Simulation of the effect of recruitment on loudness relationships in speech

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Persons suffering from perceptive deafness commonly find it difficult to understand amplified speech, and their understanding of speech is easily destroyed by competing speech or noise. One source of perceptive distortion is recruitment, which exaggerates loudness differences among the acoustical elements of speech. A transposition of these distorted loudness relationships from the deaf-subject span of hearing to the normal span is illustrated graphically, and achieved in practice with an electronic processor. A recording of processed speech, simulating for normal listeners the loudness relationships perceived by deaf subjects with recruitment, accompanies this paper. Recruitment-compensation processing for hearing aids is also simulated. The recruitment simulation is validated by an experiment with four unilaterally deaf subjects, who compared processed speech in the normal ear with unprocessed speech in the impaired ear. The simulation suggests that (1) recruitment is a sufficient cause for loss of intelligibility in the deaf, whether or not there are other causes; (2) compensation for this recruitment is a necessary, although possibly insufficient, condition for restoring that intelligibility; (3) the benefit of using both compression and post-compression equalization in a hearing aid designed to compensate recruitment is likely to be considerably greater than the arithmetic sum of the separate, limited benefits of each process; and (4) the combined processing, by restoring redundant speech-recognition cues to the subjects's perception, can increase the resistance of intelligibility to acoustical interference.

Subject Classification: 65.80, 65.64, 65.48.

INTRODUCTION

Persons suffering from perceptive deafness commonly find it difficult to understand amplified speech, and their understanding of speech is easily destroyed by competing speech or noise. It has been suggested by Huizing (1948) and others that recruitment,[1] which distorts the subject's perception of amplitude relationships among the acoustical elements of speech, is a sufficient cause for loss of intelligibility. This paper presents an analogy in which the abnormal loudness relationships created by recruitment are transposed to the normal dynamic span of hearing. The analogy suggests that recruitment can reduce both intelligibility and the resistance of intelligibility to acoustical interference.

I. SIGNAL PROCESSING TO SIMULATE RECRUITMENT

Figure 1 shows the approximate range of sound-pressure levels for conversational speech, plotted in relation to the normal span of hearing between threshold and the 74-phon equal-loudness contour. The left half of Fig. 2 shows these speech levels amplified and plotted against the average of the corresponding hearing spans of six deaf subjects, measured in a previous experiment by Villchur (1973). The reference level of the deaf-subject equal-loudness contour is related to that of the 74-phon contour by way of the deaf subjects' preferred listening levels for speech. The reduced span between the thresholds and equal-loudness contours of these subjects reflects their recruitment.

The processed speech band at the right of Fig. 2 is a projection of the amplified speech band from the abnormal to the normal span of hearing. The distorted amplitude relationships within this projected speech band are created by subjecting the speech signal to frequency-dependent volume expansion followed by high-frequency attenuation. Each audible element of the processed speech has the same relative level in the dynamic span of normal hearing as the corresponding element of the unprocessed, amplified speech has in the deaf-subject span of hearing. Those elements of speech that are below the deaf-subject threshold also fall below the normal listener's threshold. The aim of such processing is to make it possible for normal listeners to perceive the same loudness relationships among the acoustical elements of speech as are perceived by the deaf subjects.

Although the deaf-subject equal-loudness contour appears to indicate only a mild increase of hearing loss above 1 kHz, the transposed speech band shows severe high-frequency attenuation as a result of the expansion factor. The normal-hearing equivalent of the deaf subject's perceived high-frequency attenuation for speech is of the order of 40 dB/oct above 1 kHz for the highest-level speech elements and 60 dB/oct for the lowest-level elements, after an initial attenuation of the lowest speech levels relative to the highest speech levels of 28 dB at 1 kHz.

Since recruitment expands the difference in loudness between low-frequency, high-amplitude vowels and high-frequency, low-amplitude consonants, speech is subjected to effective high-frequency attenuation even if the subject's hearing impairment does not become greater at high frequencies.

II. VALIDITY OF THE ANALOGY

The analogy may or may not apply to the overall quality of speech as perceived by the deaf subject: It is derived only from the abnormal loudness relationships of that perception, and the subject may have to

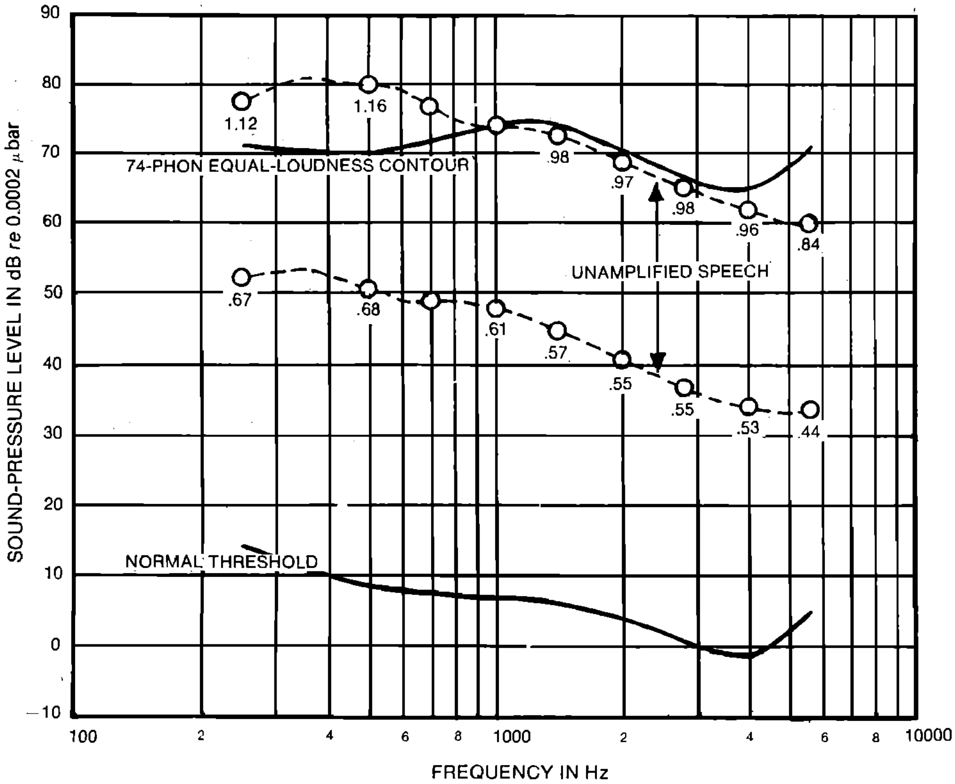

FIG. 1. Sound-pressure levels of conversational speech, as measured in $\frac{1}{2}$-oct bands by Dunn and White (1940). Numbers under open circles show the proportionate positions of speech levels relative to the amplitude range between normal threshold and the ISO 74-phon equal-loudness contour.

contend with additional perceptive aberrations. Assuming that the analogy is accurate with respect to loudness relationships, we can predict only that the intelligibility of unprocessed, amplified speech for the deaf subject is likely to be at least as bad as the intelligibility of the transposed speech is for normals.

Possible sources of inaccuracy in the loudness analogy include (1) the effect of signal duration on the sensation of loudness in deaf subjects, which may be different from the corresponding effect in normals; (2) nonlinear recruitment in deaf subjects, i. e., recruitment that varies in degree over the amplitude range; and (3) abnormal susceptibility of deaf subjects to forward masking, as reported by Gardner (1947), which could change their instantaneous loudness responses in vowel–consonant sequences.

Miskolczy-Fodor (1958) reported that the relationship between signal duration and loudness for deaf subjects with recruitment was the same as for normals. Hallpike and Hood (1959) found that measured recruitment curves did vary in shape, but that for many subjects they were essentially linear over the dynamic range with which we are concerned. Villchur's (1973) earlier experiment with the six deaf subjects of Fig. 2 provides indirect support for the general accuracy of the analogy. In that experiment speech was processed to compensate rather than to simulate recruitment. The speech signal was subjected to frequency-dependent amplitude compression followed by treble emphasis—the reverse of the process used here for the analogy—so that deaf subjects could perceive the same loudness relationships among speech elements as are perceived by normals. Speech recognition was improved significantly for the six subjects, and readjustment of the processing characteristics by each subject for maximum speech intelligibility usually confirmed the calculated values as reasonably accurate.

Figure 3 is an idealized processor that creates the transposition of signals shown in Fig. 2. The expander in each channel has an expansion ratio[2] equal to the ratio between the normal dynamic span of hearing (as defined in Fig. 2) at the frequency of that channel and the corresponding deaf-subject span of hearing. The equalization is derived in the Appendix. If test signals were routed through such a processor and presented to a subject with average-normal hearing, the measured threshold and equal-loudness levels would be the same as for the deaf subject represented. The following experiments are designed to test the hypothesis that the processing can also simulate the abnormal loudness relationships perceived by the deaf subject in a complex speech signal.

III. EXPERIMENT 1: COMPARISON OF EXPANSION/EQUALIZATION PROCESSING WITH MASKING

The loss of hearing induced in a normal listener by masking is accompanied by recruitment, as pointed out by Steinberg and Gardner (1937); the subject's loudness response increases toward normal as the amplitude of the sound stimulus is increased above his new threshold. The recruitment curve associated with masking approaches the curve of normal loudness growth asymptotically, but the departure from linearity of the induced recruitment, measured in the present subjects by loudness-balance tests, was relatively minor over the in-

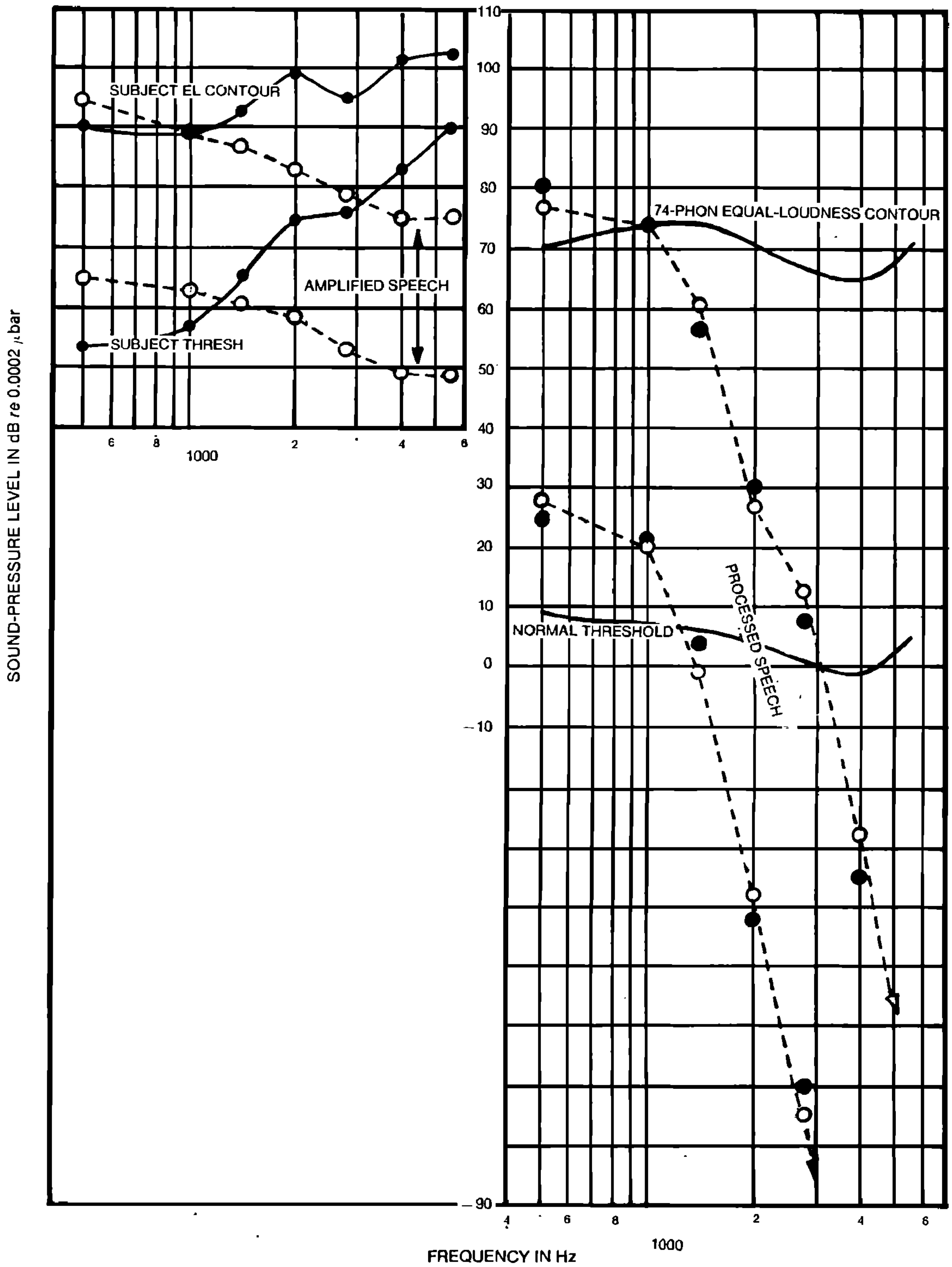

FIG. 2. Projection of the amplified speech band from the deaf-subject span of hearing to the normal span, keeping the same proportionate relationship between speech levels and corresponding hearing spans. Solid dots represent the processing actually achieved in Expt. 1.

tensity range significant to the masked speech. The noise-deafened ears were thus considered to be reasonable substitutes for deaf ears with linear recruitment so far as loudness relationships in speech are concerned.

This preliminary experiment tests the ability of a real expander/equalizer to simulate induced recruitment, using experienced listeners.

A. Equipment

A practical version of the processor of Fig. 3 was constructed, using three Krohn-Hite bandpass filters and three modified DBX Model 117 expanders. The crossover frequencies were 1.6 kHz and 4.2 kHz. The expansion ratios, in the order of ascending frequency of the channels, were 2, 3, and 4.5, and compromise values

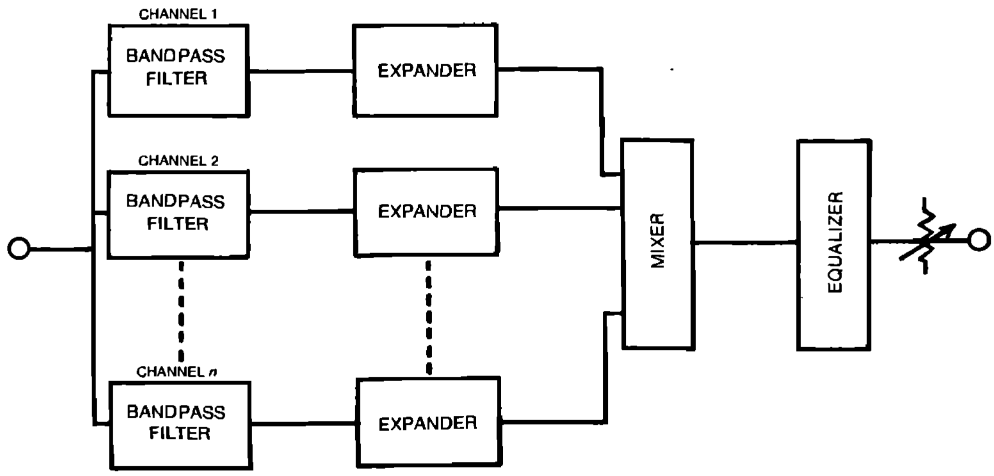

FIG. 3. Signal processor to simulate for normal listeners the abnormal loudness relationships perceived by a deaf subject with recruitment. The processor is an electrical analog of recruitment: The response of the model represents the deaf subject's loudness response relative to normal response.

for the individual-frequency expansion ratios required by the data of Fig. 2. Attack time was of the order of 1 msec, and release time was of the order of 5 msec for a 40-dB change of input level.

The equalizer was a General Radio multifilter 1925, which is adjustable in 1-dB steps for each adjacent 1/3-oct band. The combination of precise equalization and compromise expansion achieved acceptably accurate processing of discrete signals, as indicated by the solid dots in Fig. 2.

Unfortunately, if more than one input signal is present at the same time in one channel of the real processor, the expander gain in that channel will be controlled by the momentary amplitude of the higher-level signal. The expanders will thus tend to overamplify background sounds and some elements of speech. The significance of this processing inaccuracy must be determined experimentally.

All signals were presented through a Grason–Stadler speech audiometer. Signal patterns were formed by a Grason–Stadler programming system and Grason–Stadler electronic switches. Speech was recorded on a Revox A77 tape recorder, and the masking source was a General Radio random-noise generator.

B. Procedure

Two normal-hearing subjects (one of them the writer) compared unprocessed, continuous speech, presented monaurally in the presence of masking, with unmasked speech processed by the recruitment model. The masking noise was shaped by an equalizer to produce thresholds similar to the deaf-subject thresholds of Fig. 2. The expansion ratios, crossover frequencies, and equalization for the recruitment processing were calculated from the relationship between the subjects' masked and unmasked hearing spans, from thresholds to corresponding equal-loudness contours. The masked contour was derived from loudness-balance measurements against the unmasked 74-phon contour at half-octave intervals.

The subject adjusted speech levels so that the processed speech would have the same loudness as the masked, unprocessed speech, readjusted the expansion ratios, and made a subjective comparison between alternate 5-sec periods of speech in each mode. The speech was presented both with and without interfering voices in the background, at a signal-to-interference ratio of 10 dB before expansion. An attempt was also made to simulate the quality of the masked speech by subjecting an unexpanded signal to high-frequency attenuation alone.

C. Results

The quality that characterized the expanded signal—an exaggeration of the normal dynamics of speech, and an absence of high-frequency elements—was reported by both subjects, as clearly evident in the masked speech. The similarity in quality and intelligibility between the masked, unprocessed speech and the unmasked, processed speech was judged by the writer (as subject) to be about as close as that between his normal left-and right-ear perception of unprocessed speech. The judgments of each subject are shown in Table I.

For both subjects the masked and processed signals remained similar to each other after speech interference was added. The relative loudness of the interfering signal did appear greater in the processed signal as predicted, but only part of the time and not by a large amount.

In the attempt to match the masked signal with equalization-only processing, the closest overall color was achieved when high-frequency attenuation of 24 dB/oct above 1.5 kHz was added to the original post-expansion treble attenuation. The absence of exaggerated dynamics in the unmasked signal, however, kept the two signals quite different in quality, with higher intelligibility for the unexpanded signal.

D. Discussion

The three expanders worked well, so far as speech quality is concerned, in representing the infinite bank of effective expanders created by the induced recruitment. Speech interference was not grossly misrepresented by the processor, possibly because the random time pattern of speech provided enough staggering among the instantaneous levels of simultaneous voices.

TABLE I. Judgments by four unilaterally deaf subjects to evaluate accuracy of the recruitment simulation. The subject ratings refer to the similarity between unprocessed speech in the impaired ear and speech presented to the normal ear in three different modes of amplification. Unilateral recruitment in the two normal-hearing subjects was induced by masking.

	Unprocessed						Calculated processed				Adjusted processed					
Subject	BJ	RO	OJ	AB	Normal 1	Normal 2	BJ	RO	OJ	AB	BJ	RO	OJ	AB	Normal 1	Normal 2
Identical or almost identical																X
Very similar											X	X			X	
Similar								X	X				X	X		
More different than similar	X	X	X			X	X			X						
Very different				X	X											
Totally different																

IV. EXPERIMENT 2: EVALUATION OF RECRUITMENT PROCESSING BY UNILATERALLY DEAF SUBJECTS

A. Procedure

Simulated-recruitment processing calculations were made for each of four subjects with unilateral, nonconductive deafness accompanied by recruitment. The calculations were made on the basis of the relationship between the subjects' normal-ear and impaired-ear hearing spans in the same way as in the masking experiment, using threshold and loudness-balance measurements.

The subjects were asked to make judgments of the similarity or difference in quality between samples of speech heard in each ear. The impaired ear was always presented with unprocessed speech: The normal ear was presented in turn with unprocessed speech, speech subjected to recruitment processing calculated for that subject, and processed speech whose characteristics had been readjusted by the subject to make the sound in his two ears as alike as possible. The latter adjustments were not made until after the first two signal presentations in order to avoid biasing the earlier judgments, and the presentation of the adjusted processing was mixed in with repetitions of the previous comparisons. In all cases the speech level in the impaired ear was adjusted for the same loudness as speech in the normal ear.

The simulation-processing equipment was that used in Expt. 1. Signals were presented alternately to each ear in 5-sec periods. Contralateral masking with speech-spectrum noise was used for three of the subjects, whose losses were severe. Subjects reported their judgments by marking the form shown in Table I.

Tests were also conducted to determine whether speech intelligibility in the impaired ear was improved by subjecting the unprocessed signal to the reverse of the recruitment-simulation processing. The recruitment-compensation processing consisted of compression plus equalization, as described in Sec. V.

The initial plan was to use five-to ten-word sentences for the intelligibility tests, but only one of the four subjects, BJ, had impaired-ear speech recognition adequate for this task. An easier test—spondee lists (airplane, baseball, etc.)—was therefore used for the other three subjects. All tests were administered from a tape recording at levels chosen by the subject, and answers were written. The overall procedure of the testing, the compensation-processing equipment, and the method of signal-processing calculations and subject adjustments were those previously described by Villchur (1973).

B. Results

All subjects rated unprocessed speech in the normal ear as different, in varying degree, from unprocessed speech in the impaired ear, and all subjects rated the final processed speech in the normal ear as "similar" or "very similar" to unprocessed speech in the impaired ear. The individual subject judgments for each signal mode are shown in Table I.

The recruitment-compensation processing improved impaired-ear intelligibility scores for two of the subjects, as shown in Table II. These were the same two subjects who judged the recruitment simulation as "very similar" to the real thing.

TABLE II. Impaired-ear intelligibility scores for four subjects, with and without recruitment-compensation signal processing.

	Subject BJ	Subject RO	Subject AB	Subject OJ
Unprocessed	78.5%	8%	0	44%
Compressed/ equalized	97%	35%	0[a]	43.5%

[a]Subject AB recognized a few individual phonemes.

C. Discussion

The subjects may have recognized their own adjustments in the final sequence of comparisons, which would taint the ratings in the "adjusted processed" column. There was, however, some objective confirmation of the subjective ratings. First, subjects were free to adjust each of the three expansion ratios to any value including unity without knowing the significance of the dial settings, and all four subjects chose to to use expansion in all channels of the processed signal for the closest match between left and right signals, usually at expansion ratios at least as high as the calculated values. The second confirming element was that the impaired-ear intelligibility scores for two subjects were improved significantly by the reverse of the recruitment-simulation processing. The recovery or partial recovery of speech recognition implies that at least part of the perceptive distortion of these subjects is represented by the simulation processing.

The two subjects whose speech recognition was not improved by recruitment compensation may have suffered from perceptive aberrations more damaging to intelligibility than their recruitment.

The similarity between left-and right-ear signals was not evaluated by objective intelligibility tests. While a valid simulation of recruitment would duplicate the effect of exaggerated loudness relationships on intelligibility, it would not take into account perceptive impediments to speech recognition other than recruitment. Intelligibility scores for the impaired ear could thus be considerably lower than scores for the processed speech without invalidating the simulation of loudness relationships. Further, agreement between processed and impaired-ear scores would not validate the simulation, because processed-speech intelligibility can be reduced by signal processing unrelated to the subject's perceptive aberrations.

The analogy illustrated in Fig. 2 appears to be validated for the deaf subjects of this experiment. To the extent that the analogy is valid, the processor of Fig. 3 enables normal listeners to hear the abnormal loudness relationships perceived in speech by these subjects, although it does not necessarily reproduce the total quality of speech that they perceive.

The circuit of Fig. 3 is proposed as an electrical analog of recruitment in human perception. The response of the model represents the deaf subject's loudness response, relative to normal response, as the input signal varies in amplitude and frequency.[3] Signals over the reduced dynamic range of the deaf subject's residual hearing are expanded to the dynamic range of normal hearing, representing the fact that the deaf subject interprets these signals as covering the full range of loudness.

V. SIGNAL PROCESSING TO SIMULATE HEARING-AID COMPENSATION

The recruitment analogy can be used as a tool to help analyze problems in hearing-aid design. It should be clear that hypotheses rather than conclusions are properly derived from the analogy, and that such hypotheses must be tested in the real world of deaf subjects.

Ideal signal processing for a hearing aid designed to compensate recruitment would counteract precisely the expansion and treble attenuation of the recruitment model. A compromise processor is used here, with a two-channel amplitude compressor that provides approximate compensation for the infinite number of expanders representing the subject's recruitment.[4] For the deaf-subject hearing characteristics plotted in Fig. 2, the compressors are adjusted to a low-channel compression ratio of 2, a high-channel ratio of 3, and a crossover frequency of 1.5 kHz. The processing also requires post-compression treble emphasis, equal to the inverse of the high-frequency attenuation in the recruitment model divided by the expansion ratio at each frequency. (The hearing-aid processor precedes the recruitment model, following the order of signal processing in real life, and the equalization will be multiplied by the expansion ratios of the recruitment model or, in the case of a real subject, by the effective expansion ratios of the subject's recruitment.) About 14 dB of boost at 4 kHz *re* 1 kHz is used, with some extra boost above 4 kHz to make up in part for the need of a higher compression ratio at higher frequencies.

This hearing-aid processor will restore near-normal intensity relationships to the transposed speech shown at the right in Fig. 2, which means that if the recruitment analogy is valid the circuit will process normal, amplified speech to have near-normal loudness relationships for the deaf subject represented at the upper left of Fig. 2. As shown in Fig. 4, the transposed speech is reprocessed to an imperfect copy of the normal-speech band in Fig. 1, while for the deaf subject the elements of normal speech are processed to corresponding relative levels within his hearing span.

Figure 5 shows the effect of using only the equalization of this hearing-aid circuit, without compression. The low-amplitude speech elements are not amplified to useful levels. More high-frequency emphasis could be used to bring the top of the speech band to its normal relative level, but treble boost without compression must be used sparingly in a hearing aid. High-amplitude treble sounds in the real environment may be amplified to levels that are intolerably loud for the deaf subject, whose discomfort levels are likely to be either normal or lower than normal.

Figure 6 shows the effect of using only the compression of the hearing-aid circuit, without equalization. The low-amplitude elements of speech are brought into the correct loudness relationship with high-amplitude elements of the same frequency, but severe treble attenuation remains. The normal-hearing equivalent of this attenuation is of the order of 30 dB/oct above 1 kHz.

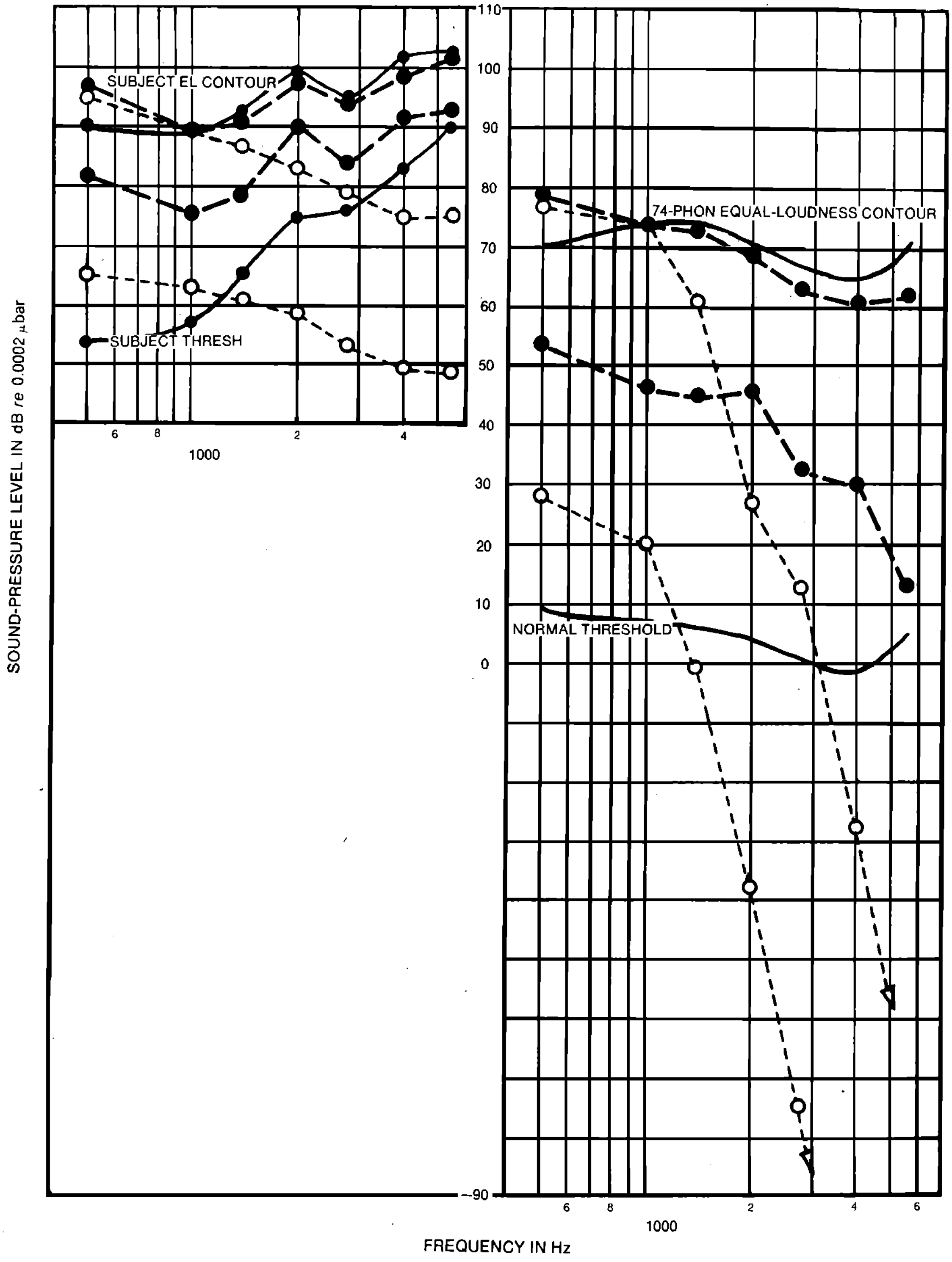

FIG. 4. Solid dots and heavy dashes show the effect of the compromise hearing-aid compensation (two-channel compression plus post-compression frequency equalization) described in Sec. V. Open circles at the right outline processed speech before compensation, as in Fig. 2, and open circles at the left outline amplified, unprocessed speech before compensation.

VI. THE DEMONSTRATION RECORDING

Each of the four bands of the accompanying disk has been dubbed through a different processing circuit from the same recorded sample of speech. The first part of each band is the recording of a single talker in quiet, who is joined by two interfering talkers in the second part. The signal-to-interference ratio in the original, unprocessed tape is of the order of 10 dB. This ratio is expanded in Bands 1 and 2 and restored in Bands 3 and 4.

Band 1 is speech processed by the 3-channel version of the recruitment model. The processing characteristics were chosen so that the loudness relationships of this speech for normal listeners would be those heard by the "average" deaf subject of Fig. 2 when he is aided only by unprocessed amplification.

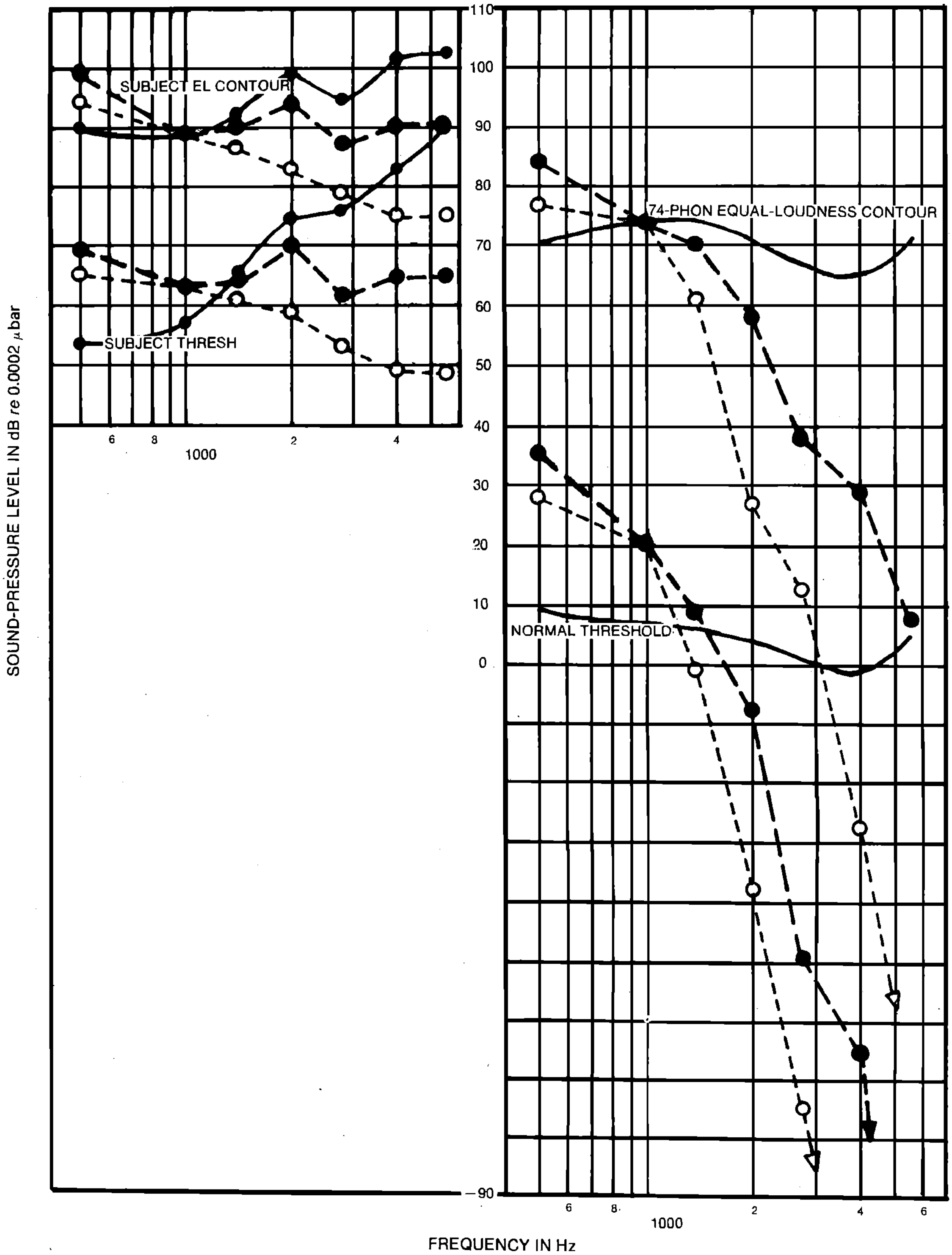

FIG. 5. Solid dots and heavy dashes show the effect of the hearing-aid equalization of Fig. 4, without compression. Open circles outline speech bands before compensation.

Band 2 is the processed speech of Band 1 compensated by the hearing-aid equalization of Fig. 5.

Band 3 is the processed speech of Band 1 compensated by the two-channel hearing-aid compression of Fig. 6.

Band 4 is the processed speech of Band 1 compensated as in Fig. 4. The same compression used in Band 3 is combined with the same equalization (applied after compression) used in Band 2.

It should be pointed out that the signal cannot properly be compensated by compression *after* expansion. The expansion theoretically increases the dynamic range of uncompressed speech to more than 190 dB, and the lower-level elements of uncompressed speech drop irretrievably into the circuit noise.

The recording was made in a normally reverberant living room. The best reproduction will be through earphones, which eliminate double reverberation.

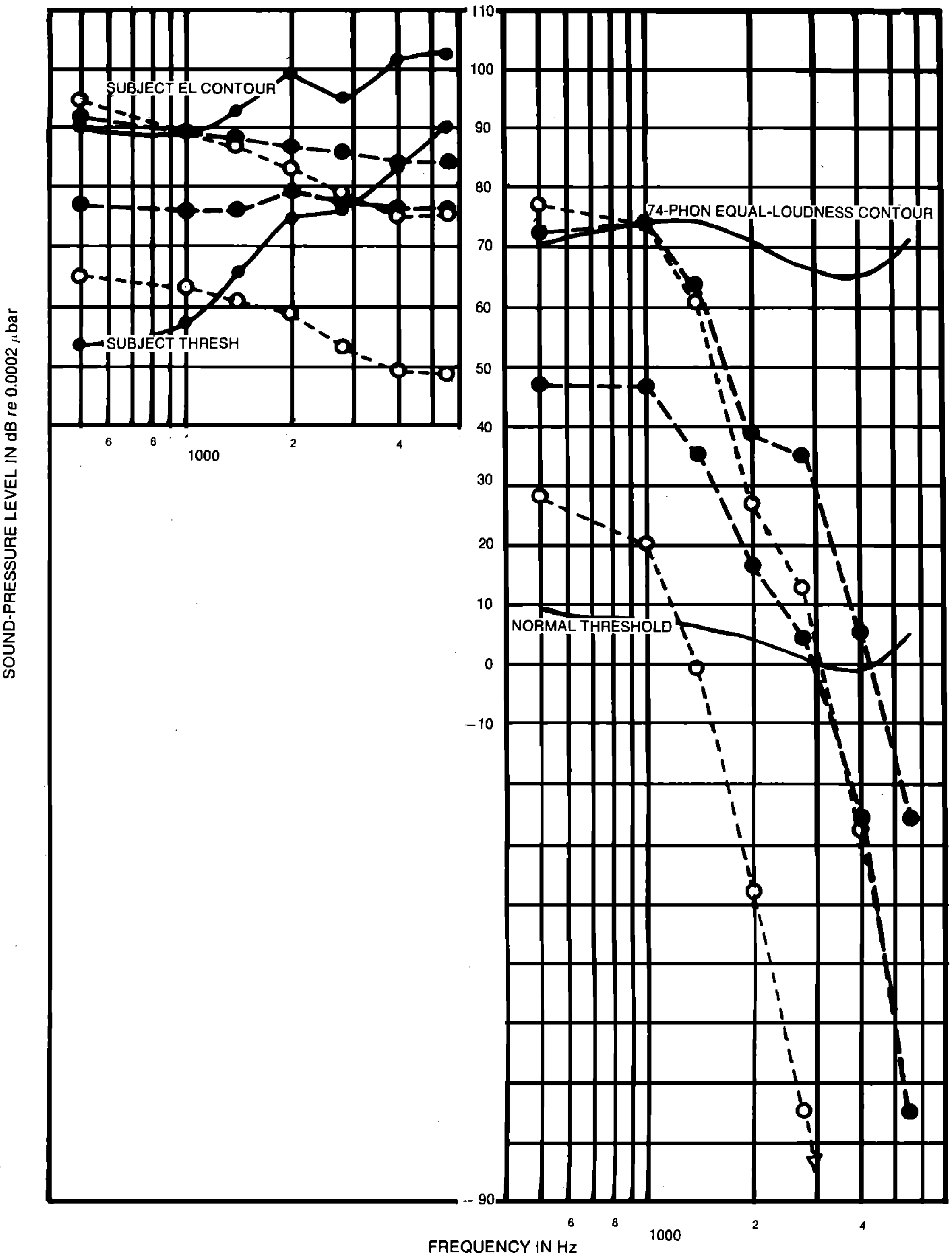

FIG. 6. Solid dots and heavy dashes show the effect of the hearing-aid compression of Fig. 4, without equalization. Open circles outline speech bands before compensation.

VII. GENERAL DISCUSSION AND CONCLUSIONS

The poor intelligibility of Band 1 implies that the recruitment represented is a sufficient cause for loss of intelligibility in the deaf, whether or not there are other causes, and that compensation for this recruitment is a necessary although possibly insufficient condition for restoring that intelligibility.

The implication of the relative intelligibility of Bands 2, 3, and 4 is that the benefit of using both compression and post-compression equalization in a hearing aid is likely to be considerably greater than the arithmetic sum of the separate, limited benefits of each process. Either process by itself may have the unrewarding effect of lifting critical elements of speech from well below the subject's threshold to just below that threshold.

Martin, Murphy, and Meyer (1956) reported that when speech was subjected to high-frequency attenuation similar to that in the compressed-only speech of Band 3, not much of the intelligibility was lost for normal listeners in a quiet environment. However, when the same treble attenuation was combined with other distortions such as multiple echoes and/or noise, each relatively innocuous by itself, intelligibility was reduced significantly. The hearing-aid user listens in an environment that includes reverberation, noise, and competing speech, and his perceptive aberrations combine with external distortions. The poor resistance to interference of the partially compensated signal of Band 3 suggests that compression without equalization in a hearing aid may restore just enough acoustic information for understanding continuous speech under ideal conditions, but not enough of the redundant information, normally present in speech, that makes it possible for the listener to tolerate destructive influences like interference. The limited number of recognition cues that have been made available to the deaf subject do not include a reserve against further losses. Note that the intelligibility of the compressed/equalized speech in Band 4, where redundant recognition cues have been restored, is much more resistant to interference than is the intelligibility of the uncompensated or partially compensated speech, even though the signal-to-interference ratio in Band 4 has been reduced from that in Bands 1 and 2 by compression. The effect of redundant cues on the resistance of speech intelligibility to noise has been discussed by Coker (1973).

Two major problems that must be considered by the designer of hearing aids for the perceptively deaf are poor subject recognition of amplified, clear speech and the abnormal vulnerability of this reduced recognition to acoustical interference. Band 4 suggests that signal processing that relieves the first problem may simultaneously relieve the second, by restoring redundant speech-recognition cues to the subject's perception. The deaf-subject experiment referred to earlier lends support to such a conclusion. When a background of competing speech was added to the speech tests of that experiment (the interference being equally subject to the processing), the benefit of compression/equalization processing to speech intelligibility was almost always maintained and usually increased.

ACKNOWLEDGMENTS

I am indebted to Dr. Peter B. Denes of Bell Laboratories, who told me about an experiment he had conducted in which unilaterally deaf subjects were used to evaluate signal processing, and to Mead C. Killion of Industrial Research Products, Inc., a Knowles Company, for first pointing out to me the limitations of an electronic expander in processing simultaneous signals. This project was conducted under the auspices of the Foundation for Hearing Aid Research.

APPENDIX. RECRUITMENT–PROCESSOR EQUALIZATION

The expanders of Fig. 3 have equal gain at a common reference level. The maximum low-channel speech-input level is convenient as the reference level: The expansion of lower-level signals may be thought of as taking place downwards from this level, in different degree for each channel. Equalization completes the work of the expanders in projecting deaf-subject threshold and equal-loudness levels to coincide with the corresponding normal levels (see Fig. 2), and the equalization cannot be calculated until the equal-gain expander level has been determined.

The frequency response required of a post-expansion equalizer is the difference between normal threshold levels and the deaf-subject threshold levels subjected to expansion only, minus the absolute difference between levels at a reference frequency chosen for plotting the response curve:

$$R_f = L_{nf} - [\, L_0 - E_f(L_0 - L_{sf}) \,] - A_{f'} ,$$

where R_f is the equalizer response at frequency f relative to the gain at reference frequency f'; L_{nf} the normal threshold level at frequency f; L_0 the reference level of equal gain for expanders; E_f the expansion ratio at frequency f; L_{sf} the deaf-subject threshold level at frequency f; $A_{f'}$ the equalizer gain at reference frequency f'; and $A_{f'} = L_{nf'} - [\, L_0 - E_{f'}(L_0 - L_{sf'}) \,]$.

All values are in decibels except E_f, which is expressed arithmetically. Pre-expansion equalization (for which R_f must be divided by E_f) may be substituted, but the expanders would then have to handle signals of increased dynamic range.

[1]Recruitment is a perceptive aberration in which changes of sound intensity produce greater-than-normal changes of loudness.

[2]The expansion ratio is expressed arithmetically, as the number of decibels of change in the output signal divided by the number of decibels of initiating change in the input signal.

[3]Linearity controls may be included in the expander circuits, so that the model can be adjusted to represent different recruitment characteristics.

[4]The compressors of this experiment are the same ones used in the previous study with real deaf subjects. The averages of the compression ratios chosen by the subjects of that study were 2.1 for the low channel and 2.8 for the high channel. The compressor attack time is less than 1 msec, and release time is 20 msec for a 40-dB drop in level.

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