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SPECIFIC LOUDNESS EXPRESSION OF QUASI-STEADY SOUNDS BY CONSIDERING PERCENTILE DISTRIBUTION

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Abstract

In previous studies, the loudness of time-varying sounds has not been defined as a function of frequency because the spectral summation takes place prior to the temporal integration in Zwicker's and Fastl's certification. However, sounds such as railway noises have both time-varying and quasi-steady state characteristics because railway noises are defined only within exposed time. In this study, the time-varying loudness of the quasi-steady sounds is derived with respect to the critical band rate by percentile value. Frequency contribution of railway noise sources will be discussed through the comparison between one-third octave filtering and specific loudness.

INTRODUCTION

The main purpose of noise abatement is to reduce the A-weighted sound pressure level because it is convenient to evaluate the noise itself and to find related noise regulations as well. However, perceived noise quantities such as annoyance are different from sound pressure level, so that noise evaluation should be properly applied based on human response. Therefore, the noise evaluation using A-weighted sound pressure level might cause inaccurate or even misleading results for planning countermeasures of noise reduction.

In psychoacoustics, loudness is defined as main factor determining sound quality based on human sensation of "loud"[3], and it is closely related to annoyance and noisiness[6]. Subjective test is generally accepted for the evaluation of loudness because auditory perception of human hearing system depends on subjective feelings. Even if the loudness of steady sounds has been standardized in ISO 532B, the information on both human hearing system and cerebrum is not yet sufficient. However, the method based on the listening test cannot be used for planning countermeasures for the improvement of sound quality because most of listeners use unphysical dictions such as “loud” rather than physical quantities. Especially, in case of long-term unsteady sounds, subject cannot exactly evaluate the degree of unphysical diction due to the limitation of human sensory memory. Therefore, it can only be used as additional factor in total sound quality evaluation.

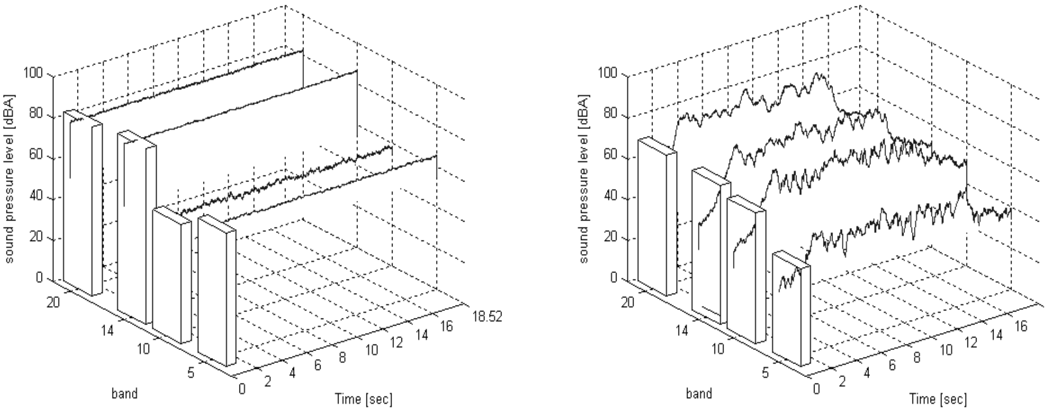
In this paper, time-varying loudness is calculated through the previous researches to evaluate quasi-steady state noise and it is applied to railway noises. For this, loudness of steady sounds according to DIN 45631[4] is considered. And then, three additional routines are applied to the standard calculation procedure to reflect the nonlinear temporal processing on the human hearing system[7][11][12]. One of them, envelope smoothing, is improved employing a Levenberg-Marquardt algorithm[1] to minimize the sum of the least-squares error between original data and smoothed data. To pattern the specific loudness of quasi-steady sounds, meanwhile, the sequence of spectral and temporal integration is reversed considering percentile loudness for each critical band rate. The change of the integration sequence is verified from the comparisons between ISO method and median value composition using time-varying routine. Finally, analyses with calculated metrics such as time-varying loudness and sharpness are compared with measurement of A-weighted sound pressure level of railway noises.

PREVIOUS STUDIES ON TIME-VARYING LOUDNESS CALCULATION

Because most natural sounds are not steady but strongly time dependent, the loudness for such sounds must be a function of time. Zwicker found out that sensation level is dependent on exposure time to sound and this is one among nonlinear effects of human auditory[2]. Zwicker’s loudness model for time-varying sound was reported the outline of algorithm but not detailed parts of algorithm. For this reason, loudness models for time-varying sound have been suggested by using various digital filter banks, consideration on more exact temporal effects[5] [9]. However, an analysis result, which is based on one of algorithms, is different from another since loudness model for time-varying sound has not been standardized yet. Three additional routines are required in the calculation of loudness model for unsteady sounds contrary to loudness model for steady sound. These are power envelope extraction from 1/3 octave-bands[12], simulation of post masking[11], and temporal integration[7].

ENVELOPE EXTRACTION FOR UNSTEADY SOUNDS

Contrary to steady sounds, the nonlinear temporal processing is required in the calculation of loudness if the time series of sounds are strongly time varying. Thus, it is necessary to extract envelope of the pass band signal for unsteady sounds because non-stationary characteristics should be reflected on loudness model. The moving average technique is popular for the extraction of time envelope as well as the evaluation of steady state condition of sound. Smoothing constant of 0.077, which is identical to 0.125 second of moving average window length, is generally applied to the signal to evaluate whether the sound is steady state or not.



(a) Steady sound *(b) Unsteady sound*

Figure 1 – Relation between time pattern and its average value at 5th, 10th, 14th, 20th band

Figure 1 shows steady state evaluation of noise signal by using moving averaging filter for each 1/3-octave band. The sound pressure levels in Figure 1 (a) are constant regardless of time, while the levels in Figure 1 (b) are fluctuant with respect to time for each 1/3-octave band, respectively. In addition, if each signal in band is constant through the exposed time, then an average value of time signal in each band can be considered as the representative of that band. The bars on the figure indicate the average value through the time at 5th, 10th, 14th, 20th band, center frequency of 63, 200, 500, and 2000 Hz. It is clearly shown that distributed values with respect to time in each band almost equal to an averaged value in each band for the steady sound. Hence, this value can represent total characteristics of band passed acoustic signal regardless of the time shown in Figure 1 (a). However, an averaged value described in Figure 1 (b), in case of unsteady sound, cannot cover total characteristics within whole range of time history. Thus, time-varying characteristics of acoustic signal should be implemented to model loudness of unsteady sound.

Zwicker assumed 2 ms of time resolution for more than 1 kHz center frequencies in time-varying loudness calculation in order to extract 1/3-octave band envelope[3]. At low frequencies below 1 kHz, filter time constants are increased for digital implementation[7]. Exponential smoothing is also used to make proper time pattern of each band noise. The formulation of this model is[8].

$$S(t) = \beta \sum_{j=1}^{t-1} (1-\beta)^{j-1} y(t-j) + (1-\beta)^{t-2} S(t-2),\ t \geq 2 \tag{1}$$

$y(t)$ indicates observed data point within current local window, and $S(t)$ denotes smoothed time series at current window or predicted time series from previous interval.

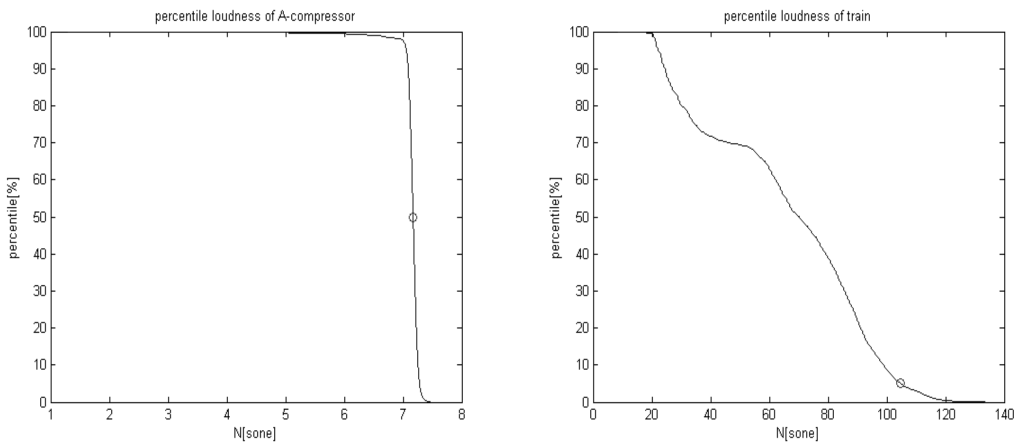
The smoothing parameter β controls the closeness of the interpolated value to the most recent observation. When fast-mode (1/8 second) smoothing is applied to the signal, the least-squares error between original and smoothed data exists so that it might lead wrong estimation of signal pattern. Consequently, the problem in envelope extraction is minimization of sum of least squares error. For this, nonlinear unconstrained optimization problem can be derived as

$$\min.\ f(t) = \frac{1}{2}\sum_{i=1}^{m} e_i^2(t)\Big|_{\beta},\ e(t) = y(t) - S(t-1),\ 0 < \beta \le 1 \tag{2}$$

Since error function $e(t)$ is defined as the difference between observed data at current window and predicted data at previous window from the algorithm, objective function $f(t)$ is non-linear function of β. This problem, which is to find optimal smoothing parameter satisfying objective function, can be solved by applying one among the most widely used optimization algorithms. The Levenberg-Marquardt Algorithm[1] is applied for the computational efficiency in this study.

THE USE OF PERCENTILE LOUDNESS FOR EACH CRITICAL BAND RATE ON THE EVALUATION OF QUASI-STEADY SOUNDS

A percentile is a value on a scale of one hundred that indicates the percent of a distribution that is equal to or below it. In practical signal processing application, percentile estimation is, in most cases, synonym for the estimation of 50 % percentile, i.e., the median[10]. In the calculation of time-varying loudness, it is important to know which value is able to represent the whole loudness over the time to find the value that is closely correlated with listening test results. Zwicker and Fastl found that N5, which is the loudness value reached or exceeded in 5 % of the measurement time, is well



(a) Steady sound distribution, circle: median *(b) Unsteady distribution, circle: N5*

Figure 2 – Percentile loudness distribution with respect to time

correlated with the judgement of loudness by subjects for the different sorts of noises such as aircraft, road traffic, and train[3]. Figure 2 shows the condition of noise using percentile loudness distribution with respect to time. The circle in the figure represents the representative value over the whole time range, which is identical to percentile loudness for each critical band rate. In loudness summation from the calculation procedure of time-varying loudness, there are two kinds of integration, namely spectral and temporal ones, so that it is important to know which one takes place first. Zwicker and Fastl assumed as spectral summation comes first from the sequence of anatomy and proved this assumption by using strongly frequency-modulated pure tones[3]. This means that specific loudness is not exactly expressed on loudness of unsteady sounds. However, the sound such as railway noise has both time-varying characteristics and quasi-steady state ones because noises emitted from the train are not only defined within exposed time but also composed of similar pattern. Considered as characteristics of steady sounds, and used by calculation routine of time-varying loudness, composition of median values for each critical band rate can be compared with the result on specific loudness using DIN 45631, as shown in Figure 3. It can be also applied in quasi-steady sounds if 5-percentile value is used instead of median because the result is almost identical.

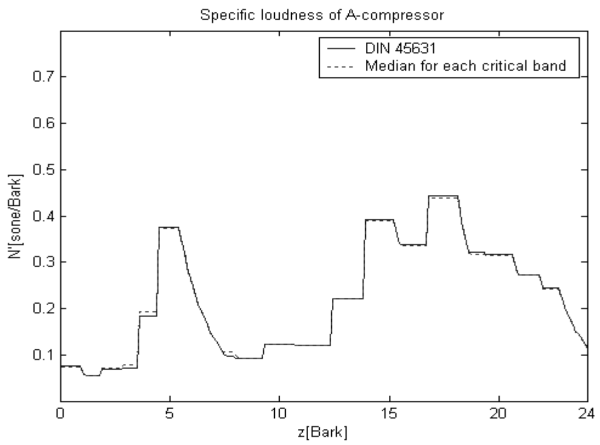
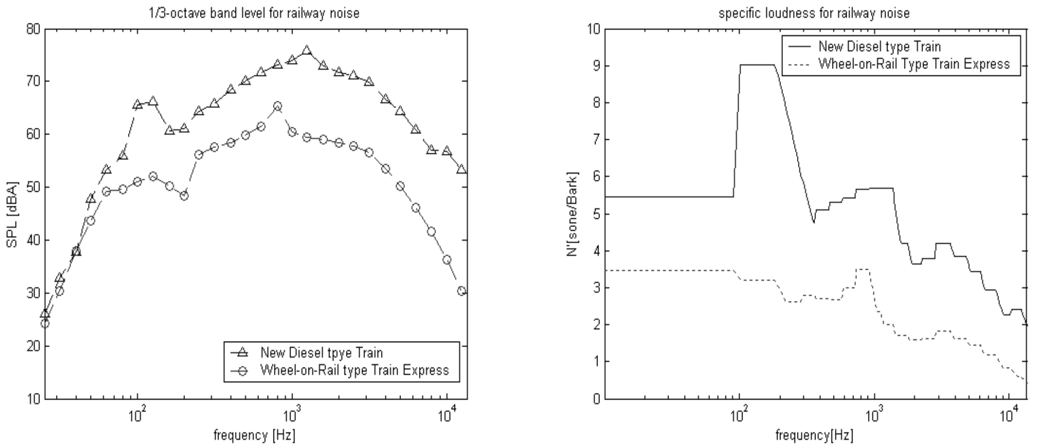


Figure 3 – Specific loudness by either DIN 45631 and median of time-varying routine

APPLICATION TO RAILWAY NOISE

Loudness comparison of spectral contribution with 1/3-octave band level

Railway noise is one of the unsteady sounds, and it has not only strongly time-varying but also quasi-steady characteristics. From the consideration of previous chapter, specific loudness can be expressed with respect to critical band rate. Figure 4 shows the comparison of spectral contribution between A-weighted 1/3-octave band level and specific loudness. For a better comparison, critical band rates are converted into frequency scales. Circles and triangles in Figure 4 (a) indicate New Diesel Car (NDC) type Train and Wheel-on-Rail type Train eXpress (WRTX), respectively. Solid and dotted lines in Figure 4 (b) identify type of train. Figure 4 demonstrates that spectral contribution is different in terms of annoyance although some of noise sources are different according to the types of train.

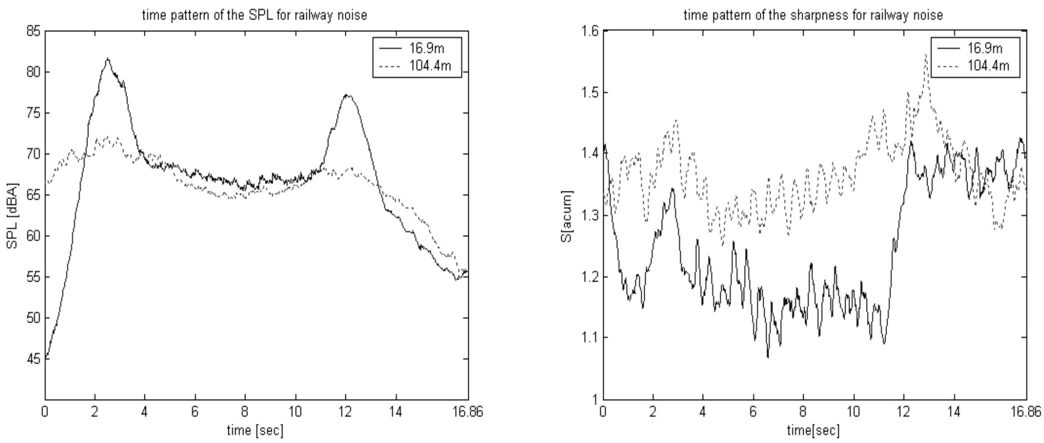


(a) 1/3-octave band level [dBA] *(b) Specific loudness [sone/Bark]*

Figure 4 – Comparison between A-SPL and N' in frequency domain

Sharpness comparison with sound pressure level in time

Since railway noises are measured according to distance in this study, characteristic with respect to distance can be considered in terms of either sharpness or sound pressure level. Figure 5 (a) shows the time pattern of sound pressure level, and Figure 5



(a) Time pattern *(b) Sharpness with respect to time*

Figure 5 – Comparison between SPL and sharpness with respect to distance

(b) shows the corresponding sharpness pattern. Solid and dotted lines correspond to distance of 16.9, 104.4 meters, respectively. It can be normally said that A-weighted sound pressure level decreases with respect to distance due to acoustic propagation. However, contrary to A-weighted sound pressure level, sharpness of noise from distant way is larger than that from close-in way. This implies that sharpness is more sensitive than sound level or loudness to people who live in relatively distant place from railway.

CONCLUSIONS

In terms of annoyance, noises emitted from products or vehicles should be evaluated by sound quality metrics rather than by A-weighted sound pressure level. For this, time-varying loudness is considered in order to compare spectral contribution, and sharpness whose input data is specific loudness is also considered for analyzing human auditory characteristic with respect to distance. One of the additional routines in time-varying loudness calculation, the envelope smoothing, is exactly reanalyzed to reduce the least squares error between original signal and smoothed signal by applying Levenberg-Marquardt algorithm.

From the assumption about that railway noise is in quasi-steady state, it is possible to define critical band rate expression of time-varying loudness. This can be certified from comparing calculation procedures for steady sounds. Median value is used in time-varying routine, and it is compared with specific loudness of standard method.

In application, spectral difference between N5 and 1/3-octave band as well as temporal difference between sharpness and time SPL are considered in terms of annoyance. Since railway noise is closely related to environmental noise especially, the approach considered in this study is more useful than conventional approach that is measurement of A-weighted sound pressure level.

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