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Information of Loudness in Aural Communication

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1. Introduction

Loudness is one of the subjective attributes of sound and is related to every aspect of sound perception. It depends not only on the intensity of sound, but also on its frequency.

If a sound is heard in the presence of noise, it is influenced by the masking of noise so that the loudness and the quality of the sound deteriorate. Therefore, the sound reproduction system used in a noisy environment should be suitably designed to compensate for such deterioration. The sensorineural hearing impaired often show characteristics of hearing similar to those of normal people who listen to signal sound in a noisy environment. Hence, the same idea can be applied to hearing aids. Even in the absence of noise, the quality of a complex sound is determined by mutual masking among its components.

The authors have been studying this subject from various points of view for some time. This paper describes some of our findings and some of the new approaches we have developed. We first discuss the optimum level of music which is listened to in a noisy environment and show the optimum frequency response characteristics used in such circumstances. Next, we touch upon a 'loudness control circuit' designed for a sound reproduction system in which the shape of frequency characteristics in terms of partially masked loudness is kept unchanged. Finally, we describe the design of advanced hearing aids for the sensorineural hearing impaired.

2. Partially Masked Loudness

If a masker sound is not intense enough to mask a signal sound thoroughly, the signal sound is still heard but is perceived as being softer than it would be in the absence of the masker. This phenomenon is called partial masking, and the reduced loudness of the signal is called masked loudness or partially masked loudness. As to a pure tone masked by a wide-band noise, Lochner and Burger gave the following equation that describes the masked loudness quantitatively [Lochner and Burger, 1961]:

$$S' = k[I^{\alpha} - I_0^{\alpha}] = S - S_0 \tag{1}$$

where S' is the loudness of a tone in sones, and k is a constant. I represents the intensity of the tone, and I_0 shows the intensity of a tone at its threshold in the presence of noise. The value of α is 0.27 for a 1000-Hz pure tone [Lochner and Burger, 1961]. Though this equation is simple, it yields a relatively good estimation of the masked loudness. In this equation, the masked loudness of a tone is obtained by subtracting the threshold in terms of loudness, S_0, from the loudness of the tone in the absence of a masker, S. It well expresses the psychological additivity of loudness.

The results of our study on the masked loudness of a complex tone with two or three components showed that the masked loudness could be explained by eq. (1) when a masker had a wide-band spectrum [Suzuki and Sone, 1981]. If a masker was a narrow band noise, however, the masked loudness of a complex tone was perceived as being a little louder than that masked by a wide-band masker as far as the calculated loudness obtained with eq. (1) is concerned.

3. Optimum Level of Music Listened to in the Presence of Noise

A study on the optimum level of music listened to in the presence of noise is a practical case in which the musical sound is partially masked by wide-band noise [Suzuki *et al.*, 1982].

3.1 *Experimental procedure*

A typical situation of listening to music in the presence of noise is met when we drive a car. We carried out an experiment, therefore, to obtain the optimum level of music listened to in the presence of simulated car noise [Suzuki *et al.*, 1982]. Figure 1 shows the 1/3 octave band sound pressure levels of three kinds of simulated car noise. A pseudo-diffuse sound field with simulated car noise was synthesized in an anechoic room through six

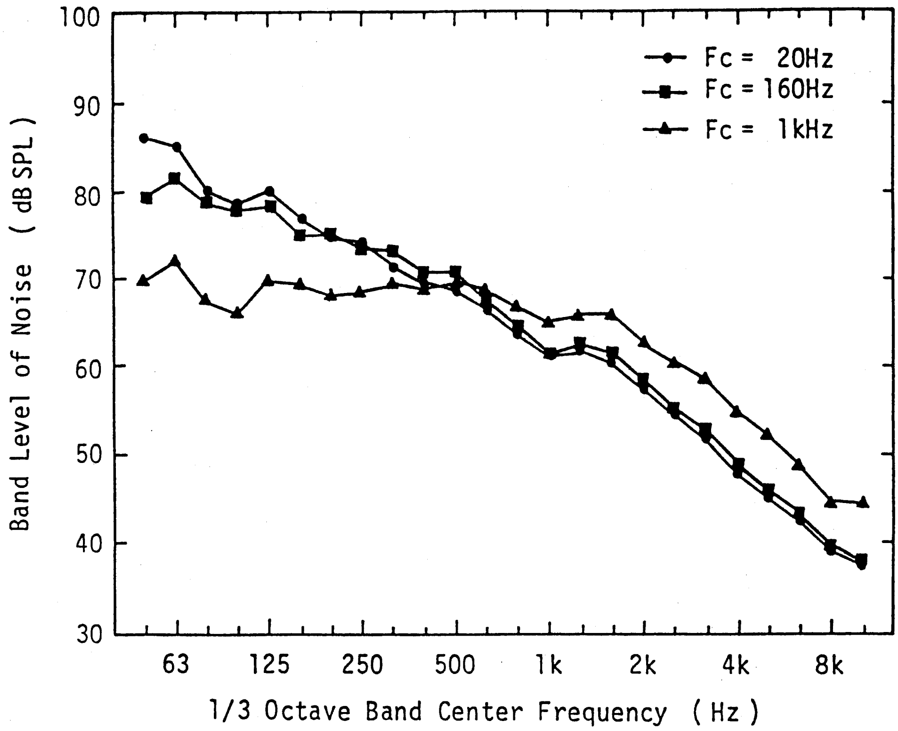

Fig. 1. 1/3 octave band sound pressure level of noise used in the experiments. The curves shown here correspond to 75 dBA of noise.

channels of independent noise sources. The five pieces of music shown in Table 1 were used in the experiment. Music was monophonically presented. The method of adjustment was used for the experiment, and the subjects were requested to adjust the level of music to an optimum level.

3.2 *Experimental results*

Figure 2 shows an example of the relation between the average optimum level of the five pieces of music and the A-weighted sound pressure level of noise. The level of music is represented by the median value, and the data shown are the values for eight subjects and their average. The dynamic range, in terms of standard deviation, for each 1/3 octave band of the music used was about 6 dB in average. Although the standard deviation of the overall sound pressure level of the music was not recorded, it is assumed to be a little less than 6 dB.

As shown in Fig. 2, the optimum listening level increases along with the increase in the level of noise, and the slope of the curve against the level of the noise is gentler for a lower level of noise than for a higher one. The effect of the sound level of noise on the optimum music level is significant beyond 0.01 level. If the difference among subjects is considered, the subjects who preferred a higher intensity of music showed slightly gentler slopes against level of noise than those who preferred a lower intensity.

3.3 *Relation between experimental results and masked loudness*

3.3.1 *Calculation of masked loudness of music*

To calculate the masked loudness of music, referring to eq. (1), the loudness of music at the threshold level in the presence of noise was subtracted from the loudness of music under noiseless conditions for each 1/3 octave band. Then the loudness of all bands was summed up using the method of Mark VII by Stevens [Stevens, 1972].

Since the subjects judged some levels of music as being most preferable, there must be some basis for their judgment. It is natural to say that a subject primarily intends to keep the loudness of music constant according

Table 1. Five pieces of music used as the source sound in the experiment.

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

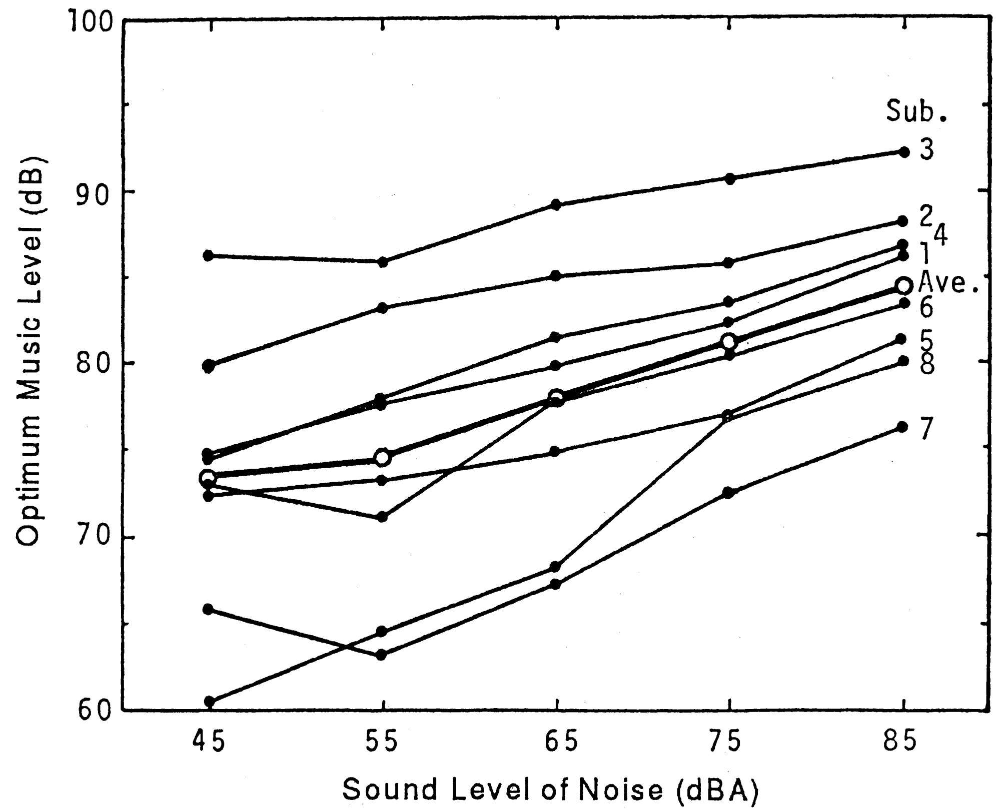

Fig. 2. Experimental results for the optimum listening level of music under a certain system. The level of music is shown in its median value.

to his preference. Subjects increase the level of music slightly against a moderate increase in the level of noise to keep the masked loudness of music at the same magnitude. Figure 3 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 seems not to be heavily dependent on the noise level. In fact, the result of a statistical test of the Analysis of Variance showed that the effect of noise level was not significant while the effect of subject was significant beyond 0.01 level. Other data show similar results [Suzuki *et al.*, 1982]. It can be said, therefore, that subjects adjust the level of music so as to keep the masked loudness of music at the same value irrespective of the level of noise.

3.3.2 *Standard level contours for music reproduction in the presence of noise*

When we consider the standard curve denoting the changing rate of the level of music in relation to the level of noise, it is convenient to use an appropriate standard signal. We chose the averaged spectrum of broad-casting signals stated in a CCIR Draft Record [CMTT, 1981]. It is assumed here that the intensity of this signal varies while its spectral shape remains unchanged and that the level fluctuates normally with a standard deviation of 6 dB, which was the average standard deviation of music level fluctuation for the five pieces of music used in the listening experiment.

Figure 4 shows the equal loudness contours of the signal against the level of noise; it corresponds to the experimental results already presented. It can be said, therefore, that this figure shows the standard curve of optimum music level against noise level in a car. When the contour of 75 PLdB (Perceived Level dB obtained using the method of Mark VII by Stevens, 1972) is considered, the change in the optimum level of music for a 10 dBA increase in the level of noise is about 3 dB for levels of noise between 55 dBA and about 4 dB near 85 dBA of noise.

4. Masked Loudness and Sound Quality

It is clear that timbre or quality of a steady sound is related to the power spectrum of the sound. In particular, as for sounds in which any fluctuation in time is not perceivable, the timbre of such sound is closely related to the power spectrum of the sound. A sound with perfect harmonic structure would be a good example of this type of sound.

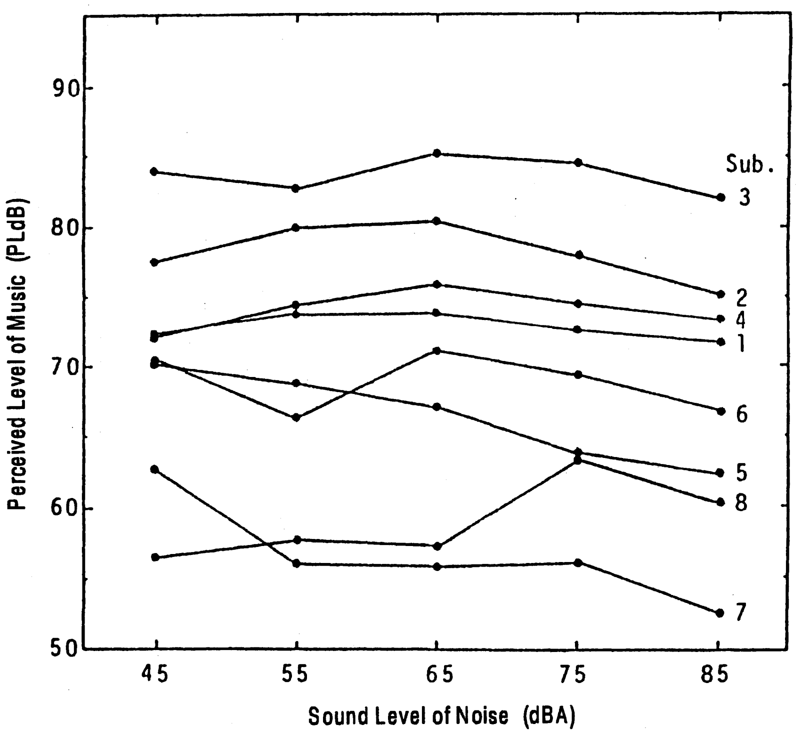

Fig. 3. The perceived level of music corresponding to the average of loudness among the samples above threshold. These results were obtained with the same system as the data shown in Fig. 2.

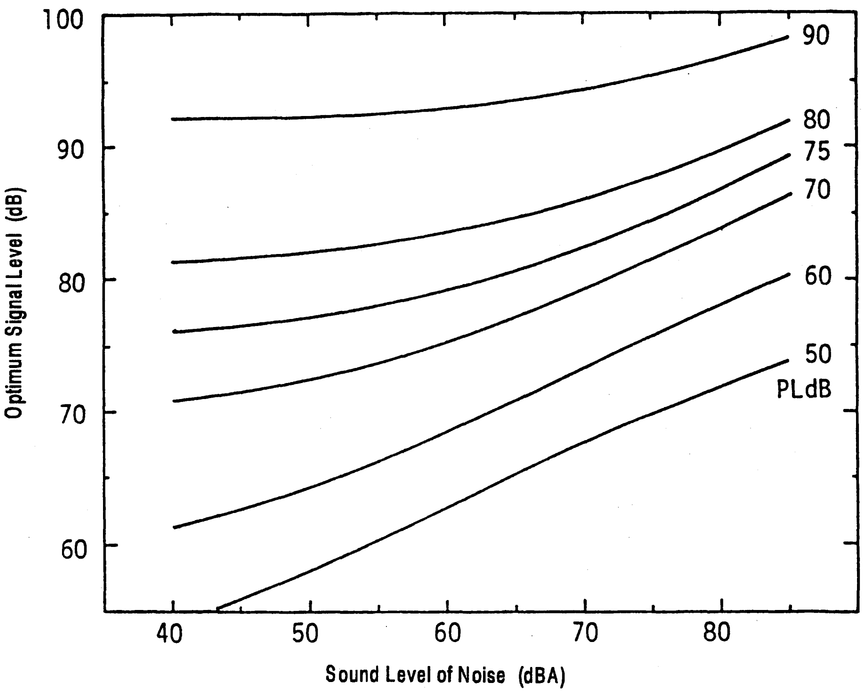

Fig. 4. The contours for equal perceived level as a function of A-weighted sound pressure level of noise. The ordinate denotes the level of signal sound shown in its median value.

However, it is irrational that the power spectrum of such sound is directly related to the timbre of the sound. In our auditory system, a sound lead into the ear canal is deformed by nonlinear processing such as (approximately) logarithmic perception of frequency, some band-filtering process such as critical bands, exponential perception in loudness, masking, lateral inhibition, subjective tones, etc. If a component of the sound is masked, for example, the component has no effect on timbre while it has some role on the power spectrum of the sound.

We assume, therefore, that there is a frequency spectrum which is directly related to the discrimination of the timbre of a sound. We tentatively call this assumed spectrum 'psychological spectrum.' This psychological spectrum is similar to the 'internal spectrum' proposed by Buunen *et al.* [1974]. On the other hand, the masking pattern as a function of frequency is sometimes considered when discussing sound perception such as speech recognition [Hirahara, 1991]. This masking pattern directly reflects tonotopical neural activity in the auditory nerves and is an intermediate stage from the physical spectrum of a sound to the sychological spectrum. We do not think that the psychological spectrum should be observable and consider that the spectrum is a virtual spectrum directly connected to the timbre discrimination process.

While many of the auditory processes mentioned above contribute to the generation of the psychological spectrum of a sound, our assumption is that the masking effect is one of the most important factors for describing the projection from the physical spectrum of the sound to the psychological spectrum. We term the frequency spectrum as 'masked frequency spectrum' in consideration of the part played by the masking effect [Sone *et al.*, 1989].

4.1 *Effect of noise on the sound quality of music*

As an example which illustrates the contribution of partial masking to the 'masked frequency spectrum,' a study on the optimum frequency response characteristics of music reproduction in the presence of noise will be presented here [Suzuki *et al.*, 1985].

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

A procedure similar to that described in the previous section is 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 obtained 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 is obtained.

Figure 5 shows the PFC's of music at the optimum listening levels obtained in the experiment mentioned above. These PFC's are the averages for five different pieces of music. It is clear from this figure that the PFC changes along with the change in the level of noise even if the music is so reproduced that the masked loudness is kept constant.

4.2 *Optimum amount of low and high frequency boost*

An experiment was carried out to obtain the optimum amount of boost in the low and high frequency regions to compensate for the deterioration of the sound quality of music caused by noise.

First, music was modified by an equalizing circuit, the frequency transfer function of which is shown in Fig. 6. In this figure G represents the amount of boost. The amount of boost was controlled by the subject and the level of music was controlled by a microcomputer. As in the previous experiment, the low-passed pink noise

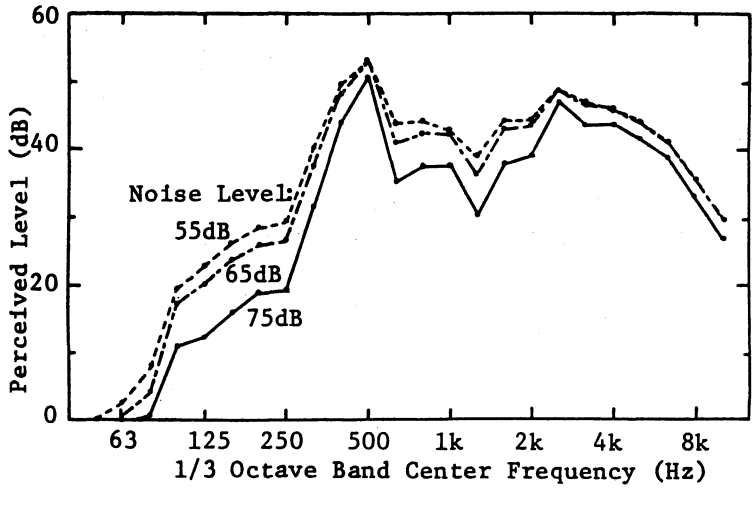

Fig. 5. The PFC's of music at the optimum listening level in the presence of noise similar to that in a passenger car.

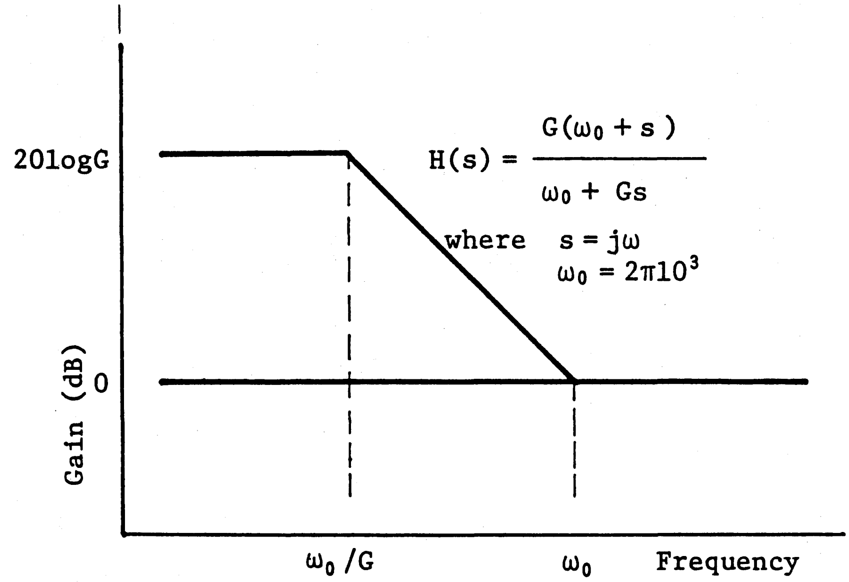

Fig. 6. The frequency transfer function of the low frequency booster. G in this figure is exponentially changeable from 1 to 10, and thus the amount of boost can be linearly changed in dB from 0 to 20.

radiated by six loud-speakers was used to compose a pseudo-diffuse noise field. A piece of music was monophonically presented to a subject in an anechoic room in the presence of simulated car noise at 45 dBA as the reference, and after a ten-second pause, the same piece of music was presented twice in the presence of simulated car noise at 55, 65, or 75 dB as test stimuli. Subjects were requested to adjust the degree of low frequency boost so that the sound quality of test stimuli would resemble that of the reference stimulus as closely as possible.

Figure 7 shows the relative PFC of music with and without optimum low-boost for three levels of noise. That is, PFC of the reference stimulus was subtracted from that of the test stimuli. Broken lines show the PFC's for a state of no boosting, while solid lines show those for optimum low boosting. The signal level is expressed by a value relative to the optimum listening level. The amount of low-boost is adjusted so that the PFC's are almost the same as that of the reference stimulus, since the solid lines approach the 0 dB line.

Next, we conducted an experiment on compensation in the high frequency region as well. Here the amount of low-boost was fixed at the average optimum level derived from the above experiment. Subjects reported that the compensation in the high frequency region was more effective in improving the sound quality of music and that the compensated sound was more natural than that in the case of a boost in the low frequency region only. Normalized PFC's were calculated again to understand the relation between the subjects judgment and the PFC of music. Figure 8 shows that the subjects adjusted the degree of high-boost so as to make the normalized PFC flat.

From these considerations, it is clear that the change in 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. On the basis of the results, by means of a computer simulation, we derived the frequency response characteristics which best compensate for the deterioration in sound quality of the standard signal [CMTT, 1981] as well as in loudness [Suzuki *et al.*, 1985].

5. Optimum Frequency Characteristics for a 'Loudness Control Circuit'

It is known that the timbre of a sound has a close relation to its waveform or frequency spectrum. The timbre of a sound, however, is determined not only by its waveform or frequency spectrum but also by its sound pressure level as stated in the previous section. Hence it is natural to introduce the general idea of a subjective or a psychological frequency spectrum of sound that can explain its timbre directly [Suzuki *et al.*, 1993].

By applying the notion of psychological spectrum of sound mentioned above to a sound reproduction system, we can design a so-called 'loudness control circuit' for it [Ozawa *et al.*, 1990].

5.1 *Role of the 'loudness control circuit'*

If a piece of music is reproduced at a reduced level, it sounds different from that reproduced at a louder level. This can be easily understood considering that the equal-loudness level contours are highly dependent on the tonal level.

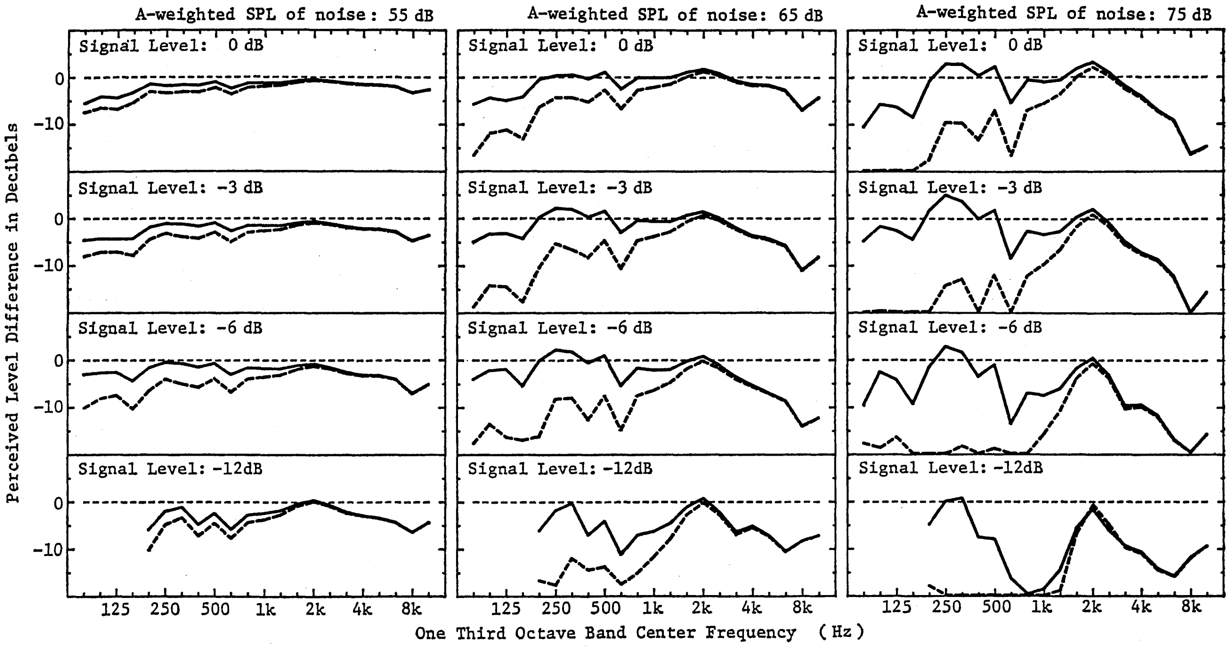

Fig. 7. The experimental PFC's of test stimuli shown in relative forms to those of the comparison stimulus. Here PFC's of the comparison stimulus are subtracted from those of test stimuli. Broken lines show the PFC's for no boosting state while solid lines show those for the optimum low boost. The signal level is expressed by a relative value to the optimum listening level obtained in the preliminary experiment.

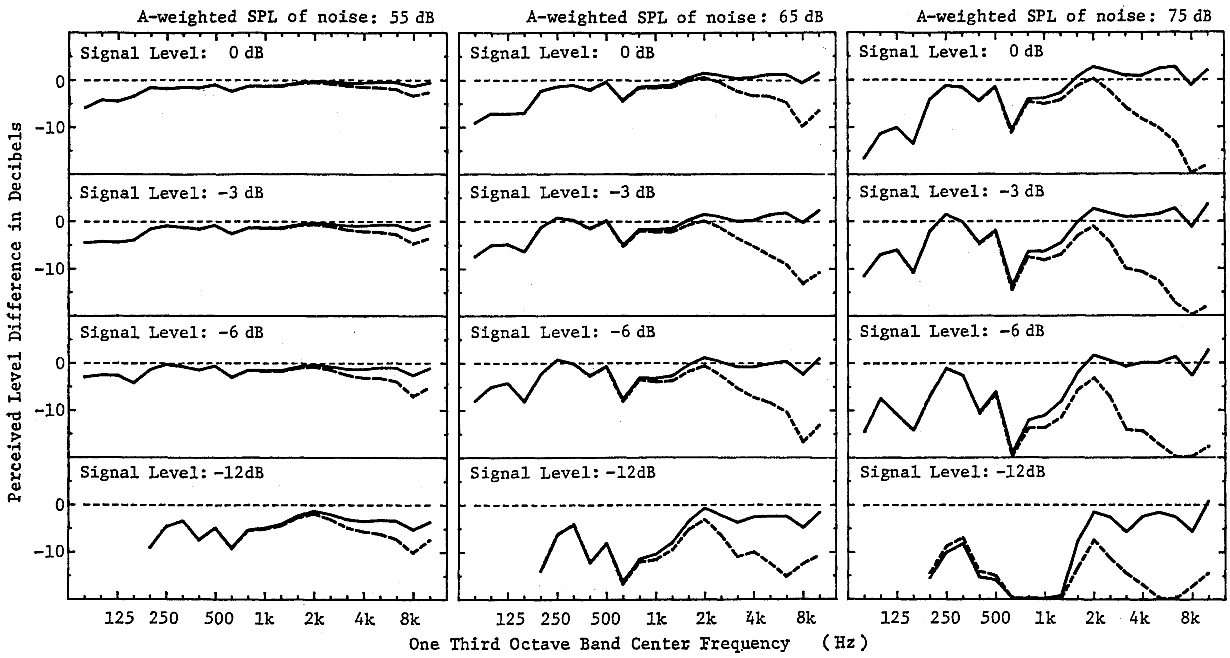

Fig. 8. The experimental PFC's of test stimuli shown in relative forms to those of the comparison stimulus. These results are for high boosting.

When the sound quality of music is kept unchanged while the reproduced level of the music is changed, what frequency characteristic should be applied? The simplest way is to use the mirror frequency characteristics of the difference of equal-loudness level contours [ISO 226, 1987]. These characteristics, however, are not always appropriate since mutual masking takes place in our auditory system among frequency components of the musical sound.

5.2 *'Masked frequency spectrum' of a sound*

The quality of a sound is closely related to its frequency spectrum. It is not directly related, however, to its physical spectrum, but rather to its 'internal' frequency spectrum representation in our auditory system [Suzuki *et al.*, 1993]. Though the internal representation is produced by various properties including non-linearity in the auditory system, we assume that the conversion from sound pressure to loudness and the masking between sound components are the dominant (decisive) factors in this process. We thus estimate the 'masked frequency spectrum' of a sound, which is a kind of frequency spectrum consisting of masked loudness of each frequency component of the sound (Section 4). If the masked frequency spectrum is considered in terms of a 1/3 octave band or Bark scale, we can use the procedure of ISO 532B [ISO 532, 1975] in estimating the masked frequency spectrum, the purpose of which is to calculate the loudness of a broad band noise. Figure 9 shows the procedure to obtain the masked loudness of a single 1/3 octave component. The masked loudness estimated for every 1/3 octave band in this manner forms the 'masked frequency spectrum' of the sound [Sone *et al.*, 1989].

5.3 *Frequency characteristics for maintaining the relative shape of the 'masked frequency spectrum'*

It is expected that the sound quality remains unchanged if the shape of the masked frequency spectrum of a sound is unchanged while the overall level of reproduced sound is altered. The solid lines in Fig. 10 show the fre-

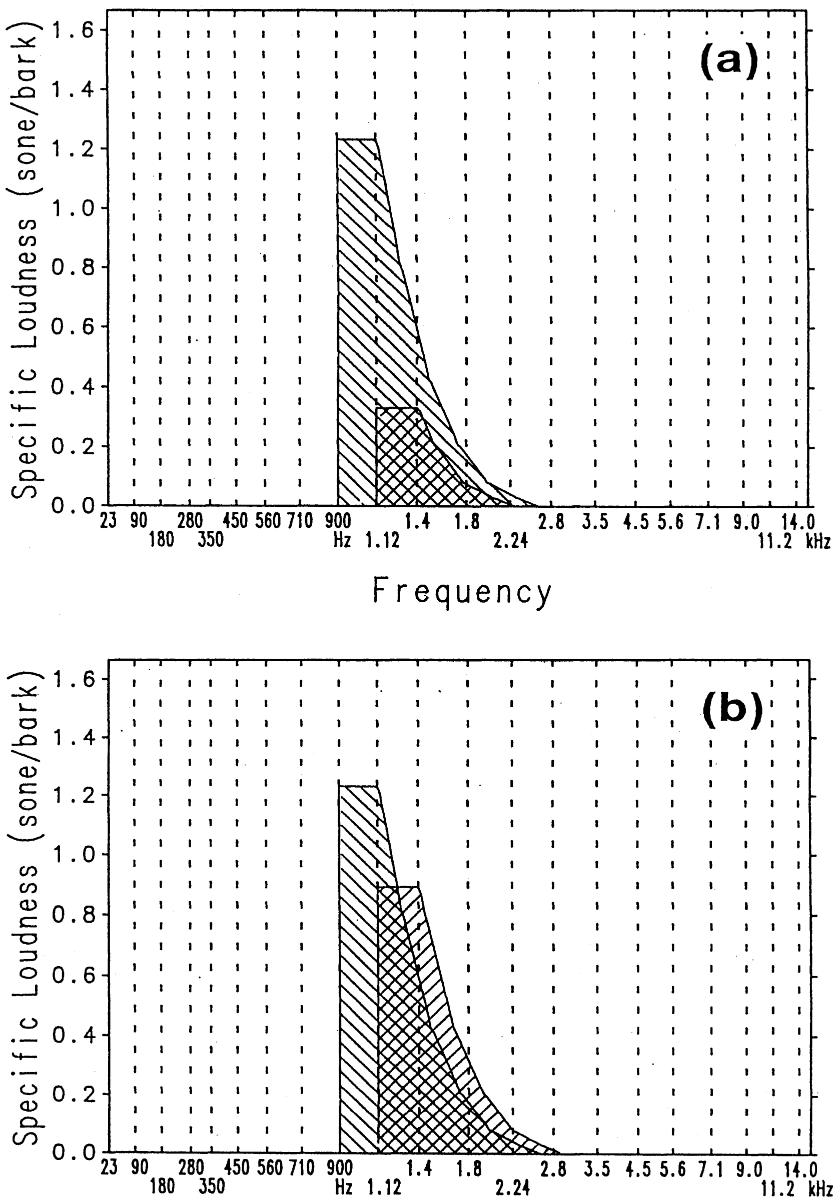

Fig. 9. Method of obtaining the masked loudness of a one-third-octave band component using Zwicker's loudness scheme [Zwicker and Scharf 1965i Paulus and Zwicker 1972]. (a) A sound with a band from 1.12 to 1.4 kHz is completely masked by a 1 kHz pure tone. (b) A sound with a band from 1.12 to 1.4 kHz is partially masked.

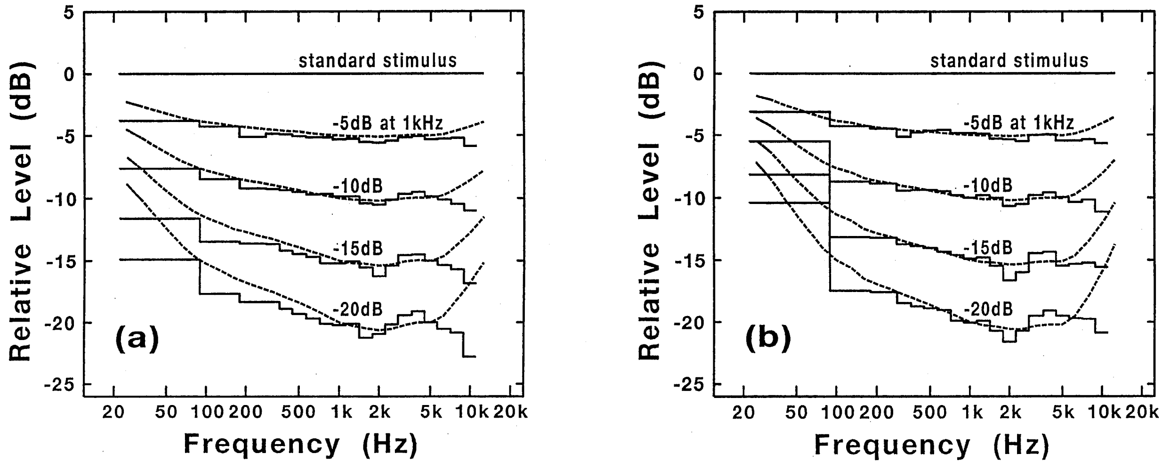

Fig. 10. Frequency-response characteristics of an optimum "loudness control circuit" obtained by simulation. (a): the standard stimulus I (pink noise) and (b): the standard stimulus II (noise with the average frequency spectrum of dance music). Broken lines indicate the frequency-response characteristics based on the equal loudness level contours (ISO 226).

quency characteristics for keeping the relative shape of the masked frequency spectrum of (a) pink noise and (b) noise with the averaged frequency spectrum of dance music unchanged. The sound level of the standard signal was 85 dBSPL. The dashed lines show the frequency characteristics derived from the equal-loudness level contours stated in ISO 226. Comparing the two curves, we can see that the frequency characteristics based on ISO 226 may require too much high-boosting.

5.4 *Experiment on the frequency characteristics to maintain sound quality for various levels of sound reproduction*

We examined whether or not the frequency characteristics for maintaining the relative shape of the masked frequency spectrum shows a good agreement with the results of the listening experiment for keeping the sound quality unchanged.

For each of two standard noises, several levels of test stimuli were presented: (1) Stimuli with an overall level 5 to 20 dB lower than that of the standard; (2) Stimuli with systematically enhanced/enfeebled lower and higher frequency components as compared with the standard as well as the overall level of 5 to 20 dB lower. The similarities of the sound quality among stimuli were tested by the method of triad combinations. The results were analyzed with Torgerson's Multidimensional Scaling Method [Torgerson, 1958]. Frequency characteristics that resulted in the highest similarity between the sound quality of the test stimuli and the standard stimuli were obtained from the result of analysis as shown in Fig. 11.

The dashed lines in Fig. 10 show the best frequency characteristics derived from the results of the experiment, and the solid lines represent the characteristics based on masked loudness. The two curves show very good agreement. This means that keeping the relative shape of the masked loudness for the 1/3 octave band constant against the change in the overall level of sound is effective in keeping the sound quality unchanged.

6. A Digital Hearing Aid for The Sensorineural Hearing Impaired Based on the Principle of Recovering Partially Masked Loudness

Sensorineural hearing impairment is characterized by a narrow dynamic range between the threshold of hearing and the uncomfortable level (UCL) or discomfort level (DL). The loudness of a sound, therefore, increases rapidly above the hearing thresholds. This phenomenon is called the 'recruitment' of loudness and is observed not only for sensorineural hearing impaired listeners but also for normal listeners in a noisy environment. This is regarded as the state in which the equal-loudness level contours are distorted by impairment or noise [Suzuki *et al.*, 1993]. This distortion causes the perceived frequency spectrum of a sound to be quite different from that perceived by those with normal hearing and results in a deterioration of speech intelligibility [Hood, 1968].

6.1 *Outline of the system*

6.1.1 *Loudness compensation function*

Restoration of the recruitment itself has been attempted by some researchers [Yund *et al.*, 1987; Kollmeier, 1991; Farassopoulos *et al.*, 1989]. If this idea is adopted for each narrow frequency band throughout the audi-

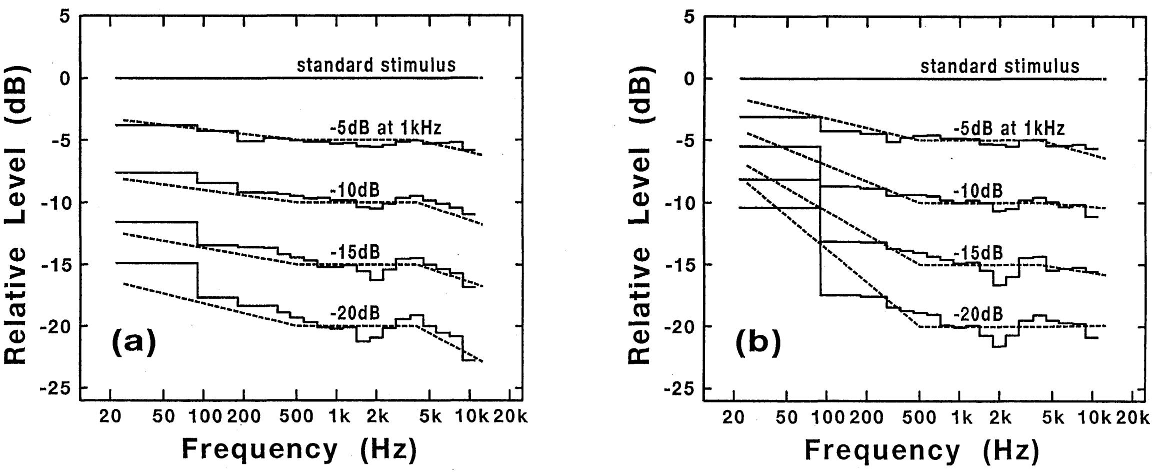

Fig. 11. Optimum frequency-response characteristics for a "loudness control circuit" obtained by a listening test (solid lines). Broken lines are the frequency-response characteristics based on the equal loudness level contours (ISO 226).

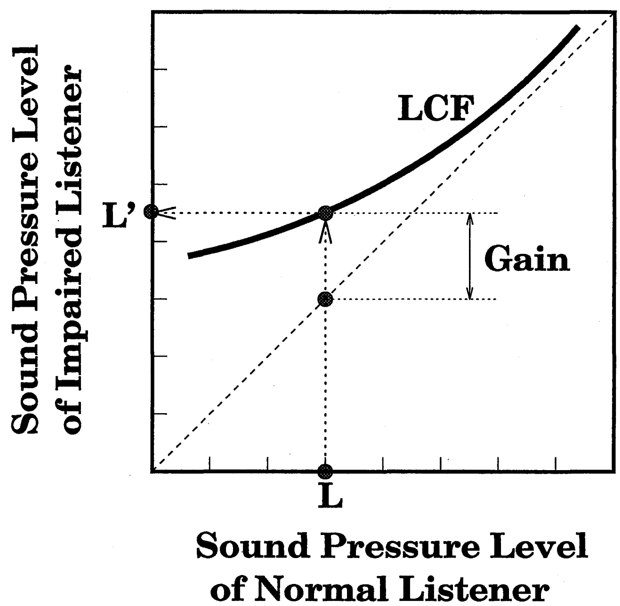

Fig. 12. An example of the loudness compensation function (LCF). Thick solid line shows the LCF, which describes the relationship between the loudness for an impaired listener and that for an average normal listener.

ble frequency range, the distortion of equal-loudness level contours can be eliminated.

To achieve this, the authors are developing a digital hearing aid called CLAIDHA—Compensating for Loudness by Analyzing the Input, Digital Hearing Aid—which is capable of restoring the distorted equal-loudness level contours [Asano *et al.*, 1991a, 1991b, 1991c; Suzuki and Sone, 1993].

To eliminate the distortion, the relationship between loudness perceived by a hearing impaired listener and that perceived by a normal listener must be known for each frequency unit. In CLAIDHA, we term this relationship a Loudness Compensation Function (LCF), and LCF's are determined for each narrow frequency band during fitting; actually for each one-octave band. Figure 12 shows an example of LCF. The thick solid line in the figure represents the LCF. As seen from the figure, the LCF for sensorineural impaired listeners shows a nonlinear character in general.

Using this function, the gains required are calculated as the following: L in Fig. 12 indicates the level of the input signal at a certain narrow band at a certain instant. For the impaired listener, the level of L′ is needed in order for the subject to perceive the same loudness as perceived by a normal listener. Thus, the difference between the LCF curve and the diagonal line gives the gain required for restoring the loudness of the impaired listener at the instant.

6.1.2 *Smoothing the frequency characteristics*

If LCF's for all of the narrow frequency bands are given for an impaired ear, the gain-frequency characteristics for restoring the equal-loudness level contours for the relevant ear can be determined according to the input signal to the hearing aid. However, how to achieve the required gain-frequency characteristic is a problem. In conventional multi-channel compression hearing aids, a filter bank consisting of 2 to 4 band-pass filters is widely used [Moore, 1987; Villchur, 1987; Engebrettson *et al.*, 1987]. If the hearing impairment varies greatly from frequency to frequency, however, the distortion cannot be satisfactorily recovered in such a way. Thus, we adopt a frequency sampling digital filter [Oppenheim, 1975] which is equivalent to a 32-channel filter bank to realize a complicated frequency characteristic.

It has been reported that a multi-channel compression system with many channels causes spectral flattening [Moore, 1987; Bustamante and Braida, 1987] and results in a performance worse than expected [Bustamante and Braida, 1987; Lippmann *et al.*, 1980]. Since the spectral flattening, however, is due to the independent control of each channel's gain, it can be reduced by using a gain characteristic smoothed over the frequency bands [Asano *et al.*, 1991a, 1991b, 1991c; Suzuki and Sone, 1993]. We are using spline functions for such smoothing.

6.1.3 *Blockdiagram of the system*

Figure 13 shows a blockdiagram of the system. The input signal is divided into 8-ms time blocks and the average frequency spectrum within the block is obtained with FFT analysis. The gains required for relevant octave-bands are then calculated from the LCF's and connected smoothly by a spline function to derive the gain-frequency characteristics over the necessary range at the instant. The calculated characteristics are then realized

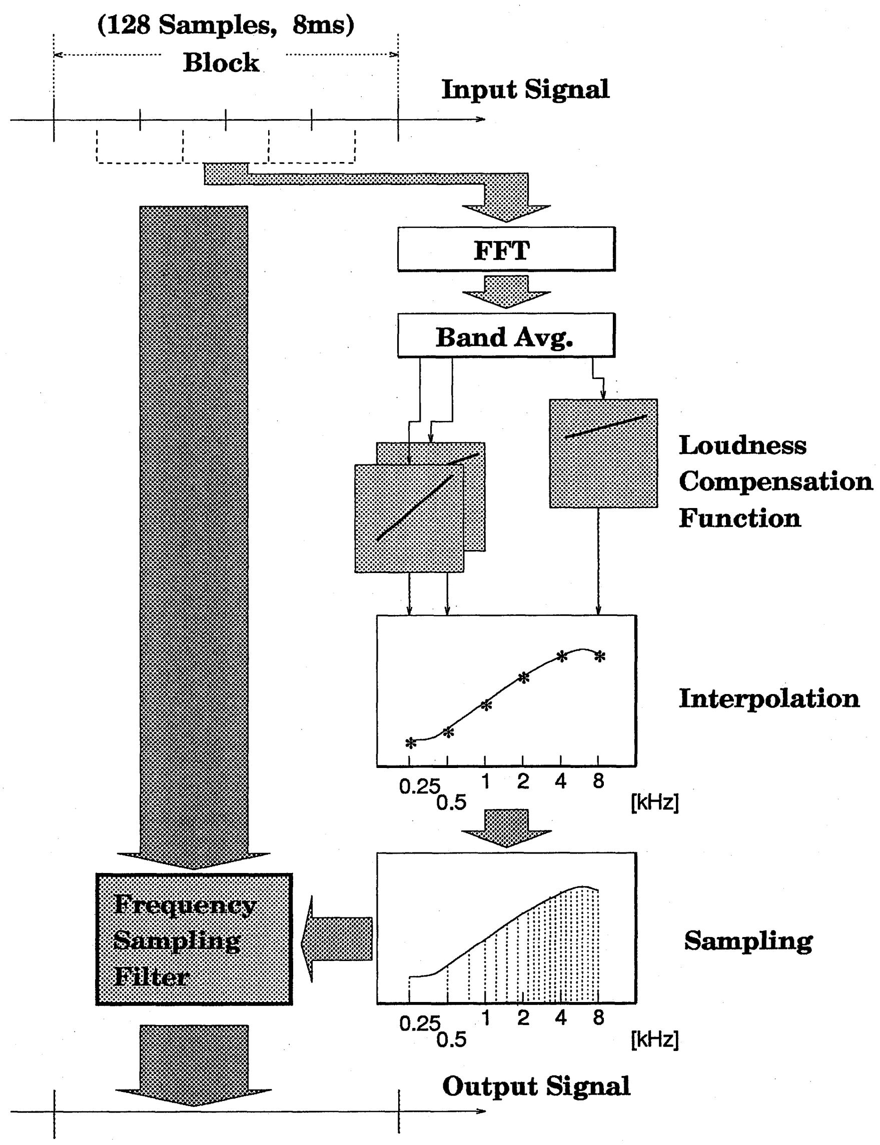

Fig. 13. Block diagram of the proposed hearing aid system (CLAIDHA).

by a single digital filter called a frequency-sampling digital filter. With the above procedure, the input spectrum is always projected onto the (narrowed) audible range of the impaired ear.

6.2 *Evaluation of the system*

Figure 14 shows an example of the signal processed. The input signal is the Japanese consonant /s/ and the vowel /a/. We can see that the frequency spectrum of the input signal is projected on the narrower audible range of the impaired listener while the fine structure of the spectrum is maintained.

A real time system using a DSP chip (Fujitsu MB 86232) is actualized on a PC. Japanese monosyllable articulation tests were conducted with the system for 22 moderate to severe hearing impaired subjects. The subjects were patients at Tohoku University Hospital and volunteers. The speech test was conducted under the following three conditions:

1. using the proposed hearing aid system
2. using linear amplification with a gain of half of the subject's hearing loss
3. without a hearing aid.

The articulation scores were improved by the proposed system for 15 of the 22 subjects. For these subjects, the scores improved over a wide range of input levels [Asano *et al.*, 1991c]. The scores for linear amplification rapidly decreased with decreasing input level. Some of the subjects reported such impressions as "clearly spoken" for speech processed with the proposed system.

6.3 *Assessment of accurate LCF's with two-step scaling method*

In the speech test described above, LCF's were approximated with linear functions in the dB-dB plane when the speech signal was processed with the proposed hearing aid system. That is, as shown in Fig. 15, at a certain frequency, a line connecting a point decided by the thresholds of hearing for normal hearing people and a subject with hearing impairment to another point decided by their uncomfortable levels was used to specify the LCF for the subject at the frequency.

If LCF's can be so decided that the lack of loudness is correctly compensated for, better restoration of hearing acuity can be expected. To achieve this, the accurate shape of the loudness function of a subject should be obtained. Therefore, we used a two-step scaling method for the derivation of loudness functions [Hellbrück *et al.*, 1986]. In this method, loudness of a sound is evaluated in two steps: coarse scaling followed by fine scaling. Figure 16 shows an example. In the first step, a subject is requested to evaluate loudness with several categories from very soft to too loud, as shown in Fig. 16(a). This step is equivalent to an ordinary scaling method such as LGOB [Allen *et al.*, 1990]. Then, to evaluate the loudness more precisely, an interval between two adjacent categories in the first step is magnified and divided into ten grades as shown in Fig. 16(b). The subject judges the loudness again in terms of these grades. This two-step scaling yields smaller variance of judgment than conventional single-step scaling.

Figure 17 shows an example of a loudness function derived by the two-step scaling method for a hearing impaired subject and a normal subject. Comparing these two functions, LCF can be calculated as shown in Fig. 18.

We are now developing a fitting system to assess loudness functions with the method mentioned above. This system enables us to install more appropriate LCF's than the linear function, resulting in, we expect, better per-

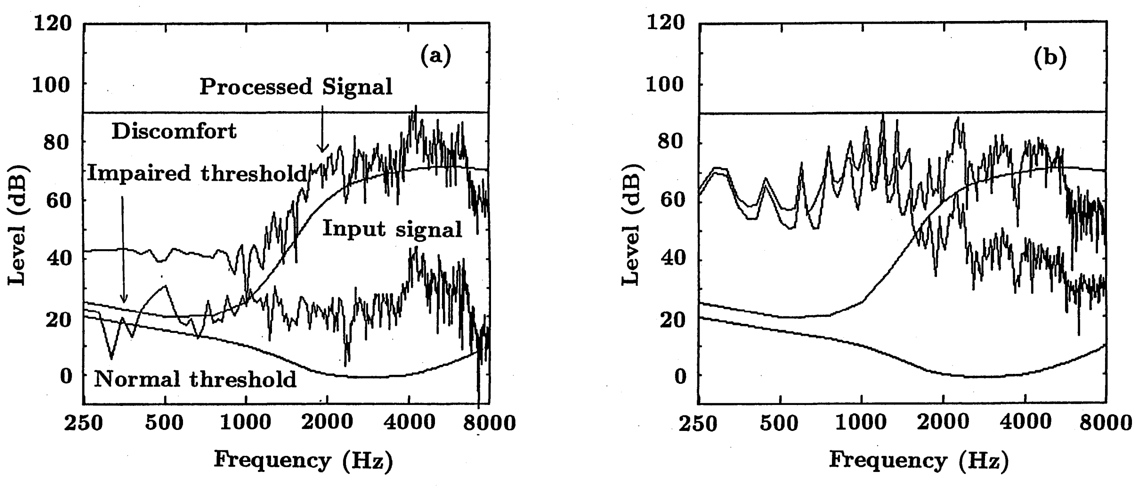

Fig. 14. An example of a processed signal using the proposed hearing aid system. The input signals are (a) the portion of consonant and (b) the portion of vowel of the Japanese monosyllable /sa/.

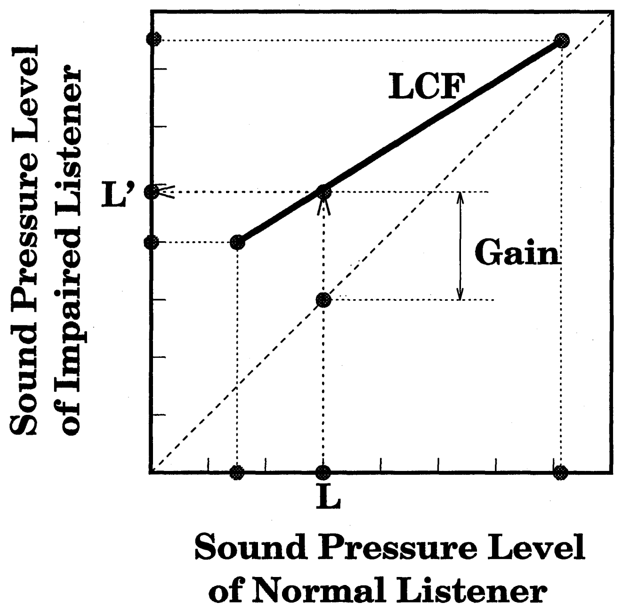

Fig. 15. An example of LCF at a certain frequency used in the listening test. The line of LCF is drawn to connect the point corresponding to the threshold of hearing straight to the point corresponding to the uncomfortable level.

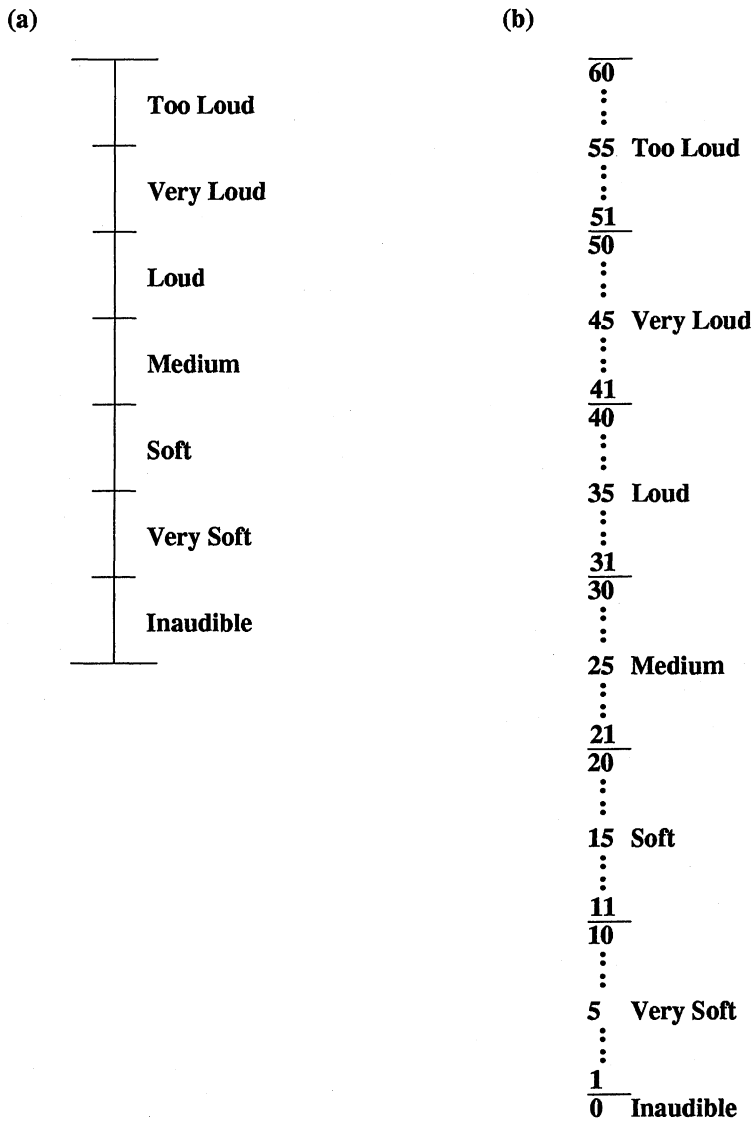

Fig. 16. Two-step rating scale for loudness judgment. The rating scale at (a) the first step and (b) the second step.

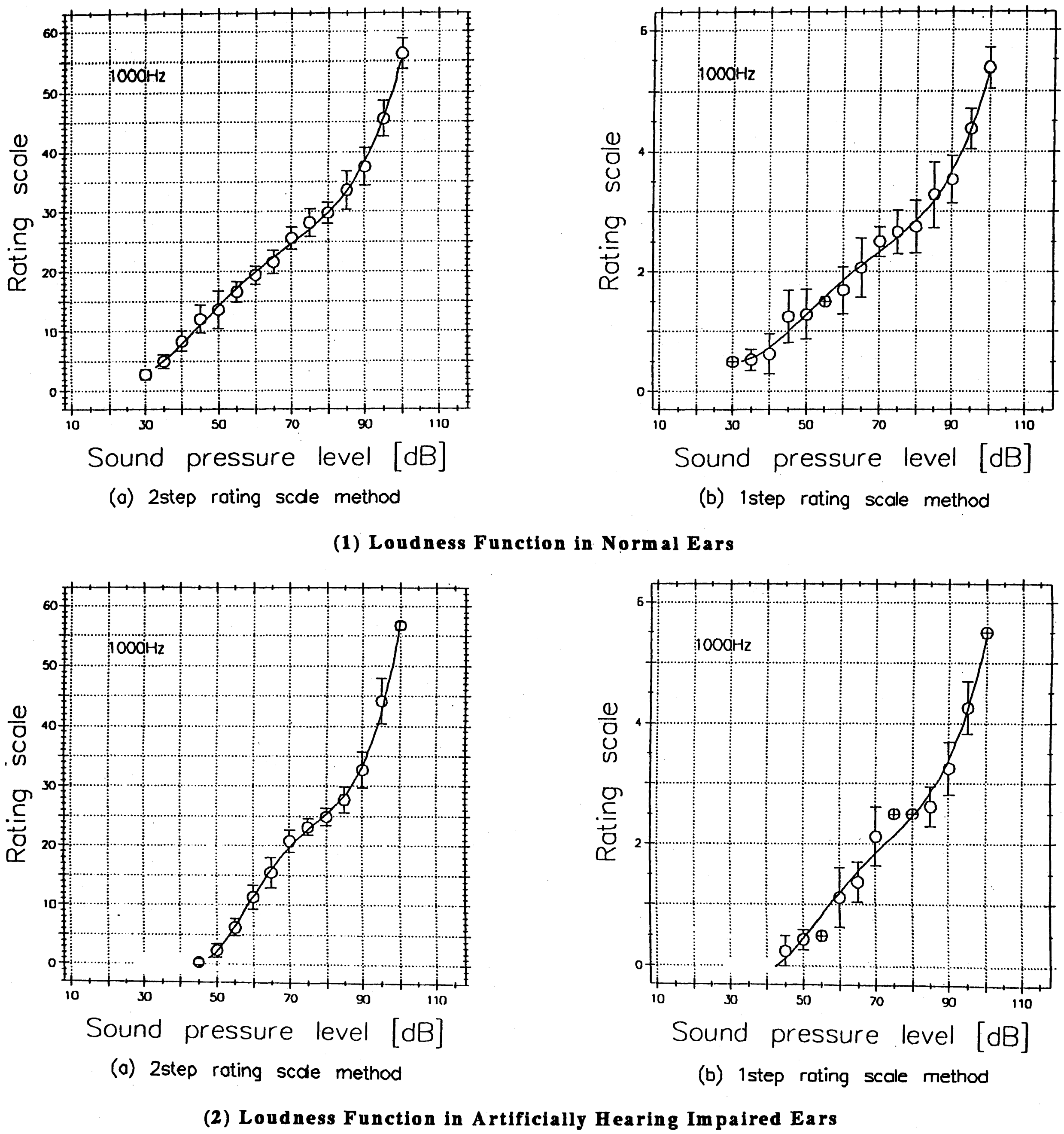

Fig. 17. Loudness functions obtained for a subject with a two-step rating scale method at 1000 Hz.

formance of the proposed system—CLAIDHA. Moreover, a new portable CLAIDHA system is being developed. An experiment on portable hearing aids adapted with the new fitting system is now being planned to evaluate the performance of the CLAIDHA algorithm in a daily life environment.

7. Conclusion

Since loudness is one of the subjective attributes of sound and is related to every aspect of sound perception, a good knowledge of loudness perception of our auditory system is useful and effective in designing acoustic systems.

In this paper, three applications of this concept proposed by the authors were discussed. The first application is a sound reproduction system used in the presence of noise. The second is a 'loudness control circuit' for maintaining sound quality even in very soft reproduction. The third is a digital hearing aid system named CLAIDHA, which is capable of restoring the distorted equal-loudness level contours of the sensorineural hearing impaired. An evaluation experiment showed that 15 of 22 subjects attained better results with the digital

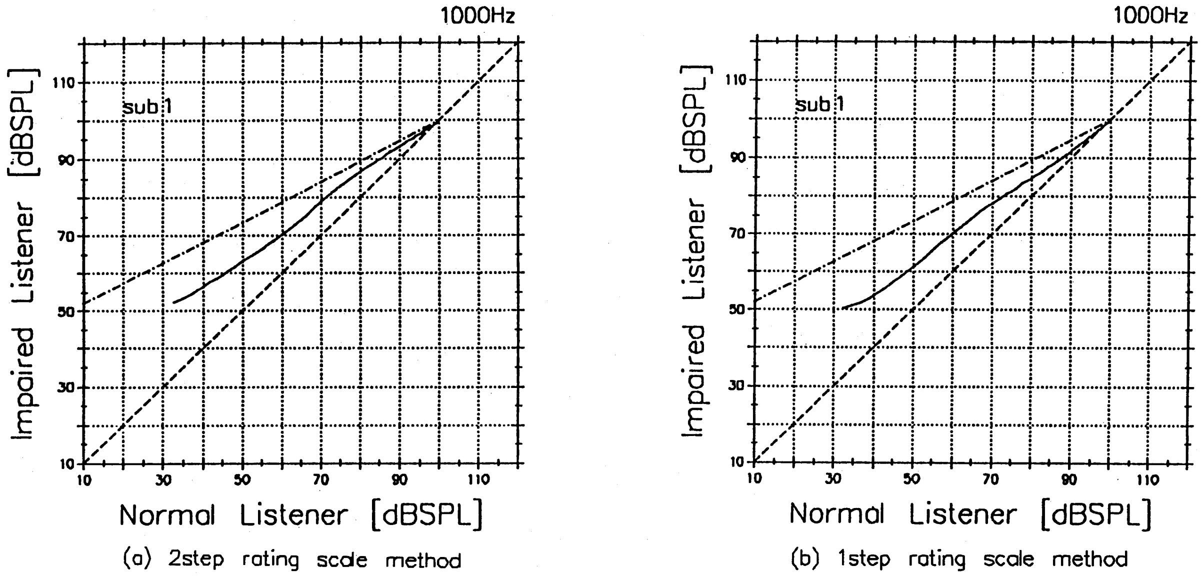

Fig. 18. LCF's obtained with (a) a two-step rating scale method and (b) a single-step rating scale method.

hearing aid than with conventional linear amplification.

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