Loudness and Perception of Sound

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ABSTRACT

This paper presents basic data on loudness level and loudness along with data obtained by the authors, and describes an application of the idea of masked loudness to perception of music in the presence of noise. It is shown that timbre or sound quality of music is well explained by masked loudness vs frequency characteristic.

I. INTRODUCTION

Loudness is one of the principal attributes of audible sound, and its expression has been studied from various points of view. The fact that loudness of sound depends upon its frequency has attracted the notice of acousticians since the beginning of the 20th century, and data on loudness of pure tones were presented in the 1920's by Kingsbury(1) and others.

In 1933, equal loudness level contours were first published by Fletcher and Munson.(2) The unit of loudness level is "phon", and a 70 phon curve, for example, shows the sound pressure levels of tones at various frequencies which have the same magnitude of sensation as that of 1 kHz pure tone (reference tone) at a sound pressure level of 70 dB. They proposed a method of calculating loudness of complex sound consisting of discrete concurrent frequency components, and a method of evaluating the magnitude of noise was studied on this basis. In 1937, the unit phon was formerly defined as an international unit of loudness level, and a noise meter for measuring loudness of noise was designed by Churcher and King(3).

In 1936, tentative standards of sound level meters for measurement of noise were published by a technical committee of the Acoustical Society of America, in which the frequency-weighting characteristics A and B were proposed.(4) An A-weighting curve was drawn based on the equal loudness level contour of 40 phon and a B-weighting curve was drawn based on the contour of 70 phon. In other words, the earliest application of loudness characteristics was to the establishment of a method of noise evaluation and that idea is still alive.

After that, S. S. Stevens of the USA began to study loudness sensation in the 1940's and the results of his work can be seen in a series of his papers, Mark I published in 1956 through Mark VI published in 1972. A little later, Zwicker of Germany developed his own method of loudness evaluation based on the excitation pattern of hearing, and his method is now recognized as being the most reliable one for loudness calculation.

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In this article the authors intend to show the basic idea of loudness first and then to mention an application of the concept of masked loudness to a practical case. All of the data cited here have been published.

II. LOUDNESS LEVEL AND LOUDNESS

2.1 Loudness level

Loudness level is defined on the basis of the relation between equal loudness vs frequency characteristics as stated above. The equal loudness level contours adopted as the present international standard, ISO 226, [5] are from Robinson and Dadson's data[6] published in 1954 and shown in Fig. 1. These contours are slightly different from those obtained by Fletcher and Munson because they show some depression at around 400 Hz. Four years ago, doubt was cast on this depression by a member body of ISO / TC43 as a result of a loudness estimation using Zwicker's method as well as a subjective experiment. After discussion, ISO / TC43 decided to undertake revision of ISO 226 as a new work item of the committee. This proposal was approved by the member bodies, and its implementation was referred to WG 1.

In response to this decision, we decided to carry out experiments to obtain new equal loudness level contours under free field listening conditions with an up-to-date measurement technique. We also planned to try to obtain the minimum audible sound field.

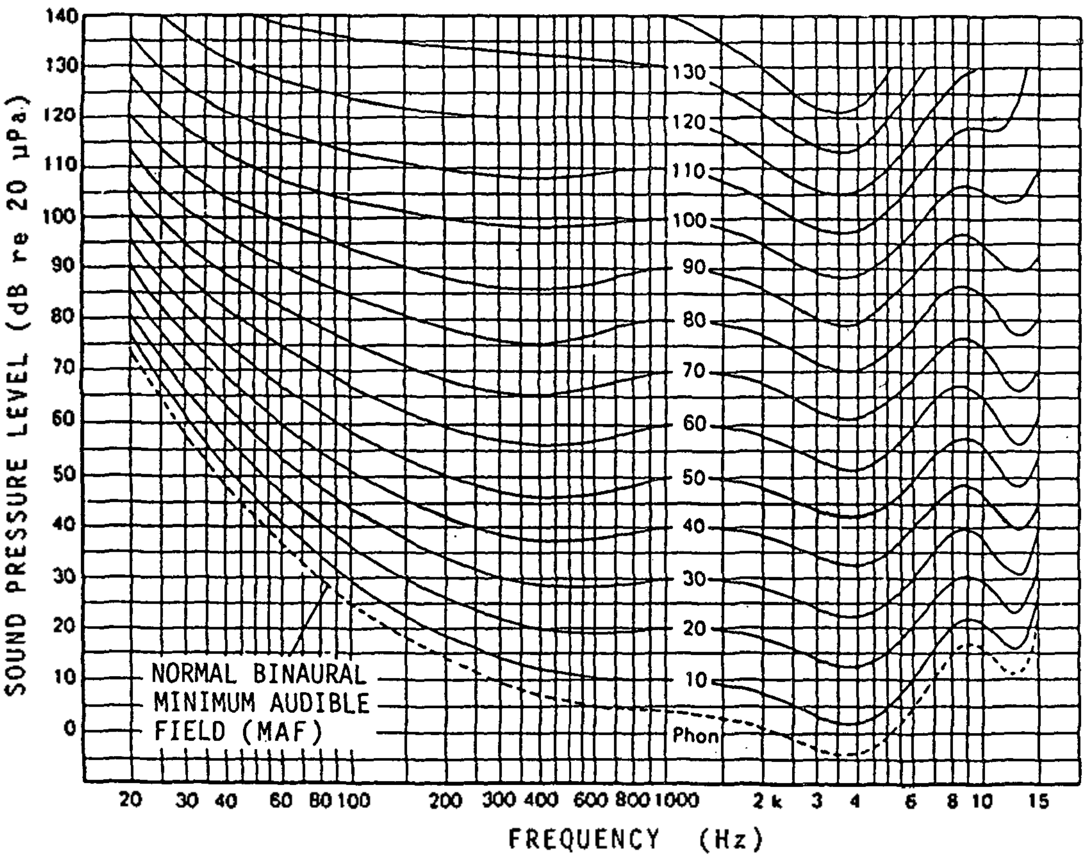

Fig. 1 Normal Equal-Loudness Contours for Pure Tones(ISO 226)
(Binaural Free-Field Listening)

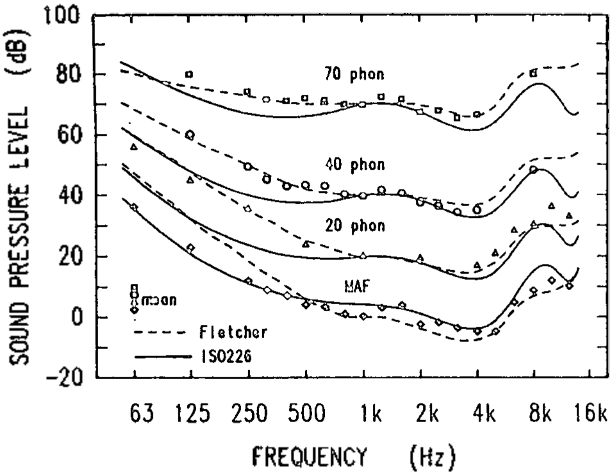

Fig.2 Equal-Loudness Relations and the Minimum Audible Sound Field for Pure Tkoes under Free Field Listening Conditions (Sone et al.)

Fig. 2 shows the results of the first stage experiment.[7] Squares, circles, triangles and diamonds represent the experimental results, while solid lines show the contours from ISO 226 and broken lines are taken from Fletcher and Munson's data. From the figure, our data can be seen to be similar to those of Fletcher and Munson up to 1 kHz rather than to ISO 226, except for the minimum audible sound field. The depression at around 400 Hz in the contours of ISO 226 disappears in our data.

In order to determine on experimental factors that might cause the difference in contours, we examined the effect of methodological differences on results. If we compare the experimental conditions of the three groups, that is, Fletcher and Munson, Robinson and Dadson, and our group, some differences can be found. In particular, we have to take note of the difference in the choice of stimulus levels for a single run of experiment.

Robinson and Dadson fixed the level of test tone and varied the level of standard tone in random order, Fletcher and Munson used a method similar to that of Robinson and Dadson, except that the level of test tone was randomly selected from among three different levels. In this way, they intended to lessen the possibility of bias error caused by presenting test tones at a constant level throughout a test run. They called the effect which might cause this type of error the memory effect. We fixed the level of reference tone and selected the level of test tone at random. In order to consider the effect of such a difference of experimental procedure on the results, we conducted two supplementary experiments.

The first one was to examine the effect of varying the level of standard or test tones. This experiment consisted of two kinds of tests. In Test 1, 1kHz standard tone was fixed at a constant level, while in Test 2 the level of test tone was fixed as in the experiment of Robinson and Dadson.

The second experiment was to examine the effect of varying levels of both standard and test stimuli, that is levels of neither the standard tone nor the test tone were fixed in this experiment.

In Test 1 of the first experiment, the level of standard tone was fixed at 70 dBSPL while the level of test tone was randomly selected from among 9 levels. In Test 2, the level of test tone was fixed at

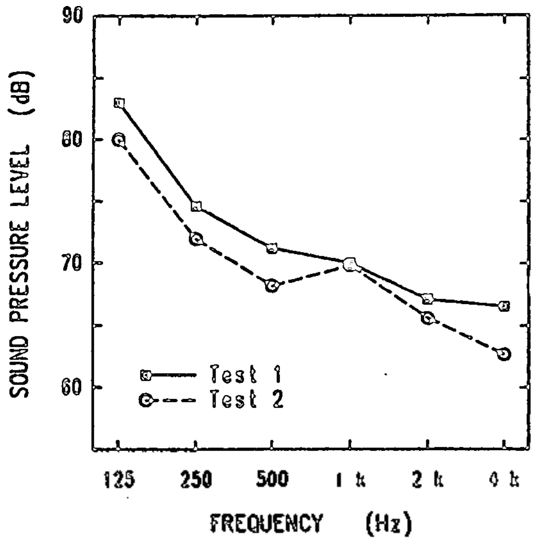

Fig.3 Results of the Supplementary Experiment
Test 1: Level of the Standard Tone is fixed.
Test 2: Level of the Test Tone is fixed.

the level of integer nearest to the 70 phon level measured in Test 1 for each subject.

Fig. 3 shows the average values in Test 1 and Test 2, respectively. In this figure, the depression at around 400 Hz is seen only in Test 2. This is very interesting because the way of selecting the level combinations of reference and test tones in Test 2 is the same as in Robinson and Dadsons's experiment. These results suggest that the subjects perceive the tone whose level is fixed to be louder than the other tone whose level is varied. According to the analysis of variance, the difference between Test 1 and Test 2 was significant beyond the 1 % level.

In the second experiment, levels of the reference tone and the test tone were randomly selected from among the lattice points shown on Fig. 4. The experimental results were similar to those of Test 1 of the first experiment. A statistical test also showed that there is no significant difference between the results of these two experiments, while the difference between the results of Test 2 and the second experiment is significant.

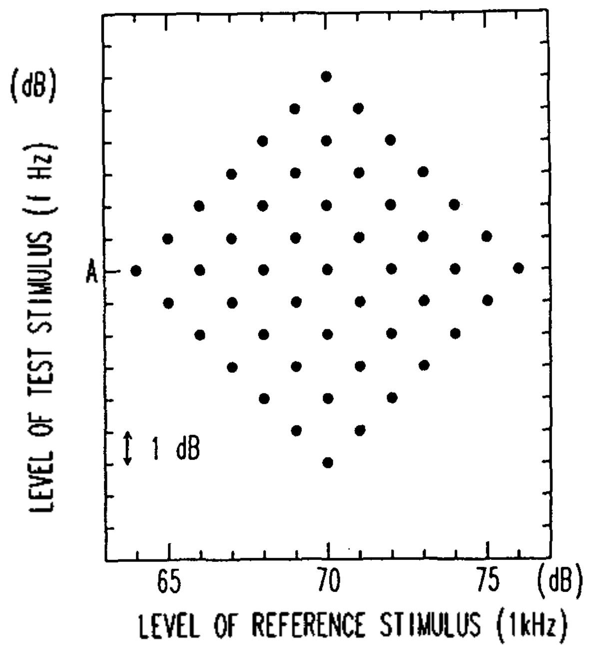

Fig. 4 Stimulus condition in the 2nd Experiment

In conclusion, test tone is felt to be louder than it is when its level is fixed and the level of standard tone is varied, while the loudness of test tone is correctly perceived when the level of standard tone is fixed and the level of test tone is varied.

2.2 Loudness

Loudness level is defined on the basis of the sensory magnitude of sound. However, the sound pressure level of 1 kHz pure tone is adopted as the reference for this, so that a doubling of loudness level does not mean a double magnitude in loudness sensation. Loudness scale was defined separately from loudness level. The unit of loudness is "sone".

The relation between loudness and loudness level was originally obtained by Fletcher and Munson and has been investigated by several researchers since then. According to those studies, a doubling of loudness corresponds to an increase of 9 or 10 phons in loudness level. The relation between loudness and loudness level is approximately expressed by equation (1), [5][8] where P is loudness level in phons, and S is loudness in sones. Loudness of a sound of 40 phons is defined as 1 sone.

$$S=2^{\frac{P-40}{10}} \tag{1}$$

According to this equation an increase of 10 phons in loudness level corresponds to a doubling of loudness in sones.

It is widely recognized that there is a relation expressed by power function between loudness and intensity of sound. Equation (2) shows that relation, where S means loudness in sones, k is a constant, and I represents a sound intensity in [W / m²]. The power is about 0.3 and it somewhat increases in the low frequency region.

$$S=kI^{a} \tag{2}$$

2.3 Method of calculating loudness

In order to obtain the loudnes of a pure tone, the sound pressure level of the tone must be changed into loudness level, and then its loudness is calculated by using equation (1).

As for the loudness of sound with a wide band spectrum, on the other hand, mutual masking among components should be taken into consideration, and several methods of loudness estimation have been proposed. Three methods of loudness estimation are mentioned here from a practical point of view.

2.3.1 The method after Stevens

The first method is Method A stated in ISO 532 [9] It is a method for calculating the loudness of steady complex sounds for which octave-band analyses are appropriate. The level in each octave band is converted into a loudness index, and then the total loudness in sones in calculated by means of an empirical formula as shown in Eq.(3), where S_t is the total loudness in sones, S_i represents the loudness index of the ith band and S_m is the greatest loudness index of all. The value F is 0.3 for octave bands, and 0.15 for one-third octave bands.

$$S_t = S_m + F(\sum S_i - S_m) \tag{3}$$

The chart and the table for converting each band level graphically to a loudness index are provided in ISO 532. The loudness level in phons may be calculated by means of Eq.(1)

In Stevens' method, the mutual masking effect is not so precisely treated as in Zwicker's method. Especially when the one-third octave band analysis is applied, Zwicker's method is recommended rather than Stevens' method. But the simple procedure for calculating loudness by Stevens' method is still attractive.

2.3.2 The method after Zwicker

Zwicker's method is mentioned as Method B in ISO 532, and it specifies a procedure for calculating the loudness of steady complex sounds to which one-third octave band analysis is applicable. In this method, a chart drawn from the excitation pattern

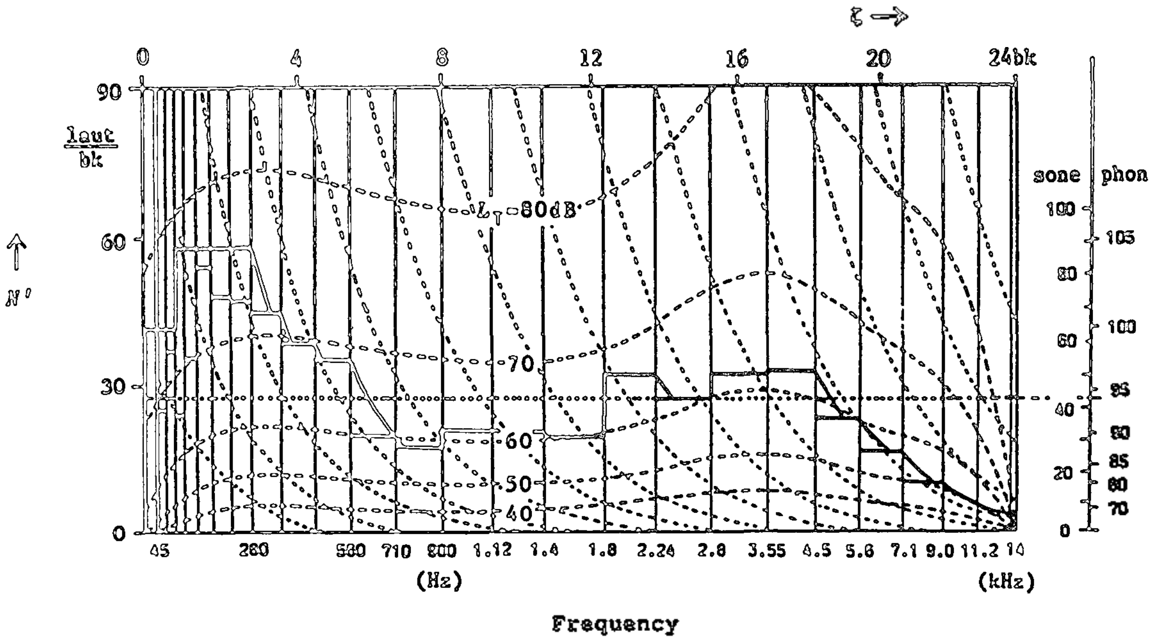

Fig. 5 An Example of Loudness Calculation by ISO 532 B

of hearing is used to obtain the total loudness of complex sound.

Zwicker defined the band width corresponding to 1.3 mm of the basilar membrane as the critical band width, and designated its unit as "bark". Then the whole audible frequency range was divided into 24 barks. Fig. 5 shows an example of the process of loudness calculation by means of a chart. Here the one-third octave band levels are transformed into parts of an area which correspond to parts of loudness. When the level in the adjacent higher frequency band is lower, the fall is drawn as a downward sloping curve interpolated between the dashed curves on the graph, starting from the right-hand end of the horizontal line. This sloping curve corresponds to an excitation pattern of hearing and shows the masking from the band in question against the adjacent higher frequency band. The area enclosed by the whole stepped figure corresponds to the total loudness. If we transform the enclosed area into a rectangle of the same area which has a base equal to the width of the graph, the height of the rectangle directly yields the loudness in sones or the loudness levels in phons, which can be read from the nomogram on the right side of the graph. A computer program for calculating loudness by this method is also provided.[10]

2.3.3 Perceived level

In the last version of a series of studies on loudness, Stevens proposed a scale designated as perceived level.[11] The features of this revision are as follows:

1) The reference sound used in the method was changed from 1 kHz pure tone to one-third octave band noise centered at 3150 Hz. The frequency at which the threshold of hearing shows the minimum value is 3150 Hz.

2) the form of power function was changed to equation (4), where S is the perceived magnitude of reference sound with intensity E, and threshold E_0 corresponds to -3 dBSPL.

$$S=k(E-E_0)^{1/3} \tag{4}$$

3) The coefficient F of equation (3) is provided as a variable depending upon the greatest loudness index, S_m.

4) Perceived level (PLdB) is obtained by means of equation (5), where S_t is the total loudness and P is the perceived level in dB.

$$S_t=2^{\frac{P-32}{9}} \tag{5}$$

2.4 Loudness of sound with short duration

2.4.1 Tone burst

That the loudness of tone burst is smaller than that of a steady tone which has same amplitude is a well-known fact. The increase in loudness of sound with an increase in its duration is considered to represent a feature of temporal summation in hearing. Although much research has been done on this problem over a long period of time, it is still not clear. For example, Munson calculated the loudness of sound with a duration up to 100 seconds on the assumption that loudness was proportional to the number of neural impulses in the auditory system.[12]

Considering the adaptation of neurons to loudness perception, he came to the conclusion that the loudness of tone burst increases with its duration up to about 300 ms and that beyod one second, the loudness decreases with an increase in duration. Experimental results which deny the existence of any adaptaion effect in loudness perception have been reported by many researchers, and indeed the view that there is not an adaptation effect in loudness summation seems to hold an advantage over the opposite view at this moment. Even though there is not an adaptation in loudness perception, however, the rate of loudness increase with tonal duration is never uniform. For example, it is said that the increase in loudness with tonal duration may be approximated by sound energy for a shorter duration than about 80 ms but that the rate of loudness increase depends upon the duration of comparison stimuli.[13]

2.4.2 Impulsive sound

Impulsive sounds, such as those from a pile driver, a car door being slammed gunfire, a typewriter and so on, are different from tone bursts in that they do not have any steady duration. In the study of loudness of simulated impulsive sounds where the rise time, steady part, decay time, peak level, carrier signal and repetition rate were taken as parameters, we came to the following conclusions:

1) Loudness of a single burst of impulsive sound shows a good correlation with the frequency-unweighted sound exposure level (LPE).[14]

2) The coefficient for linear regression of PSE on LPE is about 0.6 for loudness of repeated impulsive sound, while it is about 1.0 for noisiness.[15] This seems to show that the time constant for integration of sensation is greater for noisiness than for loudness.

III. PARTIALLY MASKED LOUDNESS

3.1 Masked loudness of pure tone

The expression that the sound A is masked by another sound B implies in general that A cannot be heard any more in the presence of B. In this case the loudness of A should be considered as zero, since it cannot be heard. There is an intermediate condition, however, in which sound A is perceived as being softer than it would be in the absence of sound B. This phenomenon is called partial masking. For example, we frequently experience situations in which speech or televised sound is difficult to hear at the instant when a train or a heavy truck passes nearby. Loudness of a specific sound partially masked by another sound is called masked loudness or partially masked loudness of the specific sound.

Masked loudness of pure tones was studied by Lochner and Burger, Zwicker, Hellman, Stevens and others. The loudness of tones which have levels near thresholds will be mentioned first, because it is closely related to masked loudness.

It has been shown that the relation between loudness and intensity of pure tones is expressed by a power function. Loudness of a pure tone near threshold, however, is not a simple function of its intensity, and the expression for loudness in such a case is written in a modified form as in equations (6), (7) or (8). Here S' is the loudness of a tone in sones, k is a constant, I represents the intensity of the tone, P is the sound pressure, and I_0 and P_0 show the intensity and the sound pressure of a tone at its threshold of hearing, respectively. The value of is about 0.3. Eq.(6) was proposed by Stevens,[11] Eq.(7) by Sharf,[16] and Eq.(8) by Lochner and Burger.[17]

$$S'=k(I-I_0)^a \quad \text{(Stevens)}, \tag{6}$$

$$S'=k(P-P_0)^{2a} \quad \text{(Sharf)}, \tag{7}$$

$$S'=k(I^a-I_0^a)=S-S_0 \quad \text{(Lochner)}. \tag{8}$$

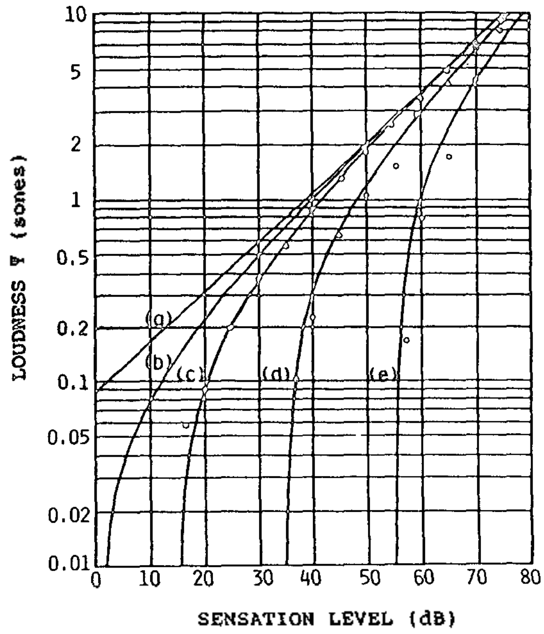

Fig. 6 Loudness of a 1000Hz pure tone in the presence of physiological noise (b) and octave band random

noise that gives pure tone threshold levels of 15dB(c), 35dB(d) and 55dB(e). The straight line (a) is the curve for $\psi = kI^n$.

It is thought that the threshold of hearing is determined by physiological noise generated inside the ear. According to this view, it is expected that partially masked loudness may also be expressed by these equations, since only the source differs between an internal noise and an external noise. Which of the above three equations is best seems to depend on the interpretation of the equations, but we have adopted Eq.(8) here. Fig. 6 shows the experimental results obtained by Lochner. Solid lines are calculated from Eq.(9). This equation means that the masked loudness of a tone is obtained by subtracting the threshold in terms of loudness from the loudness of the tone in the absence of a masker. It expresses the psychological additivity of loudness well.

3.2 Masked loudness of a complex tone and a sound with a wide band spectrum

As to the masked loudness of a complex tone, we studied sounds with two or three components. the results showed that masked loudness could be explained by Eq.(8) when a masker had a wide band spectrum, but could not always be explained when a masker was a narrow band noise. Namely, the loudness of a complex tone partially masked by a narrow band masker is perceived as being a little louder than that masked by a wide band masker insofar as the calculated loudness in the same.

A study on the optimum level of music listened to in the presence of noise is a practical case in which

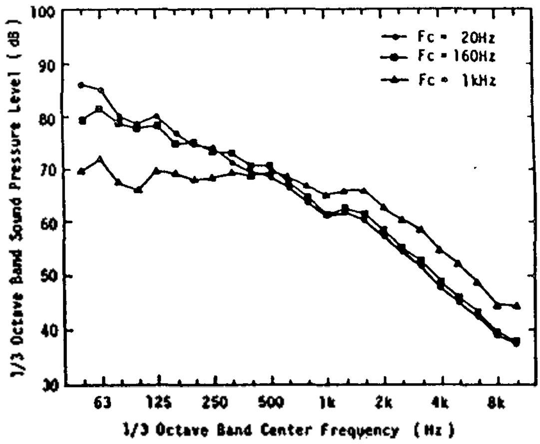

Fig. 7 1/3 octave band pressure level of the simulated car noise corresponding to 75 dBA.

the musical sound is partially masked by wide band noise.

As for listening to music reproduced through a playback system in the presence of noise, the allowable limit of noise annoyance for listening to music, the allowable limit of deterioration of the quality of music by noise, the intensity of music listened to under noisy conditions, the frequency response characteristic of the playback system, and the dynamic range of reproduced music may be important for determining the optimum listening conditions.

A typical situation of listening to music in the presence of loud noise is met when we drive a car. The noise is not steady, so that the level and the spectrum of noise can vary according to driving conditions and the property of the road surface. However, human behavior under such time-varying noise conditions can be understood to some extent if the behavior under steady noise conditions is cla-

Table 1 Five pieces of music used as source sound.

Music A:	Violin Solo	Partita for Solo Violin No. 3	J. S. Bach
Music B:	Piano Solo	Alice in Wonderland	D. Brubeck
Music C:	Vocal & Orch.	Kaze no Machi	S. Ookawa *et al.*
Music D:	Jazz Quartet	Aliança	P. Desmond
Music E:	Orchestra	The Red Pony Suite	A. Copland

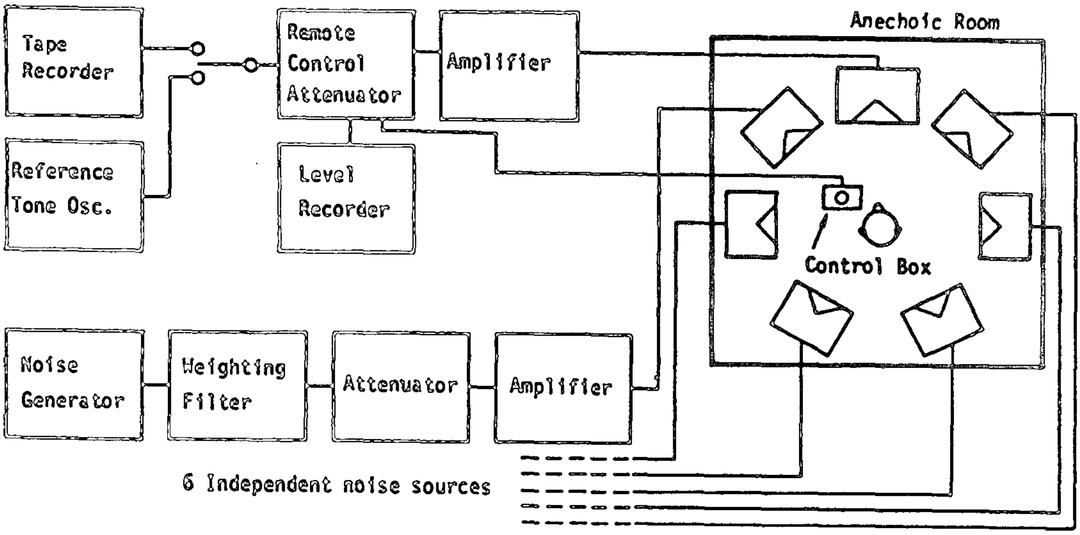

Fig. 8 Block diagram of the apparatus used for the experiment

rified. We carried out an experiment, therefore, to obtain the optimal level of music listened to in the presence of simulated car noise.[18] Fig. 7 shows the 1 / 3 octave band sound pressure levels of three kinds of simulated car noise.

The five pieces of music shown in Table 1 were used in the experiment Selection was made to include different types of music, a variety of musical instruments, a range of frequency spectrums and so forth. For subjective experiments, a pseudo-diffuse sound field of simulated car noise was synthesized in an anechoic room through six channels of a noise generating system as shown in Fig. 8. Noise radiated from six loudspeakers were uncorrelated to avoid definite sound localization. Music was monophonically presented.

In order to investigate the effect of noise caused by the difference in the frequency response characteristic of the reproduction system on the perception of music, two systems with different characteristics were used for music reproduction. The method of adjustment was used for the experiment, and the subjects were requested to adjust the level of music to the optimum loudness.

Fig. 9 shows an example of the relation between the average optimum level of five pieces of music and the A-weighted sound pressure level of noise. The numerals affixed to the different curves in the figure indicate the different subjects. The level of music is presented by its medium value. An analysis of variance of the results showed that the effects of subjects, the kind of music, the A-weighted sound pressure level of noise, the spectrum of noise, and the frequency response characteristics of the reproduction system were all significant beyond the 1%

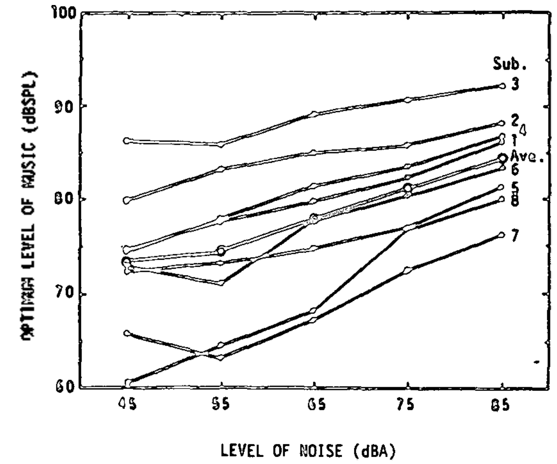

Fig. 9 Optimum listening level of music under EQ-I system. The ordinate indicates the level represented by the median.

point. The effects of subjects and the frequency response characteristics of the reproduction system, however, were far greater than those of the other factors.

It is apparent from Fig. 9 that the optimum listening level increases along with the increase in noise level and that the slope of the optimum lvevel against noise level is gentler for a lower noise level than for a higher one.

If the difference among subjects is considered, the subjects who preferred a higher intensity of music showed slightly gentler slopes against noise level than those who preferred a lower intensity. In other words, it can be said that the subjects who preferred a lower intensity of music were more affected by noise.

These results are considered from the viewpoint of masked loudness. In order to calculate the masked loudness of music, first, the loudness of music at the threshold level in the presence of noise was subtracted form the loudness of music under noiseless conditions for each critical band. Then the loudness for each band was summed up using the method of Mark Ⅶ by Stevens. However, since this method is based upon a 1/3 or 1 octave band analysis, the threshold level should be translated into a 1/3 or 1 octave band level.

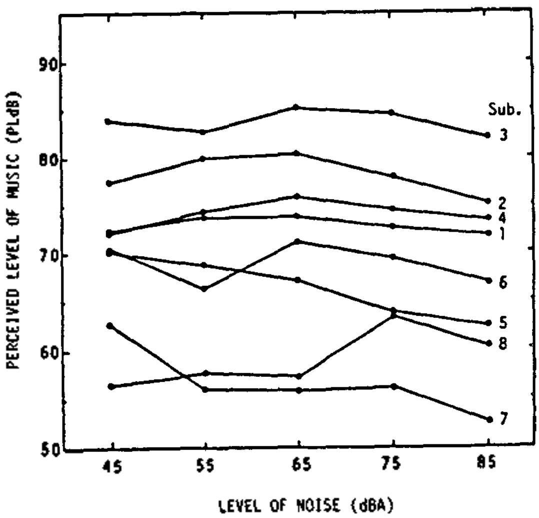

Fig. 10 Perceived level of music corresponding to its average masked loudness. (EQ-I system)

If we denote the spectral threshold level of the signal at a certain frequency by L_s, the critical band width around that frequency by W_{CB} and the 1/3 octave band width centered at the same frequency by $W_{1/3}$, then te threshold level of hearing in a 1/3 octave band level, L_b, is expressed by equation (9).

$$L_b = L_s + 10\log(W_{\frac{1}{3}}/W_{CB}) \quad (9)$$

Since the subjects judged some levels of music as being most preferable, there must be some basis for their judgment. It seems natural to say that a subject primarily intends to keep the loudness of music constant according to his preference. Subjects increase the level of music slightly against a moderate increase in noise level in order to keep the masked loudness of music at the same value.

Fig. 10 shows an example of the perceived level of music corresponding to its average masked loudness. In this figure we find that the loudness of music obtained in this way is nearly a flat function of noise level. Other data show similar results. From this consideration, it can be said that the optimum level of music is adjusted so that its loudness might be kept at a constant magnitude.

When we consider the standard curve denoting the changing rate of the level of music in relation

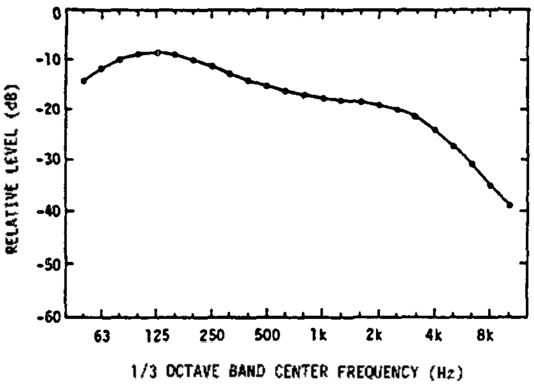

Fig. 11 1/3 octave band level of the CCIR standard signal. 0 dB means the total level.

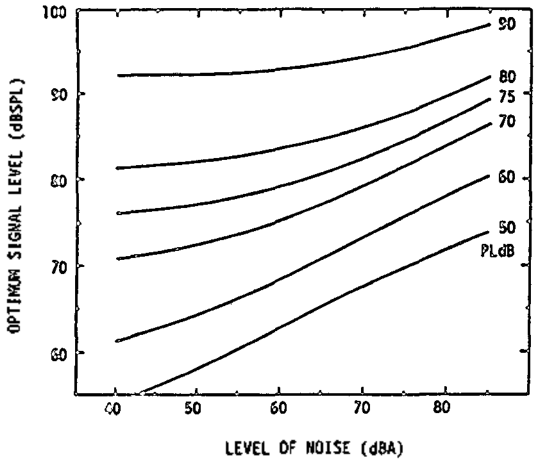

Fig. 12 The contours for equal perceived level as a function of noise level. The ordinate show the level of signal represented by the median.

to noise level, it is convenient to use an appropriate standard signal. We choose the averaged spectrum of broadcasting signals stated in CCIR Draft Record.[19] Fig. 11 shows the 1 / 3 octave band spectrum of the standard signal.

It is assumed here that the intensity of this signal varies while its spectral shape remains unchanged and that the standard deviation of the level fluctuation is 6 dB. It is also assumed that the distribution of the signal level is normal. This value of 6 dB was selected because it was the average standard deviation of music level fluctuation for the five pieces of music used in the experiments.

Fig. 12 shows and equal loudness contour for the signal against noise level: it corresponds to the experimental results already shown. It can be said, therefore, that this figure shows the standard curve of optimum music level against noise level. When the contour of 75 PLdB is considered, he change in the optimum level of music for a 10 dBA increase in noise level is about 3 dB for levels of noise between 55 dBA and 75 dBA and about 4 dB near 85 dBA of noise.

The standard curves obtained here are conveniently applicable to the situation in which automatic level control for use in the presence of noise is required.

Ⅳ. MASKED LOUDNESS AND SOUND QUALITY

When we consider the timbre or sound quality of a steady sound, it is clear that timbre is closely related with the power spectrum of the sound. Discrimination of timbre is considered to be made in accordance with intensity discrimination in a certain frequency band width.

Here, we assume that the psychological spectrum of sound is a projection of the physical spectrum onto the psychological space. It is natural to say that the physical spectrum undergoes some modifications in the process of hearing and yields a psychological spectrum. Masking is considered to be one of the main sources of this modification, and other sources may include the conversion of power into loudness, the transformation of frequency axis into logarithmic scale and so forth. We term the frequency spectrum after it has undergone such modifications as "masked spectrum".[20]

4.1 Effect of noise on the sound quality of music

As an example which suggests the contribution of partial masking to "masked spectrum", a study on the optimum frequency response characteristics of music reproduction in the presence of noise will be shown here.[21]

Even if the masked loudness of music listened to in the presence of noise is kept constant against the change of noise level, its frequency characteristics must undergo a change due to noise. To describe the timbre or sound quality of music, its 1 / 3 octave band Perceived Level vs frequency characteristics is referred to as PEC herafter.

Since the sensation of loudness of music is well described by its masked loudness averaged over the time duration of music above threshold, as shown

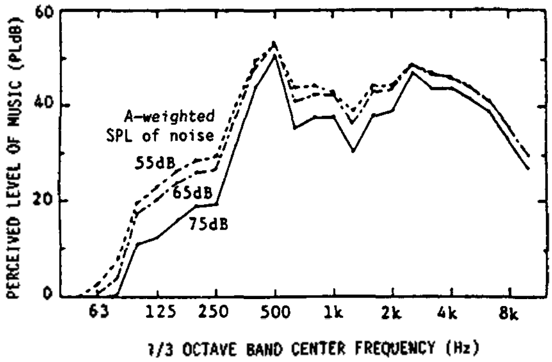

Fig. 13 The PFC's of music at its optimum listening level in the presence of noise similar to that in a passenger car.

in the previous section, the same procedure is again used here to derive PFC. This procedure is summarized as follows:

1) Masked loudness of music at each 1/3 octave band at any moment is given by subtracting the threshold loudness of music in the presence of noise from its loudness in the absence of noise at the same band.

2) This masked loudness at each band is averaged over the samples while the music is above threshold.

3) From the logarithm of these narrow band data, the PFC of music can be drawn up.

Fig. 13 shows the PFC's of music at optimum listening level obtained in the previous experiment. These PFC's are the averages for five different

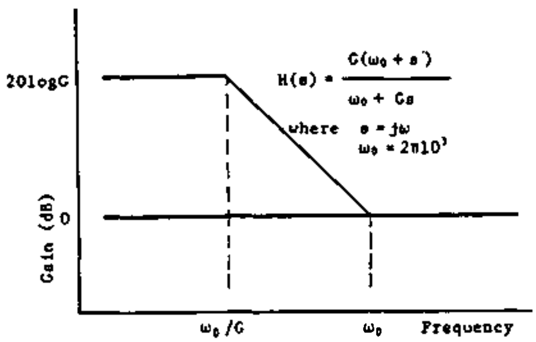

Fig. 14 The frequency transfer function of the low frequency booster. The amount of boost can be linearly changed in dB from 0 to 20.

pieces of music, and the masking noise is low-passed pink noise which is modeled after the noise in a passenger car.

4.2 Optimum amount of low frequency boost

Our experiments were carried out to obtain the optimum amount of boost in the low and high frequency regions in order to compensate for the deterioration of sound quality of music caused by noise.

First, music is modified by an equalizing circuit, the frequency transfer function of which is shown in Fig. 14. G in this figure represents the amount of boost and is exponentially variable form 1 to 10. This means that the amount of low frequency boost can be changed linearly from 0 to 20 dB. The amount of boost was controlled by a subject through voltage controlled amplifiers and the level of music was controlled by a microcomputer. Low-passed pink noise form six independent sources were radiated by six loudspeakers distributed around a subject so as to compose a pseudo-diffuse noise field. Music was monophonically pressented.

In the experiment, a piece of music was presented to a subject in an anechoic room in the presence of noise at 45 dBA as the referecne, and after a ten-second pause, the same piece of music was presented twice in the presence of ambient noise at 55, 65 or 75 dBA as test stimuli. Subjects were requested to adjust a degree of low frequency boost by sliding a knob so that the sound quality of test stimuli would resemble that of the reference stimulus as closely as possible. This adjusted value was read by a computer at the end of the test stimuli. Subjects were also asked not to use the loudness of music as a cue for adjustment, and not to pay any special attention to musical instruments or fixed parts of the music, but to listen to the total sound quality of the music.

Fig. 15 shows the relative PFC of music for three levels of noise. That is, PFC of the reference

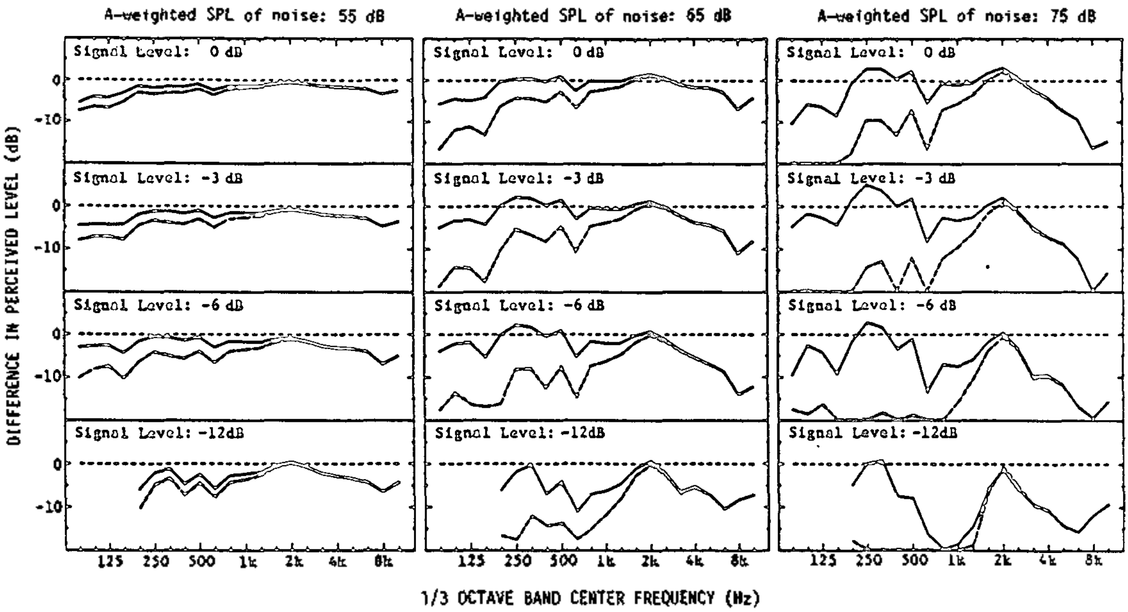

Fig. 15 PFC's of test stimuli are shown in a relative form to those of comparison stimulus. Dashed curves show the PFC's for no boosting state while solid curves show those for the optimum boost.

stimulus was subtracted form that of the test stimuli. Broken lines show the PFC's for a state of no boosting, while solid lines show those for the optimum boost. The signal level is expressed by a value relative to the optimum listening level obtained in the preliminary experiment. When the music level is high or the noise level is low, the PFC's are adjusted so as to be almost the same as that of the reference stimulus, since solid lines almost reach 0 dB in these cases. On the other

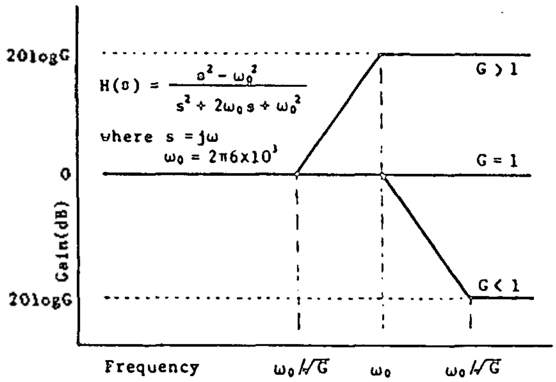

Fig. 16 The frequency transfer function of the compensation system for high frequency components.

hand, when the listening conditions are not good, compensaion for middle frequency components around 700-1500Hz is insufficient, though the compensation seems to be good for low frequency components.

4.3 Optimum amount of compensation in the high frequency region

Next, we conducted an experiment on compensation in the high frequency region as well. All conditions concerning noise and music presentation were the same as in the experiment for low frequency boost. As the interaction between compensations for low and high frequency regions was found to be negligible in the preliminary experiment, the amount of boost in the low frequency region was fixed at the averaged optimum level irrespective of subjects and music sources. The frequency transfer function of the compensation system is shown in Fig. 16.

The results are shown in Fig. 17. Subjects reported that the compensation in the high frequency

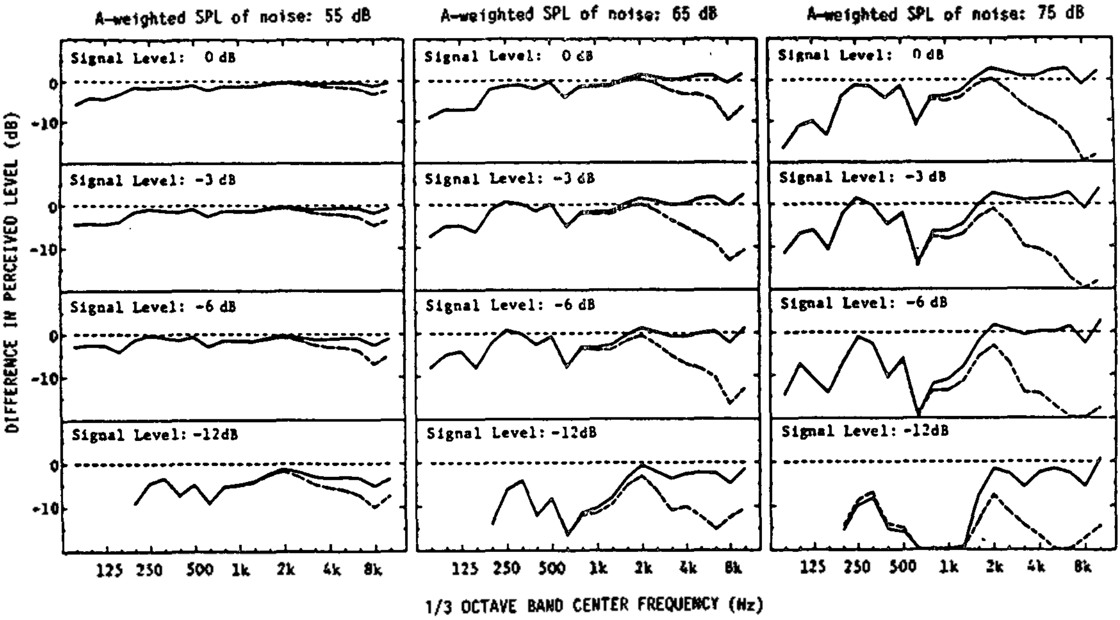

Fig. 17 PFC's of test stimuli are shown in a relative form to those of comparison stimulus. Dashed curves show the PFC's for no boosting state while solid curves show those for the optimum boost.

region was more effective in improving the sound quality of music, and compensated sound was more natural than in the case of a boost in low frequency only. Normalized PFC's were calculated again to understand the relation between the subjects' judgment and the PFC of music. Fig. 17 shows that the optimum compensation makes the normalized PFC flat. This fact implies that the sound quality of music is best compensated when the PFC of compensated music is similar to that of the reference stimulus even in the high frequency region as well as in the low frequency region. Thus, it can be said that the compensation of sound quality of music against noise is realized on the basis of a unified rule over the entire range of frequencies.

From these considerations, it is clear that the sound quality of music is well compensated for when the masked loudness of music for all narrow bands is kept unchanged irrespective of noise level. If this procedure is adaptively applied to a sound reproduction system, in other words, if the masked loudness in each narrow band is compensated for by a voltage controlled amplifier bank, we may obtain a nearly ideal compensation for sound quality. On the basis of this study, we derived the frequency response characteristics which best compensate for the deterioration in sound quality of CCIR's standard signal as well as in loudness by means of a computer simulation.

V. SUMMARY

Various aspects of loudness have been mentioned in this article. Loudness is one of the fundamental attributes of sound, which is related not only to the magnitude of hearing sensation but also to all aspects of perception of sound including timbre or sound quality

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