

Short-term temporal integration: Evidence for the influence of peripheral compression

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Thresholds for a 6.5-kHz sinusoidal signal, temporally centered in a 400-ms broadband-noise masker, were measured as a function of signal duration for normally hearing listeners and listeners with cochlear hearing loss over a range of masker levels. For the normally hearing listeners, the slope of the function relating signal threshold to signal duration (integration function) was steeper at medium masker levels than at low or high levels by a factor of nearly 2, for signal durations between 2 and 10 ms, while no significant effect of level was found for signal durations of 20 ms and more. No effect of stimulus level was found for the hearing-impaired listeners at any signal duration. For signal durations greater than 10 ms, consistent with many previous studies, the slope of the integration function was shallower for the hearing-impaired listeners than for the normally hearing listeners. However, for shorter durations, there was no significant difference in slope between the results from the hearing-impaired listeners and those from the normally hearing listeners in the high- and low-level masker conditions. A model incorporating a compressive nonlinearity, representing the effect of basilar-membrane (BM) compression, and a short-term temporal integrator, postulated to be a more central process, can account well for changes in the short-term integration function with level, if it is assumed that the compression is greater at medium levels than at low or high levels by a factor of about 4. This is in reasonable agreement with physiological measurements of BM compression, and with previous psychophysical estimates. © 1997 Acoustical Society of America. [S0001-4966(97)05606-3]

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INTRODUCTION

This study examines temporal integration, or how the threshold for detecting a signal depends on signal duration, and investigates the extent to which the results may be influenced by peripheral compression in the auditory system. Although temporal integration has been the subject of intense study since the 1940s, there is still no consensus as to the underlying mechanisms involved. The fact that thresholds for sinusoidal signals in quiet or in background noise decrease by approximately 3 dB per doubling in duration, between about 10 and 200 ms, led to the hypothesis that stimulus intensity (above a certain minimum intensity) is fully integrated (Hughes, 1946; Garner and Miller, 1947). A different formulation, in terms analogous to a simple electrical RC circuit, was proposed independently by Feldtkeller and Oetinger (1956) and Plomp and Bouman (1959). Both types of model lead to very similar predictions (Plomp and Bouman, 1959), namely a 3-dB decrease in threshold per doubling of duration up to a certain duration (between about 100 and 300 ms) followed by asymptotic behavior. More sophisticated models of temporal summation, taking neural activity into account, have been proposed by Zwislocki (1960, 1969).

As previously pointed out, most recently by Viemeister and Wakefield (1991), the fact that the auditory system can

act like an energy detector over a limited duration does not necessarily imply true integration of an intensity-like quantity. Results can often be equally well described in terms of an increase in signal duration leading to an increase in the statistical probability of detection. In a recent exposition of such a theory, known as the "multiple-looks" hypothesis, Viemeister and Wakefield (1991) presented results, involving the detection of two separate tone bursts, which cannot be accounted for by a long-term temporal integrator. Another possible approach has recently been described by Dau *et al.* (1996a,b). In their model an analysis window (template) is used, which is matched to the time pattern of the signal. With an extension of the original model, Dau and colleagues have successfully modeled the data of Viemeister and Wakefield (1991) (Dau *et al.*, 1997). Whether or not long-term temporal integration is due to a long time constant, multiple looks, or an adjustable template, it is clear that there is no "hard-wired" long time constant in the auditory system that cannot be bypassed. If this were the case, then thresholds for a very brief signal, temporally centered in a masker, would increase with masker duration for durations beyond 100 ms. In a series of experiments, Penner and her colleagues showed that in fact the threshold of a brief signal pulse, temporally centered in a broadband-noise masker, increased for masker durations up to between 10 and 20 ms, and then remained roughly constant or decreased (Penner *et al.*, 1972; Penner and Cudahy, 1973). The masker duration at which thresholds

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ceased to increase was termed the critical masking interval. These results were interpreted as reflecting a short-term integrator with an effective time constant of around 10 ms. This is broadly consistent with many other measures of temporal resolution, such as gap and decrement detection and the decay of forward masking (e.g., Buus and Florentine, 1985; Oxenham and Moore, 1994; Peters *et al.*, 1995).

We therefore adopt the position that in cases where detection is achieved by an overall change in level, *there is a short-term temporal integrator which cannot be bypassed, and which determines thresholds for signal durations up to about 10 ms.* In this paper, we refer to this as *short-term temporal integration*. For signal durations greater than this, thresholds may be determined by a multiple-looks strategy, longer time constants, an adjustable template, or a combination of these.

Even if we accept the proposition of true temporal integration for short signal durations, it is still not necessary to assume that a quantity proportional to signal intensity is integrated. Penner (1978) has shown, for instance, that using a power-law nonlinearity prior to integration, complementary pairs of nonlinearities and temporal-weighting functions can be constructed which all produce the same time-intensity trade function. Thus the slope of the temporal-integration function is not sufficient in itself to determine what quantity is integrated. From this, Penner (1978) was able to show that, for a given temporal window, *a change in the nonlinearity leads to a change in the slope of the integration function.* Specifically, the more compressive the nonlinearity, the steeper (more negative) the slope of the function [relating signal level (dB) to log (duration)]. An intuitive explanation of this relationship is given in the Appendix.

In a number of previous studies, the compression used in this type of model has been linked to peripheral auditory compression (e.g., Oxenham and Moore, 1995; Moore *et al.*, 1996; Oxenham and Plack, 1997). This approach has been stimulated by physiological measurements of basilar-membrane (BM) motion (Rhode, 1971; Sellick *et al.*, 1982; Ruggero, 1992; Ruggero *et al.*, 1995). Essentially, the response of the BM to sound at the characteristic frequency (CF) of the place of measurement appears to be highly compressive, especially for levels between about 40 and 80 dB SPL. Damage to the cochlea reduces or eliminates compression.

In terms of a short-term integration model, the nonlinearity, representing peripheral (BM) compression, is followed by a linear integrator, representing a somewhat higher processing stage. Relating the model's compression to BM nonlinearity, and always assuming an invariant temporal-window shape, produces the following two predictions. First, for normally hearing listeners, greater mid-level BM compression should produce a steeper slope in the integration function at medium levels than at low or high levels. Second, to the extent that the BM response is more linear in listeners with cochlear hearing loss, the slope of the integration function should be shallower for hearing-impaired listeners than for normally hearing listeners.

This second prediction is extremely well documented for unmasked (absolute) thresholds (e.g., Miskolczy-Fodor,

1953; Elliott, 1963; Wright, 1968; Pedersen and Elberling, 1973; Chung, 1981; Florentine *et al.*, 1988): There is indeed in general a reduction in the slope of the temporal-integration function for listeners with sensorineural hearing loss. This is not due to the generally higher levels at which the stimuli are presented to the hearing-impaired listeners (Gengel, 1972; Florentine *et al.*, 1988) and cannot be accounted for by the possible detection of spectral splatter (Florentine *et al.*, 1988; Carlyon *et al.*, 1990). It has been previously suggested that this reduction in slope over the whole range of signal durations is due to reduced peripheral compression (Moore, 1991, 1995), but to our knowledge no quantitative test of this hypothesis has been attempted.

The first prediction of the model that, for normally hearing listeners, short-term temporal integration should have a steeper slope at medium signal levels than at higher or lower levels, has much less experimental support. A test of the hypothesis would need to fulfill the following conditions. First, the number of points measured for durations of 10 ms or less must be sufficient to give a reasonable estimate of the slope of the integration function. Second, in order to avoid problems of detection by combining information across different frequency channels, the bandwidth of the signal, even at the shortest durations, must fall approximately within one critical band. This essentially limits the choice of signal frequencies to those above about 4 kHz. Third, the masker level must be chosen so that the signal level at durations of 10 ms or less lies within the level region thought to be most compressive, namely between about 50 and 70 dB SPL (Oxenham and Plack, 1997). For comparison with regions of more linear processing, masker levels must lie well below or above that level. These restrictions severely limit comparisons within the available literature. Both Florentine *et al.* (1988) and Gengel (1972) have compared the slope of integration for a sinusoidal signal in the presence of a masking noise with that in quiet. Interestingly, they were looking for evidence that the slope decreased in the presence of a masking noise, while we expect the reverse. Unfortunately, Gengel (1972) did not measure thresholds for durations less than 10 ms, even at 4 kHz. Florentine *et al.* (1988) have a number of data points which could come into consideration in terms of signal frequency and duration. However, only one of the noise levels they used (simulation of hearing-impaired listener RT) produced thresholds below 70 dB SPL for signals of 10 ms or less at 4 kHz. In Fig. 5 of their paper, for durations less than 10 ms, there is a tendency for the slope of the masked thresholds to be steeper than that of the thresholds in quiet. However, the difference is small, is based on only three durations, and is therefore far from conclusive.

The most positive evidence for a change in the integration function with level comes from a study by Stephens (1973). In that study, performance was measured in terms of percent correct for sinusoidal signals at a constant signal-to-noise ratio (in terms of overall energy), for a number of signal durations and noise-masker levels. It was found that performance at the shortest duration (2 to 3 cycles of the signal) was strongly dependent on masker levels, reaching a minimum at medium masker levels. For signal durations of 20 ms and longer, performance was independent of masker

level. This was true for signal frequencies of both 1 and 4 kHz. The results of Stephens (1973) imply that the slope of the integration function may indeed be steeper at medium levels. However, as he measured performance for a fixed signal level, it is not possible to derive the slope of the integration function from his data. Second, as his signals were switched on and off without ramps, it is not clear to what extent the integration of signal energy across frequency played a role in his experiments.

Finally, recent experiments on the loudness of sinusoids as a function of duration have shown that the difference in loudness between a short and a long tone is greatest at medium levels (Florentine *et al.*, 1996). The authors note that the results are consistent with greater BM compression at medium levels. However, quantitative analysis of on-frequency compression based on loudness judgments is difficult, due to the fact that loudness is almost certainly influenced by off-frequency excitation (e.g., Zwicker, 1960).

In summary, while there are indications that the slope of the short-term temporal-integration function may be steeper at medium levels, the available data are neither conclusive nor sufficient for quantitative analysis. In the experiment described below, thresholds for a 6.5-kHz sinusoidal signal in a broadband-noise masker were measured as a function of signal duration in both normally hearing and hearing-impaired listeners over a range of masker levels. Based on the assumption that the shape of the temporal window remains invariant with level, and that it is not altered by cochlear pathology, the basic prediction for normally hearing listeners is that the slope of the integration function at medium levels will be steeper than at low and high levels. Assuming a linear BM input-output function, the slope of the integration function should remain constant with level for the hearing-impaired listeners. The difference between this slope and that of the normally hearing listeners at low and high levels should provide an indication of the amount of compression present in normal hearing at the lowest and highest levels, always assuming the same or similar temporal windows across the two groups.

I. EXPERIMENT 1. TEMPORAL INTEGRATION AT 6.5 kHz

A. Stimuli

The masker was a bandpass-filtered Gaussian noise (Hewlett-Packard 3722A) with cutoff frequencies of 2 and 12 kHz (Kemo VBF/8/03 filter, 48-dB/oct slope). The signal was a 6.5-kHz sinusoid (Farnell DSG1). Both masker and signal were gated with 1-ms raised-cosine ramps, and the signal was always temporally centered within the 400-ms masker. Thresholds were measured for signals with half-amplitude durations ranging from 2 to 200 ms (1–199 ms steady state). For the normally hearing listeners, thresholds were measured at masker spectrum levels of -10 , 20 , and 50 dB (*re*: $20 \mu\text{Pa}$).¹ For the hearing-impaired listeners, masker spectrum levels of 30 , 40 , and 50 dB were tested. These levels were chosen to span the dynamic range of the listeners, such that thresholds for the longest duration signal (200 ms) and the lowest masker spectrum level (-10 dB for the

normally hearing and 30 dB for the hearing-impaired listeners) were about 5 dB above individual thresholds in quiet for all listeners.

For the normally hearing listeners, stimulus timing was controlled by a Texas Instruments 990/4 computer system, and the signal level was varied using a Charybdis model D programmable attenuator. Two pairs of analog multipliers (AD 534L) in series were used as gates for the masker and signal, giving an on-off ratio exceeding 100 dB. For the hearing-impaired listeners, who were tested at a later time, stimuli were controlled using a Tucker-Davies Technologies (TDT) system with a PC. The masker and signal were gated and attenuated, using two switches (TDT SW2) and two programmable attenuators (TDT PA4), before being added (TDT SM3) and passed through a headphone buffer (TDT HB6). For both groups, the stimuli were then passed through a final manual attenuator (Hatfield 2125) to one earphone of a Sennheiser HD414 headset. For the normally hearing listeners, the stimuli were presented to the left ear. For the hearing-impaired listeners, the stimuli were presented to the ear with the lower absolute threshold at 6.5 kHz. For listeners VT and DT, this was the right ear and for listeners AW and MG this was the left ear.

A trial consisted of two observation intervals, marked by lights, separated by a silent interval of 500 ms. The 400 -ms masker burst occurred in both intervals and the signal was presented randomly in either the first or the second interval.

B. Procedure

Thresholds were determined using a two-alternative forced-choice method with a three-down one-up adaptive procedure that estimates the 79.4% correct point on the psychometric function (Levitt, 1971). The initial step size was 5 dB, which was reduced to 2 dB after the first four reversals. A run was terminated after a total of 12 reversals and the threshold was defined as the mean of the levels at the last 8 reversals. Each data point reported here is the mean of three such threshold estimates. Listeners were tested individually in a double-walled sound-attenuating chamber.

C. Subjects

Four normally hearing listeners and four listeners with cochlear hearing loss participated as subjects. Two of the normally hearing listeners were authors AO and DV, one (MS) was a member of the laboratory who volunteered for the experiment, and the other (ST) was paid an hourly wage for her participation. Audiometric thresholds for all four listeners were 15 dB HL or less for octave frequencies between 250 and 8000 Hz. The ages of the normally hearing listeners ranged from 25 to 34 years. All normally hearing listeners had extensive experience in psychoacoustic tasks and were given at least 1 -h practice before data were collected.

The four hearing-impaired listeners were selected on the basis of having a sensorineural hearing loss of between 40 and 60 dB at the test frequency (6.5 kHz). All had air-bone gaps of less than 10 dB and showed normal tympanometry, indicating no conductive element. There was no sign of tone decay for any of the four listeners (tone decay is often a

TABLE I. Summary characteristics for the hearing-impaired listeners, showing their ages, genders, diagnoses, and the audiometric thresholds for the test ears, given in dB HL.

Listener	Age	Sex	Frequency (Hz)							Diagnosis
			250	500	1000	2000	4000	6000	8000	
AW	81	M	5	0	5	15	45	50	60	presbycusis
VT	61	M	20	10	5	10	50	40	50	noise induced
MG	71	F	5	10	20	50	45	55	50	presbycusis
DT	72	M	20	10	10	45	65	60	60	noise induced

symptom of retrocochlear loss) and all showed recruitment, as indicated by a smaller-than-normal range between threshold and the highest comfortable level, which is a characteristic of cochlear hearing loss. Speech discrimination was not measured.²

One listener (AW) had extensive previous experience in psychoacoustic tasks. The other three listeners were given at least 4-h practice before data were collected. Audiometric thresholds for the ears that were tested, together with each listener's diagnosis, gender, and age, are given in Table I.

D. Results

Data from the individual listeners are plotted in Fig. 1. The left and right columns show data from the normally

hearing and hearing-impaired listeners, respectively. Each panel represents a different masker spectrum level, as shown in the insets. The solid curves denote mean thresholds across listeners for each condition. The data for normally hearing listener MS (diamond symbols) in the 50-dB condition have been shifted upwards by 5 dB, for ease of comparison, and are treated in the analyses below as if they had been measured in the presence of a 50-dB masker (see footnote 1).

There are some individual differences in the data, in terms of both overall sensitivity and the slope of the integration function. For instance, hearing-impaired listener MG (right panels; circles) is generally less sensitive than the other three listeners, especially at the 40- and 50-dB masker spectrum levels, and normally hearing listener ST (left panels; squares) exhibits a shallower slope of integration than the other listeners, especially in the -10-dB condition. The shallower slope of ST does not seem to be due to an elevated absolute threshold in quiet: while ST's threshold in quiet for the 200-ms signal (14 dB SPL) was higher than that of AO (6.8 dB SPL), it was lower than that of MS (16.2 dB SPL).

Initially, single-line linear regression analyses, in terms of signal level (dB SPL) as a function of $10 \log [\text{duration (ms)}]$, were performed across all signal durations. Resulting slopes for the individual and group mean data in the different conditions are given in Table II. For the individual slopes, the three estimates for each data point shown in Fig. 1 were used. For the group mean slopes, the individual mean thresholds shown in Fig. 1 were used and converted to deviations from the mean for that condition and listener, thus compensating for differences in overall sensitivity across listeners.

In Table II it can be seen that for all the normally hearing listeners the slope of the function for the 20-dB condition is steeper than for the -10- and 50-dB conditions. The effect of masker level is not so pronounced for the hearing-impaired listeners. A within-groups comparison of the slopes for the mean data confirmed this impression: There was no significant effect of masker level for the hearing-impaired group [$F(2,93) = 1.08, p > 0.3$], while for the normally hearing group, the effect of masker level was highly significant [$F(2,93) = 21.46, p \ll 0.0001$], reflecting the steeper overall slope of the 20-dB condition. Although the slopes from the hearing-impaired group are generally shallower than those from the normally hearing group, the difference is not as great as that sometimes reported in the literature, where a difference of a factor of 2 is not uncommon for listeners with a hearing loss of 40 dB or more (e.g., Pedersen and Elberling, 1973).

Visual inspection of the data in Fig. 1 raises some doubt

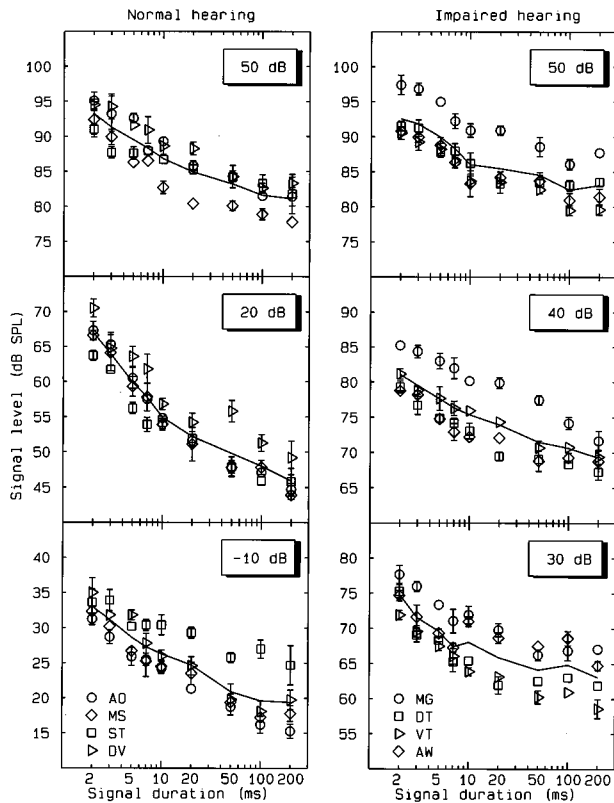


FIG. 1. Individual data from experiment 1. Signal level at threshold is plotted against signal duration on a log scale. The left and right columns represent data from the normally hearing and hearing-impaired listeners, respectively. The masker spectrum level in each condition is given in the insets. Error bars represent ± 1 standard deviation, and are omitted if smaller than the height of the symbol. The solid curves show the mean thresholds of the listeners within each condition.

TABLE II. Slopes from a single-line linear regression analysis [signal level (dB) against 10 log (duration)] of the data from Fig. 1. Masker spectrum level is in dB SPL. All slopes are negative.

Group	Masker level	Listener				Mean data
		AO	ST	DV	MS	
Normal	-10 dB	0.79	0.44	0.84	0.74	0.70
	20 dB	1.14	0.89	0.95	1.09	0.92
	50 dB	0.71	0.39	0.64	0.70	0.61
		MG	DT	VT	AW	
Impaired	30 dB	0.55	0.53	0.62	0.33	0.51
	40 dB	0.66	0.58	0.57	0.51	0.58
	50 dB	0.55	0.44	0.56	0.48	0.51

as to the appropriateness of a single-line regression. Consistent with some previous data (Green *et al.*, 1957; Stephens, 1973; Florentine *et al.*, 1988), the integration function seems to be steeper at shorter durations than at longer ones. This is especially apparent in the 20-dB condition for the normally hearing group, but is also visible, for instance, in the 30- and 50-dB conditions for the hearing-impaired group. The steepening can probably not be accounted for by the spread of signal energy outside the auditory filter centered on the signal frequency, as the 3-dB bandwidth of the shortest signal (ca. 420 Hz) is less than the estimated equivalent rectangular bandwidth of the auditory filter at 6.5 kHz (ca. 725 Hz; see Glasberg and Moore, 1990). The impression of two separate regions was tested by comparing the one-line fit with that using two lines. In the latter procedure, the data in each condition were divided by duration into two subconditions. A linear regression was carried out independently on each subcondition. In order to determine a best-fitting dividing point, all the data were pooled (across condition and group) and pairs of lines were fitted to all contiguous combinations of signal durations. The best fit was achieved using durations of 2–10 ms and 20–200 ms for the two lines. This corresponds well with our hypothesized division between short-term and long-term temporal integration. However, it cannot be taken as providing strong support for this idea, since setting the division point at the adjacent shorter or longer durations only marginally worsened the goodness of fit. Nevertheless, for all further analysis, the division between durations less than or equal to 10 ms and those greater than 10 ms is maintained.

A series of *F*-tests showed that the improvement in fit using two lines was significant for the mean data in every condition [$F(2,29) > 3.49$, $p < 0.05$]. The resulting pairs of slopes for each listener and condition are given in Table III, together with the slopes for the mean data. Consider first the hearing-impaired group. In all but one case (MG, 40-dB condition), the slope for durations between 2 and 10 ms is steeper than for durations between 20 and 200 ms. Also, there seems to be no consistent effect of masker level. The mean data from the hearing-impaired listeners show no significant effect of masker level for either the short durations [$F(2,45) = 0.95$, $p > 0.3$] or the long durations [$F(2,33) = 1.96$, $p > 0.1$]. In contrast, for the normally hearing group, there is a strong effect of masker level at the short signal durations for all four individual listeners ($p \leq 0.01$), and for the mean data [$F(2,45) = 19.28$, $p \leq 0.0001$]. As can be seen from Table III, this is primarily due to the steeper slope of the 20-dB condition. Interestingly, however, differences in slope for the mean data at the longer durations are not significant [$F(2,33) = 2.17$, $p > 0.1$]. This is also true for the individual data for three of the four normally hearing listeners ($p > 0.1$); the marginally significant effect for the exception (listener MS; $0.01 < p < 0.05$) is due to the shallower slope of the 50-dB condition at the longer durations. Finally, an across-group comparison of the mean data revealed *no* significant difference in the shorter duration slopes between the hearing-impaired group, pooled across level, and the highest and lowest levels of the normally hearing group [$F(1,78) = 0.03$, $p > 0.5$]. The effect of group for the longer durations, however, was significant [$F(1,70) = 13.75$, p

TABLE III. Slopes from a two-line linear regression analysis [signal level (dB) against 10 log (duration)] of the data from Fig. 1. Masker spectrum level is in dB SPL. The first and second numbers denote the slopes for durations 2–10 ms and 20–200 ms, respectively. All slopes are negative, unless otherwise indicated.

Group	Masker level	Listener				Mean data
		AO	ST	DV	MS	
Normal	-10 dB	0.99, 0.63	0.57, 0.40	1.22, 0.52	1.18, 0.60	0.99, 0.55
	20 dB	1.86, 0.65	1.56, 0.63	1.70, 0.56	1.79, 0.65	1.72, 0.63
	50 dB	0.95, 0.48	0.48, 0.33	0.86, 0.52	1.30, 0.26	0.89, 0.40
		MG	DT	VT	AW	
Impaired	30 dB	0.93, 0.25	1.34, 0.02	1.11, 0.40	0.70, 0.32	1.02, 0.24
	40 dB	0.70, 0.84	0.85, 0.22	0.74, 0.46	1.04, 0.30	0.84, 0.46
	50 dB	0.99, 0.38	0.80, +0.01	0.90, 0.44	1.03, 0.33	0.93, 0.28

<0.001], indicating that the expected shallower slope for the hearing-impaired listeners is observed at the longer durations.

E. Discussion

The results confirm the main prediction made in the Introduction: For all four normally hearing listeners, the slope of the short-term integration function is steeper at the medium masker level than at the higher and lower levels. For the mean data, the slope is steeper by a factor of nearly 2. This is consistent with the idea that BM compression is greatest at medium sound levels. The lack of an effect of level for the hearing-impaired listeners is consistent with the idea that cochlear impairment leads to a linear BM response at all levels.

Surprisingly, the slopes of the short-term integration functions for hearing-impaired listeners at all three masker levels are very similar to those for the normally hearing listeners at the highest and lowest masker levels. If the slope of the short-term integration function is determined by peripheral compression, this implies that the BM response of the normally hearing listeners is approximately linear at very high and very low levels. This is consistent with some physiological data (e.g., Sellick *et al.*, 1982; Johnstone *et al.*, 1986; Ruggero and Rich, 1991) and is also consistent with the psychophysical results of Oxenham and Plack (1997). An alternative possibility is that the hearing-impaired listeners have some residual BM compression, which remains constant with level and is of the same order as the BM compression of normally hearing listeners at low and high levels. This seems unlikely, however, as the hearing losses of three of the four listeners between 6 and 8 kHz were between 50 and 60 dB; this is similar to the losses exhibited by the listeners in previous studies, where no residual compression was observed (Oxenham and Moore, 1995; Oxenham and Plack, 1997).

Equal integration slopes for normally hearing and hearing-impaired listeners at short durations have not been reported before. A review was therefore made of the available literature on temporal integration in hearing-impaired listeners at high frequencies (≥ 4 kHz) and short durations. The two most comparable studies are by Pedersen and Elberling (1973) and Florentine *et al.* (1988). Pedersen and Elberling (1973) measured temporal integration for durations between about 3.5 and 1000 ms at frequencies of, among others, 4 and 8 kHz. They found slopes, fitting all durations up to 200 ms, to be significantly shallower for their group of hearing-impaired listeners. Also, in the data of two sample subjects, it is clear that the integration function remains shallower even at the shortest durations. Furthermore, their use of relatively long onset and offset ramps makes it unlikely that the detection of splatter reduced the slope of the functions. The hearing-impaired listeners studied by Florentine *et al.* (1988) at 4 kHz show varied results for short durations. Defining the amount of temporal integration as the difference in threshold between a 2-ms and a 16-ms signal, four of the six hearing-impaired listeners show normal or near-normal integration, while the remaining two, listeners DP and PG, show reduced temporal integration. All the listeners in that

study had audiometric thresholds between 50 and 70 dB HL at 4 kHz, and there seems to be no correlation between amount of hearing loss at 4 kHz and the measured amount of short-term temporal integration.

Thus one previous study indicates that the reduction in temporal integration continues to very short signal durations (Pedersen and Elberling, 1973), while the majority of listeners in the other study produced slopes very similar to those of normally hearing listeners at short durations (Florentine *et al.*, 1988). One difference between both these studies and the present one is that our signals were presented in a gated noise, while in the other two studies signals were presented in quiet. It is not clear whether this difference could have affected the slope of the function.

Returning to the results from the normally hearing listeners, the change in the slope of the integration function with level implies that the signal-to-masker ratio changes with level for at least some signal durations. This has also been shown recently by von Klitzing and Kohlrausch (1994) for one listener. Using a 5-kHz signal with a total duration of 2 ms, they found that even when the signal was temporally centered in, or at the end of, a 300-ms noise masker, signal-to-masker ratios changed nonmonotonically with level by as much as 5 dB, reaching a maximum for an overall masker level of 60 dB SPL (20-dB spectrum level). In order to gain a better impression of how the integration function changes with level, thresholds were measured for signal durations of 2, 10, and 200 ms over a larger number of masker levels than were tested in experiment 1.

II. EXPERIMENT 2. CRITICAL RATIO AS A FUNCTION OF MASKER LEVEL: EFFECTS OF SIGNAL DURATION

A. Method

Thresholds were measured for half-amplitude signal durations of 2, 10, and 200 ms. For the normally hearing listeners, masker levels of 0-, 10-, 30-, and 40-dB spectrum level were tested. For the hearing-impaired listeners, levels of 35-, 45-, and 55-dB spectrum level were used. The stimuli, procedure, and listeners were all the same as those described in experiment 1.³

B. Results and discussion

The pattern of results was similar across the listeners in each group. For this reason, only the mean data are presented. The upper panel of Fig. 2 shows the data from the normally hearing listeners. Thresholds at -10-, 20-, and 50-dB masker levels are taken from experiment 1. Thresholds are plotted in terms of the ratio of the signal level to the noise spectrum level. Thus Weber's law would predict three parallel horizontal lines for the three conditions in the figure. However, consistent with the data of von Klitzing and Kohlrausch (1994), the signal-to-noise ratio (S/N_0 or SNR) for the shortest signal (open circles) increases at medium levels, reaching a maximum at 20- and 30-dB spectrum level. The maximum difference between levels is just over 4 dB, in good agreement with the 5 dB found by von Klitzing and Kohlrausch (1994). For the 10-ms signal (asterisks), the

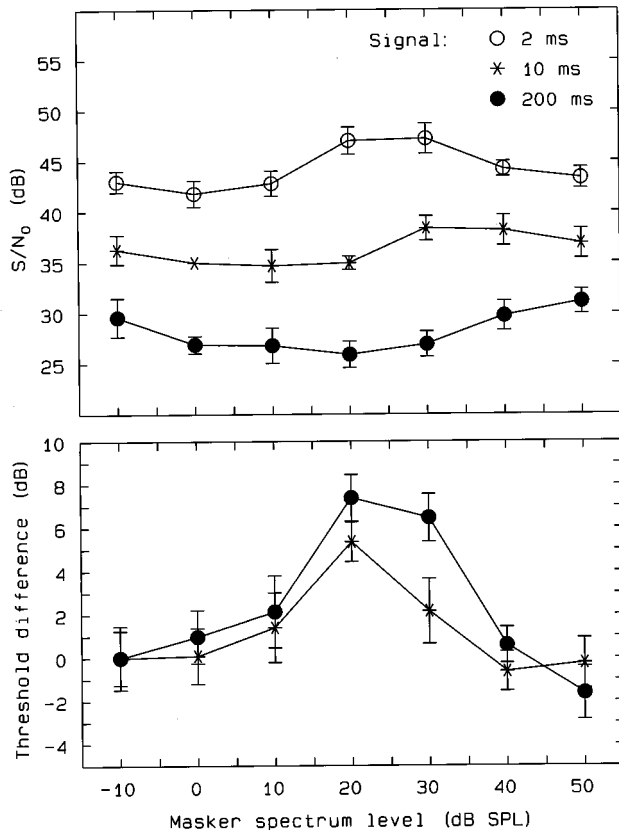


FIG. 2. Upper panel: Signal thresholds, in terms of signal-to-noise (spectrum level) ratio as a function of masker spectrum level, with signal duration as the parameter. Mean data of four listeners are shown, and error bars represent ± 1 standard error of the mean. Lower panel: Mean difference in thresholds between the 2- and 200-ms signal (filled circles) and between the 2- and 10-ms signal (asterisks), relative to the difference at -10 -dB spectrum level. Error bars represent ± 1 standard error of the mean.

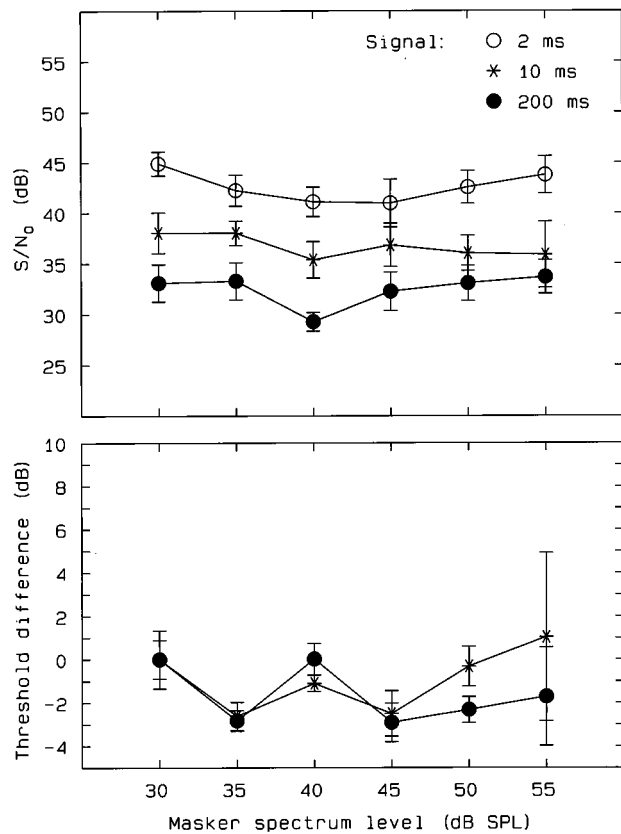


FIG. 3. As in Fig. 2, but for the four hearing-impaired listeners.

variation with level is less systematic, while for the 200-ms signal (filled circles) the SNR is maximal at the highest and lowest levels, with a maximum variation of over 5 dB. The increase in SNR at the lowest level may be due to the approach to absolute threshold. The mean threshold in quiet for the long-duration signal was 12 dB SPL, and so the threshold in the presence of the -10 -dB noise was less than 8 dB above this level. The increase in SNR at the highest two levels may be due to the increase in the effective bandwidth of the auditory filter at high levels. The increase in the critical ratio with level for long-duration high-frequency signals in broadband noise has been found previously both in humans (Reed and Bilger, 1973; Moore, 1975; Pick, 1977) and in a behavioral study of masking in cats (Costalupes, 1983).

For each masker level, the difference in thresholds between the 2-ms and the 10-ms signal provides a rough measure of the amount of short-term integration. An estimate of overall integration can be obtained using the threshold difference between the 2-ms and the 200-ms signals. These differences, relative to the difference at the lowest masker level (-10 dB), are plotted in the lower panel of Fig. 2. It can be seen that the change in the difference with level is as much as 9 dB (20-dB vs 50-dB condition) and that, with the exception of the 30-dB condition, most of the change is due to the difference between the 2-ms and the 10-ms signals (as-

terisks). This corresponds to the finding in experiment 1, that there were significant differences in slope with level for short, but not for long, signal durations.

Data from the hearing-impaired listeners are shown in the upper panel of Fig. 3. Data for the 30-, 40-, and 50-dB conditions are taken from experiment 1. There is a tendency for the SNR of the 2-ms signal to be lower for masker levels in the middle of the range tested (40–45 dB spectrum level), and this is in part mirrored by the SNRs of the 200-ms signal. More importantly, however, there seems to be no systematic effect of level on the differences between the durations, as shown in the lower panel of Fig. 3. This is in agreement with the finding of experiment 1, that the slope of the integration function for the hearing-impaired listeners seems independent of masker level over the level range tested.

While the lack of a level effect for the hearing-impaired listeners is consistent with the expected changes in BM non-linearity, the strength of the conclusions is limited by the restricted ranges of levels (25 dB) over which the hearing-impaired listeners could be tested. Another caveat concerns the age difference between the two groups. It is possible that some of the difference in performance between the two groups reflects the large difference in mean ages, independent of hearing loss. While there is no *a priori* reason why age, independent of hearing loss, should affect the slope of the temporal-integration function, we cannot rule out that possibility based on our data. Finally, large intersubject variability has often been reported for hearing-impaired listeners

(e.g., Florentine *et al.*, 1988). While our listeners show reasonably consistent results, the possibility exists that other listeners with similar audiometric configurations may show a somewhat different pattern of results.

In summary, the results of experiment 2 confirm the strong mid-level effect for the normally hearing listeners and the lack of a level effect for the four hearing-impaired listeners over the testable range of levels. The following section is concerned with determining whether known changes in BM compression with level can provide a quantitative account of the data, at least at short signal durations.

III. MODELING THE EFFECTS OF PERIPHERAL COMPRESSION

In her influential paper, Penner (1978) derived the shape of a temporal-weighting function, $h(t)$, suitable for describing time-intensity trades, as:

$$h(t) = S_0 a p t^{ap-1}, \quad \text{for } t \geq 1,$$

$$h(t) = S_0, \quad \text{for } 0 < t < 1,$$

where t is time in arbitrary units, $-a$ is the slope of the integration function, measured in terms of signal level (dB SPL) against $10 \log$ [duration (ms)], p is the power to which signal intensity is raised, and S_0 is the initial height of the integrator. Clearly, ap must be less than unity. In practice, this is achieved, as it is assumed that $p < 1$ and the slope of the integration function ($-a$) in most cases lies between 0 and -1 . As the units of t can be made arbitrarily small, the discontinuity between $t=1$ and $t < 1$ is of no practical importance. However, the equations show that for a given weighting function (from now on referred to as a temporal window), the slope of the integration function, $-a$, and the value of the power-law nonlinearity, p , are inversely proportional. An intuitive explanation of this relationship is given in the Appendix.

Within this framework, and assuming a fixed temporal window, the results from experiment 1 suggest that in order to account for the mid-level steepening in the slope of the short-term integration function, a nonlinearity is required which is about twice as compressive at medium sound levels as at high or low levels. Relating this to compression on the BM results in a prediction that, if the BM response of a damaged cochlea is linear, then the response of a normal cochlea is also linear at low and high levels and has a compressive growth, amounting to slightly more than 0.5 dB/dB at medium sound levels. This conclusion does not correspond well with the most recent physiological data. In most cases, compression resulting in growth of between 0.15 and 0.2 dB/dB at medium levels has been reported (e.g., Ruggero, 1992; Yates *et al.*, 1990; Murugasu and Russell, 1995). Also, other psychophysical experiments suggest that compression in human hearing is comparable to that measured physiologically (Oxenham and Moore, 1995; Oxenham and Plack, 1997). This difference may be related to the different stimuli used in the experiments. For instance, it is possible that the presence of broadband noise in the present experiment reduces the measured amount of compression. However, we know of no physiological measurements of the

overall BM response to broadband stimulation. Another reason for the apparent discrepancy may be found in one assumption of the Penner model, which requires closer consideration.

The assumption is that the output of the integrator is linearly related to the signal intensity raised to the power p , for a given signal duration. This condition is fulfilled when the signal is presented alone, as was assumed by Penner. However, in the case where a signal is presented simultaneously with a masker, the change in the output of the integrator due to the addition of the signal is no longer linearly related to the compressed signal intensity, I^p (for $p \neq 1$). Instead, the relationship between signal level and the integrator output due to the combination of signal and masker depends on the signal-to-noise ratio, on the amount of compression applied, and on the ratio of signal duration to temporal-window duration (assuming the masker is longer than the window). Hence, the equations derived by Penner (1978) are not valid for situations in which a masker and signal interact in the auditory periphery. In practice, this rules out Penner's model for all simultaneous-masking experiments.

A mathematical derivation of a revised analytical model, taking into account nonlinear interactions, is not attempted here. Instead, simulations were carried out using the model described below, in an attempt to find the change in nonlinearity with level necessary to account for our data within the context of the model.

A. Description of the model

The generic model we assume has been used many times in the past (e.g., Rodenburg, 1977; Viemeister, 1979; Buus and Florentine, 1985; Forrest and Green, 1987; Moore *et al.*, 1988; Plack and Moore, 1990) and consists of a bandpass-filter centered around the signal frequency (to simulate peripheral auditory filtering), a nonlinearity (rectification, followed by a power-law device), a short-term temporal integrator (or low-pass filter), and a decision device. The temporal window used here is a two-sided exponential window, as used by Oxenham and Moore (1994) to account for nonsimultaneous masking and by Peters *et al.* (1995) and Moore *et al.* (1996) to account for decrement detection. A symmetric shape is assumed, with the decay of the window on each side determined by a single time constant T , and given by the weighting function

$$W(t) = \exp(-|t|/T).$$

An asymmetric shape would be more realistic (Oxenham and Moore, 1994), but would not affect the outcome here. Data from temporal-integration experiments do not provide strong information as to the size of the temporal window. Therefore, we took the mean value of the "equivalent rectangular duration" (ERD), defined as $2T$, of 9.5 ms, from across a number of studies using stimulus frequencies of 4 kHz or greater (Oxenham and Moore, 1994; Peters *et al.*, 1995; Moore *et al.*, 1996).

The decision device used is also the same as, or similar to, that used in a number of previous studies (e.g., Plomp, 1964; Buus and Florentine, 1985; Oxenham and Moore, 1994). The output of the integrator due to the signal and

masker is compared with that due to the masker alone. If at any one time the difference between these two exceeds a criterion amount (in dB), the signal is “detected.” This criterion level provides the model with one free parameter.

Prior to compression, the stimuli within the model are represented simply by their envelopes, as it is thought that the auditory system has no access to stimulus fine structure at frequencies above about 4 kHz (Rose *et al.*, 1967). These envelopes are assumed to be flat for the steady-state portions of the stimuli and to have onset and offset ramps as used to gate the stimuli in the experiments. The noise masker and sinusoidal signal are assumed to add incoherently, i.e., the addition of two stimuli of equal level leads to a 3-dB increase in the overall level. The level of the masker envelope is initially derived by calculating the “effective” level of the broadband-noise masker within the equivalent rectangular bandwidth (ERB) of an auditory filter centered around 6.5 kHz (Glasberg and Moore, 1990). However, for the data from the hearing-impaired listeners, and for the 50-dB data from the normally hearing listeners, this value was increased by 2 dB to take account of the presumed broadening of the auditory filters in these conditions; a 2-dB increase corresponds to a broadening by a factor of about 1.6. The assumed masker level for the -10 -dB condition for the normally hearing listeners was also increased by 2 dB. This was done to model the effect of an internal noise, which is assumed to be added independently and to be responsible for absolute threshold. Thus only the 20-dB condition for the normally hearing listeners retained the original effective masker level; all other conditions were simulated using a masker level 2 dB higher. Any smoothing of the envelope due to the auditory filter at 6.5 kHz is assumed to be negligible compared to the smoothing of the temporal window.

In using a flat temporal envelope to represent the masker, we ignore the noise’s variability in level, as well as its envelope distribution. Regarding the first point, as long as a fixed temporal integrator is assumed (as we do here), the slope of the predicted integration function is the same whether or not the random level fluctuations of the noise are taken into account (Eddins and Green, 1995). The second point concerns the effective (long-term) level of a time-varying stimulus once it has been compressed. Two stimuli of equal energy, one with a flat and the other with a modulated temporal envelope, will have different mean levels after being compressed: The time-varying stimuli will have a lower mean level, as the peaks of the envelope will be compressed. This is also true when comparing a sinusoid with a Gaussian noise. However, initial simulations taking into account the envelope distribution of a Gaussian noise and the envelope distribution for a sinusoidal signal in Gaussian noise (van de Par and Kohlrausch, 1995) showed very little difference between this and using a flat temporal envelope to represent the noise. Using “realistic” signal-to-noise ratios and compression values, the maximum difference in absolute predictions reached 0.8 dB, and the maximum difference in the predicted slope of the integration function was 0.5%. Thus, for simplicity, in all the simulations presented below, the noise was represented by a flat temporal envelope.

Finally, possible dynamic (time-variant) effects in the

auditory system, such as the large onset response in the auditory nerve, are not taken into account here. While BM compression is thought to be near instantaneous (and hence time-invariant), the rate-intensity functions of auditory nerve fibers are dependent on prior levels of adaptation. Some previous psychoacoustic models have incorporated an approximation to this response (Zwislocki, 1969; Dau *et al.*, 1996a), although its importance in perception is not well understood. Nevertheless, the *change* in overall response due to changes in BM compression may not be affected by the response at the level of the auditory nerve. Thus the exclusion of this aspect of auditory processing from the model may not affect the main conclusions.

B. Model predictions

In order to derive the best-fitting criterion parameter for the model, the group mean data from experiment 1 for durations up to 10 ms were used from all conditions except those from the 20-dB (normally hearing) condition. Recall that the slopes of these data were not significantly different from each other. We fitted these data with the model by assuming that the signals are processed linearly before being integrated. “Linear” in these terms is with respect to intensity, rather than amplitude, for two reasons. First, Oxenham and Moore (1995) found that for hearing-impaired listeners, data from the additivity of nonsimultaneous masking could be accounted for well by assuming linear additivity of intensity. Second, physiological studies by Yates and colleagues (Yates *et al.*, 1990; Yates, 1990) have found that, in the absence of BM compression, the rate-intensity function of auditory-nerve fibers is a linear function of stimulus intensity.

By simulating all the conditions, and by comparing the predictions with the mean data from the experiment, the best-fitting (least-squared error) criterion value was selected. The mean data from experiment 1 for durations between 2 and 10 ms are replotted in Fig. 4, together with the model predictions (solid curves). The best-fitting decision criterion was equivalent to a steady-state level difference of 5.6 dB.

Next, the same model was used to fit the data from the remaining 20-dB condition. Here, the value of the nonlinearity was varied to produce the best-fitting predictions, while the time constant and the decision criterion were held constant. The decision criterion was set in terms of an equivalent long-duration (steady-state) level difference (in dB) prior to compression. The level difference prior to compression, rather than the “internal” (compressed) level difference, was chosen in order to maintain an approximation to Weber’s law for long-duration stimuli. This was done for empirical reasons and implies more efficient coding at medium levels than at high or low levels. As discussed above, the model of Penner (1978) predicts a best-fitting exponent of about 0.54, as the slope of the 20-dB condition is about 1.84 (the reciprocal) times steeper than that of the other conditions. For the present model, however, the best-fitting exponent was 0.25, indicating a compression ratio of 4:1. This value is more in line with the physiological estimates of BM nonlinearity, as mentioned above. Overall, the fit is very good, and lies within one standard error of the mean for 27 out of 30 data

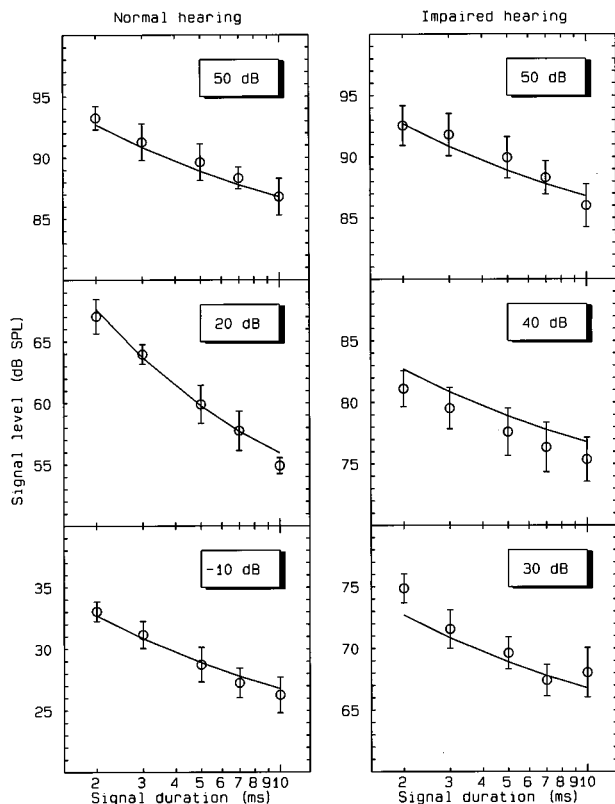


FIG. 4. Mean data from Fig. 1 re-plotted for signal durations between 2 and 10 ms (symbols) together with the predictions of the model described in the text (solid curves). Error bars represent ± 1 standard error of the mean.

points. The predictions for the 40-dB condition of the hearing-impaired listeners lie consistently between 1 and 1.5 dB above the data points. This reflects the fact that, for our four listeners, the critical ratio in this condition seems to be lower than in the other conditions. If we allowed the decision criterion to vary across levels, the fit would improve.

As stated in the Introduction, longer-term integration, for signal durations of more than 10 ms, may be due to longer time constants (in parallel with, or following, our hypothesized short-term integrator), a multiple-looks mechanism, an analysis window of variable duration, or a combination of these. At present, there seems to be no good way of distinguishing between these possibilities. Similarly, if we accept that the BM response in the normal cochlea is completely linear at low levels, then the reason for the reduced temporal integration at longer durations in the hearing-impaired listeners remains unclear. One of this study's initial aims of accounting for differences between normal and abnormal temporal integration therefore remains partially unfulfilled.

One hypothesis for explaining the difference in temporal integration at longer durations has been proposed by Carlyon *et al.* (1990). They found that the psychometric functions for single 5-ms 1-kHz tone pulses were steeper for hearing-impaired listeners than for normally hearing listeners. Psychometric functions were also steeper for a series of ten such pulses, separated by 80 ms. Thus while temporal integration, measured in terms of the level difference at threshold be-

tween one and ten pulses, was reduced in the hearing-impaired listeners, the change in detectability, in terms of d' , was the same as for normally hearing listeners. It may be, therefore, that a change in the underlying psychometric function can account for changes in the slope of the longer duration temporal-integration function, although this has not been tested for tones of different durations.

C. Concluding remarks

The model presented here provides a good description of short-term temporal integration, and shows that a change in nonlinearity consistent with that found in physiological studies can be applied to account for changes in temporal integration with level. However, the data are not suitable for deriving precise estimates of the weighting function for the temporal integrator. In this study we used as a time constant the mean value derived from other studies, but even a doubling of the value of this time constant had only a small effect on the mean squared error of the predictions. This is because the data we fitted only extend to durations of 10 ms, and increases in the ERD beyond values of about 10 ms have an increasingly small effect on predictions. Furthermore, the quality of the predictions is not dependent on the exact form of the temporal window. A 10-ms rectangular window, a Hanning window with a total duration of 20 ms, and a low-pass filter with a cutoff frequency of about 50 Hz all provide reasonably good fits to the data. In all these cases, the best fit to the data from the 20-dB condition was achieved with a compressive exponent of between 0.2 and 0.3. In general, a longer-duration temporal window results in a smaller required change in the nonlinearity for a given change in slope.

IV. SUMMARY

(1) For a signal frequency of 6.5 kHz, the slope of the short-term temporal-integration function for normally hearing listeners is steeper at medium levels than at high or low levels by a factor of nearly 2. This is consistent with the hypothesis that BM compression is greatest at medium levels, between about 50 and 70 dB SPL. For the hearing-impaired listeners, no effect of level was found over the 25-dB range tested. This is consistent with the idea that cochlear damage can lead to a linear BM input-output function.

(2) For signal durations of 10 ms and less, there was no significant difference in slope between the data from the hearing-impaired listeners and those from the normally hearing listeners at the low and high masker levels. While this is inconsistent with the results of one previous study (Pedersen and Elberling, 1973), four of six hearing-impaired listeners in a later study (Florentine *et al.*, 1988) show results similar to ours. If future studies confirm this finding, it suggests that the normal BM input-output function at low and high levels may be similar to that of listeners with cochlear hearing loss and so may be approximately linear.

(3) For signal durations of 20 ms and more, the temporal-integration functions for the hearing-impaired listeners were generally shallower than for the normally hear-

ing listeners. This is in agreement with the literature. However, the mechanisms underlying this effect remain unclear.

(4) The model proposed by Penner (1978), which predicts that the slope of integration is inversely proportional to the amount of compression for a given temporal window, is shown not to be applicable to any simultaneous-masking conditions.

(5) Simulations using an established model of temporal resolution indicate that the change in compression with level necessary for the model to account for the data from normally hearing listeners is similar to that found in many physiological measurements of BM compression and other psychophysical tasks.

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APPENDIX: COMPRESSION AND INTEGRATION

Here, an intuitive explanation is given for why compression prior to integration produces a steeper integration function. Assume a rectangular temporal-weighting function (integrator) of longer duration than the signal, operating on stimulus intensity (I^p , where $p=1$), and consider a signal at threshold. If the signal duration is doubled, the signal intensity must be halved to maintain the same (threshold) output of the integrator. This corresponds to a decrease in level of 3 dB. If instead the rectified signal amplitude is integrated, when the duration is doubled, the *amplitude* of the signal must be halved, leading to a decrease in level of 6 dB for the same change in duration. Since the amplitude is proportional to a compressed version of the intensity (I^p , where $p=0.5$), compression produces a steeper slope of the integration function for a given temporal window. In these terms a halving of the exponent produces a doubling of the slope of the function.

¹One listener (MS) reported that he found the loudness of the 50-dB masker uncomfortable. For this listener, the spectrum level of the masker was reduced to 45 dB before data were collected.

²In this study, the term hearing impaired is used for listeners with absolute thresholds higher than 20 dB HL. If age is taken into account, however, the hearing of listener AW falls within the "normal" range for men aged 80 and above (Morrell *et al.*, 1996).

³For the normally hearing listeners, thresholds for the 10-ms signal were measured at a later time than the other thresholds. At that time, listeners AO and ST were tested on different equipment. However, replications of thresholds from other conditions revealed no consistent differences, with deviations no greater than 1.2 dB.

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