History and description of loudness models

Loudness Toolbox for Matlab

GENESIS S.A.
www.genesis.fr

December 2009
Document IB/RP/10003

Contents

1 Document purpose

This document details the loudness models and psychoacoustics indicators that allow the calculation of overall loudness. The case of time-varying sounds and impulsive sounds is also taken into account. Thus, the loudness Toolbox for Matlab allows the loudness calculation of any type of sound using standardized or published models in the scientific literature.

2 Introduction

This section is dedicated to a history of researches made during the last decades in the field of measure and calculation of perceived sound level (loudness).

2.1 Hearing system

Loudness is a subjective magnitude corresponding to the perceived sound pressure level. It depends on the sound pressure level, the frequency, and the duration of sounds. It is thus correlated to the functioning of human auditory system.

Figure 1 is a representation of the human peripheral hearing system.

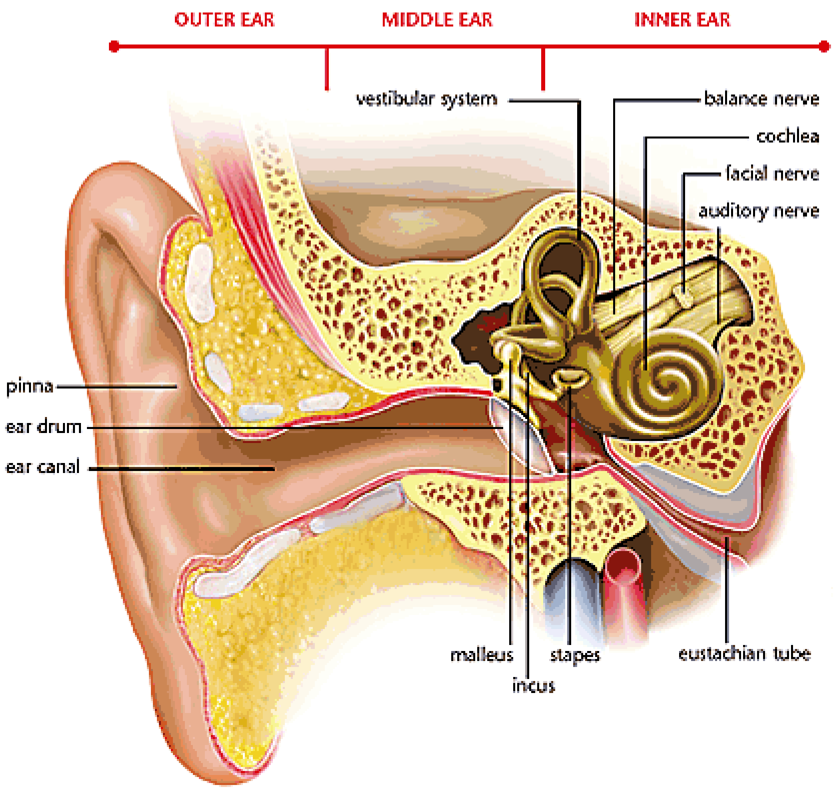

Figure 1: Human peripheral hearing system.

The outer ear is the first part of the hearing system. The pinna and the ear canal have an influence on the amplitude of sound vibrations. Frequencies are more or less amplified. A sound reaching the ear is filtered by the outer ear and transformed into a mechanical vibration at the ear drum. This vibration is transmitted through the middle ear by three tiny bones. These are known as the malleus, incus and stapes (sometimes called hammer, anvil and stirrup). The middle ear is used as an impedance adapter to limit the energy loss at the transmission of the vibration to the inner ear.

In the inner ear, the vibration is transmitted inside the cochlea all the way to the basilar membrane. This membrane acts as a spectral analyser, and is the place where frequency masking phenomena occur. The mechanical vibration is then transformed into a nervous impulse inside the organ of Corti, by the inner and outer hair cells. Eventually, the nervous impulse is transmitted by the auditory nerve to the brain.

2.2 Metrology

Loudness is a subjective magnitude which unit is the Sone. Sone is based on a sensitive scale. This scale was established from a method of measure called magnitude estimation (Stevens, 1956). In this method, subjects are to assign a number proportional to the stimulus loudness, for several stimuli presented at different sound levels.

The relationship between sound pressure level and loudness, called loudness function (see Figure 2), is a power function (S (sones) = $k\, P^{0,6} = l\, I^{0,3}$).

By convention, 1 sone is allocated to the loudness of a 1-kHz and 40 dB SPL sound. A sound with a loudness of 2 sones is perceived twice as loud as a sound of 1 sone.

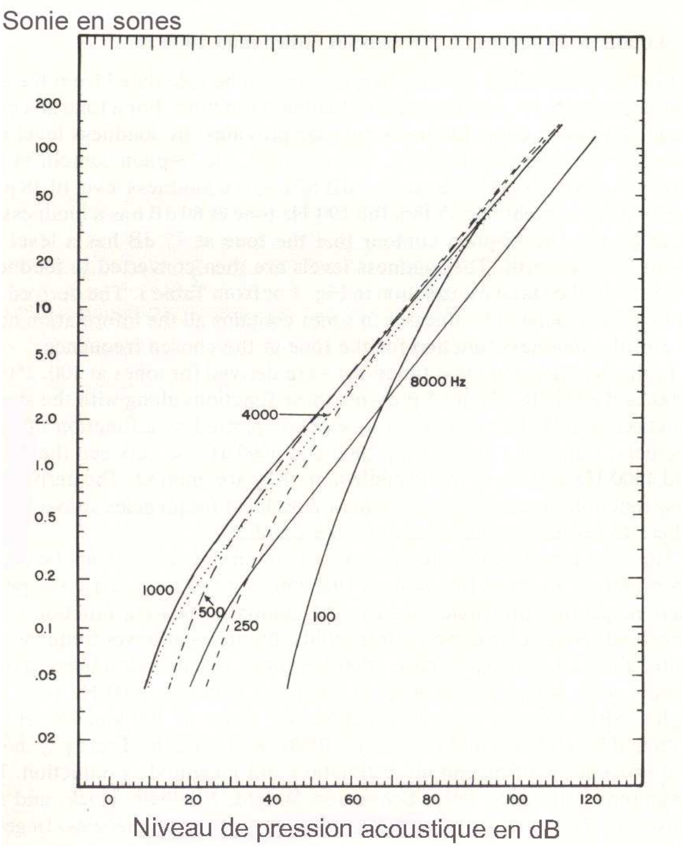

Figure 2 : Loudness functions of pure tones for several frequencies: 100, 250, 500, 1 000, 4 000 and 8 000 Hz (Scharf, 1978).

Another method for measuring loudness consists in asking subjects to determine the loudness of a « test » sound by comparison to a "standard" 1-kHz tone.

The loudness level is obtained in phons. It corresponds the sound pressure level of the « reference » 1-kHz tone when it is perceived as loud as the « test » sound. Then, the Phons scale, established from methods of adjustment (Fletcher and Munson, 1933), matches with dB SPL scale for a 1-kHz tone. The equal-loudness contours obtained (see Figure 3) highlight the variation of ear sensibility with frequency.

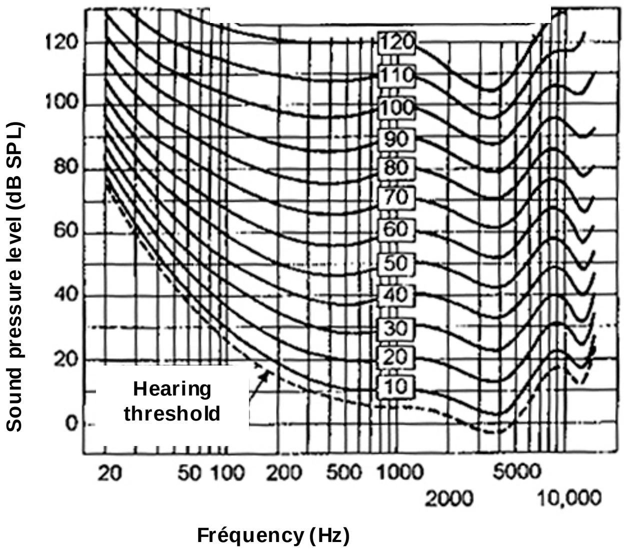

Figure 3 : Equal-loudness contours (Robinson and Dadson, 1956). Phons scale matches with dB SPL scale for a 1-kHz tone.

Loudness depends not only on sound level but also on frequency spectrum and sound duration. For sounds with same amplitude, loudness increases up to a duration called critical duration (which varies between 50 and 400 ms according to authors). This phenomenom is called temporal integration. Beyond critical duration and up to several seconds, loudness is constant.

To estimate loudness, physical magnitudes were developed. The most common and used one is dB(A), which takes into account the variations of ear sensitivity with frequency. dB(A) is a sound level measure weighted by the equal-loudness contours at 40 phons. This is why measures in dB(A) under-estimate the perceived sound level for high level complex sounds.

There are also dB(B) and dB(C), associated to the weighting of mid-level sounds (70 phons) and high-level sounds (100 phons) respectively. Nevertheless, dB(A) is most commonly used for industrial and environmental acoustics. It is important to note that A, B, and C weightings do not take into account physiological phenomena such as frequency masking or the filter bank functioning of human ear, and thus remains insufficient to correctly estimate loudness.

Sor far, two loudness models have been designed for stationary sounds: Zwicker and Moore's models. Zwicker's model was first described in a publication of 1958. In 1975, a graphic method allowing the calculation of loudness was the subject of an international standard (ISO532B) and the implementation of this standard in BASIC was published in 1984. More recently, Zwicker's model was the subject of a German standard DIN45631 (1991) which contains a BASIC program based on the graphic method of the standard ISO532 method B. Moore's model was firstly published in 1996 and then revised in 1997 to include the calculation of partial loudness. It was also the subject of an American standard (ANSI S3.4-2007). The ANSI S3.4-2007 standard replaced the ANSI S3.4-1980 standard, based on the graphic method of the ISO532-A standard stemming from Steven's works (1961). The international ISO 532 standard is under revision and should end-up in a new version by 2010, keeping Zwicker's approach and including that of Moore.

Concerning time-varying sounds (that is which temporal and frequency characteristics evolve with time), two models have been developed. The first one has been developed by Zwicker and Fastl (1999) and the second one by Glasberg and Moore (2002). A project of international standardization is being elaborated. Nevertheless a German standard DIN45631 / A1 (2008) based on Zwicker and Fastl's works (1999) allows the calculation of time-varying loudness.

The case of the impulsive sounds - which temporal aspect can be characterized by a fast transient response, an absence of landing at the maximum of amplitude and a certain decay time - was studied by Boullet (2006). The model developed allows the estimation of global loudness for this type of sound.

All these models allow the loudness calculation of sounds emitted in frontal incidence in free field, from an acoustic signal in Pascal. It is important to note that the modelling of hearing system through the various loudness models results in an estimation of loudness. The only way to know the real loudness of sounds remains the establishment of psychoacoustic tests.

3 Loudness models

In this section, the main loudness models are detailed. Three different sections are proposed, which consider the case of steady sounds, time-varying sounds and impulsive sounds.

3.1 Loudness models for steady sounds

The models presented in this section allow the calculation of loudness for steady sounds, that is sounds presenting stable temporal and spectral statistical properties over time. Two models are detailed here: Zwicker and Moore's models.

Zwicker's model (1991) is based on the calculation of basilar membrane excitation by critical bandwidth. In this purpose, it takes into account:

- a model of transmission of the acoustic signal through the outer and middle ear,
- the phenomenon of frequency masking (calculation of specific loudness),
- the integration of specific loudness over 24 critical bands.

Moore's model (1997) is close to Zwicker's (1991). The differences are:

- filters calculation,
- field corrections,
- the calculation of the excitation pattern.

Figure 4 illustrates the main steps in the calculation of loudness for both models, which are described in the following.

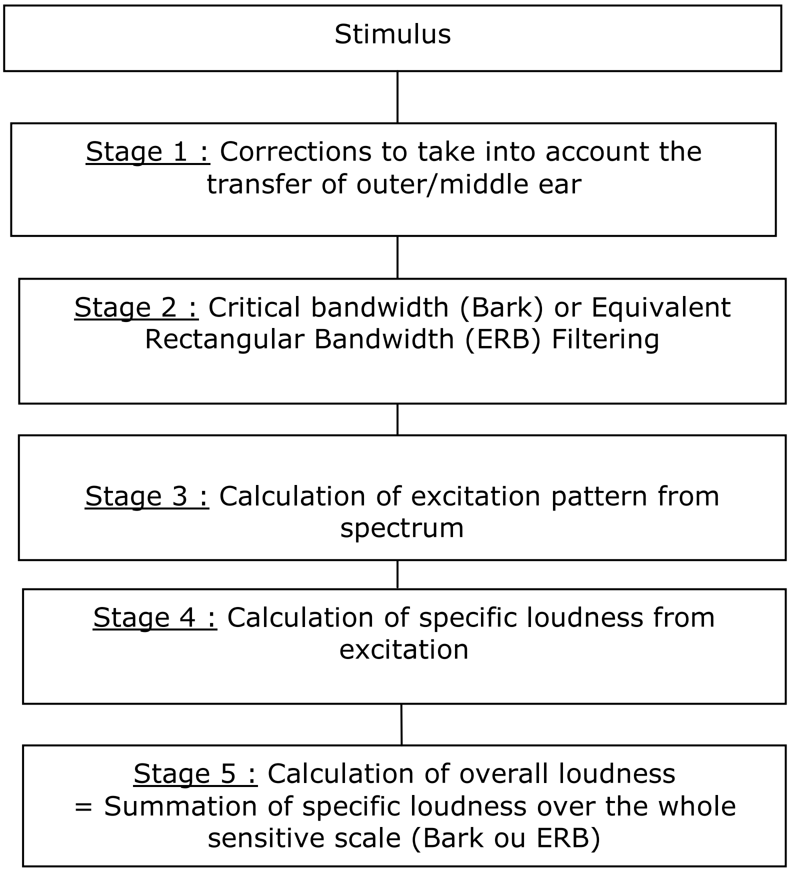

Figure 4: Stages of the loudness calculation for steady sound.

Stage 1: Transmission through outer and middle ear

Zwicker

A transmission factor called a_0 by the author (see Figure 5) takes into account the transformation between free field and inner ear, and is to be added to the original signal.

Zwicker assumes that this transmission function is similar to the absolute hearing threshold but inverted beyond 1 000 Hz (inner ear is equally sensitive to all frequencies), and constant below 1 000 Hz (the raising of the absolute threshold at low frequencies is only explained by the ear internal noise).

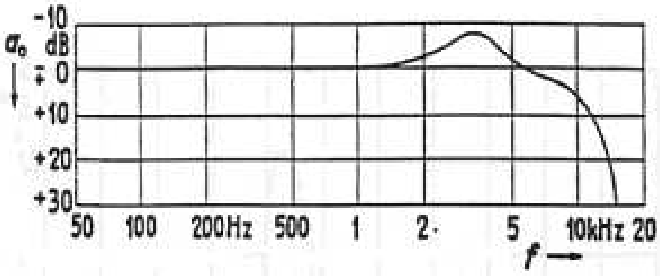

Figure 5: Transmission factor (ao) values needed in the free-field condition, as a function of frequency (Zwicker and Feldtkeller, 1981, p. 111).

Moore

Moore *et al.*. (1997) do not associate the raising of absolute threshold to the only presence of internal noise. Hence their model uses a correction called Equal Level Contour (ELC) based on two assumptions (Figure 6):

1) Beyond 1 000 Hz, inner ear has the same sensitivity to all frequencies. Thus the variation of absolute threshold with frequency is due to the outer and middle ear filtering. The transmission function curve is similar to the absolute threshold but inverted. The correction, called Minimum Audible Field (MAF), is the same as Zwicker's.

2) Below 1 000 Hz, the transfer function is assumed to have the same shape as the equal-loudness level contour at 100 phons, but inverted.

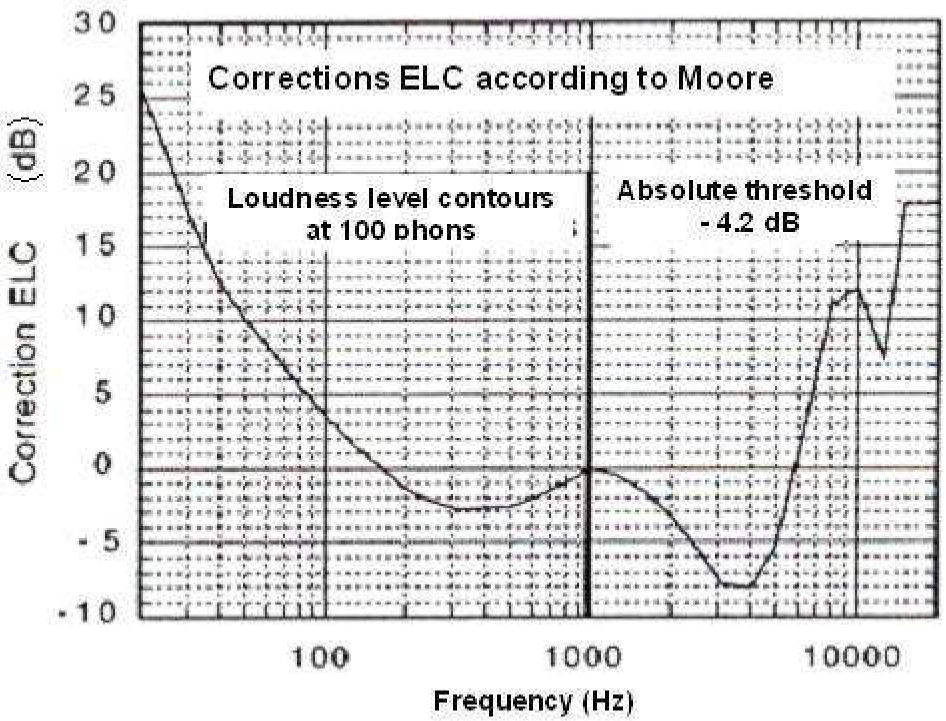

Figure 6: Transmission factor corrections (ELC) in dB as a function of frequency according to Moore's model (1997).

Stage 2: Modelling and calculation of auditory filters

Zwicker models the basilar membrane functioning by a filter bank.

These "physiological" filters are modelled by bands called critical bands. These bands are spectral adjacent bands which unit is Bark. Inside a critical band, the loudness is constant. Beyond, the loudness increases with bandwidth.

Below 500 Hz, the critical bandwidth is constant (100Hz). Over 500 Hz, the bandwidth increases with frequency. Values are given in Table 1 and Figure 7.

Band number	Centre frequency (Hz)	Critical bandwidth (Hz)	Upper cut off frequency (Hz)
1	50	80	100
2	150	100	200
3	250	100	300
4	350	100	400
5	450	110	510
6	570	120	630
7	700	140	770
8	840	150	920
9	1000	160	1080
10	1170	190	1270
11	1370	210	1480
12	1600	240	1720
13	1850	280	2000
14	2150	320	2320
15	2500	380	2700
16	2900	450	3150
17	3400	550	3700
18	4000	700	4400
19	4800	900	5300
20	5800	1100	6400
21	7000	1300	7700
22	8500	1800	9500
23	10500	2500	12000
24	13500	3500	15500

Tableau 1: 24 crititical bandwidths according to Zwicker (Zwicker and Feldtkeller (1981), p. 71).

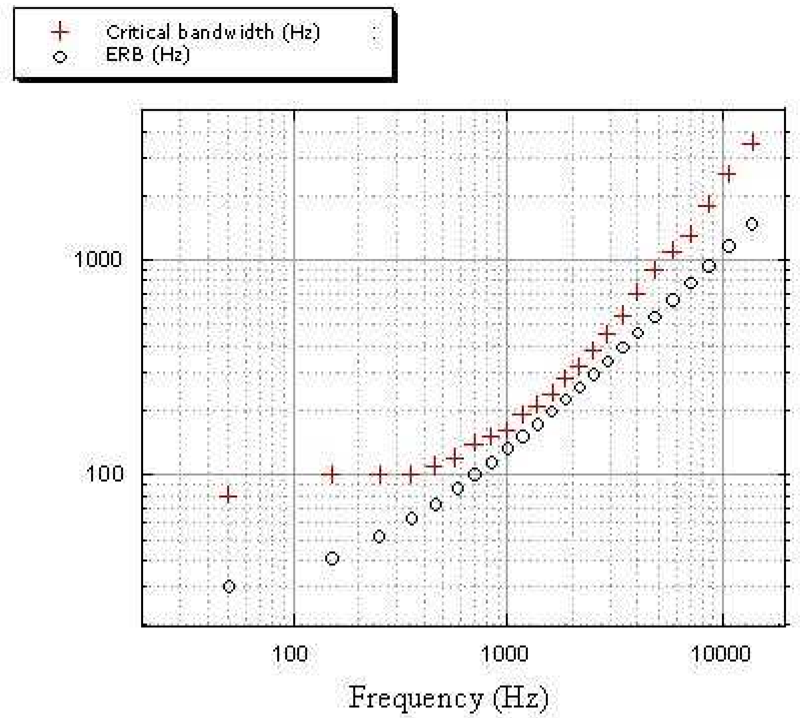

Figure 7: Critical bandwidth (cross) and ERB (circle) as a function of centre frequency in Hz (Zwicker, 1961). ERB, in Hz, is given by 24,7(4,37f+1), where f is in kHz.

Moore

Moore *et al.* (1997) model the auditory filters by Equivalent Rectangular Bandwidth (ERB) bands, that are close to critical bands (see figure 7).

Stage 3 : Excitation calculation

The excitation pattern of a sound is calculated from the sound level in each band defined previously (critical bands or ERB bands), after having taken into account the transmission factor of outer and middle ear (see Stage 1). The output amplitude of each filter is called excitation pattern, and is calculated according to the centre frequencies of the corresponding filters. The Calculation of the excitation pattern is an important part in loudness models, and differs according to the method (Zwicker or Moore).

The notion of excitation is related to the phenomenon of sound masking. The masking of a sound by another one results in the decrease of audibility of this sound. The masking can be partial (the masked signal is still audible) or total (the masked signal is then inaudible). Experiments led to study the masking phenomenon consist in measuring the hearing threshold of a signal in the presence of another masker signal. From these experiments, some masking curves were highlighted (see figure 8). These curves show how a signal can potentially mask another signal.

Zwicker

According to Zwicker, the excitation pattern reflects the masking pattern of a pure tone masked by a narrow band noise. Zwicker supposes that the excitation patterns and the masked detection threshold curves are identical. The masking curves highlight the ear selectivity and they depend on the centre frequency and the sound level (figure 8).

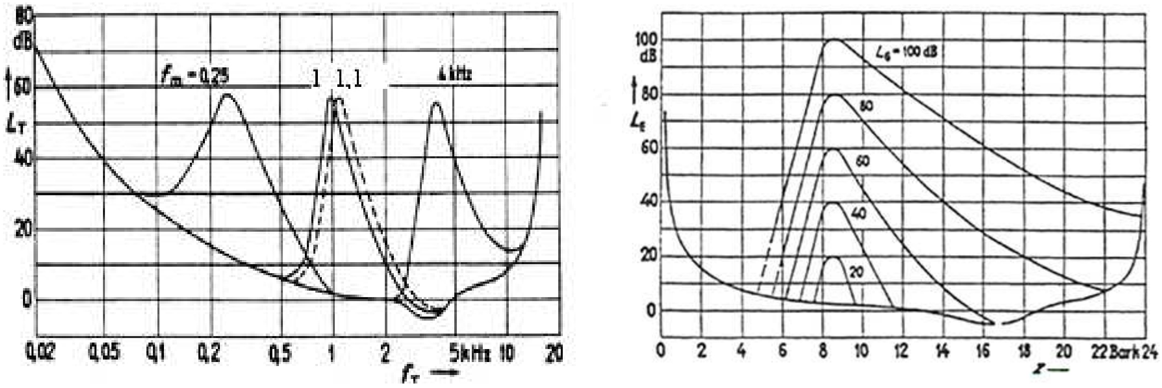

Figure 8: On the left: Level of test tone masked by critical-band wide noise with level 60 dBSPL and centre frequency of 250 Hz, 1 kHz, 1.1 kHz and 4 kHz.
On the right: Level of test tone masked by a critical-band wide noise centred around 1 kHz (bandwidth = 160 Hz). Sound pressure level of the band noise is Lg : 20, 40, 60, 80 and 100 dB SPL (Zwicker and Feldtkeller, 1981).

Moore

According to Moore, the excitation pattern is calculated from the output of the auditory filters centred on the frequencies composing the sound. Let us consider for example a 1 kHz pure tone. The upper diagram of figure 9 shows the frequency response of the filters centred in the neighbourhood of 1 kHz for which there is a contribution at 1 kHz (i.e which curves cross the vertical dotted line). The vertical dotted line represents the frequency of the pure tone. On the figure below, abscissas correspond to the central frequencies of the nearby filters (cf. upper figure), and ordinates represent the frequency response at 1 kHz of these filters. The excitation pattern is built by the curve that links these points, brung back to every centre frequency of the nearby filters.

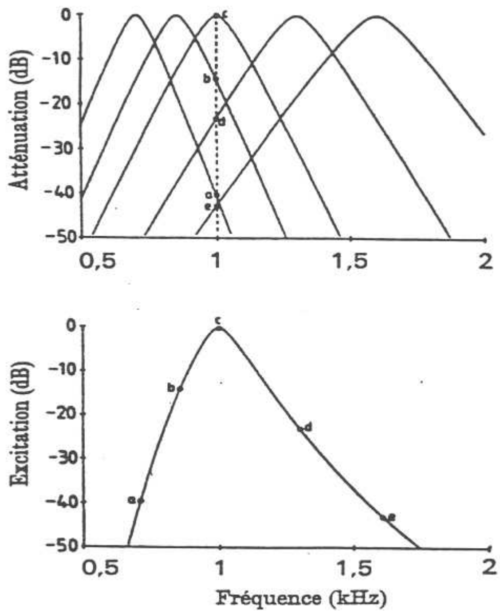

Figure 9: Calculation of a 1 kHz tone excitation pattern according to Moore.

For a complex sound, excitation is obtained by repeating the operation described above for all the frequencies in the signal. The final excitation is then the sum of partial excitations.

Stages 4 and 5: Calculation of specific and overall loudness

To calculate overall loudness, specific loudness (N') has to be calculated from the excitation in each spectral band (critical band or ERB). Specific loudness corresponds to a sensory spectrum.

According to Stevens, the relationship between excitation and specific loudness is a power function (relation 1):

$$N' = c.\ E^{\alpha} \qquad (1)$$

where c and α are constants (determined from experiments). The exponent α, lower to 1, highlights the non-linear and compressive relation between excitation and specific loudness.

Zwicker

Zwicker calculates specific loudness from excitation. For each critical band, an excitation level is calculated (see figure 8), called main excitation. In this purpose, some adjustments are made: the shape of the upper slope is globally preserved in the model (under a certain constraint described further) and depends on the level and the critical band number. The shape of the lower slope is transformed into a vertical line. The area under the curves has to remain the same before and after adjustment. In the algorithm of loudness calculation, the calculation of specific loudness is split into a calculation of a main loudness determined from the excitation level and a calculation of specific loudness (Figure 10).

Main loudness is calculated from the equation below (Zwicker and Fastl, 1999):

$$N' = c.\ (E_{Thq})^{\alpha}.[(0{,}5 + 0{,}5\ .\ (E_{Stimulus})/(E_{Thq}))^{\alpha} - 1] \quad (2)$$

The exponent α (equal to 0,23) is that of the straight line corresponding to the loudness function of a uniform-exciting noise (upper curve in dash-dotted line in Figure 11). $E_{Stimulus}$ is the excitation produced by the stimulus. E_{Thq}, given by Zwicker, corresponds to the excitation at threshold in quiet.

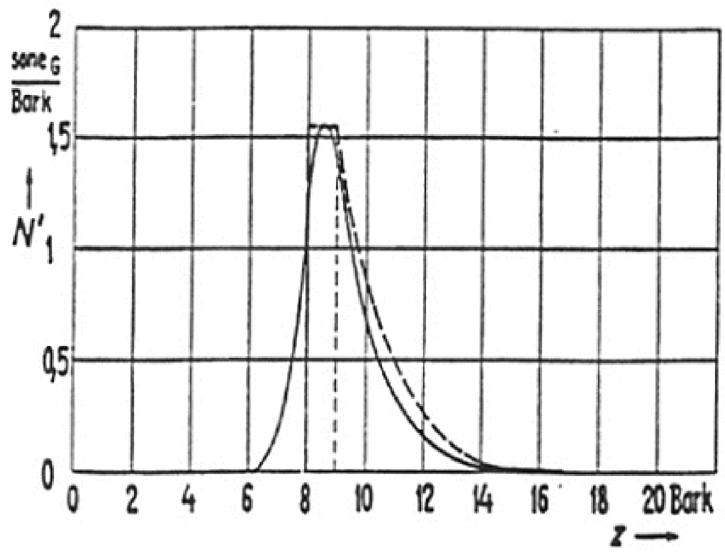

Figure 10: Specific loudness as a function of critical-band rate (Bark) for a 1 kHz tone with level 60 dB SPL. The broken curve is the approximation used for loudness calculation (Zwicker and Feldtkeller, 1981).

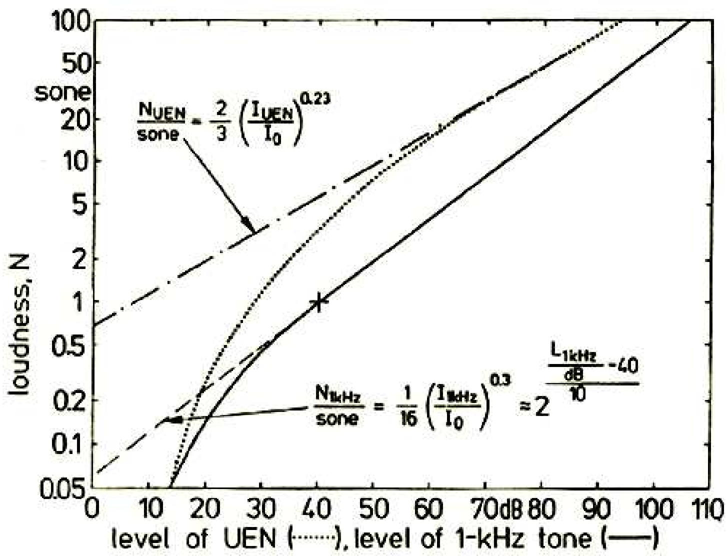

Figure 11: Loudness functions of a 1 kHz tone (solid line) and a uniform-exciting noise (UEN, in dotted line). Approximations using power laws are indicated by the broken and dash-dotted lines along with their corresponding equation. Loudness is shown as a function of sound pressure level (Zwicker and Fastl, 1999).

In the loudness calculation procedure, specific loudness of standard patterns used in ISO532B (Figure 12), were approximated by segment lines (Figure 13), which slopes depend on excitation level. The higher the excitation level, the higher the main loudness and the steeper the first segment.

The specific loudness obtained is the continuation of main loudness and\or the steep segments in all critical bands. If the main loudness in a critical band **n** is located below the segment value of the previous critical band (**n-1**), the main loudness of critical band **n** is masked and the specific loudness of critical band **n-1** is taken into account to determine the specific loudness in critical band **n** (see Figure 12).

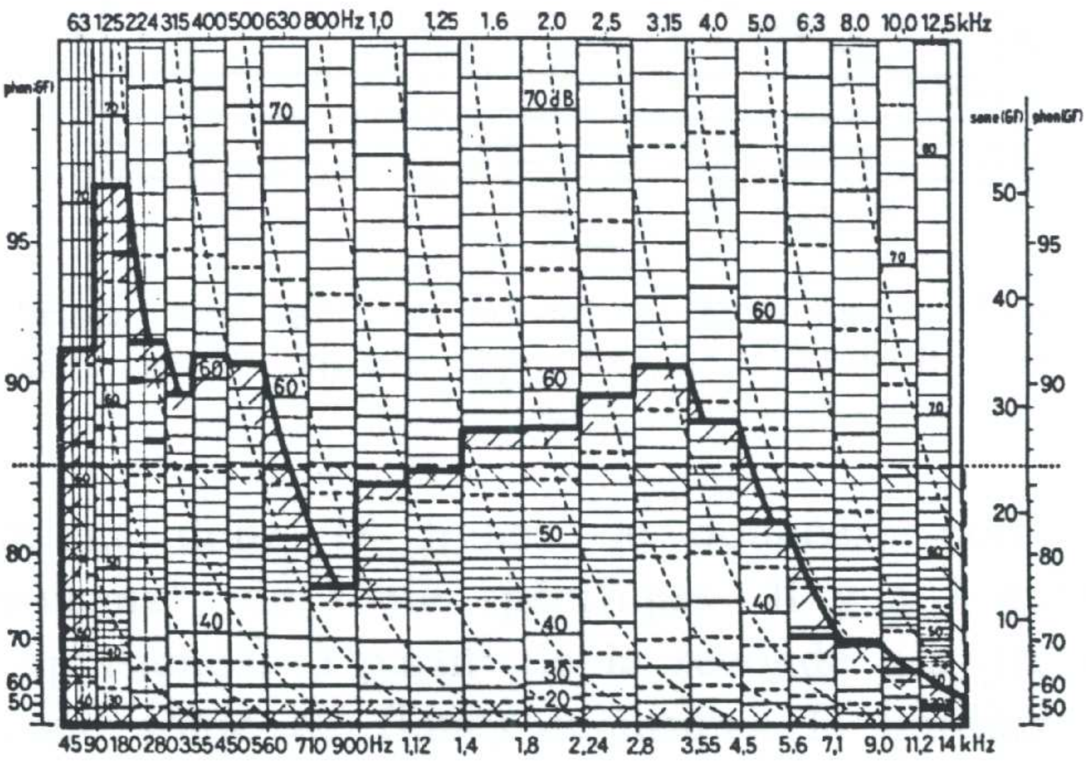

Figure 12: Example of loudness pattern (specific loudness) as a function of critical-bandwidth for a machine noise. Overall loudness is the area under the bold line and corresponds to the dash line ordinate value (loudness level is 86 phons).

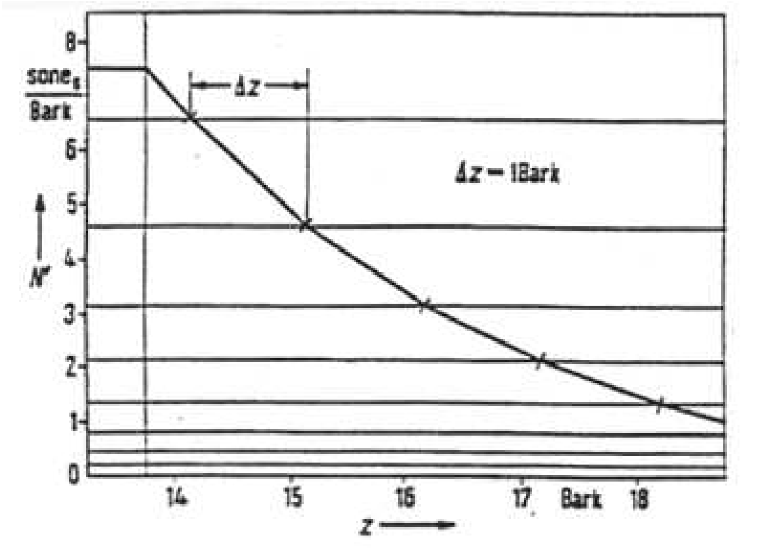

Figure 13: Example of a segment line used to approximate specific loudness in the loudness calculation procedure.

The overall loudness is obtained by summing specific loudness over the 24 critical bands.

Moore

Specific loudness is calculated from the excitation diagram (see Figure 9). Moore *et al.* (1997) assume that the ear background noise is inaudible. They model this phenomenon by subtracting the specific loudness produced by internal noise to the specific loudness of the stimulus:

$$N' = N'_{Stimulus} - N'_{InternalNoise} = c.(E_{Stimulus})^{\alpha} - c.(E_{Thq})^{\alpha}$$

$$N' = c. \left[(E_{Stimulus})^{\alpha} - (E_{Thq})^{\alpha} \right]$$

Overall loudness is obtained by summing specific loudness over ERB bands.

3.2 Loudness models for time-varying sounds

These models of loudness for time-varying sounds are similar in their principle to steady sounds models. Nevertheless, they take into account temporal masking, and the loudness is calculated as a function of time and not in a global way.

Figure 14 shows the main stages in loudness calculation for both models.

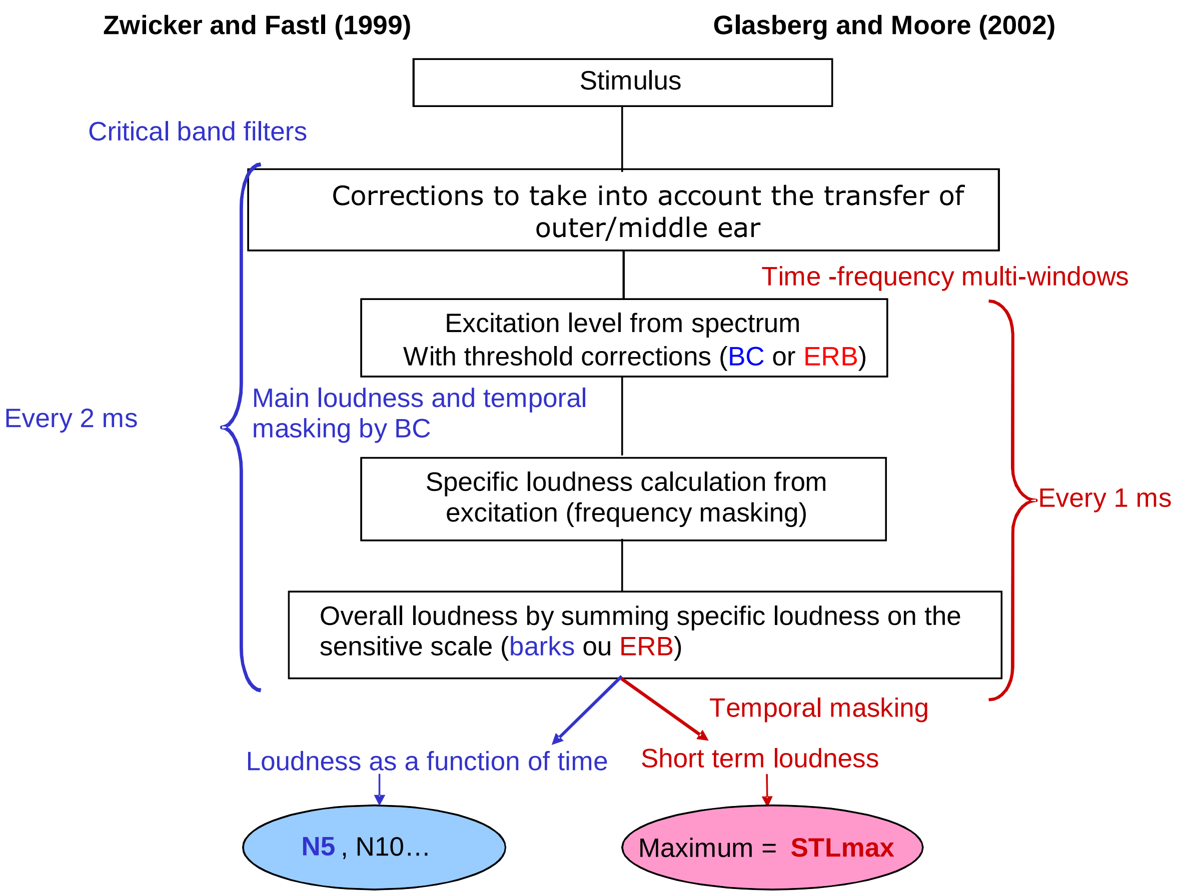

Figure 14: Stages of the loudness calculation for time-varying sounds.

Zwicker and Fastl (1999)

A loudness model for time-varying sounds was elaborated by Zwicker and Fastl (1999). Temporal masking is taken into account in this model. Indeed, a signal can be masked (or difficult to detect) if it is closely preceded by another signal (posterior masking). A signal can also be masked by a sound that closely follows it (previous masking, or retroactive). However, The latter effect is not taken into account in Zwicker's model.

Zwicker uses an equivalent scheme of quadrupole (Figure 15) to model posterior masking as a function of signal intensity and duration. In other words, depending on signal intensity and duration, the masking effect lasts for a more or less long time after the end of the signal. Temporal masking is modelled by a discharge of capacitors.

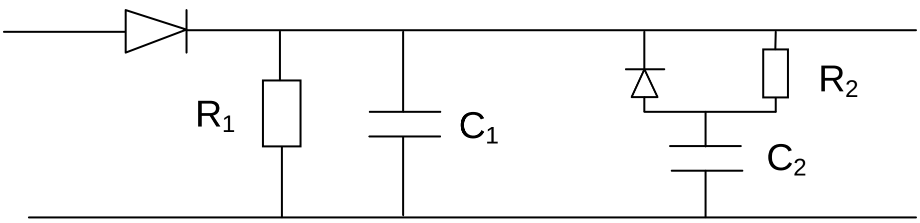

Figure 15: Network RC of quadrupole simulating temporal masking. R1=35kΩ; C1=0,7 µF ; R2= 20 kΩ; C2=1 µF (Zwicker, 1984).

The capacitors charge and discharge depend on input and output voltage. Let us suppose that the voltage of the input signal is a step. C1 is immediately charged and C2 is charged with a time constant T2 = R2*C2 equal to 20 ms. C2 is considered charged after a duration between 3*T2=60 ms (capacitor 95 % charged) and 5*T2=100 ms (capacitor 99 % charged).

When the signal is interrupted, capacitors discharge according two cases:

1. Case where the signal duration is lower than 100 ms:

Capacitor C2 doesn't have enough time to charge itself completely. C1 discharges through R1 and charges C2 by R2.

2. Case where the signal duration is greater than 100 ms:

Capacitor C1 and C2 are completely charged and the system is in equilibrium state. C1 and C2 discharge then through R1. This discharge, being made with a time constant equal to R1 * (C1 + C2), is slower than in case 1.

Figure 16 shows an example of temporal masking of a 5 kHz tone burst, with a duration of 10 or 100 ms. Zwicker's model allows the calculation of loudness as a function of time (Figure 16 (c)), but it does not give the overall loudness of time-varying sounds.

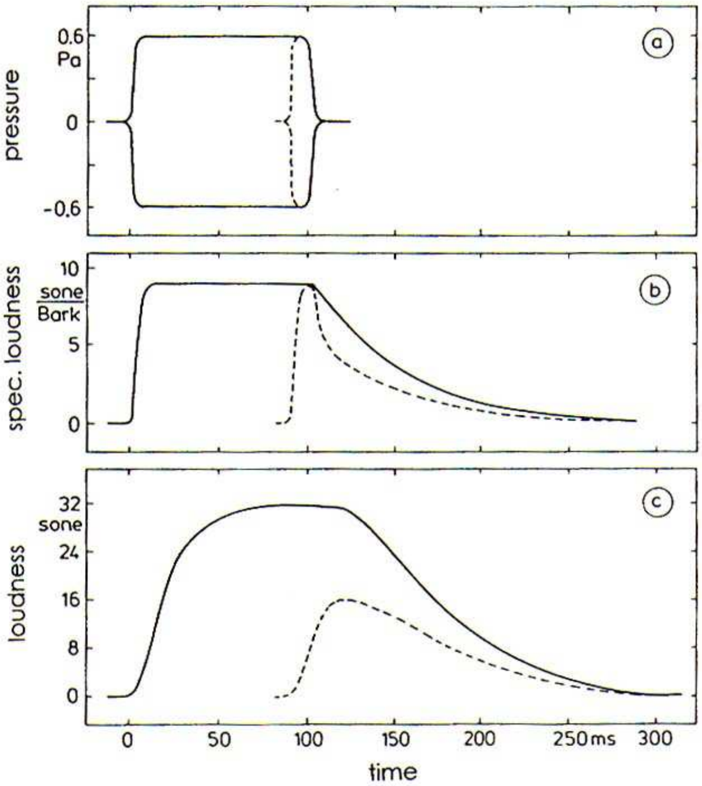

Figure 16: Establishment of the loudness of a 5 kHz tone burst with duration of 100 ms (solid line) and 10 ms (dotted line) (a). temporal envelope of both tones (b), specific loudness in the 19th critical band. (c), Loudness as a function of time (Zwicker and Fastl 1999).

Glasberg and Moore (2002)

Glasberg and Moore (2002) also developed a loudness model for time-varying sounds. Like previous models, this model takes into account the transmission through outer and middle ear, but using a finite impulse response filter (FIR). The transfer function of the FIR filter modelling the transmission from free field to inner ear is given in Figure 17.

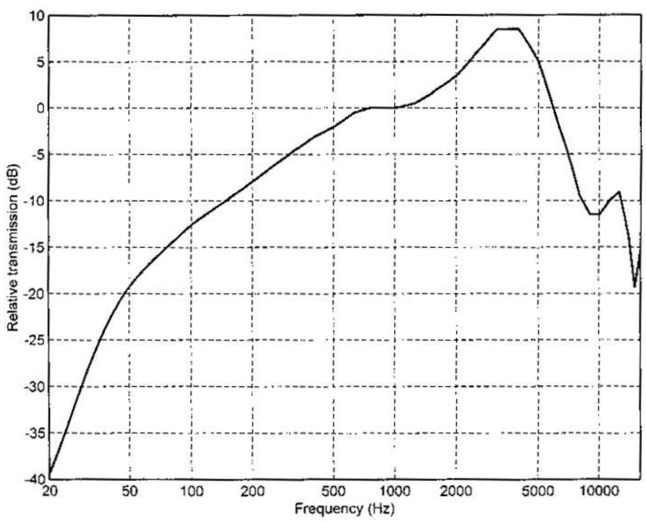

Figure 17: Transfer function of the FIR filter used to simulate the effects of outer and middle ear. Gain is set to be 0 dB at 1000 Hz (Glasberg and Moore, 2002).

Cochlea is assumed to act as a bandpass filter bank which centre frequencies going from 50 to 15 000 Hz. The filters bandwidth increases along with their centre frequency. Indeed, for a 100 Hz centred filter, the equivalent rectangular bandwidth (ERB) is 35 Hz. For a 10 kHz centred filter, the ERB is 1100 Hz.

The filters shape depends on sound level. As a matter of fact, the filters slopes at low frequencies are getting less steep as level increases. The curve giving the output amplitude of each filter for a given level models the excitation diagram.

Moore's method to calculate the excitation diagram is based on a multi-resolution spectral analysis by Fourier Transform. In order to have a resolution at low frequencies comparable to that of the auditory system, the analysis is made on relatively long segments (64 ms). For high frequencies, the authors use shorter windows (2 ms).

Six FFTs (Fast Fourier Transform) are then calculated in parallel, with temporal windows durations of 64, 32,1 6, 8, 4, ans 2 ms. These FFT calculations are used to measure the level in bands 20-80 Hz, 80-500 Hz, 500-1250 Hz, 1250-2540 Hz, 2540-4050 Hz, and 4050- 15000 Hz respectively. The excitation diagram being calculated from the spectrum each 1 ms, there is some temporal overlap in the analysis.

The next step in the model is the calculation of what Moore and Galsberg call "instantaneous" loudness. The assumption is that instantaneous loudness is not something conscious in perception. It would correspond to the overall activity inside the auditive nerve measured on a very short period of time (around 1 ms). The calculation of instantaneous loudness is made from the excitation diagram, as in the stationary model (Moore *et al.*, 1997). The excitation diagram is then transformed into specific loudness, and the area under the specific loudness curve gives the instantaneous loudness for the considered period of time.

Glasberg and Moore (2002) calculate then the short-term loudness (STL) from the instantaneous loudness. STL corresponds to the loudness perceived during a short segment of sound (a syllable for example). To calculate STL, instantaneous loudness is modified by an operation close to an automatic gain control, in order to take into account the temporal integration phenomenon (with a time constant about 100 ms) and temporal masking (loudness decline after the end of the stimulus). STL describes the loudness perceived at each instant. The authors recommend taking the maximum of the short time loudness (called STLmax) as an indicative value of overall loudness for a short sound.

This model also calculates the long-term loudness (LTL) from the short term loudness (using again a temporal integration, but with higher time constants). The long-term loudness represents the fact that the global loudness of a sound is kept in memory for several seconds after its end and in the absence of a new stimulus. The maximum of long-term loudness (LTLmax) is recommended by Glasberg and Moore to estimate overall loudness of sounds varying slowly in time.

3.3 Loudness model for impulsive sounds

Boullet (2006)

The french standard NF S31-010 (1996) defines an impulsive noise as a "noise consisting of one or several impulses of acoustic energy, each having a duration lower than 1 second and separated by time intervals longer than 0.2 seconds".

Boullet (2006) studied more precisely the loudness of the impulsive sounds presenting a very fast transient response and immediately followed by an exponential decay. The model developed during its works allows the calculation of loudness for this type of sounds, and is schematized on Figure 18 and described in what follows.

First of all, the acoustic signal is filtered to obtain a temporal signal in each critical band. The following step of the model consists in calculating the energy and the decay time for each critical band. Main loudness is the product of energy (exponent alpha) and duration of the impulse decay (exponent beta). Finally, the effect of simultaneous masking is taken into account as in Zwicker and Fastl's model (1999). Specific loudness obtained this way is then summed over the 24 critical bands to obtain the overall loudness of the impulsive sound.

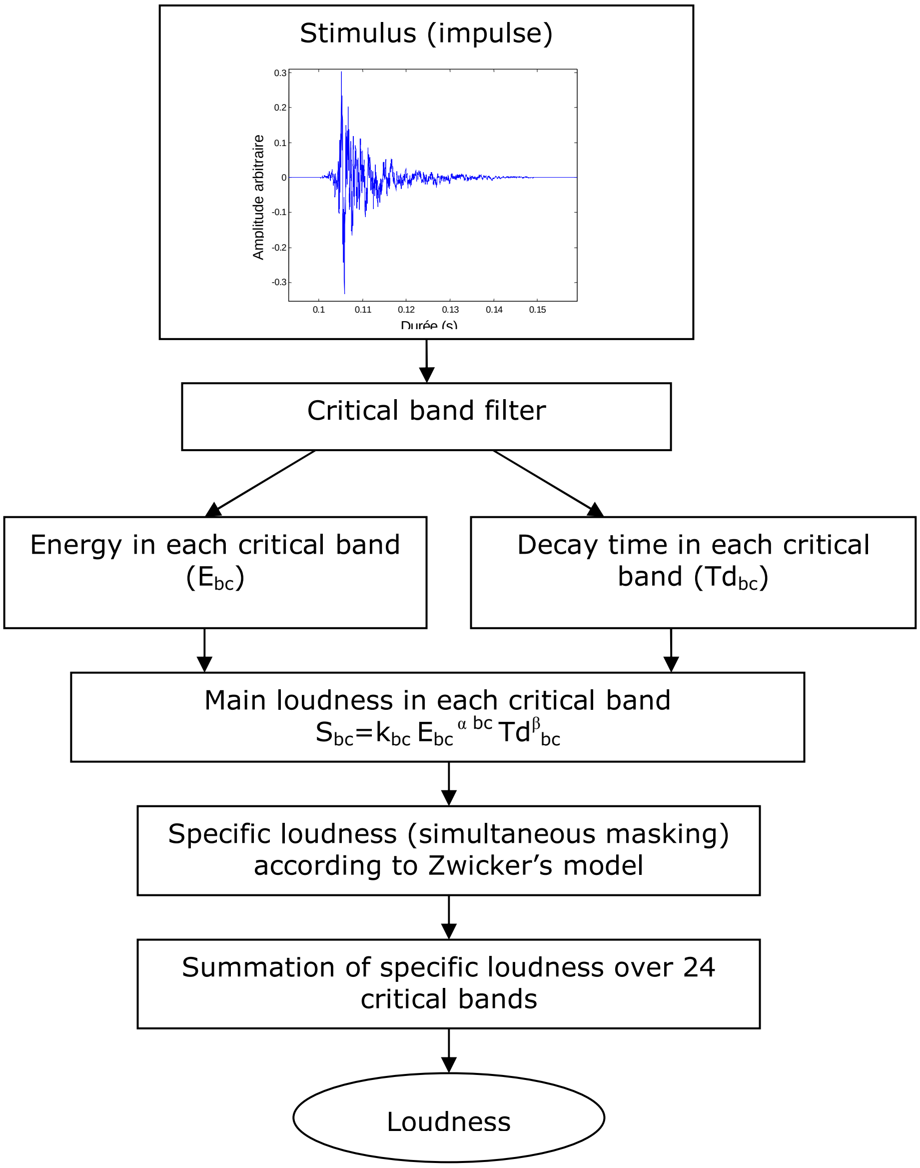

Figure 18: Stages of the loudness calculation for impulsive sounds.

4 Loudness indicators for time-varying sounds

Both models presented in section 3.1 and 3.2 calculate loudness as a function of time. The authors recommend statistical indicators to evaluate global loudness.

4.1 N4, N5 or N7 according to Zwicker and Fastl (1999)

Zwicker and Fastl's model allows the calculation of loudness as a function of time. The authors suggest the percentile loudness N7 (used for speech), N5 (used for environmental sounds) or N4 (used for road traffic).

In a generic way, indicator Nx represents the loudness value exceeded during x percent of time.

4.2 STLmax and LTLmax according to Glasberg and Moore (2002)

Glasberg and Moore's model, presented in section 3.2, allows the calculation of loudness as a function of time for time-varying sounds. The authors recommend calculating the maximum of short-term loudness (STLmax) to estimate overall loudness of time-varying sounds. STL is the loudness perceived at each instant.

The maximum of the long-term loudness (LTLmax) is recommended by Glasberg and Moore to estimate overall loudness of steady sounds or of sounds varying slowly in time. Long-term loudness (LTL) renders the fact that the overall loudness of a sound is kept in memory for several seconds after its end, in the absence of a new stimulus.

4.3 LMIS according to Boullet et al. (2006)

LMIS (Loudness Model for Impulsive Sounds), presented in section 3.3, allows the calculation of overall loudness of impulsive sounds. It has been validated on a set of impulsive sounds with peak levels contained between 56 and 92 dB SPL and with durations varying from 10 to 1620 ms.

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ANNEX 1 : References

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