] IEC 60268-4: Sound System Equipment - Part 4: Microphones

[2] (Sennheiser), Messung der Popp-Empfindlichkeit, Internal Paper, Sennheiser, 1966

[3] Werner, E., Dependence of Microphone Pop Data on Loudspeaker Properties, J. Audio. Eng. Soc., Vol. 38, No. 6, June 1990

[4] Wienhöfer, W. & J. Sennheiser, Measuring Equipment for the Estimation of „Pop“ Sensitivity of Microphones, J. Audio. Eng. Soc., Vol. 36, No. 6, June 1988

[5] Wollherr, H. & H. Ball, Meßtechnische Bestimmung der Pop-Empfindlichkeit von Mikrofonen, Fortschritte der Akustik, DAGA 1991

[6] Wollherr, H., Messung des Pop-Effektes bei Mikrophonen, IRT discussion paper, July 1991

[7] Schneider, M., Pop Measurement, Low Frequency Response and Microphone Construction, 104th AES Convention, Amsterdam, 1998, to be published

[8] Bartlett, B., Hum, Pop, Thump and Other Microphone Noises, db Magazine, Dec. 1980

[9] Schulein, R.B., Development of a Versatile Professional Unidirectional Microphone, J. Audio. Eng. Soc., Vol. 18, No. 1, Feb. 1970

[10] Wuttke, J., Microphones and Wind, J. Audio. Eng. Soc., Vol. 40, No. 10, November 1992

[11] Peus, S., On the Development of a new Vocalist Microphone, Lecture held at AES Italy Meeting, October 1991, Georg Neumann GmbH

[12] Weber, W., Das Schallspektrum von Knallfunken und Knallpistolen mit einem Beitrag über Anwendungsmöglichkeiten in der elektroakustischen Meßtechnik, Akust. Zeits. 4, Nov. 1939, English Translation by Beranek, L.L., in: Jour. Acous. Soc. Amer., 12, 210-213 (1940)

[13] Cremer, L., Über das Ersatzschema des Kugelknalls, Akust. Zeits., 5, 1940

[14] Peus, S., Impulsverhalten von Mikrophonen, Internal Paper, Georg Neumann GmbH

[15] Peus, S., Microphones and Transients, db Magazine, May 1977

[16] Wuttke, J., Betriebsverhältnisse von Mikrophonen bei Wind und Pop, Fernseh- und Kino-Technik, Nr. 7, 1987, p.300- 306