Applied Acoustics 77 (2014) 11–19

Contents lists available at ScienceDirect

Applied Acoustics

journal homepage: www.elsevier.com/locate/apacoust

Effects of signal processing on the measurement of maximum sound pressure levels

M. Robinson, C. Hopkins *

Acoustics Research Unit, School of Architecture, University of Liverpool, Liverpool L69 7ZN, UK

ARTICLE INFO

Article history:
Received 8 April 2013
Received in revised form 20 August 2013
Accepted 19 September 2013

Keywords:
Maximum sound pressure level
Signal processing
Sound level meter
Transient
Impulse

ABSTRACT

Maximum sound pressure levels are commonly used for environmental noise and building acoustics measurements. This paper investigates the signal processing errors due to Fast or Slow time-weighting detectors when combined with octave band filters, one-third octave band filters or an A-weighting filter. For 6th order Butterworth CPB filters the inherent time delay caused by the phase response of filters is quantified using three different approaches to establish the following rules-of-thumb: (1) time-to-gradient/amplitude matching occurs when $Bt \approx 1$, (2) time-to-peak matching occurs when $Bt \approx 2$ and (3) time-to-settle matching occurs when $Bt \approx 4$ for octave band filters, and when $Bt \approx 3$ for one-third octave band filters. Four different commercially-available sound level meters are used to quantify the variation in measured maximum levels using tone bursts, half-sine pulses, ramped noise and recorded transients. Tone bursts indicate that Slow time-weighting is inappropriate for maximum level measurements due to the large bias error. The results also show that there is more variation between sound level meters when considering Fast time-weighted maximum levels in octave bands or one-third octave bands than with A-weighted levels. To reduce the variation between measurements with different sound level meters, it is proposed that limits could be prescribed on the phase response for CPB filters and A-weighting filters.

1. Introduction

Maximum sound pressure levels are commonly measured with a sound level meter when assessing environmental noise and building acoustics. The A-weighted maximum sound pressure level is of particular relevance to environmental noise because links have been made with sleep disturbance [1–3]. Current guidelines from the World Health Organisation identify a threshold for sleep disturbance using the A-weighted maximum sound pressure level [4]. In building acoustics, measurement of impact sound insulation using a heavy impact source such as the rubber ball require measurements of maximum sound pressure levels in octave bands or one-third octave bands [5]. Inside buildings there are numerous transient sounds from footsteps, dropped objects on floors, building machinery, doors slamming, plumbing systems and wall-mounted sockets which can be assessed using maximum sound pressure levels. Quantifying the error in the measurement of maximum sound pressure level is therefore important for standards and regulations. Maximum levels are also required for the measurement of vehicle noise emissions in Directive 97/24/EC [6].

A sound level meter has four main components. These are input filters to remove unwanted frequencies from the signal that is being measured, Constant Percentage Bandwidth (CPB) filters [7], an A-weighting filter [8], and a time-weighted level detector [8] to convert an AC signal into a DC signal. The final stages involve statistical, time-based averaging or peak detection components to determine the desired parameter. Typical architecture for a sound level meter is shown in Fig. 1.

The focus of this paper concerns signal processing errors in the measurement of maximum sound pressure levels, specifically due to the time-weighted level detector (Fast or Slow time-weighting) and CPB filters (octave band or one-third octave band filters). Experimental work is conducted on the CPB filters and the time-weighted level detectors of a software-based sound level meter and four commercially-available sound level meters. The frequency range under consideration corresponds to the building acoustics frequency range which covers one-third octave bands from 50 Hz to 5 kHz. In addition, consideration is given to errors in the A-weighted maximum level due to the combination of the A-weighting filter and the time-weighted level detector.

2. Equipment used for the signal processing

All signal processing was carried out using either a software-based sound level meter or commercially-available sound level meters. This paper considers the errors in the signal processing and not the transducer; hence the findings are equally applicable

* Corresponding author. Tel.: +44 151 794 4938.
E-mail address: carl.hopkins@liverpool.ac.uk (C. Hopkins).

0003-682X/$ - see front matter © 2013 Elsevier Ltd. All rights reserved.
http://dx.doi.org/10.1016/j.apacoust.2013.09.017

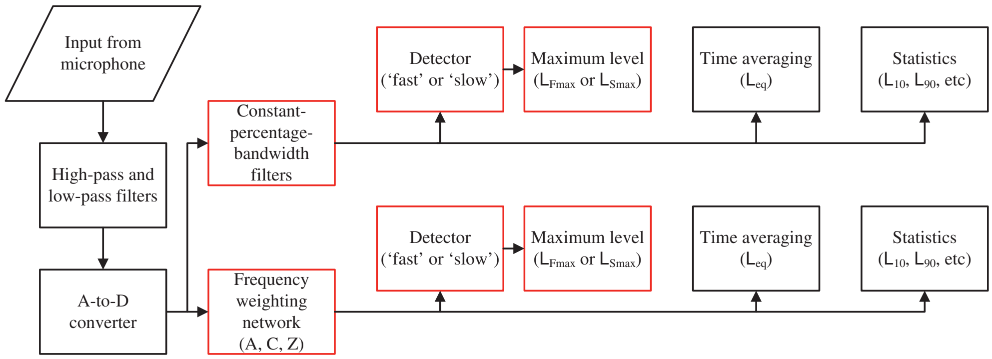

Fig. 1. Typical architecture of a sound level meter.

to the measurement of maximum vibration levels that use the same combination of filter and detector.

2.1. Software-based sound level meter

A software-based sound level meter has been implemented in the software package Matlab. A particular advantage of this software-based meter is that the output signal from each component can be analysed, specifically the AC output signal from the filter can be accessed before it reaches the detector to allow an assessment of the filter time response. The meter comprises the following components; input filters, CPB filters, and a time-weighted level detector (see Fig. 1). The meter was used to simulate the individual responses of the CPB filters and time-weighted level detectors. Octave band and one-third octave band filters were implemented using 6th order Butterworth filters. These were validated according to EN 61260 [7] by checking the magnitude response of the filters against maximum and minimum requirements. Similarly the Fast and Slow time-weighted level detectors were validated according to EN61672 Parts 1 and 2 [8,9] using a 4 kHz tone burst of various lengths and then checking the level response from the detectors, with shorter tone bursts giving lower level responses from the detectors.

2.2. Commercially-available sound level meters

To assess the signal processing in commercially-available sound level meters, four meters were chosen. These meters are referred to as A, B, C and D and their details are given in Table 1. The four meters complied with the standards that applied at their time of manufacture for the general specification [8,10] and the octave band and one-third octave band filter specification [7,11]. The meters do not allow access to the AC filter output signal and therefore all errors were reported using either the Fast or Slow time-weighted maximum sound pressure levels, denoted as L_{Fmax} or L_{Smax} respectively.

The filter and detector response of the sound level meters is assessed by supplying the pre-amp with a voltage from the signal generator that produces the transient excitation. This requires bypassing the microphone capsule for sound level meters A, C and D. Hence for these experiments it is necessary to use all four meters without a calibrated input transducer. For this reason, absolute values for L_{Fmax} or L_{Smax} are not available and only relative values in decibels are shown in the results. This requires normalisation procedures that depend upon the type of transient excitation and these are described in Section 4.

3. Filter time delay

Octave band and one-third octave band filters have an inherent time delay in their response and this has the potential to affect the measurement of maximum levels. Given the nature of filter design it is possible to create filters with the same magnitude response which satisfy the requirements of EN 61260 [7] but which result in filters with different phase responses [12]. This is because filter design is focused on the attenuation of frequencies outside the pass-band rather than the effect this has on the phase. It is this phase response that determines the time delay in CPB filters. Due to the inverse relationship between the time and frequency domains, the time delay increases as the roll-off in the filter magnitude response becomes steeper. This time delay is most conveniently investigated here with a steady-state sinusoidal signal which will then provide a basis on which to consider tone bursts as a simple form of transient in Section 4.1.

For a steady-state sinusoidal signal, the time, t, in seconds for a filter to respond, i.e. the time for the amplitude of the filter output signal to reach that of the input signal, is often quoted as occurring when $Bt = 1$ where B is the filter bandwidth [13]. To test the applicability of this rule, three different parameters are used to assess the response time of CPB filters using the 6th order Butterworth filters in the software-based sound level meter. Note that these parameters are not based on a detector output, only on the AC input and AC output of the filter. Time-weighted detectors average the response over time; hence they cannot be used to measure a filter's response time. For this reason, an RMS detector is implemented that gives an instantaneous response with no phase shift so that it can accurately measure the filter's behaviour. The input signal is a sinusoid corresponding to the filter band centre frequency. The following three matching parameters are used to assess the filter output signal:

(1) Time-to-gradient/amplitude matching using the filter's AC output before the detector. This is determined by finding the time at which the gradients and amplitudes of the filter input and output signals are equal within a specific tolerance. It requires finding the normalised difference between the input and output instantaneous gradients across the whole signal and the normalised difference in the input and output amplitudes across the whole signal. Time-to-gradient/amplitude matching is defined using a 1% tolerance; hence the first sample is identified where there is <1% difference for both gradient and amplitude matching.
(2) Time-to-peak matching using the detector output signal. This is determined by finding the maximum level in the detector output signal.
(3) Time-to-settle matching using the detector output signal. This is determined by calculating the gradient between the last point in the detector output signal and every other point in the signal. This 'gradient signal' is divided by the maximum gradient found in order to normalise the gradient signal to values between 0 and 1. The final stage is to identify the first sample in the normalised 'gradient signal' with <1% difference between consecutive gradients; this defines the time-to-settle.