Chapter 7

Modeling loudness perception in the hearing impaired. II.

Abstract

Two approaches to extending a loudness model, originally proposed by Zwicker, are compared which attempt to account for sensorineural hearing impairment. The model should account for several perceptual alterations due to cochlear hearing impairment: raised absolute threshold, loudness recruitment, reduced loudness summation and reduced frequency selectivity. Reduced frequency selectivity is modeled in both approaches by the dependence of "normal" auditory filters on level. Thus, it is assumed that auditory filter bandwidth of the hearing impaired is the same as that of normal–hearing subjects at the same sound pressure level (dB SPL) but differs when compared at the same sensation level (dB SL). In the first approach, hearing impairment is modeled by an inaudible, internal noise. It is assumed that hearing impairment resembles a masking condition in normal–hearing subjects. Raised absolute threshold and recruitment are modeled by one single component ("one–component approach"). In the second approach, both components, raised absolute threshold and recruitment, are taken into account separately ("two–component approach"). Raised absolute threshold is achieved by a frequency dependent attenuation of the excitation pattern, while loudness recruitment is modeled by increasing the exponent in the power law, which relates specific loudness to excitation patterns. Both approaches are applied to model the data measured in chapter 5 with narrowband and broadband stimuli. The one–component approach describes the data on average quite well but fails to predict individual data correctly. The two–component correctly describes the data, if the exponent of the power law is adjusted to the individual data obtained with narrowband stimuli. The model also predicts the loudness functions obtained with broadband stimuli correctly. Hence, the physiologically motivated two–component approach appears to model different aspects of cochlear hearing loss in a more realistic way than the one-component approach.

7.1 Introduction

People suffering from a cochlear hearing loss often show different performance than normal in several different auditory functions. Firstly, they show reduced sensitivity in detecting pure tones in quiet, which corresponds to a raised absolute thresholds. A second change, which is quite often used for diagnostic purposes, concerns the perception of loudness: the rate of growth of loudness with increasing stimulus level is usually steeper than in normal–hearing subjects; the same range of subjective loudness as normal is achieved over a smaller range of stimulus levels than normal. This effect, known as recruitment (Fowler, 1936), is usually not seen in subjects suffering from a purely conductive hearing loss. A third change is that hearing–impaired subjects usually show reduced frequency selectivity, i.e., broadened auditory filters (Glasberg and Moore, 1986; Tyler, 1986). Lastly, the perception of the loudness of broadband stimuli is altered in the hearing impaired. In normal–hearing listeners, the loudness of a sound of fixed overall intensity increases with increasing bandwidth beyond a certain bandwidth, called the critical bandwidth for loudness. This effect, called "loudness summation", is reduced or even completely absent in the hearing–impaired, as discussed in chapter 5. These perceptual changes seldom occur separately and might thus reflect damage of a common underlying mechanism, i.e., changes in cochlear processing.

Current models of cochlear processing (cf. chapter 2 and 6) assume that the highly nonlinear compressive inner ear mechanics is due to some "active" process within the cochlea. The most likely site of this active mechanisms is the outer hair cells (OHC), while it is assumed that inner hair cells (IHC) are responsible for encoding of auditory information but do not contribute to the active mechanism. Therefore, damage to IHCs and OHCs might cause very different alterations in cochlear processing. While damage to OHCs probably causes both loss of sensitivity and loss of the nonlinear processing mechanisms, damage to IHCs solely might cause a loss of sensitivity. Therefore, two components contributing to cochlear hearing loss can probably be distinguished. Firstly, a "sensitivity loss" which may be thought of a linear attenuation and secondly, a "compression loss" which is due to alterations in the signal processing characteristics of injured cochleae. The raised audiometric thresholds of hearing–impaired subjects, are probably a consequence of the former component, while loudness recruitment, reduced loudness summation and reduced frequency selectivity probably reflect some of the perceptual consequences of the latter component. In order to model loudness perception in the hearing impaired, these alterations in auditory functions as well as the probable components in impaired cochlear processing have to be taken into account.

In the previous chapter, two very different approaches were discussed, which qualitatively explain or quantitatively describe loudness perception in normal–hearing and hearing–impaired listeners. The latter approach was the so–called purely "psychological", which does not account for physiological findings but relates perceived loudness to stimulus level using a power law. The former was the so–called purely "physiological" which does not relate stimulus level to subjective loudness but explains the underlying physiological mechanisms of normal and impaired inner ear mechanics. Neither approach correctly accounts for the physiological findings *and* calculates perceived loudness on a subjective scale. Furthermore neither approach can explain spectral effects in loudness perception, for instance, loudness summation.

Zwicker and coworkers proposed an extended version of the simple power law model which was adapted to model loudness perception of broadband stimuli in normal–hearing subjects (Zwicker, 1960; Zwicker and Scharf, 1965; Paulus and Zwicker, 1972; Zwicker and Fastl, 1990). They combined Fletcher's filter–bank concept (Fletcher, 1940) and Stevens' power law (and its variations) within a new model; see also (Moore and Glasberg, 1995; Moore and Glasberg, 1986). Basically, Zwicker's model consists of a bank of auditory filters with nonlinear processing carried out on the output of each filter band by applying a power law. Thus, Zwicker's model might be viewed as a link between the pure psychological and the pure physiological approaches. Zwicker's loudness model currently is the model which best accounts for loudness perception in normal–hearing subjects. Therefore, it is probably most promising to extend this model to also account for hearing impairment. In this study, two approaches that extend Zwicker's model for sensorineural hearing impairment, are compared.

The basic structure of the loudness model for normal–hearing subjects is briefly described in the next section. A more detailed description of Zwicker's model is presented in appendix A.

7.2 Loudness model for normal–hearing listeners

A schematic plot of the different stages involved in Zwicker's loudness model for stationary signals is shown in Fig. 7.1. This model assumes that loudness is not directly related to stimulus intensity but is related to the level within different auditory filters. The bandwidth of these filters depends on level; it increases with increasing level. The outputs of the auditory filters are converted to the so–called excitation pattern. In this study, calculation of the excitation pattern was performed using the algorithm proposed by Moore and Glasberg (1990). It is much easier to broaden auditory filters in this approach than in the origianl approach proposed by Zwicker. Broadening of filters could be important for modeling hearing impairment.

Next, the specific loudness N' is calculated from the excitation level E in each band separatly, assuming a power law relationship, but with an exponent slightly less than in Steven's power law. In the original model, Zwicker applied equation 7.1 for calculating specific loudness from the excitation pattern, while Moore and Glasberg (1995) used equation 7.2, which yielded a better description of the results of partial masking experiments.

$$N' = C \cdot \left(\frac{E_0}{E_{ThQ}}\right)^{\alpha} \cdot \left(\left(0.5 + \frac{E}{E_{ThQ}}\right)^{\alpha} - 1\right), \tag{7.1}$$

$$N' = C \cdot \left(E^{\alpha} - E_{ThQ}^{\alpha}\right), \tag{7.2}$$

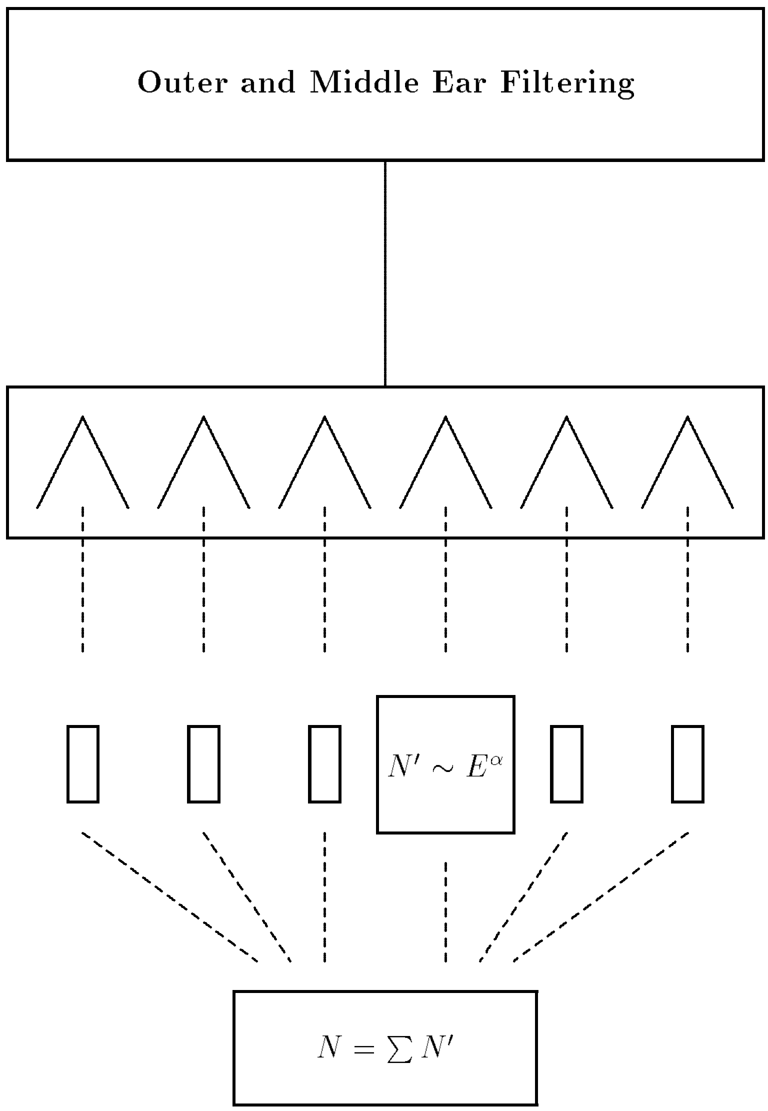

Fig. 7.1: *Schematic representation of the different stages of Zwicker's loudness model. A fixed filter representing transfer through the outer and middle ear is followed by an auditory filter bank from which excitation patterns E are calculated. In the third step these excitation patterns are transformed into specific loudness N' by a compressive power law relationship. Summing the specific loudness across auditory filters yields the overall loudness N.*

with E_{ThQ} = threshold in quiet, $\alpha = 0.23$, $C = 0.064$ in Eq. 7.1 and $\alpha = 0.21$, $C = 0.086$ in Eq. 7.2. It is assumed that this power law relationship reflects the compressive processing mechanisms within the cochlea. Finally, the specific loudness is summed across bands to give the overall loudness. The physiological correlate of the summation of loudness across critical bands might be the summation of neural activity, i.e., loudness may be related to the total neural activity evoked by the sound, summed across the whole activity pattern.

Zwicker's loudness model calculates loudness values in units of sones, i.e., using a ratio scale. Hohmann (1993) proposed an empirical transformation between measured categorical loudness and theoretical loudness values derived from Zwicker's loudness model. He suggested the following empirical relationship for transforming loudness calculated in sones to categorical units:

$$CU = 17.6 \cdot log(2.5 \cdot N) \tag{7.3}$$

where CU stands for categorical unit (range 0 to 50) and N for loudness in sones. This logarithmic relationship between the two scales simply reflects the different shapes (curved versus linear) of the loudness functions obtained when using the different scaling techniques, i.e., a ratio scale versus a categorical scale with few categories. This relationship will be used in subsequent sections to transform loudness from a sone scale to a categorical scale. Different studies in the literature also supposed a logarithmic relationship between the results of categorical scaling experiments and the results applying a ratio scaling techniques underlying the sone scale (Stevens, 1957; Luce and Krumhansl, 1988; Poulton, 1989; Hellbrück et al., 1991; Hellbrück, 1993).

7.3 Loudness model for hearing–impaired listeners

When extending the loudness model for normally hearing to describe loudness perception of the hearing impaired, the above mentioned perceptual alterations due to cochlear hearing loss (raised absolute threshold, recruitment, reduced loudness summation and reduced frequency selectivity) have to be accounted for. Before describing the two approaches to extending the loudness model described above in more detail, the influence of the perceptual alterations on loudness perception and the different ways to account for them in Zwicker's model, are briefly discussed.

Recruitment–like effects can be produced in two different ways. Some authors (Florentine and Zwicker, 1979; Humes and Jesteadt, 1991; Moore, 1995) have simply modified the assumed absolute threshold E_{ThQ} in Eq. 7.2 and 7.1, without changing the exponent of the power law. Increasing E_{ThQ} in either equation, yields a steeper than normal increase of the loudness function. Thus, audiometric threshold shift (sensitivity loss) and recruitment (compression loss) are modeled by one single component, an "internal noise". Another possibility, which has until now not been discussed in the literature, is to account for recruitment and loss of sensitivity by two separate components rather than by a single one. Reduced sensitivity could be achieved by a frequency dependent attenuation of the excitation level within each auditory filter, and less compression by increasing the exponent in the power law. Increasing the exponent in either, Eq. 7.2 or 7.1, also produces a steeper than normal increase of the loudness function. Different exponents (and thus different slopes of loudness functions) might then be viewed as reflecting individual differences in cochlear mechanics in hearing–impaired subjects. A similar approach has been applied by Leijon (1991) to predict the gain function of linear hearing aids in a more realistic way.

Another factor which should be taken into account is the reduced frequency selectivity in the hearing impaired, i.e., the broadened auditory filters (Tyler, 1986; Glasberg and Moore, 1986). Reduced frequency selectivity can affect loudness because the auditory filters are broader than normal, so there are more critical bands across which excitation level is summed. It appears, that reduced frequency selectivity can to a large extent be accounted for by the broadening of "normal" auditory filters with increasing level (cf. Fig. 4.3, p. 31). In other words, it is assumed that, to a first order approximation, the auditory filter bandwidths of normal–hearing and hearing–impaired subjects are equal when compared at the same sound pressure level (dB SPL), while they differ when compared at the same sensation level (dB SL). Thus, calculating excitation patterns at the respective sound pressure level (dB SPL) accounts for broadened auditory filters in the hearing impaired by the dependence of "normal" filter bandwidth on level.

Reduced frequency selectivity can influence the way that loudness changes with bandwidth, i.e., loudness summation. This effect is reduced or even completely absent in the hearing impaired. This reduction in loudness summation is often too large to be explained by reduced frequency selectivity alone, as discussed in chapter 5. It appears that reduced compression in the cochlea, for instance by increasing the exponent α when calculating the specific loudness from the excitation patterns, also has to be taken into account. However, no further component has to be introduced in the model in order to account for reduced loudness summation, since reduced frequency selectivity as well as reduced compression are already taken into account.

In summary, it appears that accounting for only two components might be sufficient to explain the principle effects. when the loudness model for the normal–hearing subjects is extended for hearing impairment: raised absolute threshold and loudness recruitment. In the following, two ways of modifying the above described loudness model for normal–hearing subjects are discussed in more detail.

7.3.1 One–component approach to modeling hearing impairment

In the "one–component approach" it is assumed that hearing impairment is similar to a masking condition in normals. This assumption is based on the finding that loudness functions in normal–hearing subjects increase with level at a higher rate if a continuous, broadband masker is presented simultaneously with the signal to be judged (Hellman and Zwislocki, 1964; Scharf, 1978; Zwicker and Fastl, 1990). This so–called "partial masking" condition resembles the recruitment phenomenon in hearing–impaired listeners. Combined with the steeper increase of normal loudness function near threshold, this observation led to the approach of modeling recruitment in hearing–impaired listeners by an inaudible, internal noise. Increasing the threshold E_{ThQ} in the calculation of specific loudness from excitation pattern, in Eq. 7.1 and 7.2 actually steepens the loudness function. In Fig.

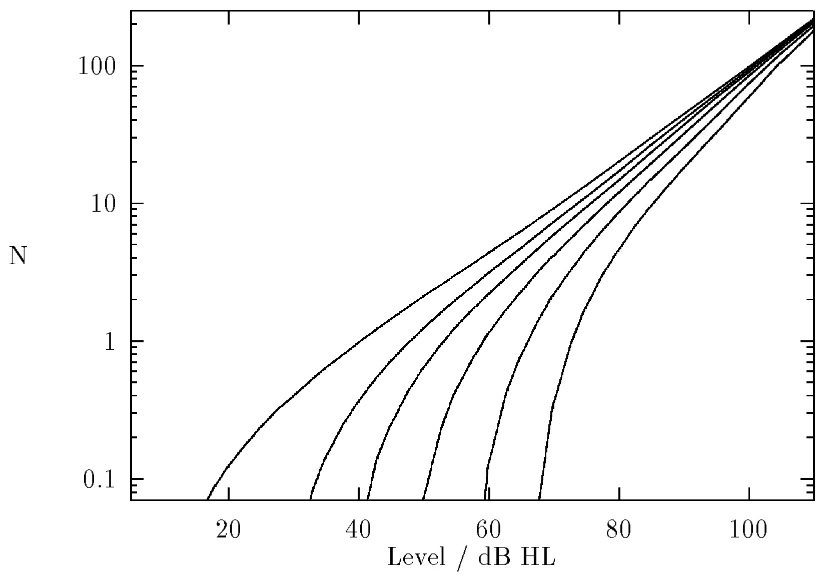

Fig. 7.2: *The loudness functions in units of sones are plotted versus stimulus level with different thresholds E_{ThQ} as parameter. Note that the loudness functions grow at a higher rate when threshold E_{ThQ} is increased. $E_{ThQ} = 10, 30, 40, 50, 60, 70$ dB, curves from left to right, respectively.*

7.2 loudness functions in sones N are plotted versus level with different thresholds E_{ThQ} as parameter. Clearly, near threshold the loudness function grows at a higher rate than at high levels. However, it does not matter in principal, whether Eq. 7.2, 7.1 is applied, since both equations produce similar shaped and steepened loudness functions, although the increase of both functions near threshold differs slightly. At mid to high sound pressure levels they yield the same loudness functions. Therefore, it is not crucial which of the two formulae is applied when using the one–component approach for modeling hearing impairment. Note, that the loudness function for the partially masked or recruitment–like condition approaches the normal loudness function at high levels. This "catching–up" is well known from complete recruitment. In summary, it appears that the modification of one single component, threshold E_{ThQ}, suffices to model both components of hearing loss, sensitivity loss and compression loss. Thus, threshold is the key variable in this approach.

Moore (1995) applied this model to describe the data measured by Miskolczy–Fodor (1960) using a loudness balancing technique. The calculated loudness function models on average the individual data quite well. However, the data show a variability of up to 15 – 20 dB since the individual data are not distinguished. It is therefore difficult to estimate the accuracy of the model when dealing with individual data.

Florentine and Zwicker (1979) applied this approach to describe the results of loudness balancing experiments with hearing–impaired listeners using stimuli of differing bandwidth. They measured the level difference required to obtain equal loudness between a 709 Hz wide noise band and a 5909 Hz wide noise band, both centered at 4 kHz in nine hearing–impaired listeners. The thresholds of the hearing–impaired subjects were about 45 dB. In accordance with our results presented in chapter 5, they reported less loudness summation in the hearing impaired. Indeed, loudness summation is already reduced when modeling hearing impairment by an internal noise. This might partly be due to the broadened auditory filters at higher levels. However, adding the internal noise alone did not suffice for modeling reduced loudness summation. Florentine and Zwicker considered how to further modify the model described above, to account for steepened loudness functions as well as for reduced loudness summation. They proposed to further increase the auditory filter bandwidth by a constant factor of three. By incorporating broadened auditory filters, the model predictions came much closer to the measured data. Note, that the main effect of modifying the filter bandwidth is to produce a parallel shift of the obtained loudness function. The slope of the loudness function is not strongly altered by changing the filter bandwidth. This is demonstrated in Fig. 7.3. The solid curve represents the loudness function computed with "normal" auditory filters while the dotted curve indicates that calculated with filters which are two times broader than normal. Obviously, the shape and the slope of the loudness function are not markedly altered by the broadening of filters. A similar result was reported by Moore (1995).

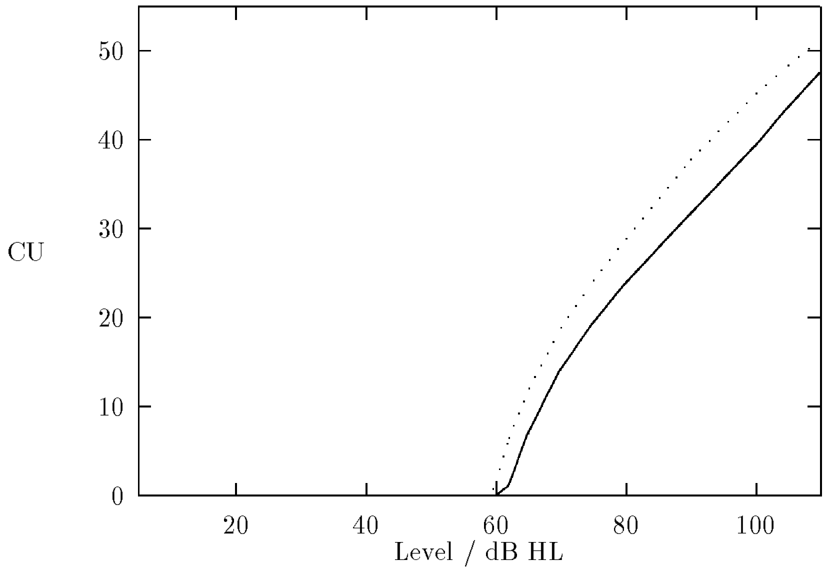

Fig. 7.3: *Calculated loudness functions (loudness in categorical units (CU) versus level) simulating hearing impairment applying the one–component approach. The solid curve represents the function calculated with "normal" auditory filters while the dotted curve indicates that calculated with filters which are two times broader than normal.*

In summary, this one–component approach seems to describe mean loudness functions of hearing–impaired listeners quite well. Furthermore, it is possible to model reduced loudness summation in the hearing impaired. However, until now, this approach has not been applied to model individual loudness growth functions obtained with a direct scaling technique like magnitude estimation or categorical scaling. This is carried out in section 7.5.

7.3.2 Two–component approach to modeling hearing impairment

The exponent α of the power law (Eq. 7.1, 7.2) incorporated in the loudness model for normal hearing subjects, probably partly reflects the nonlinear, compressive characteristics of inner ear mechanics. One important alteration in the hearing impaired is the less compressive signal processing in the inner ear. Thus, it appears reasonable to modify this exponent when modeling sensorineural hearing impairment. In Fig. 7.4 the influence of increasing the exponent of the power law is shown schematically. The solid curve represents the function

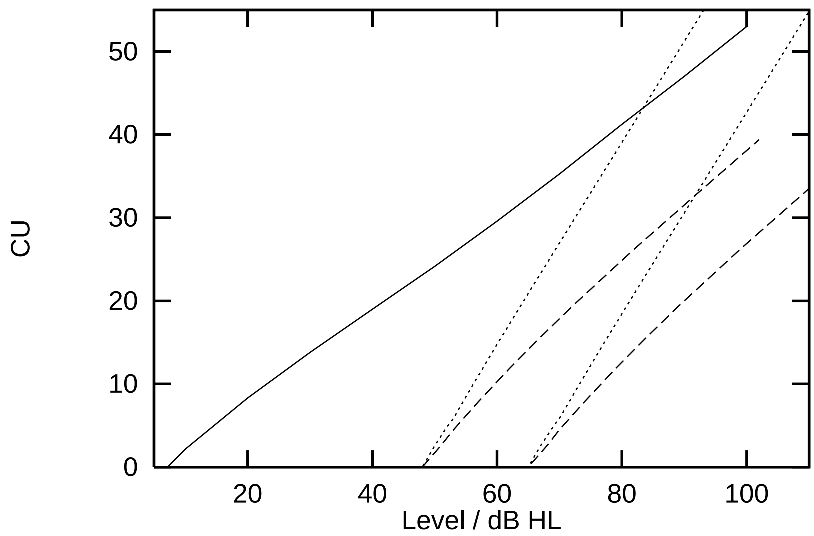

Fig. 7.4: *Calculated loudness functions (loudness in categorical units (CU) versus level) simulating hearing impairment applying the two–component approach. The loudness functions were calculated for two different absolute thresholds. The solid curve represents the function calculated with "normal" compression, while the dotted and the dashed curve indicate those calculated with increased exponents. Dotted: $\alpha = 0.8$, dashed: $\alpha = 0.6$.*

calculated with normal compression ($\alpha = 0.23$), while the dotted ($\alpha = 0.8$) and the dashed ($\alpha = 0.6$) curve indicate those calculated with increased exponents. Raising the exponent yields a steeper loudness function but has no influence on the absolute threshold. Therefore, raised audiometric threshold has to be taken into account by a second component. It is modeled by a linear attenuation of the excitation level per critical band but after calculating the excitation patterns. Increasing threshold yields a parallel shift of the loudness function as is evident from Fig. 7.4. Note, that only if the raised audiometric threshold is modeled in a different way than in the previously described one–component approach, increasing the exponent causes an increase in the slope of the loudness function. Otherwise, the increase in slope due to increased exponent is always "masked" by the increase in slope due to raised threshold.

Thus hearing impairment is modeled by two rather than one component. Firstly, reduced sensitivity is accounted for by a frequency dependent attenuation of the excitation level within each auditory filter using the audiometric threshold of the respective subject. This reflects a sensitivity loss component of cochlear hearing loss mentioned in chapter 2. Secondly, increasing the exponent α when calculating specific loudness from excitation patterns accounts for less compressive processing in inner ear mechanics. This resembles a compression loss component of cochlear hearing loss. Thus, in contrast to the one–component approach it is assumed that the two components (sensitivity and compression component) of cochlear hearing losses are independent of each other. From a physiological point of view this is only partly true, since damage to OHCs causes both reduced sensitivity and loss of compressive processing. However, it appears to be a reasonable assumption, since correctly accounting for this interdependence would imply correctly modeling the signal processing in the cochlea, i.e., corrctly modeling the active processes.

In the next sections, both approaches are applied to modeling the results of the loudness scaling experiments described in chapter 5. Neither of the two approaches has until now been applied to model the data of individual subjects. In contrast to the one–component approach, the two–component approach has also not been applied to model average data.

7.4 Modeling and fitting procedure

The loudness functions of the hearing–impaired subjects have been obtained employing a categorical scaling technique with 10 categories. Bandpass filtered noises with differing bandwidth were employed as stimuli. The center frequencies were between 1300 and 3000 Hz and the bandwidth varied from 1–6 critical bands. Details of the measurement technique and the stimuli employed and the individual results of the 14 hearing–impaired subjects involved in the study were presented in section 5.2, Tab. 5.1 and section 5.3, Tab. 5.3.

Procedure

For calculating the excitation pattern of a stationary stimulus, the algorithm proposed by Glasberg and Moore (1990) was employed. The dependence of auditory filter bandwidth on level in the hearing impaired was assumed to be the same as in normal–hearing subjects, as given in appendix A, Eq. A.2. Thus, reduced frequency selectivity is taken into account by the dependence of filter bandwidth on level. The specific loudness is calculated from excitation level using the formula originally proposed by Zwicker (1960), i.e., Eq. 7.1. The absolute thresholds of the subjects were taken from the results of the loudness scaling experiments with the narrowband stimuli. The individual thresholds of the 14 hearing impaired listeners are given in appendix C.

a) One–component approach

In this approach, the internal noise E_{ThQ} in Eq. 7.1 reflecting the threshold of hearing is given by the absolute threshold of the individual subjects. No further parameter was modified. A prediction of the individual loudness growth functions is derived for both the narrowband and the broadband data.

b) Two–component approach

In this approach, raised threshold is modeled by attenuating the excitation level within each auditory filter by the audiometric threshold before specific loudness is calculated. From this reduced excitation level, specific loudness is calculated using eq. 7.1. Calculated loudness was fitted to the measured loudness values by adjusting the exponent α in Eq. 7.1 using solely the results obtained with the narrowband stimuli. In order to account for a hearing loss depending on frequency, the exponent α was multiplied by a factor β depending on frequency f:

$$\beta = a \cdot f^2 + b \cdot f + c.$$

Thus, the exponent α is adjusted depending on frequency by adjusting the parameters a, b, c. For fitting these parameters, a least–squares technique was applied (Press et al., 1992). Due to the narrow spectral range of the signals employed in chapter 5, the quadratic parameter a was 0.0 in all but one subject (JC). Thus fitted, the model was applied to predict the loudness values obtained with the broadband stimuli. Note that, when the loudness of a signal spreading across different auditory filters is calculated, a different exponent (depending on frequency) is applied within each auditory filter for calculating the specific loudness from the excitation patterns.

c) Evaluation of the model predictions

Measured and calculated data were compared using the nonlinear correlation coefficient B_{nl} according to Schach and Schäfer (1978) and Press et al. (1992). It is a nonlinear deviation measure defined for n measured data points y_i with mean $\bar{y}$ and the respective calculated values $\hat{y}_i$:

$$B_{nl} = 1 - \frac{\sum_{i=1}^{n}(y_i - \hat{y}_i)^2}{\sum_{i=1}^{n}(y_i - \bar{y})^2}$$

For $B_{nl} = 1$ an optimal fit is obtained, i.e., the "quality of fit" is best. $B_{nl} = 0$ is obtained when the mean $\bar{y}$ of all measured data points is used for all calculated values $\hat{y}_i$. $B_{nl} < 0$ means that the prediction of measured data by the calculated curves is worse than that by simply predicting each y_i by the mean value $\bar{y}$.

7.5 Results

The individual results of all 14 hearing impaired subjects from comparing the model predictions with the measured data (B_{nl} values for all employed frequencies and for both approaches) are given in Tab. 7.1. However, before discussing the individual results in detail, the two approaches are compared on the basis of several figures showing the individual results for two subjects (HF, JKn).

For subject HF, the measured and the calculated loudness functions using the one–component approach are shown in Figs. 7.5 and 7.6. The one–component approach yields a very good description of the data for low to medium levels for all the stimuli. At medium to high levels, however, the measured data and the calculated curve differ markedly. At these levels, the description of the data by the one–component approach is rather poor. Specifically, the slopes of the measured and calculated loudness functions differ markedly. The calculated loudness functions underestimate the slopes of the measured loudness functions. For subject JKn, the measured and the calculated loudness functions using the one–component approach are shown in Figs. 7.9, 7.10. The one–component approach models the measured data rather poorly. The measured data and the calculated curves, differ markedly for all different stimuli and at all levels. Specifically, the one–component approach overestimates the slopes of the measured loudness functions.

In contrast, the loudness functions calculated by applying the two–component approach, describe the measured data rather well. It is evident from Figs. 7.7 and 7.8 for subject HF and Figs. 7.11, 7.12 for subject JKn and from Tab. 7.1, that the two–component approach accounts more accurately for the measured data of both subjects than the one–component approach. Note also that, without modifying any further parameter such as the auditory filter bandwidth, reduced loudness summation is also correctly described by the two–component approach, since the loudness functions determined with the broadband stimuli are also correctly modeled.

The fitted parameters, a, b and c, when applying the two–component approach are given in Tab. 7.2. The factors by which the exponent α was multiplied were between 1 (no alteration) and 4.5. Thus, the exponent α takes values between "normal" compressive (0.21) and linear (1.0).

The individual results of all 14 hearing–impaired subjects from comparing the model predictions with the measured data (B_{nl} values for all employed frequencies and for both approaches) are given in Tab. 7.1. The B_{nl} values obtained with the two–component approach are markedly larger than those obtained with the one–component approach for all but one loudness function (subject AW, 2210 Hz). This further indicates that the two–component approach provides a better description of both narrowband and broadband individual data sets. Furthermore, even for subject AW, the mean B_{nl}, averaged across frequency, is 0.877 for the two–component approach and 0.828 for the one–component approach. This indicates that the former again provides a better description of the overall data.

For almost every subject at least one loudness function is found for which the one–component approach yields a reasonable fit of the data. For an individual subject this

		Narrowband Stimuli						Broadband Stimuli				
VP		1375.	1600.	1860.	2160.	2500.	2925.	1480.	1635.	1795.	1985.	2210.
AP	TWO	0.748	0.887	0.887	0.901	0.873	0.878	0.850	0.899	0.887	0.801	0.753
	ONE	0.516	0.724	0.724	0.630	0.801	0.684	0.776	0.615	0.405	0.025	-0.203
AW	TWO	0.853	0.934	0.934	0.896	0.880	0.928	0.828	0.872	0.876	0.869	0.776
	ONE	0.662	0.844	0.844	0.883	0.811	0.879	0.814	0.867	0.864	0.838	0.804
CS	TWO	0.710	0.926	0.926	0.799	0.841	0.354	0.920	0.843	0.885	0.911	0.829
	ONE	0.492	0.809	0.809	0.375	0.388	-2.366	0.812	0.482	0.261	0.425	0.029
EH	TWO	0.933	0.889	0.889	0.820	0.835	0.262	0.780	0.944	0.937	0.925	0.856
	ONE	0.811	0.818	0.818	-0.531	-0.904	-5.544	0.587	0.860	0.839	0.897	0.772
HF	TWO	0.977	0.982	0.975	0.974	0.939	0.929	0.983	0.964	0.984	0.971	0.974
	ONE	0.866	0.821	0.857	0.936	0.910	0.850	0.854	0.861	0.923	0.912	0.949
Hk	TWO	0.968	0.979	0.979	0.964	0.921	0.909	0.793	0.896	0.830	0.829	0.465
	ONE	0.949	0.918	0.918	0.546	0.550	0.383	0.700	0.737	0.538	0.467	0.068
JC	TWO	0.943	0.961	0.961	0.915	0.966	0.933	0.974	0.975	0.882	0.896	0.871
	ONE	0.917	0.920	0.920	0.265	0.208	0.351	0.879	0.867	0.679	0.758	0.703
JKl	TWO	0.812	0.901	0.901	0.933	0.839	0.817	0.955	0.948	0.957	0.945	0.981
	ONE	0.728	0.621	0.621	-3.290	-1.271	-2.205	0.954	0.781	0.857	0.804	0.696
JKn	TWO	0.992	0.990	0.990	0.947	0.930	0.865	0.985	0.967	0.976	0.991	0.981
	ONE	0.886	0.749	0.749	-1.007	-0.877	-0.488	0.925	0.904	0.708	0.599	0.821
MU	TWO	0.891	0.843	0.843	0.889	0.664	0.767	0.973	0.977	0.974	0.960	0.928
	ONE	0.594	0.402	0.402	-4.214	-5.958	-4.203	0.683	0.511	0.082	-0.301	0.124
RB	TWO	0.900	0.945	0.945	0.966	0.905	0.886	0.791	0.840	0.800	0.792	0.846
	ONE	0.860	0.685	0.685	0.460	0.221	0.854	0.610	0.530	0.380	0.185	0.493
RS	TWO	0.938	0.908	0.908	0.916	0.868	0.710	0.956	0.934	0.935	0.923	0.894
	ONE	0.907	0.883	0.883	0.405	-0.250	0.435	0.937	0.903	0.835	0.785	0.771
RW	TWO	0.955	0.957	0.957	0.976	0.955	0.874	0.858	0.694	0.574	0.484	-0.055
	ONE	0.550	0.414	0.414	0.307	0.762	0.616	0.178	-0.422	-1.018	-1.467	-2.397
UH	TWO	0.961	0.835	0.835	0.739	0.585	0.701	0.854	0.791	0.623	0.583	0.520
	ONE	0.746	0.716	0.716	-1.230	-4.406	-2.095	0.610	0.253	-0.687	-1.129	-0.606

Table 7.1: *Nonlinear correlation coefficient B_{nl} for both approaches to model loudness perception. TWO indicates the values for the two–component approach and ONE those for the one–component approach. Results of 14 hearing–impaired listeners.*

approach describes the data fairly well for some frequencies but yields a rather poor fit at others (cf. subjects HK, JC, JKn). Specifically, the scatter in B_{nl} for an individual subject across stimuli is larger for the one–component than for the two–component approach, where the quality of the fit is more or less constant across stimuli. For two subjects (AW, HF) the one–component approach describes the measured functions of both data sets, narrowband as well as broadband, and across all stimuli reasonably well. However for both

VP	a	b	c
AP	0.0	-0.086	4.27
AW	0.0	0.015	1.99
CS	0.0	-0.162	5.44
EH	0.0	-0.191	6.15
HF	0.0	-0.075	4.49
HK	0.0	-0.101	3.78
JC	-0.177	7.063	-65.34
JKl	0.0	-0.230	6.86
JKn	0.0	-0.115	4.12
MU	0.0	-0.300	8.81
RB	0.0	0.035	0.82
RS	0.0	-0.228	6.70
RW	0.0	-0.003	1.70
UH	0.0	-0.160	5.1

Table 7.2: *Fitted parameters a (quadratic), b (linear) and c (constant) for modifying the exponent across frequencies in the two–component approach.*

subjects, the two–component approach still provides a better description of the measured data. Furthermore, the one–component approach accounts nearly correctly for reduced loudness summation, since the predicted difference between the level of a narrowband and a broadband signal needed to produce equal loudness is reduced to a similar magnitude as expected from the experiments. For all subjects, the difference between prediction and observation with this approach is less than 3 dB for the signal with the largest bandwidth, i.e., the noise band with a bandwidth of 6 critical bands centered at 2210 Hz. Thus, the one–component approach correctly describes the measured effect of reduced loudness summation (cf. chapter 5).

In chapter 5 we mentioned three subjects (EH, MU, JKn) who exhibited some residual loudness summation although suffering from a large amount of sensorineural hearing impairment. Even their loudness functions are rather well described by the two–component approach. In all three subjects the one–component approach fits the data rather poorly. Measured and calculated loudness functions using both approaches are shown in Figs. 7.9 – 7.12 for subject JKn.

Although the two–component approach yields a rather good description of the measured data, the results of two subjects (RW, UH) need further consideration. For subject RW, the two–component approach fits the narrowband data very well but describes the broadband data rather poorly. Specifically, with increasing center frequency the quality of fit decreases. The loudness functions of subject RW are shown in Fig. 7.13 – 7.14. Note that thresholds of his loudness curves increase with increasing center frequency more for broadband than for the narrowband stimuli. Since the broadband stimuli were presented in ascending order of the center frequencies, an increase in center frequency is directly related to increasing duration of the experiment. This suggests auditory fatigue or a reduction in concentration as the experiment proceeded. Since this increase in threshold is not taken into account by the model a deviation occurs between predicted and observed broadband loudness functions. For subject UH, a similar finding, although less pronounced, is observed.

Overall, although the one–component approach describes to some extent the properties of impaired loudness perception, the two–component approach accounts more accurately for the measured individual data. Furthermore, it describes the experimental results across different stimulus conditions more consistently.

Subject: HF, narrowband data, one–component approach

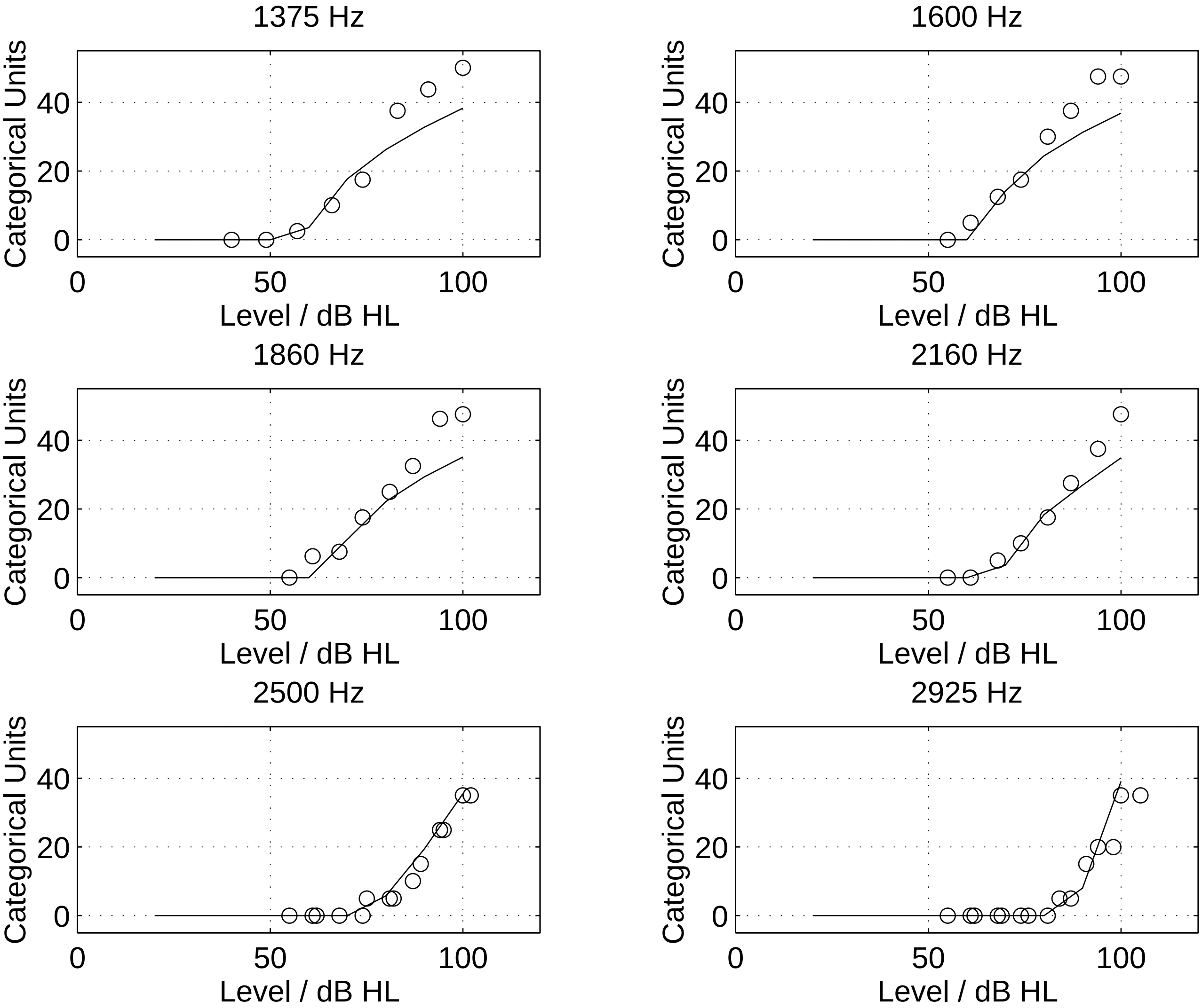

Fig. 7.5: *Measured (○) and calculated (—) loudness data are plotted versus level for one subject (HF) for six narrowband noises with different center frequencies. The center frequencies of the stimuli are indicated above each panel. Hearing loss was modeled using the one–component approach.*

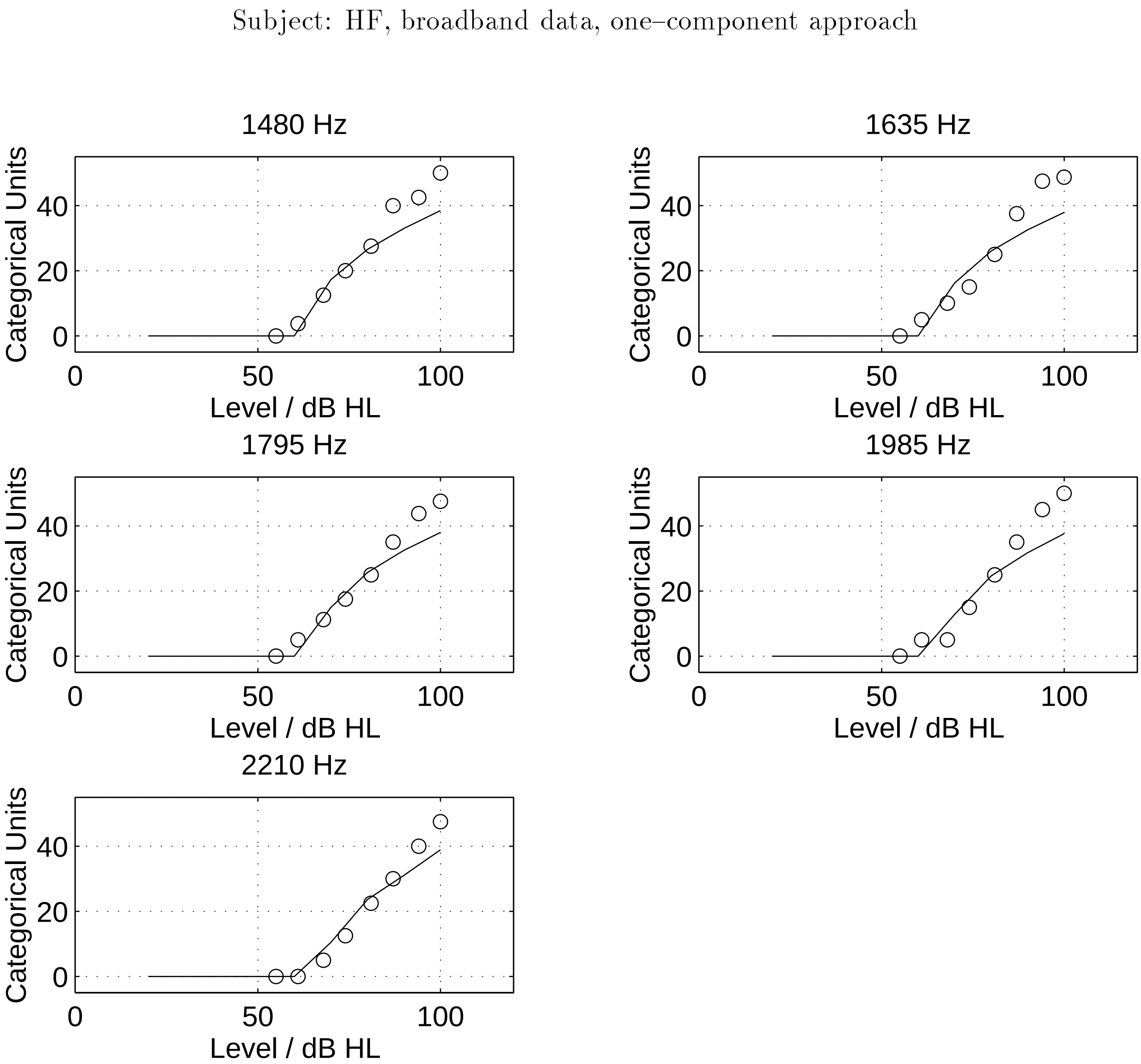

Fig. 7.6: *Same representation as Fig. 7.5 but for broadband signals. The center frequencies of the stimuli are indicated above each panel. The bandwidths of signals were two critical bands (1480 Hz) to six critical bands (2210 Hz).*

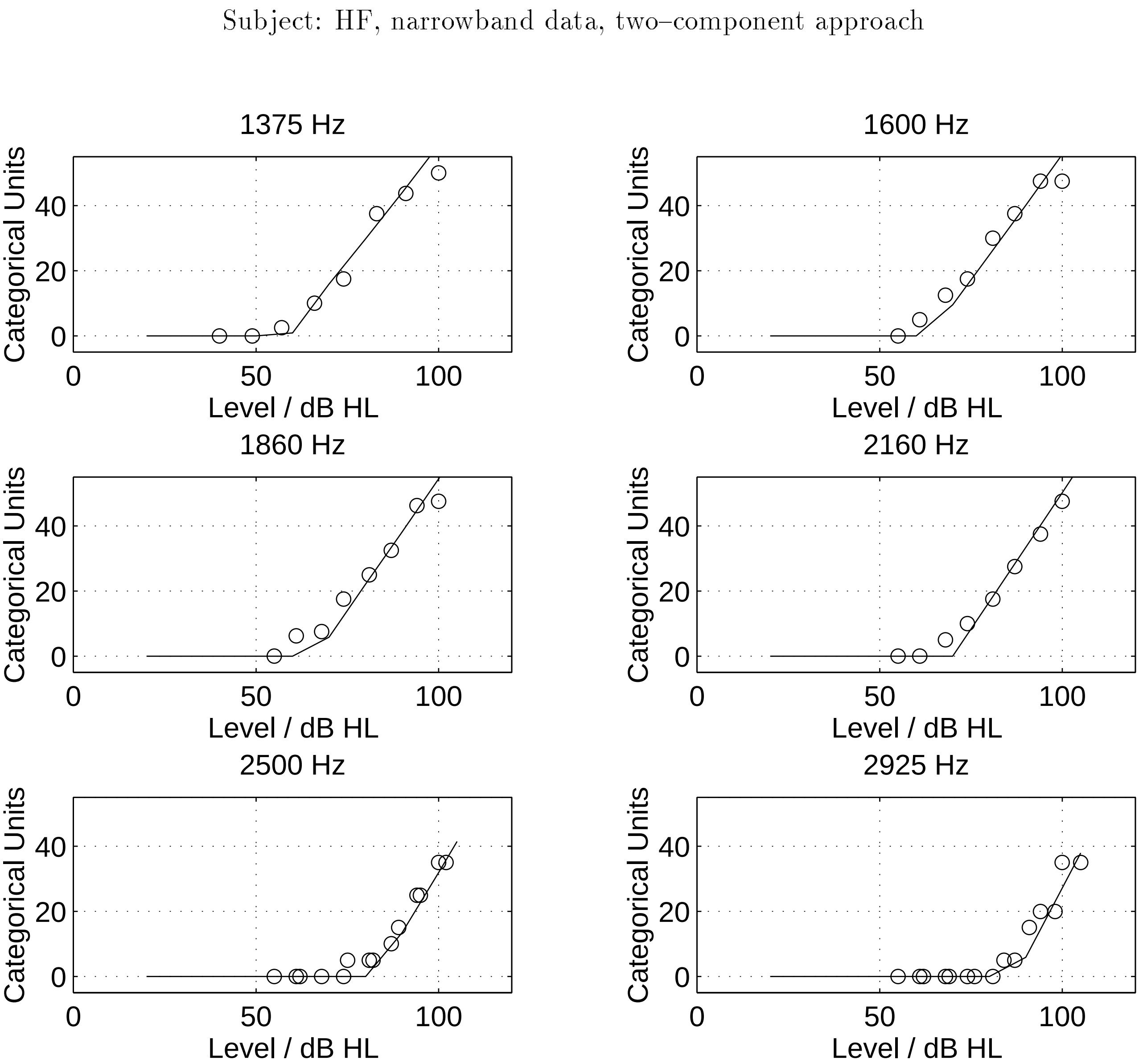

Fig. 7.7: *Measured (○) and calculated (—) loudness data are plotted versus level for one subject (HF) for six narrowband noises with different center frequencies. The center frequencies of the stimuli are indicated above each panel. Hearing loss was modeled using the two–component approach.*

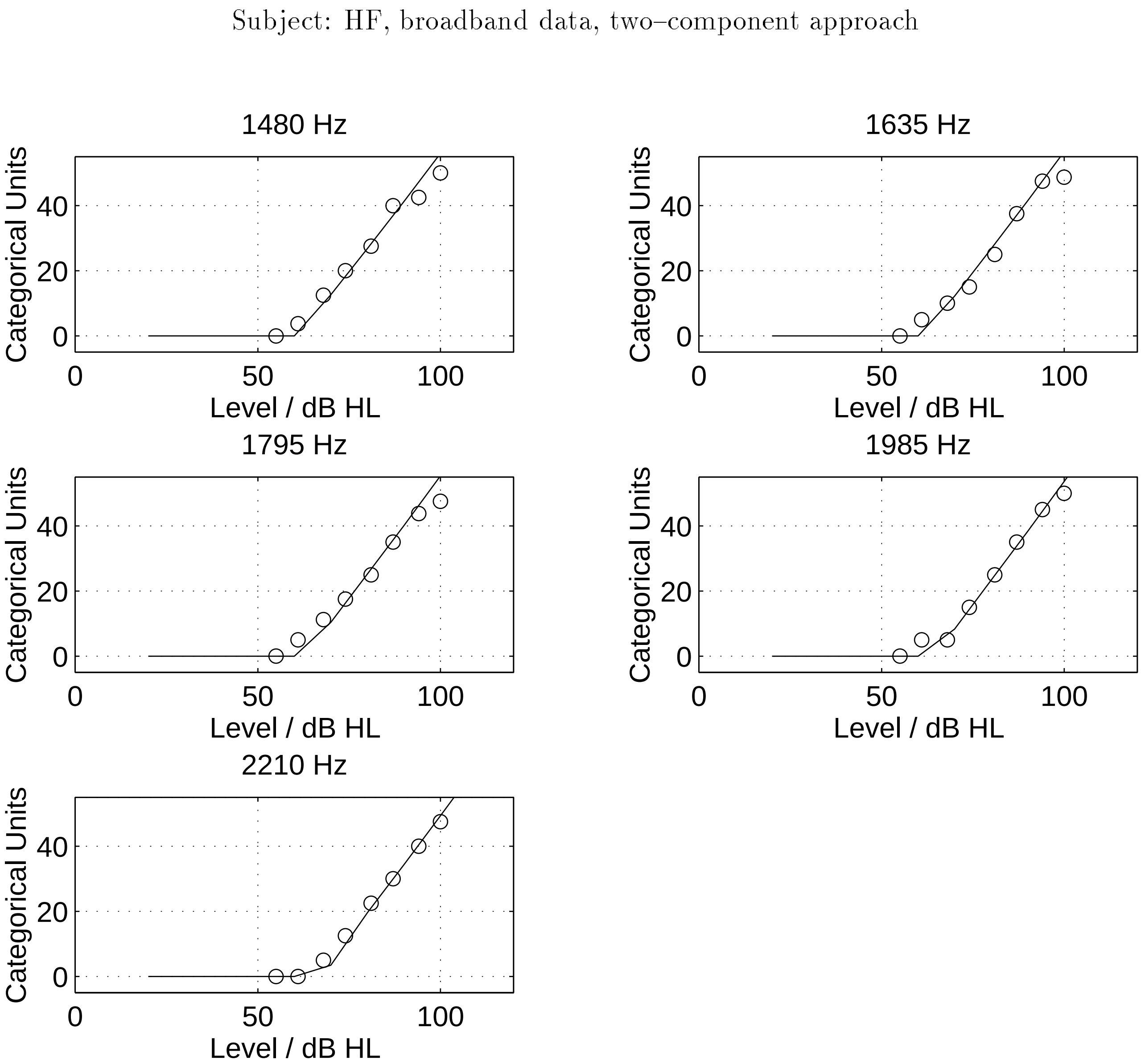

Fig. 7.8: *Same representation as Fig. 7.7 but for broadband signals. The center frequencies of the stimuli are indicated above each panel. The bandwidths of signals varied between two critical bands (1480 Hz) and six critical bands (2210 Hz).*

Subject: JKn, narrowband data, one–component approach

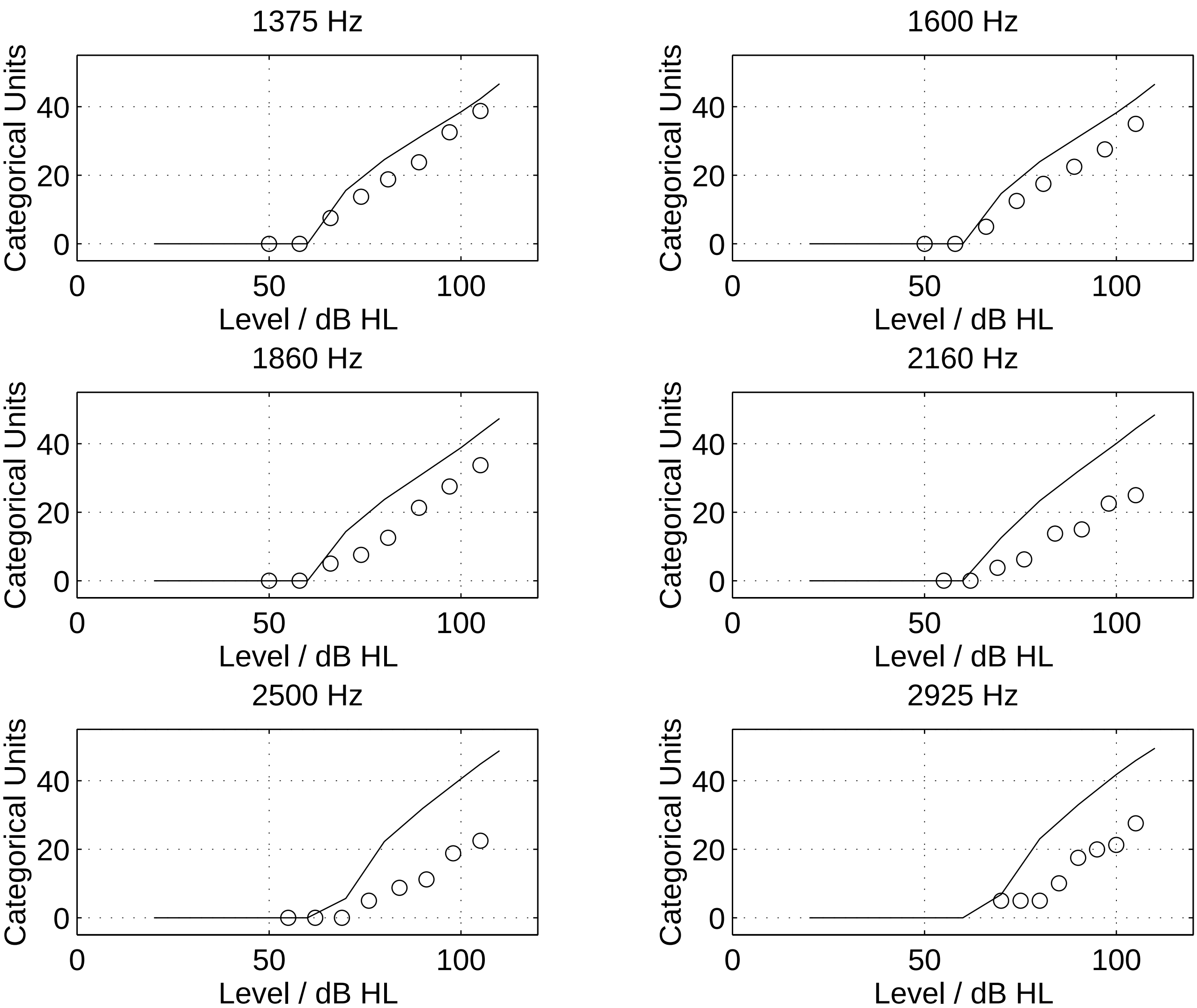

Fig. 7.9: *Measured (○) and calculated (—) loudness data are plotted versus level for one subject (JKn) for six narrowband noises with different center frequencies. The center frequencies of the stimuli are indicated above each panel. Hearing loss was modeled using the one–component approach.*

Subject: JKn, broadband data, one–component approach

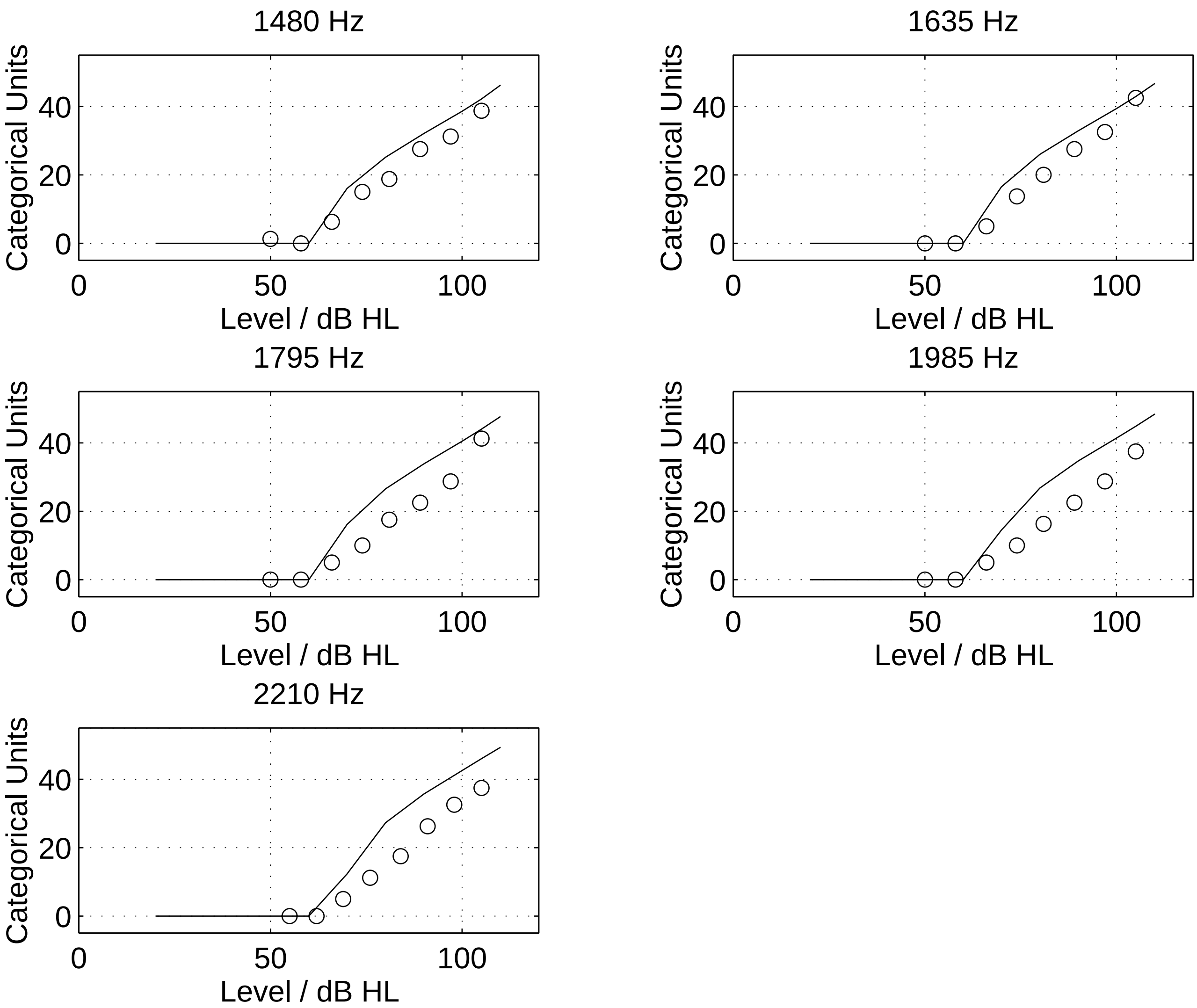

Fig. 7.10: *Same representation as Fig. 7.9 but for broadband signals. The center frequencies of the stimuli are indicated above each panel. The bandwidths of signals varied between two critical bands (1480 Hz) and six critical bands (2210 Hz).*

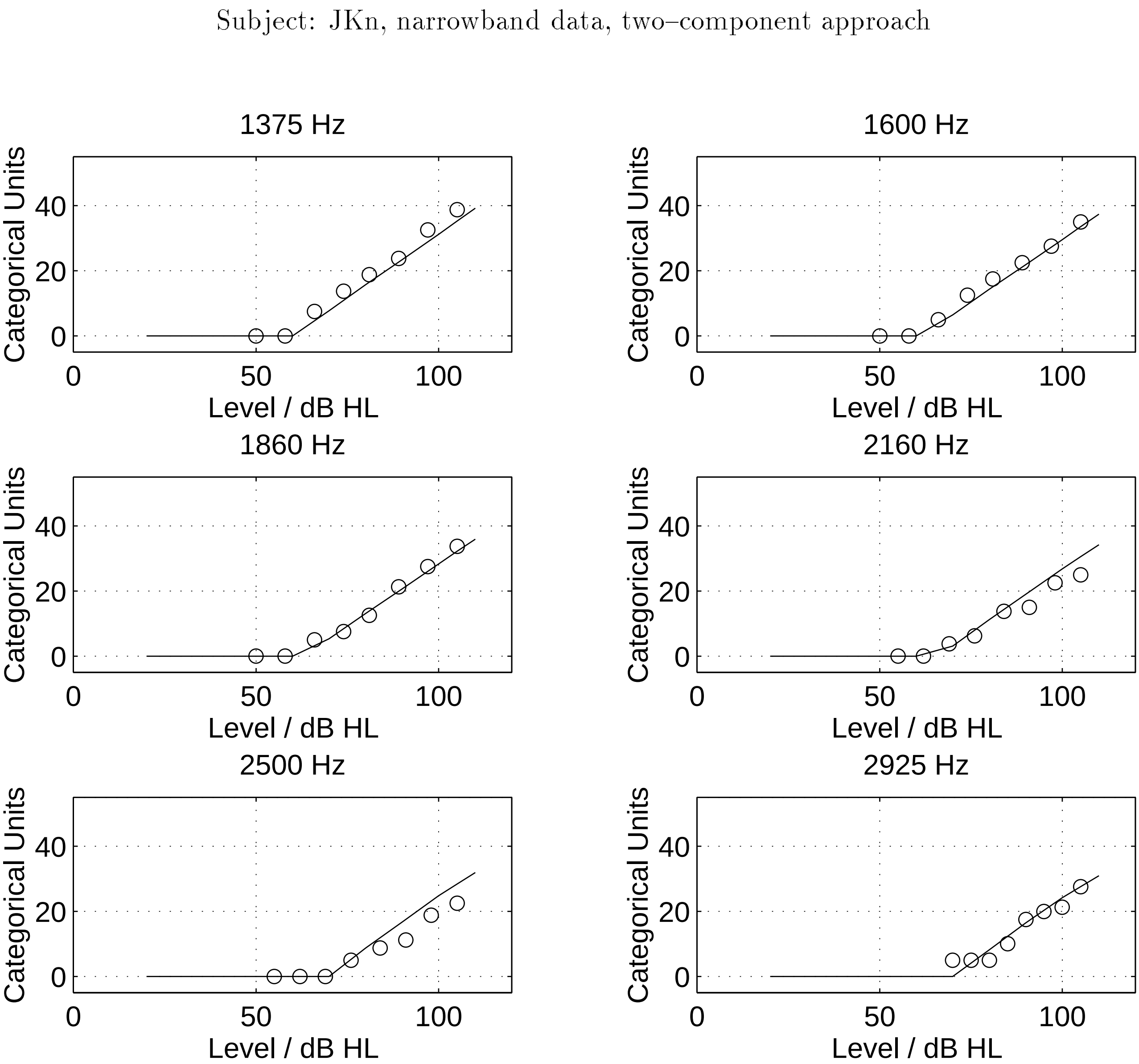

Fig. 7.11: *Measured (○) and calculated (—) loudness data are plotted versus level for one subject (JKn) for six narrowband noises with different center frequencies. The center frequencies of the stimuli are indicated above each panel. Hearing loss was modeled using the two–component approach.*

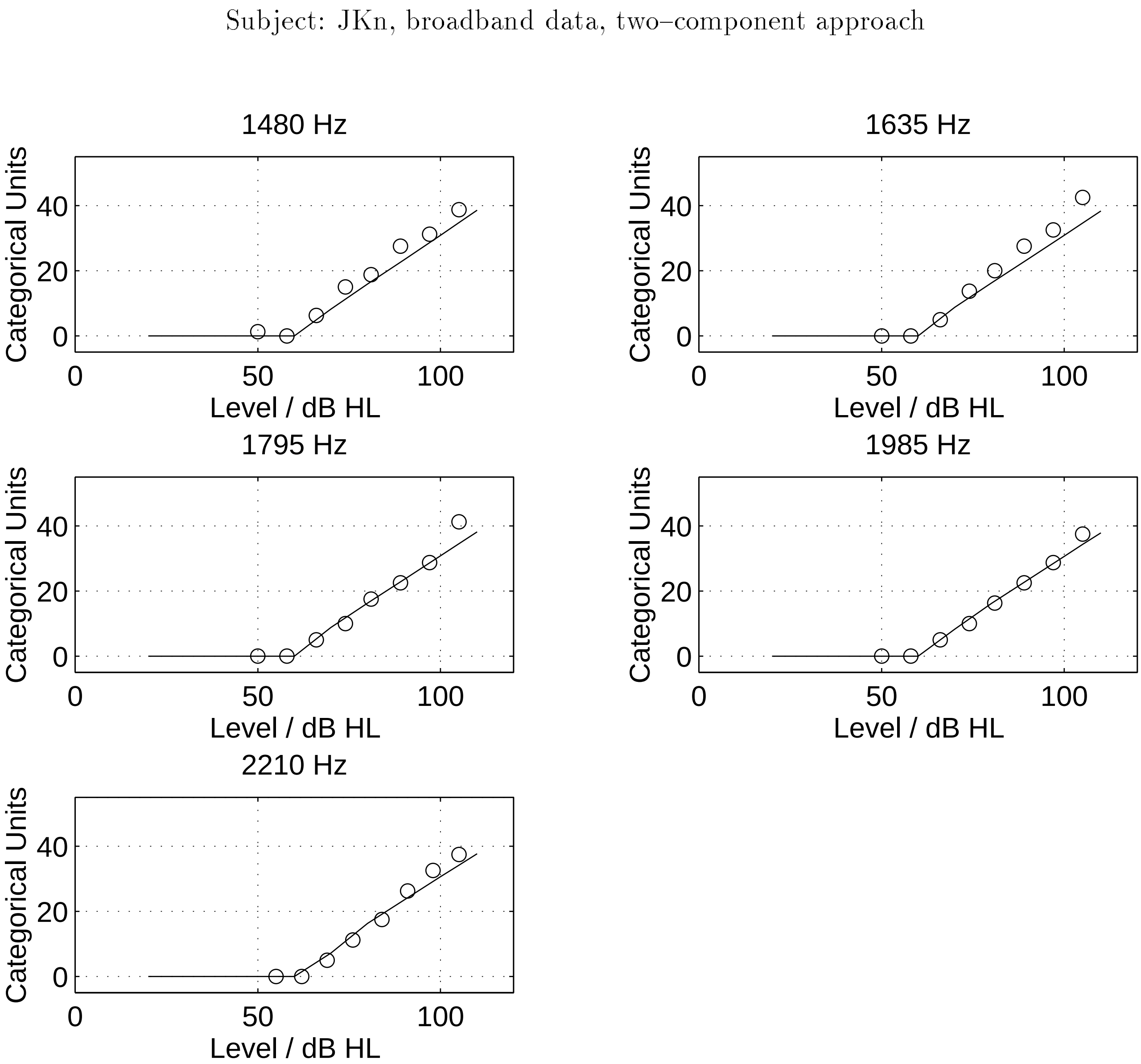

Fig. 7.12: *Same representation as Fig. 7.11 but for broadband signals. The center frequencies of the stimuli are indicated above each panel. The bandwidths of signals varied between two critical bands (1480 Hz) and six critical bands (2210 Hz).*

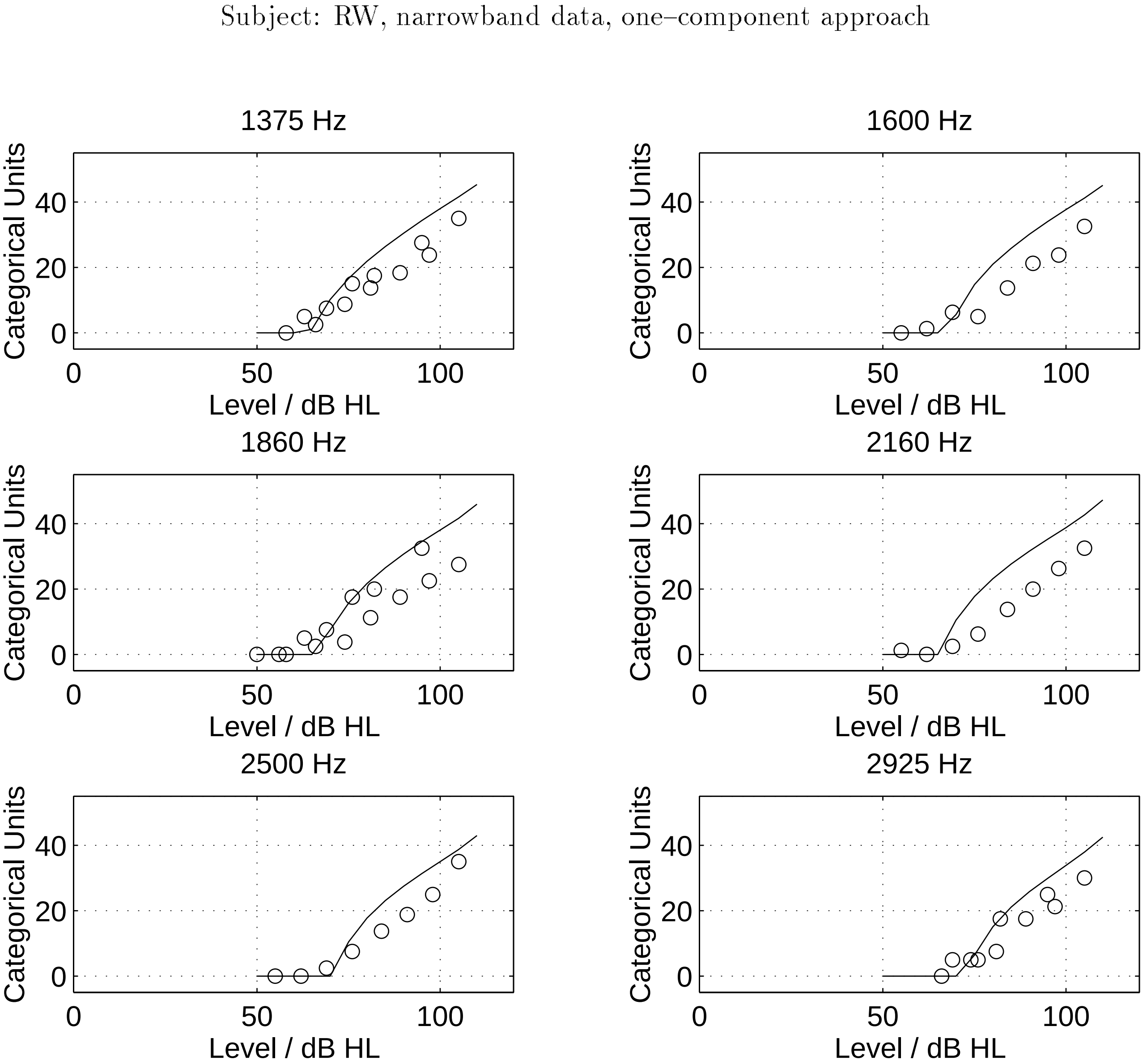

Fig. 7.13: *Measured (○) and calculated (—) loudness data are plotted versus level for one subject (RW) for six narrowband noises with different center frequencies. The center frequencies of the stimuli are indicated above each panel. Hearing loss was modeled using the one–component approach.*

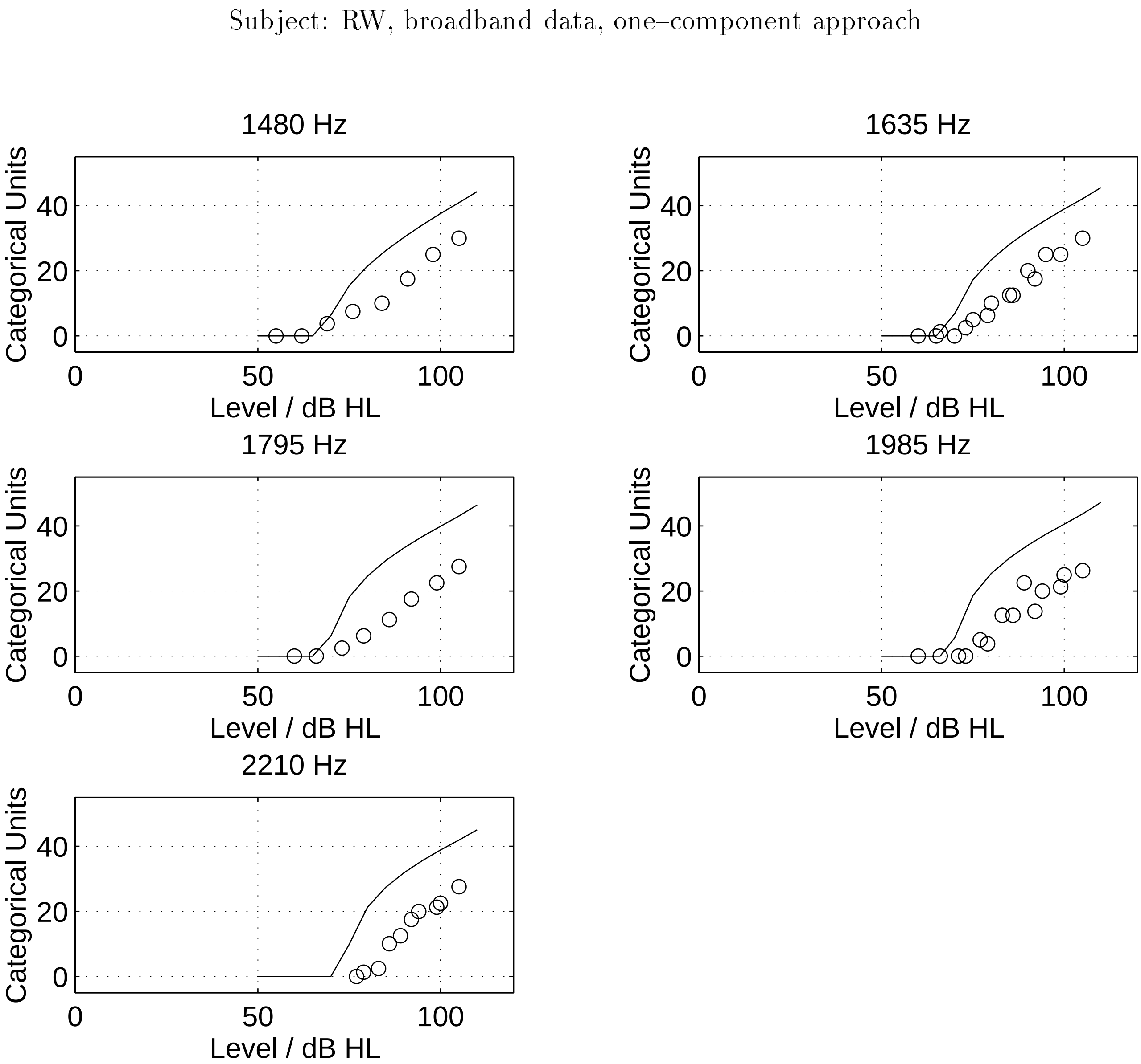

Fig. 7.14: *Same representation as Fig. 7.13 but for broadband signals. The center frequencies of the stimuli are indicated above each panel. The bandwidths of signals varied between two critical bands (1480 Hz) and six critical bands (2210 Hz).*

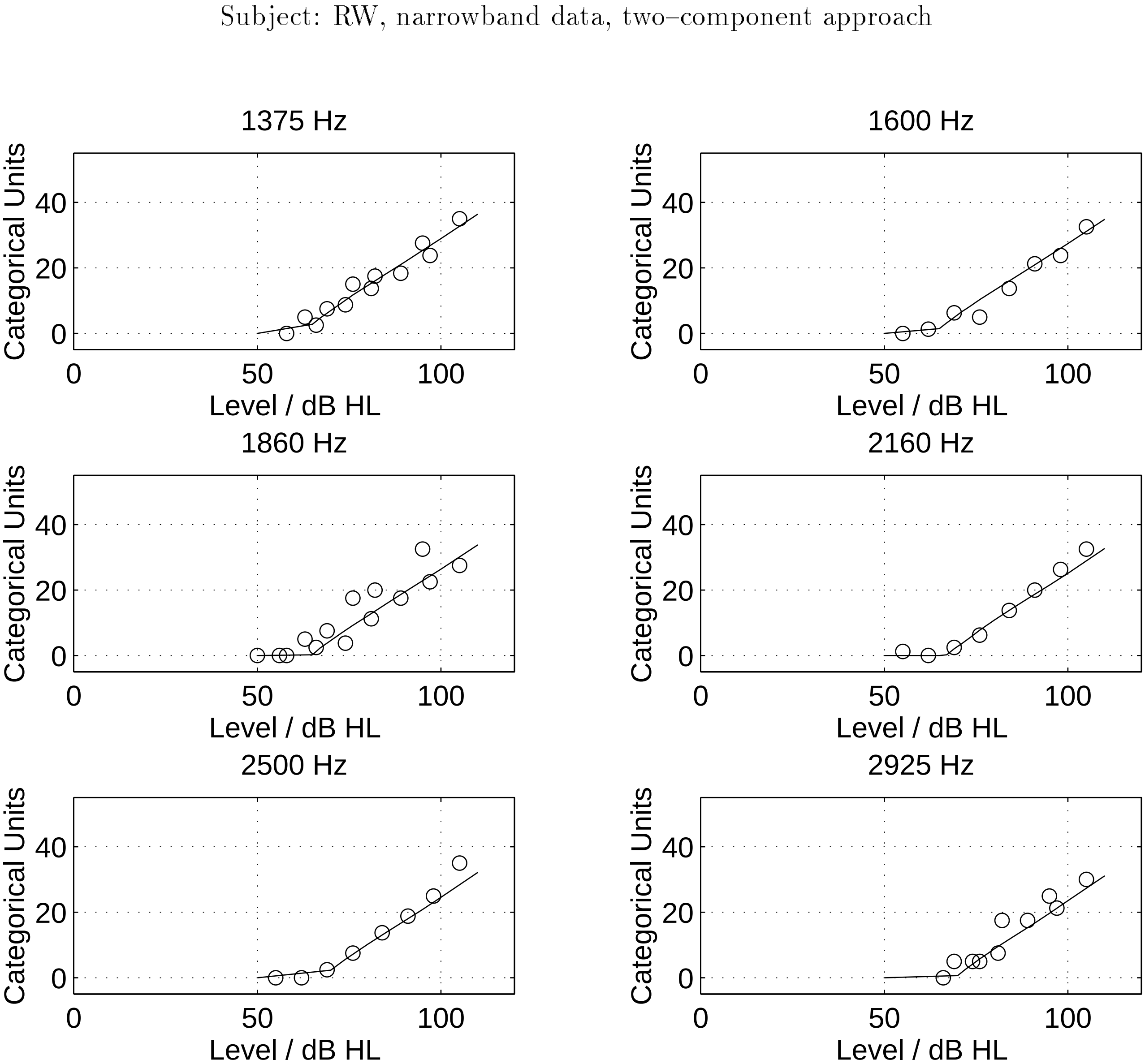

Fig. 7.15: *Measured (○) and calculated (—) loudness data are plotted versus level for one subject (RW) for six narrowband noises with different center frequencies. The center frequencies of the stimuli are indicated above each panel. Hearing loss was modeled using the two–component approach.*

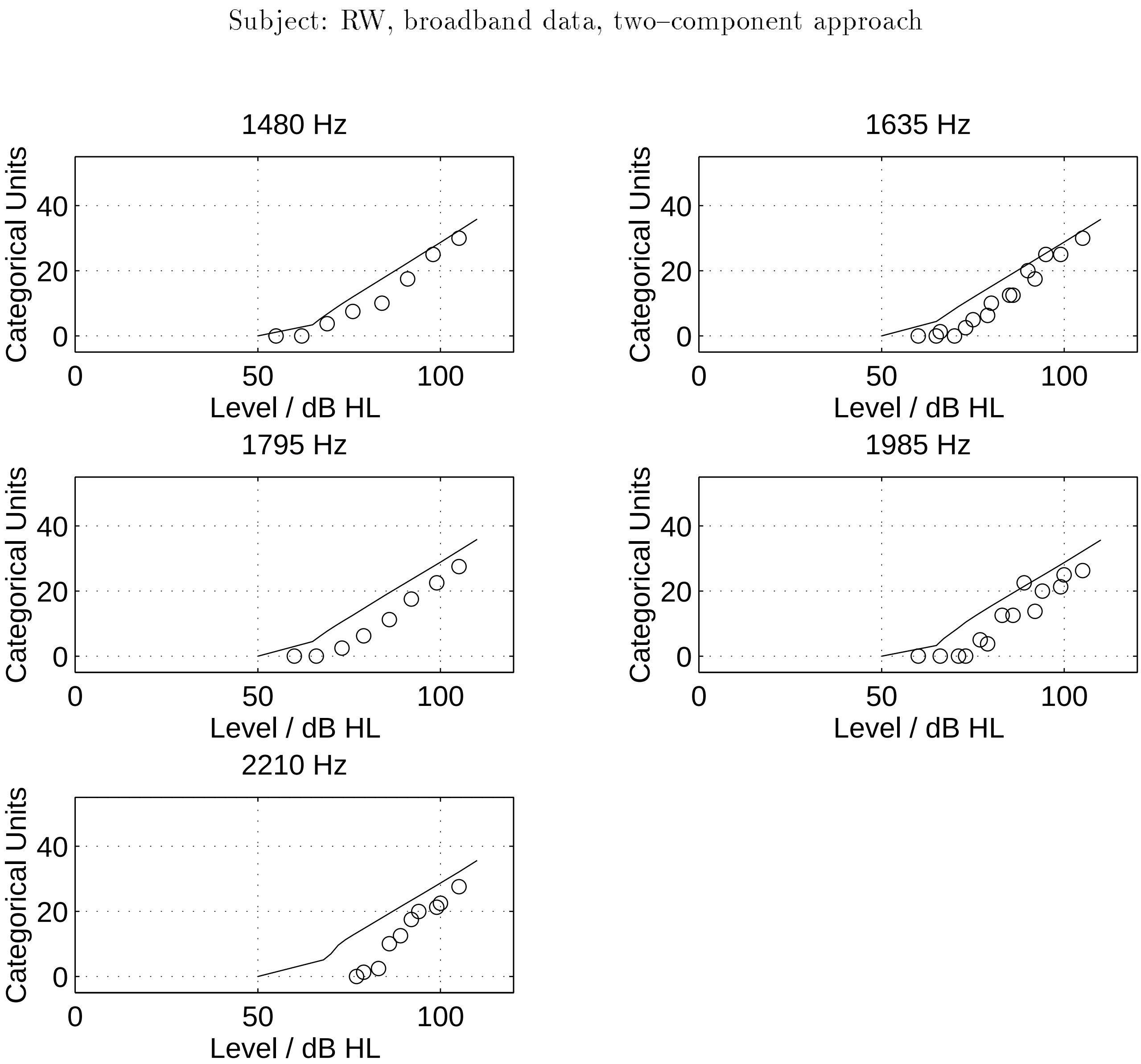

Fig. 7.16: *Same representation as Fig. 7.15 but for broadband signals. The center frequencies of the stimuli are indicated above each panel. The bandwidths of signals varied between two critical bands (1480 Hz) and six critical bands (2210 Hz).*

7.6 Discussion

Two approaches to extending Zwicker's loudness for hearing impairment have been applied to modeling loudness functions of individual hearing–impaired subjects. It was shown that the one–component approach sometimes yields a reasonable description of the measured loudness functions. However, since large intersubject variability is observed in the individual data, this approach sometimes overestimates and sometimes underestimates the measured loudness functions. Specifically, it does not predict the slope of most of the measured loudness functions correctly. This is due to the assumption inherent in this approach that individuals suffering from a similar hearing loss should show similar changes in psychoacoustic performance, specifically a similar slope of the loudness function. However, as discussed in chapter 4, this is not the case. Large intersubject variability is generally observed in different auditory parameters and especially the slopes of the loudness functions. As a consequence, the individual data cannot be correctly described by applying the one–component approach. Furthermore, the differences between measured and calculated loudness functions cannot be reduced by further modifying other parameters such as the auditory filter bandwidth, since this parameter has only a small influence on the slopes of the loudness functions. Therefore, adjusting this parameter does not produce a better description of the data. Overall, in accordance with the results reported by Moore (1995), this approach appears to account for measured loudness functions on average correctly but fails to predict individual loudness growth functions.

In contrast to the study of Florentine and Zwicker (1979), the one–component approach had not to be further modified in order to account for reduced loudness summation, that is no further broadening of filters was necessary. However, it is not clear how auditory filters in the hearing impaired were calculated in the Florentine and Zwicker study. The auditory filters applied in our study might have been broader than their filters. Thus, no further broadening was necessary. Note that the one–component approach accounts for reduced loudness summation although it provides a poor fit for both of the loudness functions, for narrowband and broadband, stimuli. Loudness summation, however, reflects the difference between two functions (narrowband and broadband). Hence, the difference obtained from the calculated functions might be correct even if both functions deviate substantially from the measured data.

Various other problems raise serious doubt as to whether the one–component approach is an appropriate way of modeling individual loudness data. Very different mechanisms underly masking and cochlear impairment (Phillips, 1987). For instance, noise–masked normals, i.e., simulated hearing–impaired subjects, and "true" hearing–impaired subjects show different performance in loudness summation. While loudness summation still occurs in noise–masked normals (Scharf, 1978) it is strongly reduced in the hearing impaired. This indicates, that masking is not an appropriate way of modeling hearing impairment. Thus, there is no physiological correlate to the internal noise in an impaired cochlea. Furthermore, this approach implicitly assumes a strong correlation between audiometric threshold shift and the alterations in pyschoacoustic performance. This further implies that inner ear mechanics in all hearing–impaired subjects are altered in the same way. However, no such correlation has up to now been reported in the literature, neither for loudness perception nor for auditory filter bandwidths or any other auditory parameter. In chapter 4 several sources for the large interindividual variability usually observed in psychoacoustic experiments with hearing–impaired listeners, have been discussed. It was concluded that inner ear mechanics in individual hearing–impaired subjects might not be comparable to each other. In other words, the large variability might reflect systematic differences in the cochlear mechanics of hearing–impaired listeners.

Modeling hearing impairment using the two–component approach avoids the above mentioned problems related with the one–component approach. The two–component approach does not assume that hearing impairment is comparable to a masking condition, i.e., it is not assumed that threshold is the only key variable. Instead, it is supposed that the exponent is an additional key parameter. Furthermore, this approach is built on the assumption that the large intersubject differences observed in the experimental results partly reflect systematic differences of cochlear processing between individual subjects and may not be due to chance as discussed in chapter 4. In order to account for the large individual differences, the exponent of the power law was adjusted to describe the individually measured loudness functions using narrowband stimuli. One might argue that it is not very surprising that fitting a power law function to a more or less linear function by adjusting the exponent, yields a good description of the measured data. However, if the model is adjusted in this way, it also predicts the individual loudness functions obtained with broadband stimuli rather well. Thus, the broadband condition might be viewed as a “test” for the model. Further different conditions have to be tested in order to evaluate this modeling approach. Lastly, there is a clearcut physiological motivation for this modeling approach, since the two–component approach more closely resembles current physiological models of impaired cochlear mechanics. The assumption whether the exponent of the power law reflects the nonlinear inner ear mechanics, cannot be proven or further justified. However, Moore (1995) also suggested that the exponent reflects properties of inner ear mechanics. Furthermore, a similar approach of reducing the compressive nonlinearity of a power law to account for the data of hearing–impiared subjects in nonsimultaneous masking, was applied by Oxenham and Moore (1995). However, the two–component approach is based on the assumption that the two components contributing to cochlear hearing losses are independent of each other. From a physiological point of view this is only partly true, as has already been discussed. This assumption might be justified in hindsight by the good agreement between measured data and calculated curves. Furthermore, in order to correctly account for this interdependence of components, psychophysical models are needed which model the cochlear signal processing in a more realistic way. Currently several authors are working on this problem in the contexts of loudness (Carney, 1994; Verhey et al., 1995) and frequency selectivity (Stone, 1994). Yet another approach was proposed by Moore (personal communication): Moore suggested combining the two approaches. However, this combination requiers a model describing this interdependence which is currently not available.

In addition, it appears that modeling reduced frequency selectivity in the hearing impaired is less crucial than the correct modeling of the loss of compression when dealing with loudness perception. In both approaches, one–component and two–component, reduced frequency selectivity was accounted for by an "average" broadening of auditory filters. It appears that this average broadening suffices to account for reduced frequency selectivity in the individuals.

In the work presented above, it is assumed that the compression used within the model is linked to the compression often observed in cochlear processing. It may be that higher levels in the auditory pathway are also responsible in part for the compression and hence, in the hearing impaired, for recruitment. In chapter 6, it was pointed out that higher levels might indeed contribute to recruitment although no clear–cut results have been reported to date.

7.7 Summary and conclusions

In this chapter two different ways of modeling loudness perception in hearing–impaired listeners by modifying a loudness model similar to Zwicker's loudness model for normally hearing, were described. The loudness functions were measured using a categorical scaling technique with bandpass filtered noises with different bandwidths as stimuli. The first way (one–component approach) assumed that hearing impairment is similar to a masking condition in normal–hearing subjects. It is assumed that audiometric threshold shift and recruitment can be modeled by one component, by an internal noise. In this approach, threshold is the key variable. However, this model cannot be applied for describing individual data. This is due to the large intersubject variability of the slopes of loudness functions as a function of audiometric threshold.

Therefore, another way of modeling hearing impairment was proposed, which has until now not been discussed in the literature. In this approach (two–component approach) it is assumed that raised threshold and recruitment should be accounted for separately: raised threshold by a frequency specific attenuation and recruitment by increasing the exponent of the power law incorporated in the loudness model. This way of modeling hearing impairment is adapted from physiological finding that the main alteration in the mechanics of impaired cochleae is the less compressive processing characteristic. Increasing the exponent accounts for this alteration.

The exponent of the power law was fitted to the measured loudness functions of individual hearing–impaired subjects obtained with narrowband stimuli. Adjusted in this way, the model also accounted correctly for data measured with broadband noises.

It is concluded that the two–component approach is a more appropriate way of modeling loudness perception in the hearing impaired than the one–component approach. Furthermore, the exponent of the power law and the threshold appear to be the key variables in modeling loudness perception in the hearing impaired.