Auditory filter shapes at 8 and 10 kHz

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Auditory filter shapes were derived from notched-noise masking data at center frequencies of 8 kHz (for three spectrum levels, $N_0 = 20$, 35, and 50 dB) and 10 kHz ($N_0 = 50$ dB). In order to minimize variability due to earphone placement, insert earphones (Etymotic Research ER2) were used and individual earmolds were made for each subject. These earphones were designed to give a flat frequency response at the eardrum for frequencies up to 14 kHz. The filter shapes were derived under the assumption that a frequency-dependent attenuation was applied to all stimuli before reaching the filter; this attenuation function was estimated from the variation of absolute threshold with frequency for the three youngest normally hearing subjects in our experiments. At 8 kHz, the mean equivalent rectangular bandwidths (ERBs) of the filters derived from the individual data for three subjects were 677, 637, and 1011 Hz for $N_0 = 20$, 35, and 50 dB, respectively. The filters at $N_0 = 50$ dB were roughly symmetrical, while, at the lower spectrum levels, the low-frequency skirt was steeper than the high-frequency skirt. The mean ERB at 10 kHz was 957 Hz. At this frequency, the filters for two subjects were steeper on the high-frequency side than the low-frequency side, while the third subject showed a slight asymmetry in the opposite direction.

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INTRODUCTION

The measurement of signal thresholds in a notched-noise masker provides a well-established method for estimating the shape and bandwidth of the auditory filter under the assumptions of the power spectrum model of auditory masking (Patterson, 1976; Patterson and Nimmo-Smith, 1980; Moore and Patterson, 1986). Estimates of the equivalent rectangular bandwidth (ERB) of the auditory filter between 0.1 and 6.5 kHz are reasonably consistent across studies (Moore and Glasberg, 1983), but we are aware of no measurements for center frequencies above 6.5 kHz. One reason for this omission is the inherent variability in the interaction between the acoustic delivery system and the peripheral auditory system; another is the problem of calibrating the stimuli at high frequencies and of producing stimuli with the desired spectral characteristics. The difficulties encountered are legion and are well reviewed by Buus *et al.* (1986) in a paper that reported measurements of psychophysical tuning curves at frequencies above 10 kHz. To the list of problems they quote may be added the great variation seen in the size and shape of individual ear canals.

The measurement of auditory filter shapes at high center frequencies is of interest for both theoretical and practical reasons. For example, the high-frequency region of the cochlea seems to be particularly susceptible to damage by noise, anoxia, or ototoxic agents. Measurement of the auditory filter shape at high center frequencies might provide a sensitive way of detecting early signs of such damage. Obviously, data would be required for normally hearing subjects before any clinical applications could be investigated. Another area in which knowledge of the auditory filter shape would be useful is that of auditory localization. It is known that the pinnae modify the spectra of incoming sounds in a way that depends on the direction of the sound source (Blauert, 1983); the changes occur mainly above 6 kHz. There is little detailed knowledge, however, about the ability to detect and discriminate these spectral changes (Moore *et al.*, 1989), and measurement of the auditory filter shape at high frequencies would make it possible to predict the detectability of such spectral changes.

In the present study we estimated auditory filter shapes at 8 and 10 kHz, using insert earphones designed to produce a flat frequency response at the eardrum for frequencies up to 14 kHz (see Sec. I). Each subject was fitted with an individual earmold, and, once inserted, the earmold and associated earphone remained in place for the whole of an experimental session. Although this arrangement was intended to minimize the problems of variability produced by earphone placement and head movement, it obviously could not control for the effects of individual differences in canal morphology.

I. METHOD

A. Stimuli

In pilot experiments, a two-alternative forced-choice task was used, with the signal and masker gated synchronously. We found, however, that it was sometimes very difficult to "hear out" the signal from the masker, even when the signal-to-masker ratio was very high, since it was difficult to discern a tonal quality associated with the high-frequency signal. This problem was alleviated by gating the signal on and off within the presentation time of the masker. That was the procedure adopted in the main experiment.

The 8- and 10-kHz sinusoidal signals (Farnell DSG1

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signal generator) had 20-ms raised-cosine ramps and an overall duration of 200 ms at the 6-dB down points; the notched-noise masker also had 20-ms ramps and an overall duration of 500 ms. The signal started 250 ms after the start of the masker and terminated 50 ms before the end of the masker. The interstimulus interval was 500 ms. The masker consisted of two independent noise bands, one above and one below the center frequency f_c. Each was produced by multiplying (Analog Devices 534L) a sinusoid (Farnell DSG1) by a band of noise (Hewlett–Packard 3722A) that had been low-pass filtered (Kemo VBF8, 96 dB/oct). Measured at the half-power points, each noise band had a width of 0.4 f_c. At the widest notch widths, the width of the lower frequency band was adjusted to avoid "wrapping around" zero frequency.

At 8 kHz, three noise spectrum levels (N_0) were used: 20, 35, and 50 dB (*re*: 20 μPa). At 10 kHz, only the 50-dB spectrum level was used since pilot results at the two lower spectrum levels indicated that hardly any masking occurred at the larger notch widths. This can be attributed to the rapid rise in absolute threshold above 8 kHz (see Sec. II).

The notch width of the noise masker is specified as the deviation of the nearer edges of the upper or lower noise bands from the center frequency f_c, divided by f_c; it will be denoted by the symbol Δ. In one set of conditions, the notch was symmetrically placed around f_c and values of Δ for each band were 0.0, 0.1, 0.2, 0.3, 0.4, 0.5, and 0.6. At the lower levels a value of Δ of 0.05 was also included. In a second set of conditions, the near edge of the upper masker band was 0.2 farther away from f_c than the near edge of the lower band. Values of Δ for the lower/upper bands in this situation were: 0.05/0.25, 0.1/0.3, 0.15/0.35, 0.2/0.4, 0.3/0.5, 0.4/0.6, 0.5/0.7, and 0.6/0.8. The third set of conditions was the mirror image of the second, with the lower band being farther away from the signal frequency than the upper.

The timing of all stimuli was controlled by a Texas Instruments 990/4 computer. Analog multipliers (AD534L) were used as gates, gating voltages being derived from two 12-bit digital-to-analog converters. Two multipliers were used in series to give an on–off ratio exceeding 100 dB. The computer varied the signal level via Charybdis model D programmable attenuators; the spectrum level of the noise bands was adjusted by means of manual attenuators (Hatfield 2125). Signal and masker were combined in an adder (Analog Devices 507) before being passed to a sound-attenuating chamber and a final manual attenuator.

Stimuli were delivered using Etymotic Research ER2 insert earphones. Only one ear of each subject was tested (i.e., presentation was monaural). An earmold was made for each subject using silicone impression material. This was drilled out and a 25-mm length of number 16 tubing inserted, as recommended by the manufacturer. This was connected to the acoustic output tube of the earphone, the latter being clipped to a collar or lapel. The manufacturer's data showed this earphone to have a frequency response that was flat within +/− 1 dB between 1 and 14 kHz when measured in a Zwislocki coupler; measurements made on an ear simulator (Bruel and Kjaer type 4157) confirmed this. This measure corresponds rather well to the average eardrum-pressure response in normal ears (Killion, 1984). The response measured in the ear simulator remained reasonably flat for frequencies up to 18 kHz, but variability in the response would be greater in individual ear canals at frequencies above 14 kHz.

Stimuli were monitored with a Hewlett–Packard spectrum analyzer (3582A) and a Gould digital storage oscilloscope (OS 4020).

B. Procedure

Signal thresholds were measured using an adaptive two-alternative forced-choice task. Observation intervals were marked by an amber light, and feedback was provided by means of red and green lights. Each run started with the signal level 10–15 dB above the estimated threshold. The signal level was decreased by 2 dB after two consecutive correct responses, and increased by 2 dB after one incorrect response. A turnaround was defined as a transition from a decreasing to an increasing level (or vice versa), and each run consisted of 16 turnarounds; the mean of the levels at the final 12 turnarounds was taken as the threshold estimate for that run. This procedure estimates the 70.7% point on the psychometric function (Levitt, 1971). Each threshold reported is the mean from at least two runs. Additional runs were obtained if the thresholds for the two runs differed by more than 2 dB, and all thresholds were averaged. Standard deviations of the threshold estimates for a given condition and subject were typically around 2.5 dB; standard errors were usually between 1.5 and 2.0, although occasional higher values occurred.

The absolute thresholds of the subjects were measured at 8, 10, 12, 14, and 16 kHz (except DE at 16 kHz) using the same 2AFC task.

Once the subject's earmold had been inserted and the transducer clipped to their collar, earmold and transducer remained in place for the whole of a 2-h experimental session. During each session, testing was restricted to a single center frequency and noise level, and thresholds were obtained for as many different notch widths as possible. It typically took four to five experimental sessions to gather a complete data set for a given center frequency and level.

C. Subjects

The subjects tested at 8 kHz were three of the authors. One (MS, aged 48) was highly experienced in psychoacoustic tasks; the other two (SH, NW, both aged 21) were given extensive practice before the collection of data. At 10 kHz, two highly experienced subjects (MS, and DE, aged 46) and one young subject (AS, aged 22), who was given extensive practice, took part. All subjects had absolute thresholds better than 10 dB HL at all audiometric frequencies.

D. Analysis

The method of deriving filter shapes from the data is similar to that described by Patterson and Nimmo-Smith (1980), Glasberg *et al.* (1984), and Glasberg and Moore (1986), but with some significant modifications. These modifications are described in detail in Glasberg and Moore

(1990) and in the companion paper (Moore *et al.*, 1990), and we will give only brief descriptions here.

Patterson *et al.* (1982) have described a family of models that can be used to characterize auditory filter shapes and to derive filter shapes from notched-noise data. For data sets where the dynamic range of the thresholds is not too large, and where the masked thresholds are well above the absolute threshold, the roex (p, r) model generally gives a good fit to the data. In this model, each side of the auditory filter is assumed to have the form

$$W(g) = (1 - r)(1 + pg) \exp(-pg) + r, \quad (1)$$

where p is a parameter determining the slope of the filter skirts, r a parameter that flattens the filter at frequencies remote from the center frequency thereby placing a dynamic range limitation on the filter and g is the deviation from the center frequency expressed as a proportion of center frequency. The value of p is allowed to differ for the upper and lower halves of the filter; the upper and lower p values are called p_u and p_l, respectively. The value of r is usually assumed to be the same for the two sides of the filter, so this model has three free parameters.

When the dynamic range of the thresholds is large, and when some of the masked thresholds approach absolute threshold, the roex (p, w, t) model sometimes gives a better fit to the data (Glasberg *et al.*, 1984). In this model, each side of the filter is assumed to have the form

$$W(g) = (1 - w)(1 + pg)\exp(-pg) + w(1 + tg) \exp(-tg), \quad (2)$$

where p is a parameter determining the shape of the main passband of the filter, t is a parameter determining the shallower tail of the filter, and w determines where the tail takes over from the passband. In its most general form, this model has six free parameters, three for each side of the filter. However, it has been found that the shallower side of the filter can be described as a "stretched" version of the sharper side (Patterson and Nimmo-Smith, 1980; Glasberg *et al.*, 1984). This is incorporated in the model by making w have the same value for each side of the filter and setting $t_u/t_l = p_u/p_l$. This reduces the number of free parameters to four.

We fitted our data with both of these models. Although the roex (p, r) model fitted most of the data reasonably well, there were some systematic discrepancies of the data from the fitted values, particularly for the data obtained at high noise levels. The roex (p, w, t) model fitted the data significantly better, and there were no systematic deviations of the data from the fitted values. Hence, the subsequent analysis will concentrate on results obtained with the roex (p, w, t) model.

One problem in analyzing the results is that the absolute threshold varies strongly with frequency at high frequencies. This can obviously have a significant influence on measures of frequency selectivity, and it is not entirely clear how it should be taken into account. One possibility—at least in young normally hearing listeners—is that the variation of absolute threshold with frequency reflects a frequency-dependent attenuation resulting largely from the transfer function of the middle ear. The data reviewed by Dallos (1973), Zwislocki (1965, 1975), and Lynch *et al.* (1982) support this view, at least for frequencies in the range 2–10 kHz. The auditory filter is usually conceived of as resulting from processes occurring after the middle ear, and hence the variation of absolute threshold with frequency could reflect a frequency-dependent attenuation applied to all stimuli before auditory filtering takes place. If this is the case, then the frequency-dependent attenuation should be taken into account in the fitting procedure.

According to the power-spectrum model, the threshold P_s for a signal in noise is

$$P_s = K \int_0^\infty N(f) W(f) df, \quad (3)$$

where K is a constant, $N(f)$ represents the long-term power spectrum of the masker at the input to the filter, and $W(f)$ represents the intensity weighting applied by the auditory filter as a function of frequency (Patterson and Moore, 1986). When the noise spectrum is flat within its passbands, $N(f)$ may be treated as a constant, and the signal threshold can be predicted by integrating the equation for the filter shape over the frequency range covered by the noise. If, however, a frequency-dependent attenuation is applied before auditory filtering, $N(f)$ can no longer be treated as a constant. To predict the signal threshold in this case, it is necessary to calculate the spectra of the stimuli at the input to the filter, and then to evaluate the integral in Eq. (2) by numerical methods. In the present paper, the form of the frequency-dependent attenuation was assumed to correspond to the average absolute threshold curve measured in our three young normally hearing subjects. The curve was extrapolated to a value of 90 dB SPL at 20 kHz.[1] At center frequencies below 8 kHz, the minimum audible pressures at the eardrum as estimated by Killion (1978, Fig. 1) were used; we will refer to this frequency-dependent attenuation as the MAP correction. The absolute thresholds of all subjects for frequencies between 8 and 16 kHz are given in Table I.

It is possible, of course, that part of the variation of absolute threshold with frequency arises from processing subsequent to the filters and/or the inherent properties of the filters themselves. For example, the filters may have an internal noise whose level varies with center frequency. So, for comparison purposes, we also derived auditory filter shapes without allowing for the variation of absolute threshold with frequency.

TABLE I. Absolute thresholds for the individual subjects at frequencies of 8, 10, 12, 14, and 16 kHz, expressed in dB SPL. The table also gives the mean absolute thresholds for the three youngest subjects; these means were used to define the MAP correction (see text for details).

Subject	Frequency, kHz 8	10	12	14	16
MS	18.7	36.5	52.5	63.9	80.3
DE	19.0	27.1	56.5	72.0	···
NW	9.8	29.9	35.4	44.3	60.7
SH	10.8	17.8	28.3	41.1	52.0
AS	9.0	26.3	34.5	47.0	78.4
Mean (NW, SH, AS)	9.9	24.7	32.7	44.1	63.7

One other modification to the fitting procedure should be mentioned. It was assumed that, when the filter shifts in center frequency so as to improve the signal-to-noise ratio, it gets slightly sharper for a downward shift and slightly broader for an upward shift.[2] The form of the variation with center frequency is given in the companion paper (Moore *et al.*, 1990) and is justified in Glasberg and Moore (1990).

II. RESULTS AND DISCUSSION

A. The effect of including the MAP correction

We start by describing the effect of including the MAP correction for a representative set of data, namely, the mean data for a center frequency of 10 kHz and a spectrum level of 50 dB. These data are shown in both panels of Fig. 1. The lines in the figure show the thresholds fitted to the data by the roex (p, w, t) model. In the left panel, no correction for the absolute threshold was applied; in the right panel, the MAP correction was applied. The fit to the data is better with the MAP correction than without; the root-mean-square (rms) deviation of the data from the fitted values is 1.8 dB for the left panel and 1.6 dB for the right panel. At 10 kHz, the fit to the data for the individual subjects was consistently better with the MAP correction than without; the rms deviation of the data from the fitted values had an average value of 2.4 dB without the MAP correction and 1.8 dB with the MAP correction. At 8 kHz there were no consistent differences in goodness of fit with and without the MAP correction.

The fitting procedure also gives an estimate of the value of K, which represents the signal-to-masker ratio required at the output of the filter to achieve threshold. The values of K obtained at 10 kHz with no correction were often rather high compared to values that have been reported at lower center frequencies (e.g., Glasberg and Moore, 1986; Moore, 1987); see Table II for details. The values of K obtained with the MAP correction were generally lower and more in line with values obtained at lower center frequencies. In summary, at 10 kHz, the model gave a better fit to the data using the MAP correction and gave more reasonable values of K. Hence, in the rest of this paper, we will concentrate on the filter shapes derived with the MAP correction. For completeness, the parameters of the filters derived without any correction will also be given.

B. The shapes of the derived filters

Figures 2–4 show the individual data and the mean data for a center frequency of 8 kHz, for noise spectrum levels of 20, 35, and 50 dB, respectively. The curves give the thresholds fitted to the data using the roex (p, w, t) model with the MAP correction. Figure 5 shows the filter shapes derived from the data at 8 kHz, using the MAP correction. Thresholds that were within 2 dB of absolute threshold were excluded from the fitting procedure. Table II gives the parameters of the derived filters both with and without the MAP correction. It also gives the rms deviation of the data from the fitted values for each subject and condition and the value of K in Eq. (3).

At the lowest level used, $N_0 = 20$ dB (solid lines in Fig. 5), the filters have sharply tuned tips, with equivalent rectangular bandwidths (ERBs) of about 7%–10% of the center frequency. The filter skirts flatten off at large values of g, but this probably reflects the approach of masked thresholds to absolute threshold at large notch widths, rather than being an inherent part of the filter characteristics. Subject MS

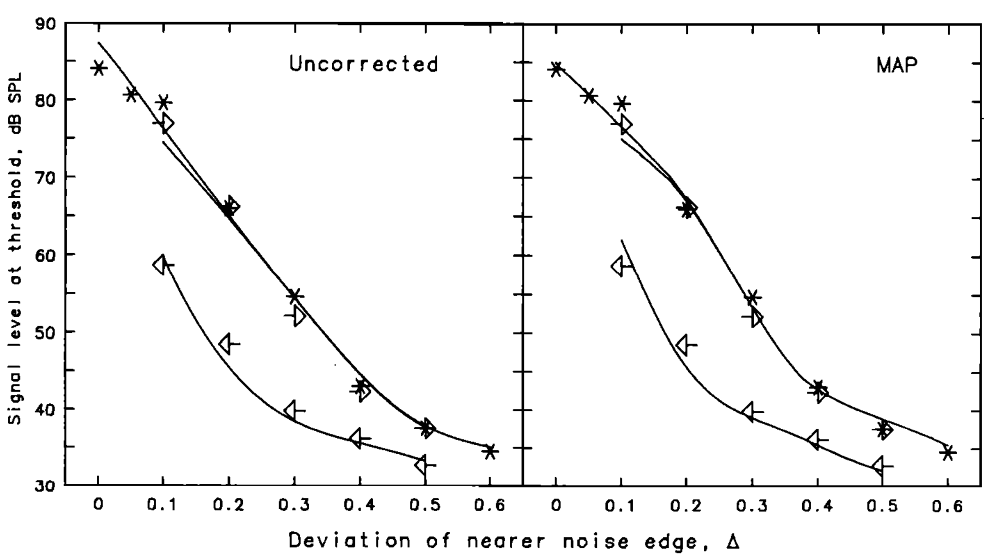

FIG. 1. Mean data of the three subjects for a center frequency of 10 kHz and a noise spectrum level of 50 dB. Asterisks denote conditions where the noise bands were symmetrically placed about the signal frequency; right-pointing arrows indicate that the notch was centered above the signal frequency, and left-pointing arrows that it was below the signal frequency. In the latter asymmetric conditions, Δ refers to whichever edge of the two noise bands was closer to the signal frequency; in such cases, the other band was always 0.2 units farther away. The same data are shown in each panel. The lines are the thresholds fitted to the data by the roex (p, w, t) model. In the left panel, there is no correction for changes in absolute threshold with frequency; in the right panel, the MAP correction, based on the mean absolute thresholds of three young normally hearing subjects, was applied during the fitting procedure (see text).

TABLE II. Summary of the parameters of the auditory filters fitted to the data at 8 and 10 kHz. The table also shows the values of K, representing the signal-to-masker ratio at the output of the filter required for threshold, and the root-mean-square (rms) deviation of the data from the fitted values. The upper number of each pair was obtained using the MAP correction; the number in parentheses was obtained without any correction. For each frequency and level, the first mean is the mean value of the parameters across subjects. The second mean (mean*) gives the parameters of the filters fitted to the mean data.

f_c /level	Subject	ERB	p_l	p_u	t_l	t_u	w	K	rms
8 kHz 20 dB	MS	0.067 (0.061)	72.4 (63.0)	50.5 (67.5)	0.1 (2.1)	0.0 (2.2)	− 27.2 (− 32.6)	− 1.8 (− 2.5)	0.6 (0.6)
	NW	0.099 (0.079)	61.9 (56.1)	30.0 (46.6)	2.6 (2.2)	1.2 (1.9)	− 31.1 (− 38.4)	− 1.8 (− 2.2)	1.2 (0.8)
	SH	0.088 (0.071)	69.5 (63.9)	33.6 (49.9)	2.9 (3.7)	1.4 (2.9)	− 36.4 (− 42.4)	1.3 (0.7)	0.9 (0.9)
	mean	0.085 (0.070)	67.9 (61.0)	38.0 (54.7)	1.9 (2.7)	0.9 (2.3)	− 31.6 (− 37.8)	− 0.8 (− 1.3)	0.9 (0.8)
	mean*	0.091 (0.071)	63.9 (60.4)	33.4 (52.3)	1.6 (4.6)	0.8 (4.0)	− 31.1 (− 35.2)	− 1.4 (− 1.5)	0.5 (0.5)
8 kHz 35 dB	MS	0.073 (0.065)	53.5 (52.5)	56.2 (75.5)	3.7 (2.6)	3.9 (3.7)	− 37.3 (− 46.0)	− 0.9 (− 0.2)	1.9 (1.8)
	NW	0.082 (0.067)	67.4 (63.5)	38.6 (56.4)	4.9 (5.5)	2.8 (4.9)	− 40.6 (− 46.9)	2.2 (2.0)	1.6 (1.6)
	SH	0.084 (0.068)	60.3 (58.8)	39.4 (58.2)	6.0 (7.1)	3.9 (7.0)	− 43.1 (− 48.5)	2.1 (2.5)	1.6 (1.4)
	mean	0.080 (0.067)	60.4 (58.3)	44.7 (63.4)	4.9 (5.1)	3.5 (5.2)	− 40.3 (− 47.1)	1.1 (1.4)	1.7 (1.6)
	mean*	0.088 (0.069)	57.0 (56.4)	39.9 (60.3)	4.7 (4.9)	3.3 (5.2)	− 39.8 (− 46.8)	0.5 (1.0)	1.4 (1.2)
8 kHz 50 dB	MS	0.101 (0.079)	37.8 (42.1)	41.2 (62.9)	6.0 (5.3)	6.5 (7.9)	− 40.2 (− 48.5)	1.4 (3.0)	2.2 (1.9)
	NW	0.142 (0.100)	35.7 (38.6)	23.2 (41.5)	8.9 (10.0)	5.8 (10.7)	− 32.1 (− 36.8)	− 5.0 (− 4.3)	2.2 (2.1)
	SH	0.136 (0.103)	28.9 (32.9)	29.8 (47.3)	8.0 (7.6)	8.2 (10.9)	− 40.3 (− 46.6)	− 1.6 (− 0.6)	2.1 (2.1)
	mean	0.126 (0.094)	34.1 (37.9)	31.4 (50.6)	7.6 (7.6)	6.8 (9.8)	− 37.5 (− 44.0)	− 1.7 (− 0.6)	2.2 (2.0)
	mean*	0.133 (0.096)	32.4 (37.0)	28.0 (47.5)	8.0 (7.9)	6.9 (10.2)	− 35.3 (− 42.3)	2.8 (− 1.3)	1.7 (1.6)
10 kHz 50 dB	MS	0.061 (0.079)	41.5 (29.8)	157.6 (174.0)	6.6 (2.6)	25.2 (15.4)	− 45.8 (− 47.8)	6.7 (9.4)	1.6 (2.0)
	DE	0.068 (0.077)	47.0 (35.3)	77.5 (100.0)	7.0 (3.1)	11.5 (8.7)	− 56.0 (− 58.2)	6.6 (9.2)	2.2 (2.4)
	AS	0.158 (0.125)	29.6 (26.9)	22.1 (39.7)	0.7 (0.8)	0.6 (1.2)	− 68.0 (− 68.9)	0.1 (7.5)	1.7 (2.7)
	mean	0.096 (0.094)	39.4 (30.7)	85.7 (104.6)	4.8 (2.2)	12.4 (8.4)	− 56.6 (− 58.3)	4.5 (8.7)	1.8 (2.4)
	mean*	0.102 (0.099)	38.3 (28.5)	40.3 (69.5)	9.4 (3.9)	9.9 (9.5)	− 42.4 (− 51.5)	3.1 (7.4)	1.6 (1.8)

had a higher absolute threshold at 8 kHz than the other two subjects, and his filter flattens off more quickly than the filters for NW or SH. The filters at this noise level have steeper lower skirts than upper skirts.

For $N_0 = 35$ dB (dotted lines), the shapes of the filters around the tips do not change very much, but the filters have wider dynamic ranges. This reflects the fact that the masked thresholds only approached absolute threshold at the largest notch widths used.

At the highest level, $N_0 = 50$ dB (dashed lines), the filters become broader, particularly on the low-frequency side. This trend is consistent with data at lower center frequencies showing that the auditory filter becomes broader with increasing sound level (Weber, 1977). On the other hand, although earlier work at lower center frequencies suggests that the high-frequency side of the filter changes only slightly with level, or even increases in sharpness with increasing level (Lutfi and Patterson, 1984, Moore and Glas-

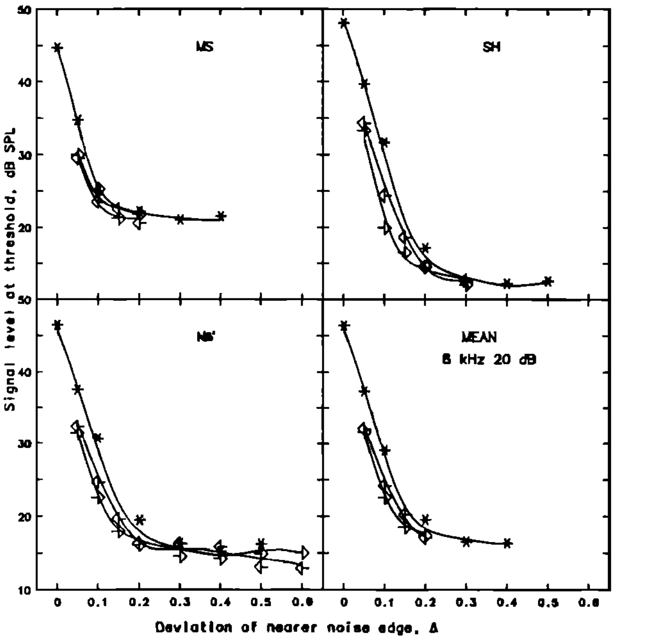

FIG. 2. Individual and mean data for a center frequency of 8 kHz and a spectrum level of 20 dB. The results are plotted in the same way as for Fig. 1. The lines are thresholds fitted to the data by the roex (p, w, t) model using the MAP correction.

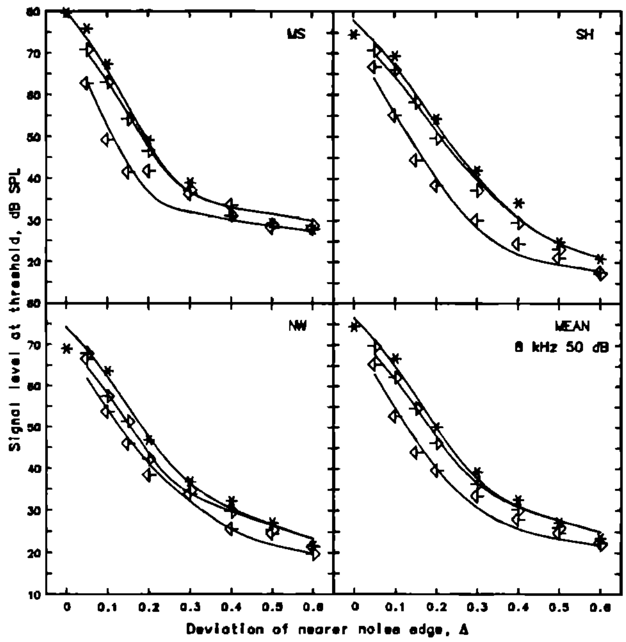

FIG. 4. As in Fig. 2 but for a spectrum level of 50 dB.

berg, 1987), the present results show a slight broadening on the high-frequency side at the highest level. The present results also differ from earlier work in that the filters tend to be less sharp on the high-frequency side than the low-frequency side, particularly for subject NW.

Figure 6 shows the individual and the mean data at $f_c = 10$ kHz. Figure 7 shows the corresponding filter shapes. Individual variations are greater than found at 8 kHz. Two subjects (MS and DE) show filters with very steep high-frequency skirts and narrow bandwidths of about 6%–7%, while the third (AS) has a filter with a slight asymmetry in the opposite direction and a greater bandwidth (15.8%). To check on the possibility that these differences result from the way in which individual subjects' absolute thresholds vary at high frequencies, filter shapes were also fitted using "corrections" based on the individual subjects' absolute thresholds rather than the MAP correction. The filters derived in this way differed only slightly from those derived using the MAP correction. Thus the individual differences in filter shape do not appear to result from differences in absolute threshold,

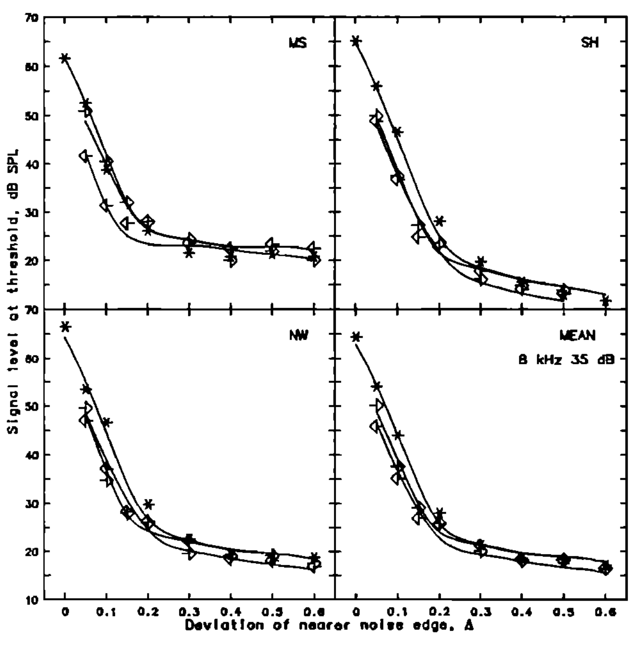

FIG. 3. As in Fig. 2 but for a spectrum level of 35 dB.

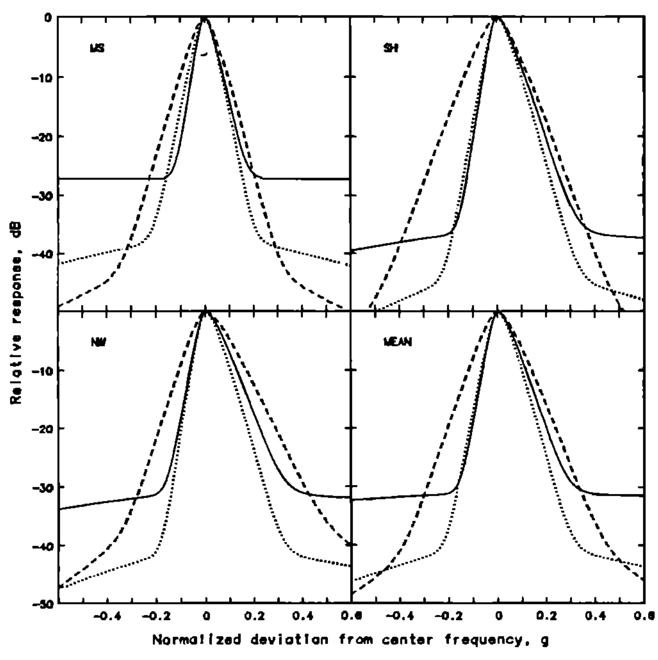

FIG. 5. Filter shapes derived from the individual data and the mean data for a center frequency of 8 kHz. Filter shapes are shown for noise spectrum levels of 20 dB (solid line), 35 dB (dotted line), and 50 dB (dashed line).

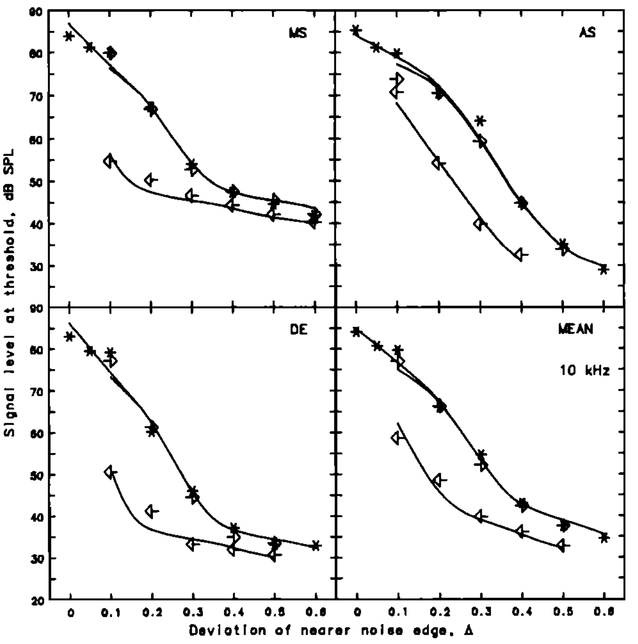

FIG. 6. As in Fig. 2 but for a center frequency of 10 kHz and a spectrum level of 50 dB.

but rather reflect genuine differences in the filters themselves. It is interesting that subject AS, who shows the broadest filter at 10 kHz, also has a wider filter than average at 1 kHz.

C. Comparison with earlier bandwidth estimates

The ERBs of the filters (637 Hz at 8 kHz, middle level, and 957 Hz at 10 kHz) are somewhat smaller than might have been expected from earlier work. At medium center frequencies, the ERB of the auditory filter is typically about 15% smaller than classical critical bandwidth (CB) values (Moore and Glasberg, 1983). Scharf (1970) gives values for the CB of 1800 Hz at 8.5 kHz and 2500 Hz at 10.5 kHz, which would lead us to expect ERBs of 1530 and 2125 Hz, respectively. The CBs given by the equation of Zwicker and Terhardt (1980) for center frequencies of 8 and 10 kHz are 1706 and 2305 Hz, which would lead to ERBs of 1450 and 1959 Hz, respectively. Zwicker *et al.* (1957) and Greenwood (1961) have suggested that each CB may correspond to a constant distance along the basilar membrane. Similarly, the ERB may correspond to a constant distance along the basilar membrane (Moore, 1986). The closest prediction of our ERBs is given by a modification of Greenwood's (1961) equation suggested by Moore and Glasberg (1986, p. 253). That equation predicts ERBs of 962 and 1198 Hz at 8 and 10 kHz, respectively. When expressed as a proportion of center frequency, our ERBs are comparable to, but slightly smaller than, values found at medium center frequencies. It should be noted that the ERBs obtained without any correction (shown in parentheses in Table II) were smaller than those obtained with the MAP correction. The discrepancy with earlier estimates would therefore be even greater if no correction were applied.

D. Implications of individual differences

In the Introduction, we mentioned the possibility that measurement of the auditory filter shape at high center frequencies might provide a sensitive measure of the condition of the cochlea and might be useful for detecting the onset of damage. While this may still turn out to be the case, the rather large individual differences among normal subjects found in the present study mean that a large deterioration would have to be found for a classification as "abnormal" to be made.

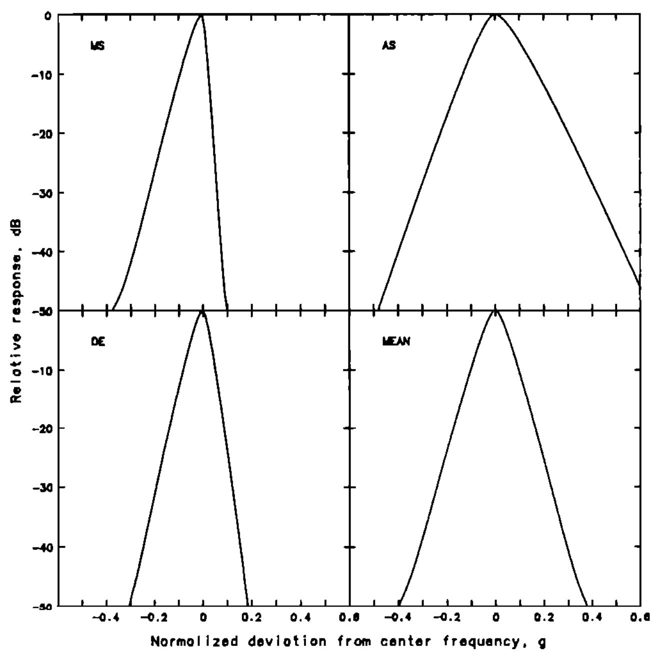

FIG. 7. Filter shapes derived from the individual data and the mean data for a center frequency of 10 kHz.

III. CONCLUSIONS

(1) The ERB of the auditory filter varies somewhat across subjects and tends to be smaller than expected from "classical" CB values at 8 and 10 kHz, but broadly in line with previous measures of the ERB obtained at medium center frequencies (when expressed as a proportion of center frequency). Mean ERBs at 8 kHz, expressed as a proportion of center frequency, were 0.085, 0.080, and 0.126 at noise spectrum levels of 20, 35, and 50 dB, respectively. The mean ERB at 10 kHz, for a spectrum level of 50 dB, was 0.096. These ERBs were obtained using a correction in the fitting procedure to allow for the variation of absolute threshold with frequency. The derived ERBs were smaller without this correction.

(2) At 8 kHz, the lower skirt of the filter tended to be steeper than the upper skirt at the two lower noise levels. The filter was more symmetric at the highest noise level. At 10 kHz, two subjects had sharp filters with very steep high-frequency skirts, and the other had a broader filter with a slight asymmetry in the opposite direction.

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