Proceedings of the Institute of Acoustics

REFINEMENT OF SLM DESIGN TO FURTHER REDUCE MEASUREMENT UNCERTAINTIES

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1 INTRODUCTION

The paper describes the design philosophy for a new handheld sound level meter: Nor-118. The specifications for the sound level meter are based on the proposal for the new International standard for sound level meters IEC 61672. Easy to operate and high precision in the measurements are the main concerns.

2 EASY TO OPERATE

2.1 Parallel measurements of a number of values

As a manufacturer of sound level instrumentation, we some times get a call from a customer: "What shall I do? I have measured noise for a week, and now afterward, I realise that I have used a wrong time constant. How will that affect my maximum values"? Some times when we know the type of noise we can advise the customer that the normal difference for such type of noise may be a certain number of decibels, but in most cases the user have to make new measurement for another week.

How can such accidents be avoided? First by having a clear indication of the set-up of the instrument. Furthermore it should be as few settings as possible to adjust before the measurement is started. Modern instruments have a lot of measurement capacity, which may be used to obtain flexibility in the way the instrument is used. However, flexibility has one very serious cost: Every decision that has to be made before the measurement is started, can be a wrong decision. It can be wrong because the operator was not aware of the requirements, but in most cases the user has forgotten to set the parameters correctly.

In the design of the new sound level meter, the design group has focused on that only a limited number of measurement parameters should be set before the measurement was started. Therefore the instrument measures a number of parameters simultaneously. When the measurement is finished, the result may be automatically stored and the wanted parameters can afterwards be picked. This is usually done later when the user downloads the result in his or her laboratory.

The new sound level meter always measures the sound with two frequency weightings: A and C or A and Z. The Z-weighting is a new weighting defined in the proposal for the new International standard for sound level meter IEC 61672. The weighting may be called flat or unweighted, as the design goal attenuation is zero decibels from at least 16 Hz to 16 kHz. For each of the frequency weightings, the signal is measured with three different time-weightings: F, S and Impulse.

The maximum and minimum level is measured for each of the time weightings as well as the instantaneous value during the measurement. In addition, for each of the frequency weightings, the integrated-averaging-value, or equivalent-value Leq, the sound exposure level and the peak value are measured. The equivalent level and the sound exposure level are also calculated for the impulse time weighting. This adds up to a total of 22 measurement values after a single measurement period.

The instrument may be delivered with an option for frequency analysis in the range 6,3 Hz to 20 kHz. If this is the case, all parameters listed above, except the peak value, are measured for each octave-band or one-third-octave band as well.

If the instrument is delivered with the option for distribution, the distribution or percent levels based on the F time weighting are measured for each frequency weighting and frequency band.

All results may be stored in the instrument. And it certainly is easier and safer to make the right selection after the measurement than before. And in case the user did a wrong selection, he or she just have to make another readout; the user don't have to start the measurement over again.

2.2 Get rid of the level range control

The level range control, which traditionally has been an integrated part of any sound level meter, may be difficult to set correctly. This is especially the situation for unattended measurements on sources with changing noise levels. Only after the measurement you are able to tell if the selected range was the best. Modern instruments with a large measurement range make it easier than using old sound level meters with a very limited range.

In the design of the new sound level meter we wanted to get rid of the level range control completely. The instrument has one large measurement range in excess of 120 dB and covers levels from the noise floor of the microphone to levels were the preamplifier start to overload. The large dynamic range has been obtained by using two measurement chains in parallel: one for low and medium levels and one for medium and high levels. The instrument automatically and instantaneously selects the most reliable range in a way so it looks like one huge measurement range for the user. This even works for the frequency analysis. The result is that there is no need for a range control and the instrument comes with a display range control instead. In contrast to a range switch, the displayed range may be changed during an ongoing measurement without affecting the measurement result.

2.3 User interface

A relative large, graphic display facilitates an easy readout of the result. When the results of a measurement are stored, the instrument will automatically assign a filename. It consists of the day the measurement was started, and a consecutive serial number. Together with the measurement result, the time of the day is stored as well as all important set-up parameters.

3 MEASUREMENT ACCURACY

3.1 Digital circuitry

During the last ten years analogue design in sound level meters has gradually faded away and has been replaced by digital circuitry. The gain is the precision, which earlier only was available for larger and more expensive laboratory equipment.

The sound varies over a large dynamic range and even a high precision voltmeter is usually unable to measure the short impulses, often found in a sound signal, with sufficient accuracy. In order to indicate the effective value of a signal, the signal has to be squared. Most analogue detection methods suffer from limited crest factor capabilities. When the microphone signal has been converted to a digital signal, the squaring is just a computational procedure, which can be made to any required accuracy if the digital signal processor has sufficient capacity.

The new International standard for sound level meters requires much tighter tolerances for the response to short impulses than the present standard. With digital detection, these requirements are more easily satisfied.

Although the input circuitry and analogue to digital converter is still an analogue circuit, the rest of the sound level meter is digital. Digital realisations of weighting networks and filters have the implication of no influence from the environmental parameters as temperature and humidity.

3.2 The combined level range setting

The signal from the microphone preamplifier is fed into two input amplifiers in parallel: one with low gain for high level signals and one with high gain for lower signals. Each amplified signal is converted in one dual channel analogue-to-digital converter. The effective sampling frequency of the converter is 48 kHz, and converts each signal with a resolution of 18 bit. The analogue-to-digital converter applies the sigma-delta principle. This means that sampling the signal at a few megahertz and then filtering and decimating the digital result obtain high resolution. The anti-aliasing filtering can thereby be solved by a combination of digital and simple analogue means.

The two digital signals are scaled and combined to obtain a dynamic range of more than 120 dB. If the signal from the channel with the highest gain is overloaded, the resulting sample is always taken from the channel with the lower gain. When a microphone with the normal sensitivity is used, the maximum peak level that can be measured is 140 dB, the A-weighted noise with the input short-circuited is less than 10 dB or more than 130 dB below the maximum peak level

3.3 Compensating the self-noise

The lower limit for the level range is normally set by the self-noise of the microphone and preamplifier. For the normal 1/2" microphone, the A-weighted noise floor is typically below 18 dB. If you try to measure an acoustic source with the same level, the result will be 3 dB higher due to the addition of the levels from the acoustic source and the electrical noise. The instrument can be set up to subtract the electrical noise level and thus increase the linearity in the measurement of levels close to the noise floor. Although the lower limit for level measurement is in general specified as 24 dB and based on no self-noise compensation, good measurements can be made down to 18 dB when the compensation is active.

3.4 High pass filter

Noise from wind and slamming of doors are normally dominated by very low frequencies. In order to remove these signals as early as possible in the measurement chain, the input circuitry of the sound level meter is equipped with an third order maximum flat high pass filter. Figure 1 shows the frequency response.

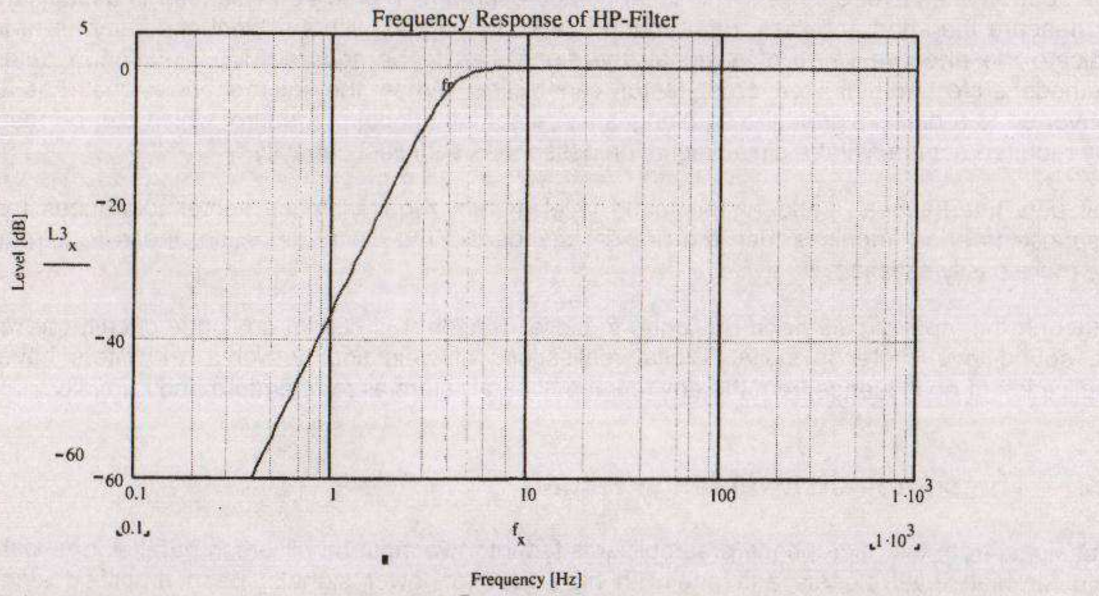

Fig 1*. An analogue high pass filter of third order is used for removing the low-frequency noise.*

4 THE ACOUSTIC ENVIRONMENT

4.1 Instrument casing

It is in general difficult to measure sound without disturbing the field to be measured. In order to reduce the effect of placing a sound level meter in the field, the sound level meter should be small and have a design, which reduces the influence from case reflections. Figure 2 shows a photograph of the new sound level meter. The length of the preamplifier reduces the effect of the case reflections as observed by the microphone. The effect of the case reflection from plane waves approaching the microphone along the main axes is shown on figure 3.

Fig. 2 *The Nor-118 instrument casing*

4.2 Windshield

A windshield is often placed on the microphone to reduce the noise from wind and for protecting the microphone from dirt. Even if the modification of the frequency response is low, especially at lower frequencies, the change may bring the response to or even above the required tolerance limits. A typical change in the frequency response with and without windshield is shown on figure 4.

In order to improve the measurement accuracy, the user may select a compensation network to compensate for the effect of the windshield.

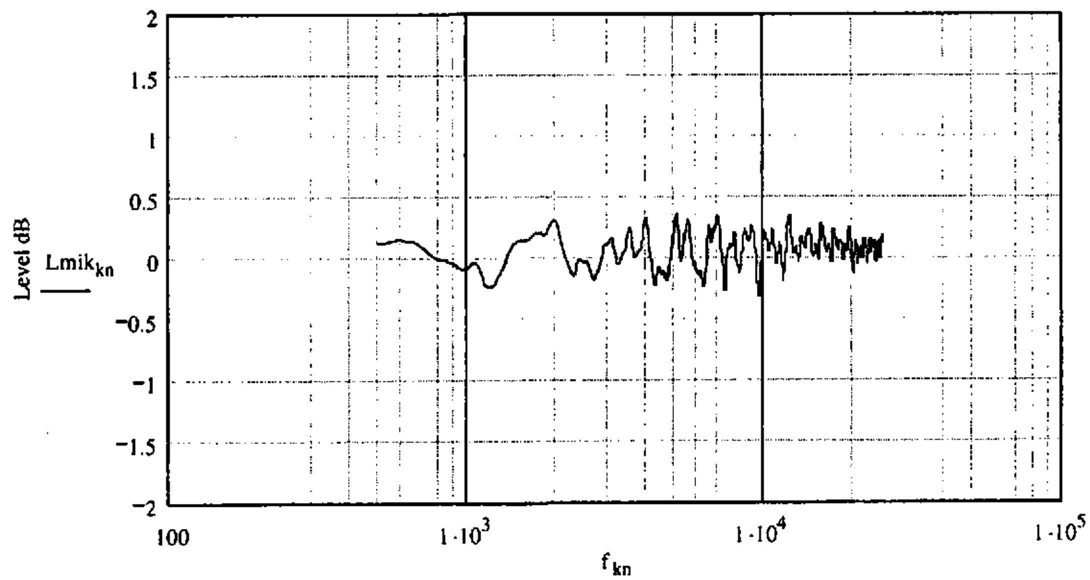

Fig. 3 *Measured effect of reflections from the Nor-118 instrument casing. The response is the free-field response for plane progressive waves approaching the microphone from the front along the axes of symmetry for the microphone-preamplifier.*

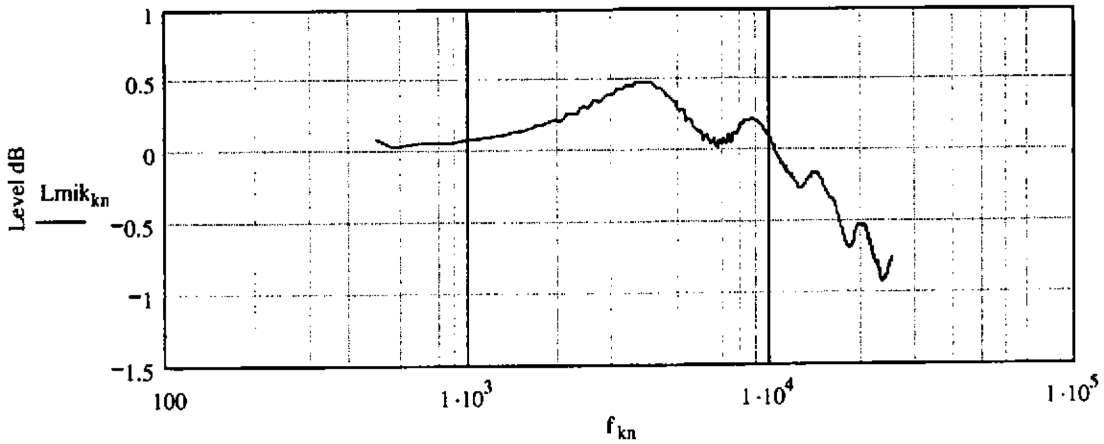

Fig. 4 *The diagram shows the change in the frequency response when the sound level meter is fitted with a windshield. The response is measured for plane waves approaching from the front of the microphone.*

4.3 Random response

The normal 1/2" microphone supplied with the sound level meter is designed to have a flat frequency response for sound approaching the microphone from the front – free field conditions. If used to measure diffuse sound, the sound will approach the microphone from all directions (random incidence) and the integral response will loose some sensitivity for the highest frequencies. Using a microphone designed to have a flat random response may solve this problem. However, in the Nor-

118 sound level meter the normal free field microphone may be used, and the wanted frequency response may be obtained by using a random correction network. As with other correction networks, the status is always shown on the display during the measurement.

5 MEASUREMENT OF HIGH LEVELS

Even if the basic instrument has a single, very large measurement range, some applications may require even higher levels to be measured. This may be obtained by using a less sensitive microphone. The sensitivity of the sound level meter may be set to adopt a large range of microphone sensitivities.

The normal microphone supplied with the sound level meter is able to measure peak levels up to 150 dB. However, the preamplifier will limit the useful range to 140 dB as described above. As an option, the sound level meter may be delivered with the possibility to reduce the polarisation voltage and thereby reduce the microphone sensitivity. In this way peak levels up to 150 dB may be measured with the normal microphone.

When the polarisation voltage is reduced from the normal 200 volt to about 70 volt, the sensitivity is reduced by approximately 10 dB. However, the reduced polarisation voltage will also reduce the electrostatic traction between the diaphragm and the back-electrode. This leads to an increase in the distance between the electrode and the diaphragm and a reduction in the damping of the diaphragm. The result is a small change in the frequency response of the microphone. Therefor, when the high level option is activated, a compensation network is automatically also activated to compensate for the change in the microphone response.

6 SUMMARY

The design off a new digital sound level meter is described. Most functions are measured simultaneously and thus to a large extent makes it unnecessary to select the requested parameters before the measurement. For special applications, the accuracy may be further improved by using additional features as compensation for self-noise and applied windshield.

7 REFERENCES

Draft for IEC 61672-1: Electroacoustics – Sound level meters – Part 1: Specifications