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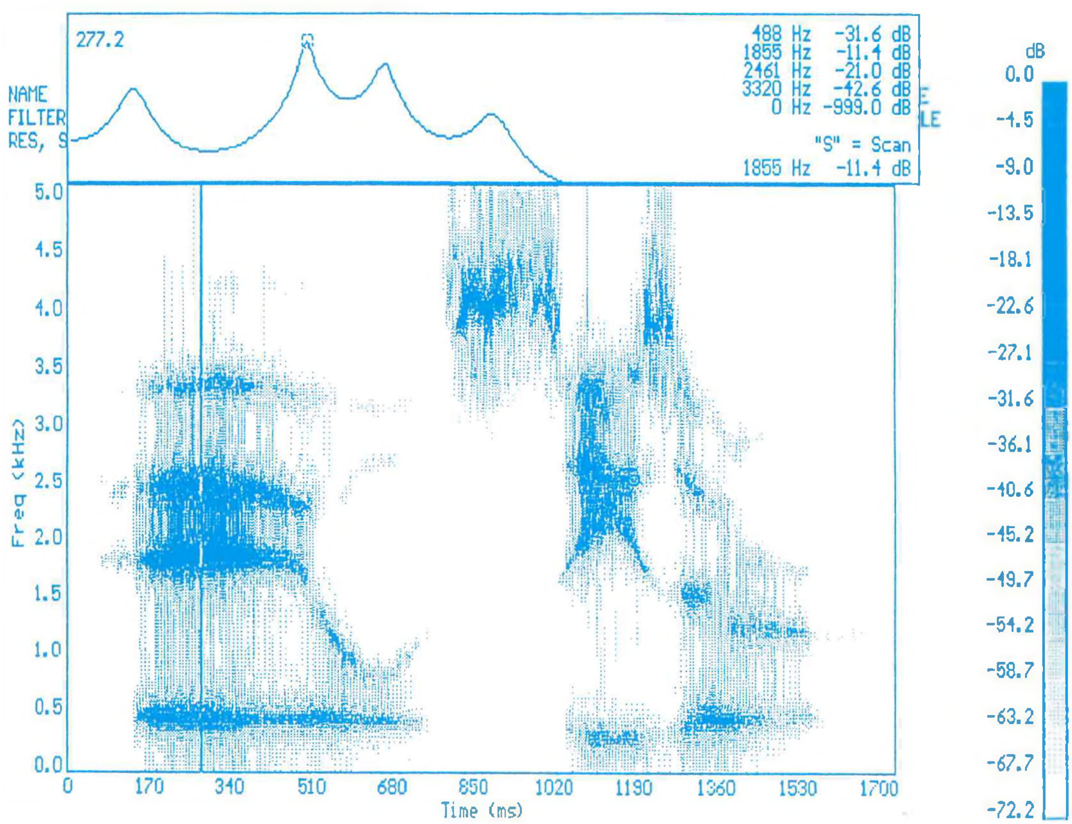

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EDITORIAL

Voici le dernier numéro de *Canadian Acoustics / Acoustique canadienne* pour lequel j'agis à titre de rédacteur-en-chef. Cette fonction m'a beaucoup appris sur la communauté des acousticiens du Canada et sur la vitalité de notre Association. J'ai pu prendre conscience de la diversité importante des domaines de recherche et d'intervention professionnelle ayant trait à l'acoustique. J'ai pu réaliser pleinement le rôle que joue notre périodique comme outil de communication entre les membres de l'Association et comme tribune pouvant assurer une visibilité minimale aux travaux des nombreux chercheur-e-s intéressés. Nos efforts pour rejoindre ces gens et pour les inviter à faire connaître certains de leurs résultats de recherche ont heureusement porté fruits. Un nombre soutenu de manuscrits nous ont été soumis, particulièrement au cours de cette dernière année, quelques uns provenant de l'étranger. Il s'agit vraisemblablement d'un signe indiquant que notre périodique est en train d'acquérir ses lettres de créance. Il reste évidemment beaucoup à faire. Resserrer les contacts avec d'autres périodiques nationaux traitant de l'acoustique, élargir encore le répertoire des domaines faisant l'objet d'articles de recherche, assurer une meilleure distribution à l'étranger de même qu'une couverture adéquate par les services de documentation, susciter davantage de manuscrits auprès de chercheur-e-s dont la compétence est déjà bien établie, et quoi encore. Je laisse à la nouvelle équipe de rédaction le soin de définir ses priorités, avec la conviction que les progrès accomplis sont maintenant des assises sur lesquelles on peut s'appuyer. Les changements dans l'équipe de rédaction s'opèreront d'ailleurs sous le signe de la continuité puisque Murray Hodgson, le nouveau rédacteur-en-chef, a déjà largement contribué aux résultats auxquels nous sommes parvenus.

Par le biais de la fonction que j'ai assumée, j'ai surtout appris à mieux connaître et à apprécier un très grand nombre de personnes qui appartiennent à cette communauté somme toute importante de scientifiques et de professionnels ,et c'est précisément ce qui m'a le plus enthousiasmé dans ce rôle. Je remercie les membres de l'Association pour la confiance qu'elle m'a accordé.

R. Hétu

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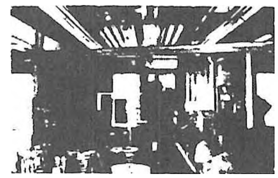

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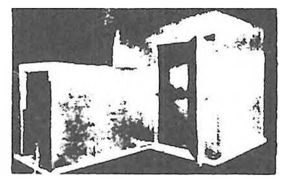

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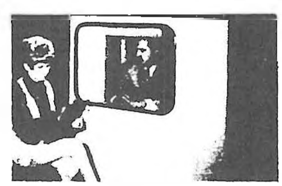

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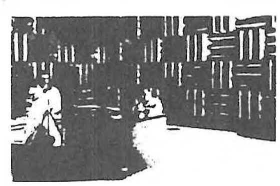

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QUANTIFYING THE PERCEIVED URGENCY OF AUDITORY WARNINGS

Elizabeth Hellier & Judy Edworthy

Department of Psychology
Polytechnic South West
Plymouth, England

ABSTRACT

Advanced auditory warning design based on Patterson's guidelines (1982) allows a degree of matching between a warning sound itself and the subjective response it elicits. One important parameter along which a warning and the subjective response can be matched is that of perceived urgency. In order to do this matching successfully, it is important to know the nature and the power of the effects of many spectral and temporal parameters. Three experiments are reported in which the effects of number of repetitions, warning speed and length upon perceived urgency were investigated. Results show that increases in all three parameters individually increased the perceived urgency of the stimulus. A fourth experiment tested a set of stimuli which varied in both number of repetitions and speed, whilst the length was held constant. This showed that large differences in the perceived urgency of a warning can be achieved over a fixed period of time. Steven's Power Law (1957) was applied to the data, enabling the power of the relationship between the objective value of the stimuli (number of repetitions, speed or length) and the subjective values of the warnings (the perceived urgency) to be quantified.

SOMMAIRE

La conception d'avertisseurs sonores fondée sur les propositions de Patterson, permet d'établir une certaine relation entre le son avertisseur lui-même et la réponse subjective qu'il engendre. Une relation entre le son avertisseur et la réponse subjective qu'il provoque peut être établie à partir d'un paramètre important à savoir la perception de l'urgence. Afin d'établir cette relation, il est important de connaître la nature et la force des effets de plusieurs paramètres spectraux et temporels. Trois expériences portant sur l'effet du nombre de répétitions, de la vitesse et de la longueur du signal sur la perception de l'urgence sont rapportées. Les résultats démontrent qu'un accoissement sur l'un ou l'autre des paramètres étudiés se traduit par un accroissement de l'urgence véhiculée par le signal. Une quatrième expérience avait pour but de tester un ensemble de stimuli qui variaient à la fois en fonction du nombre de répétitions et de la vitesse, alors que la durée du signal était maintenue constante. Les résultats démontrent que des différences majeures dans la perception de l'urgence d'un avertisseur sonore peuvent être mesurées même si la durée de l'avertisseur est maintenue constante. La loi de puissance suggérée par Stevens a été appliquée aux données dans le but de quantifier la relation de puissance entre les paramètres physiques des stimuli (nombre de répétitions, vitesse et durée) et les valeurs subjectives (perception de l'urgence).

Auditory warnings have found their way into many working environments and have obvious advantages over visual warnings. However, there are many problems associated with the use of auditory warnings; many are too loud, aversive and insistent and often too many warnings are found in one working environment. It is possible, however, to predict the appropriate level for many warnings to ensure that they are neither too loud nor too quiet. It is also possible to design warnings that do not startle and which are non-aversive (Patterson, 1982). Warnings of this sort inform, rather than alarm. Sets of warnings based on Patterson's guidelines for ergonomic auditory warning construction have now been made for helicopters (Lower, Wheeler, Patterson, Edworthy, Shailer, Milroy, Rood & Chillery, 1986), fixed-wing aircraft and hospitals (Patterson, Edworthy, Shailer, Lower, & Wheeler, 1986).

The method of construction of these advanced warnings is as follows: first, the appropriate level for the components of the warnings is predicted using theoretical notions based on the auditory filter (e.g. Patterson, 1976). When this is done, individual pulses of sound lasting from 100-300 ms in length are designed, one of which will form the basis of each warning required. These pulses then form the building-block of an auditory warning burst, in which the pulse is played several times, at different pitches, amplitudes and with different time intervals between the pulses. The burst typically lasts from one to two seconds, and is like an atonal melody with a rhythm. Several bursts are then assembled, together with silent intervals, to form a complete warning. The resulting warnings are non-aversive and allow people to communicate at the very time that communication is vital.

Another advantage of warnings created in this way is that a degree of matching can take place between the warning sound itself and the situation which it is signalling. This matching could take place along many subjective parameters, one of which is urgency. Auditory warnings are intended to alert the hearer; if it is possible to convey the degree of urgency with which the hearer should respond, or to convey the different priorities of urgency when many warnings go off together (which sometimes happens), then a large step forward in auditory warning design will have been made. The level of urgency can only be conveyed through the warning sound itself, so it is essential to establish these design principles at an early stage in work of this sort. At Polytechnic South West we are beginning to formulate these principles by carrying out a range of experiments on perceived urgency and other subjective parameters important in auditory warning design.

One of the cues contained in many of the complex sounds in our environment is an indication of urgency. We can even tell when a situation is increasing in its urgency by several cues such as increasing loudness, pitch, speed and so on. It is possible to make changes of this sort in warnings designed in the way described by Patterson (1982), as the type of warnings produced lend themselves to change in loudness, speed, pitch and other temporal and musical parameters which lead to changes in subjects' assessment of their urgency. This should enable some matching to take place between auditory warning and the situation for which it is designed.

The first part of our research program was to take many of the most important sound parameters used in auditory warning design (e.g. pitch, speed, pitch range, rhythm, number of repetitions, harmonic content and so on) and to establish the nature of the relationship between objective and subjective variables. We confirmed many intuitive opinions about the relationship between the sound parameters and perceived urgency. For example, we found that an increase in pitch increased the urgency of a stimulus. Such findings are demonstrated elsewhere (Edworthy, 1988; Edworthy, Loxley, Geelhoed & Dennis 1989; Edworthy, Loxley, Geelhoed & Dennis 1988). Many of our findings confirm received opinion, but some of our findings were counterintuitive as has been shown by earlier work (Halpern, Blake & Hillenbrand 1986).

We are now charting the relationship between objective change in the most important sound parameters and subjective change in urgency in more detail. The work reported here summarises four experiments relating three temporal variables to perceived urgency. These are three variables which are often naturally confounded; number of repetitions, speed and overall warning length. Our hypotheses, based on our earlier findings and our own intuitions, was that often-repeating, fast warnings (with a necessarily long length) will be perceived as more being urgent than slow, non-repeating warnings. We expected a gradation of urgency between these two extremes. Our experiments show this to be the case. In each experiment, a number of levels of the parameters under investigation were played to subjects, who were asked both to rank and to rate the urgency of the stimuli. This allowed us not only to confirm the predicted order of urgency, but to describe the power of the relationship between objective change and subjective change within the framework of Steven's Power Law (1957). Steven's Power Law takes the form

$$S = kO^{m}. \qquad (1)$$

S is the subjective value, O is the objective value, and k and m are the intercept and the slope of the line of best fit drawn when the logarithmic values of objective and subjective values are plotted as xy coordinates. It is the usual method of describing the objective/subjective relationship. In the case of perceived urgency, the slope of the line (m) indicates the ability of the parameter under investigation to produce changes in perceived urgency - the steeper the slope, the more effective the parameter should be. Deriving the power function will indicate more formally the degree of change in each of the parameters that is required to produce a predetermined change in perceived urgency. In addition, the power functions derived should enable us to make comparisons between parameters in order to establish which changes would be most effective in increasing the urgency of a warning. For example, merely lengthening a warning may make it more urgent, but it might be more effective to hold the length constant and communicate urgency through another parameter. For this reason, we imposed a 2.5 second limit on the stimuli.

In each experiment, a multiple comparisons procedure (Gulliksen & Tucker, 1959) was carried out whereby each stimulus was effectively compared with every other, producing paired-comparisons data without the usual fatigue problems. The consistency of subjects' responses is measured by a corrected form of Kendall's W (W') which is calculated by examination of the number of inconsistent responses given by each subject. An inconsistency occurs when a subject ranks Stimulus C as more urgent than Stimulus A, when they have previously ranked Stimulus A as being more urgent than B, and B as more urgent than C. W' has a maximum value of 1, which indicates completely consistent performance. Subjects were also asked to carry out a magnitude estimation task on the urgency of the stimuli previously ranked, from which the power functions (the relationship between objective and subjective change) were derived.

METHOD

Subjects.

Twelve subjects, two males and ten females, participated in this study. All of the subjects were undergraduate psychology students whose ages ranged from 18 to 27 years, none reported having a history of hearing problems. The study was a within-subject design run in two half-hourly sessions which were held two weeks apart. In the first session subjects completed Experiments One and Two, in the second session, Experiments Three and Four. The order in which the two experiments were presented in each session alternated between subjects.

Apparatus.

The experiment was conducted in a two-roomed sound-attenuated laboratory. In Laboratory One a Tandon microcomputer linked to a Cambridge Electronic Design 1401 interface and 1701 low-pass filters set at a cut off of 4 kHz was used to produce the experimental pulses and bursts. Laboratory Two housed a BBC B microcomputer which controlled the experimental design. The two computers were connected via a serial line.

Stimuli.

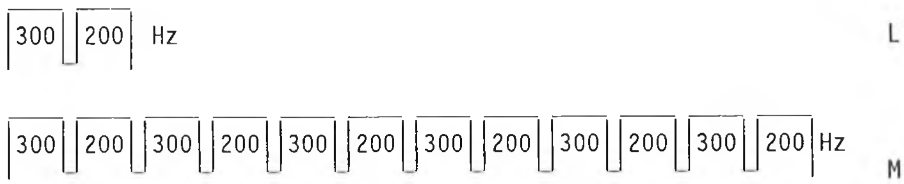
A single experimental pulse lasting 200 ms with a 20 ms onset and offset envelope was used in all four experiments as the basis of the experimental bursts. This pulse possessed a fundamental frequency of 300 Hz with 15 regular harmonics. Bursts lasted approximately 2 seconds in length and were played at the same loudness level. Seven bursts were generated for each experiment (Figure 1). They were constructed in the following way:

EXPERIMENT ONE (Number of repetitions): Two pulses, the first played at 300 Hz, the second at 200 Hz, represented one unit of repetition. The seven bursts contained 1, 2, 3, 3.5, 4, 5 and 6 such units. The stimuli were thus different total lengths.

EXPERIMENT TWO (Speed): The inter-pulse intervals of the seven stimuli were as follows: 950, 475, 237, 118, 59, 29 and 9 ms. All stimuli were approximately the same total length, with each pulse played at a fixed fundamental of 300 Hz.

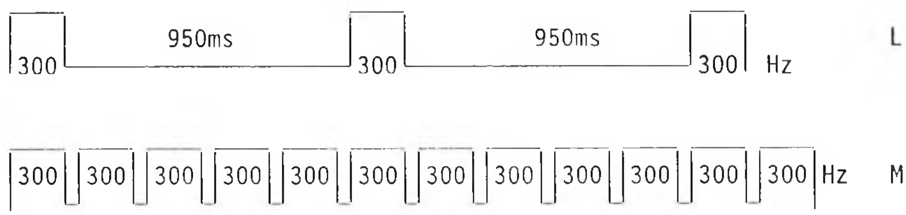
Figure 1: Least and most urgent experimental stimuli

Experiment 1



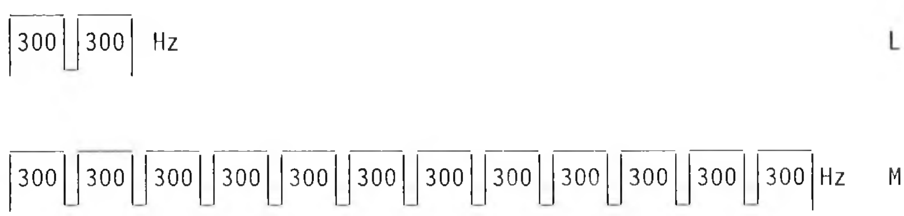
5ms gap between pulses for both L and M

Experiment 2



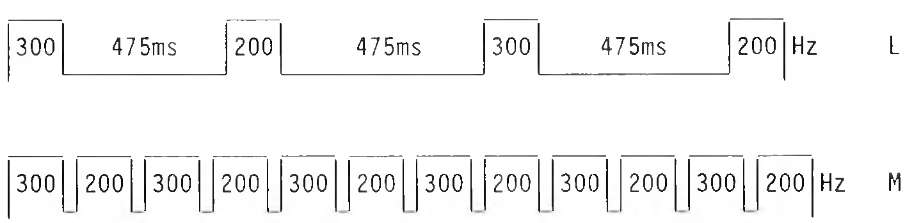
9ms gap between pulses for M

Experiment 3



5ms gap between pulses for both L & M

Experiment 4



9ms gap between pulses for M

L = Least urgent

M = Most urgent

EXPERIMENT THREE (Length): The number of pulses in the seven bursts was 2, 4, 6, 7, 8, 10 and 12, resulting in stimuli that became increasingly longer. All pulses were played at a fixed frequency of 300 Hz, and at a fixed speed.

EXPERIMENT FOUR (Number of repetitions and speed): The seven bursts varied along two parameters, speed and number of repetitions. The number of repetitions/speed combinations were as follows: 2/475 ms, 2.5/300, 3/237, 4/118, 4.5/59, 5/50 and 6/9. All stimuli were approximately the same length, and the basic unit of construction was the two-pulse 300 Hz/200 Hz unit described in Experiment One. The values of the repetitions and interpulse intervals used were not identical to those in Experiments One and Two in order to minimise the use of incomplete units of repetition.

Procedure.
Subjects were required both to rank and rate the bursts according to their urgency. The procedure was identical for all four experiments. In each trial they heard three bursts separated by a gap of approximately 1 second, presented at a comfortable, fixed level. They were required to select the most, and then the next most urgent of the three bursts. Seven trials were presented to each subject, during which each burst was compared with every other. The rankings stage of the experiment was over when subjects had completed all seven trials. In the ratings stage of the experiment, a magnitude estimation task was employed. Subjects were given the following instructions: "Imagine the most urgent sound possible. Let this represent 100 on an urgency rating scale. Let 0 represent the other end of the continuum (the most non-urgent sound that you can imagine). You are required to assign a number between 0 and 100 to each of the following sounds". They were asked to preserve ratios between urgencies in the numbers that they assigned to the stimuli.

RESULTS

Mean rankings (from 1, the most urgent to 7, the least urgent) of the stimuli for all twelve subjects are shown in Table 1. These were ranked in the predicted order - stimuli which contained more units of repetition, and/or were faster or were longer, were assigned higher rank orders, that is, two of the stimuli were not ranked exactly as predicted.

TABLE 1: Mean rankings of stimuli, Experiments One to Four.

EXPT. 1	REPETITION RATE (IN UNITS)							W'
6	5	4	3.5	3	2	1		
RANKING.	1	2	3	4	5	6	7	0.849
EXPT. 2	SPEED (INTER-PULSE INTERVAL)							
9	29	59	118	237	475	950		
RANKING.	1	2	3	4	5	6	7	0.843
EXPT. 3	LENGTH (M. SECS)							
2455	2045	1635	1430	1225	815	405		
RANKING.	1	2	3	4	5	6	7	0.890
EXPT. 4	REPETITION RATE (IN UNITS)							
6	5	4.5	4	3	2	1		
	SPEED (INTER-PULSE INTERVAL)							
9	50	59	118	237	300	475		
RANKING.	1	2	3	4	5	6	7	0.939

In all four Experiments Kendall's coefficient of concordance, W', was high and very significant. For Experiment One, F = 61.744, (df = 5,64, p <.001); for Experiment Two, F = 59.212, (df = 5,64, p <.001); for Experiment Three, F = 88.941, (df = 5,64, p <.001); and for Experiment Four, F = 167.927, (df = 5,64, p <.001). There was thus a high level of agreement between subjects as to the rank orderings of the stimuli. Each subject's ranking of each stimulus was correlated with their magnitude estimation (rating). The means of these correlations are shown in Table 2.

TABLE 2: Mean Spearman correlations between rankings and ratings.

EXPT. 1 Ranking v Rating	EXPT. 2 Ranking v Rating	EXPT. 3 Ranking v Rating	EXPT. 4 Ranking v Rating
0.841	0.855	0.790	0.940

The high correlations between subjects rankings and ratings of the stimuli indicated that the magnitude estimation task had produced a reliable measure of perceived urgency. As in the rankings task, stimuli that contained more units of repetition, and/or were faster or were longer, were perceived as being more urgent.

In the final stage of analysis the ratings data from Experiments 1-3 was fitted to Steven's Power Law (Figure 2). This requires a logarithmic transformation of the data. For Experiment 1 the objective values of the stimuli were calculated by taking the logarithm of the number of repetitions. In Experiment 2 the logarithm of 2500 (maximum stimulus length in milliseconds) divided by the inter-pulse intervals provided an objective measure of stimulus speed, giving the number of pulses possible in a unit of time. Here, smaller figures indicate a slower speed. In Experiment 3 the logarithm of stimulus length was calculated to represent the objective value. In all three experiments the mean subjective value plotted for each stimulus was the logarithm of its magnitude estimation. The data from Experiment 4 was omitted from this analysis because the stimuli varied along two objective parameters, repetition rate and speed.

Application of the Power Law quantified the relationship between the objective values of the stimuli (number of repetitions, speed, or length) and the subjective values (perceived urgency). Thus for Experiments One, Two and Three:

$$\text{Perceived urgency} = 22.13 \ \text{Number of repetitions}^{0.69} \tag{2}$$

$$\text{Perceived urgency} = 18.32 \ (2500/\text{pulse-to-pulse time})^{0.61} \tag{3}$$

$$\text{Perceived urgency} = 1.65 \ \text{length}^{0.49} \tag{4}$$

Repetition rate and speed are therefore the stimulus parameters that are most powerfully related to perceived urgency. A linear regression showed that the data from all three Experiments was well fitted by a straight line when plotted in log-log co-ordinates, as predicted by Steven's Power Law. The percentages of variance accounted for by the straight line were 99.8% (Expt. 1); 99.7% (Expt. 2) and 99.0% (Expt. 3).

DISCUSSION

Our experiments show that, individually, the number of repetitions, the speed and the length of an auditory warning each have individual effects on perceived urgency. Increasing the number of repetitions without increasing the speed (although necessarily increasing the length) of a warning increases its urgency; increasing the speed of a warning without increasing either the number of repetitions nor the length of a warning also increases its urgency; and simply increasing the length of a stimulus appears to increase its urgency. In practice, these three variables would tend to be confounded - indeed there is no way to separate number of repetitions from overall stimulus length if speed is held constant - but our experiments separate the parameters as far as possible. Moreover, the high values of W' for the ranking tasks suggests that subjects are very sure about the direction of

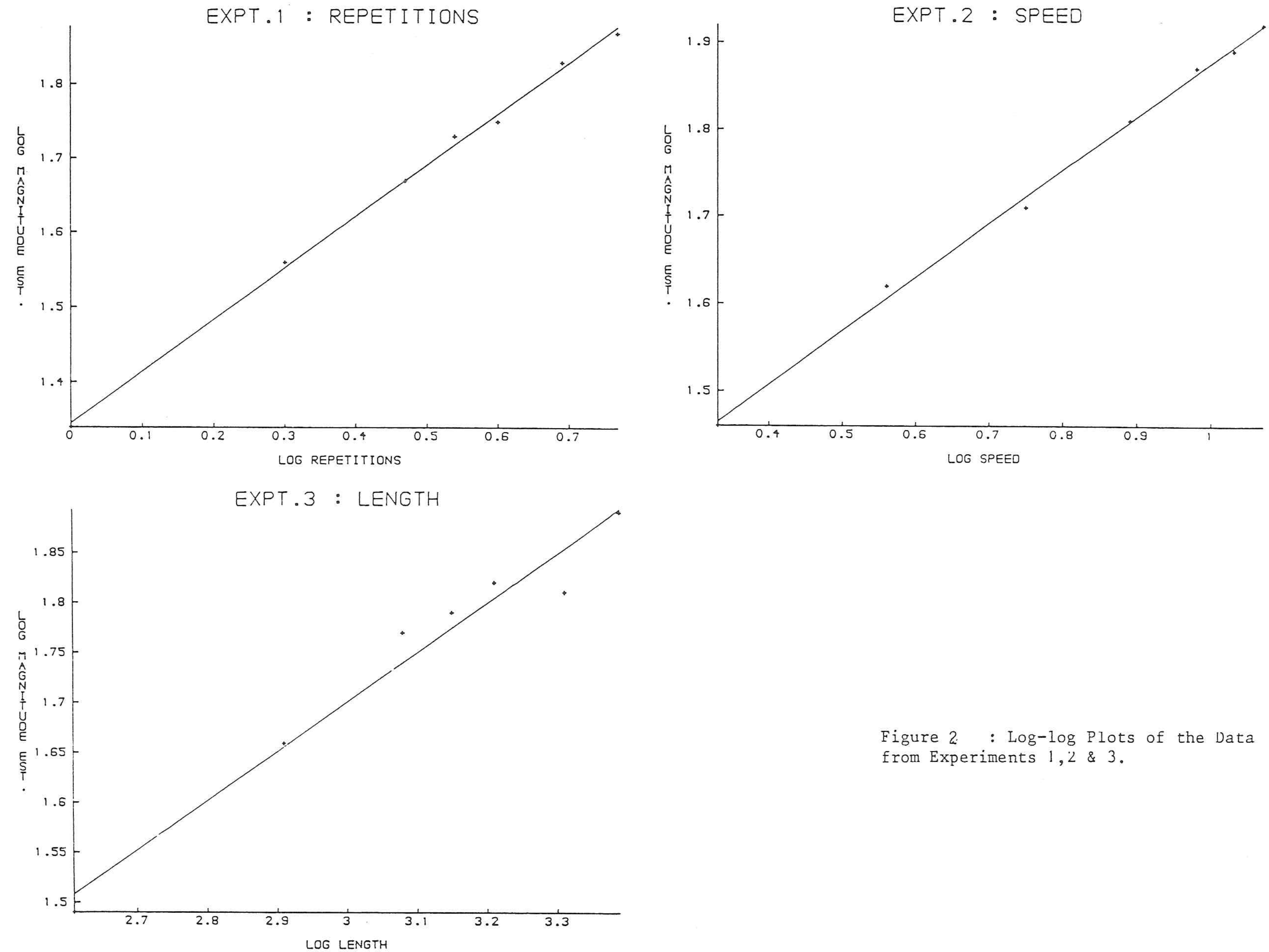


Figure 2 : Log-log Plots of the Data from Experiments 1,2 & 3.

change in perceived urgency for all three parameters. The power functions derived for the three parameters suggest that the number of repetitions and the speed of the warning are more powerful in producing changes in perceived urgency than is a simple length change. When all three parameters are combined (Experiment Four), subjects are even more sure about the direction of the change in perceived urgency as evidenced by the value of W' of .939, the highest for the experimental series.

The high correlations obtained between the rankings and the ratings suggests to us that magnitude estimation is an effective way of measuring perceived urgency. Other experimenters have supported the use of magnitude estimation techniques (for example, Haverland, 1979; Kowal, 1987).

These results suggest that the perceived urgency of an auditory warning is readily manipulated when the warnings are of the pulse-burst variety proposed by Patterson (1982), and the power functions begin to suggest the contributions of individual sound parameters to the overall urgency of a warning, which may be of use to the manufacturer with only one or two parameters at his or her disposal. Our results will eventually lead to a set of design principles for use in advanced auditory warnings work.

We are continuing our work on auditory warning design by looking in more detail at the measurement of the objective/subjective relationship and at auditory signals other than warnings such as auditory icons (e.g. Gaver 1989) where urgency is no longer the most important dimension, but where the signal is an analogue of a mechanical or other function.

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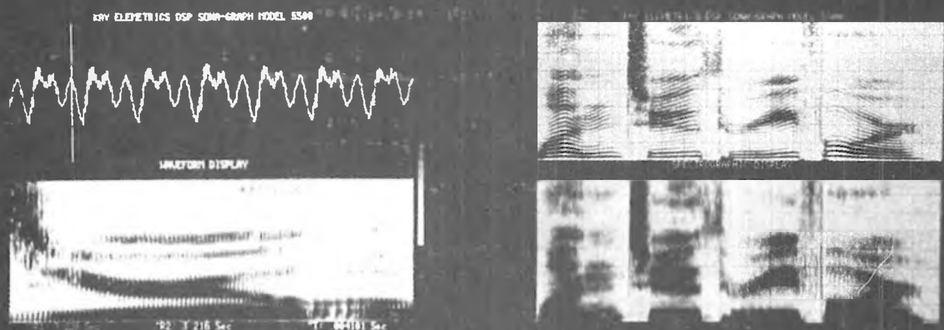

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A RECIPROCITY BASED TECHNIQUE FOR INVESTIGATING CONTRALATERALLY STIMULATED OTO-ACOUSTIC EMISSIONS

K. R. Tough and H. Kunov
Institute of Biomedical Engineering
University of Toronto
Toronto, Ontario, M5S 1A4

ABSTRACT

The existence of the olivo-cochlear efferent neural system has led to theories of feedback mechanisms in the auditory system, possibly involving the contralateral ear, and exerting physiological control over cochlear micromechanics. Oto-acoustic emission phenomena can be used as a sensitive method of monitoring cochlear micromechanical properties in the search for such a feedback mechanism. One such phenomenon is stimulus frequency emission, well documented ipsilaterally but as yet undocumented contralaterally. A novel technique relying on the principle of acoustic reciprocity was designed for detecting the existence of any active physiological source at the frequency of a contralateral acoustic stimulus. The results of preliminary experiments carried out on normally hearing subjects with and without spontaneous oto-acoustic emissions (SOAE) do not indicate obvious correlation between the deviation from passive linearity and SOAE activity. The results do not support the hypothesis of contralateral emissions with levels significantly greater than the body conducted component of the stimuli levels used here. The experimental results, however, are limited by the sensitivity of the reversible transducers used.

INTRODUCTION

There is increasing evidence that efferent fibres entering the cochlea (i.e., nerves conducting signals towards the cochlea from the brainstem) form tonotopic connections between the cochleae (Fex, 1962; Robertson, 1985; Warr & Guinan, 1979). The afferent neural responses to stimulation of these fibres is also well documented (Klinke, 1969; Murata, 1978; Bonfils, 1986; Gifford, 1987). The crossed olivo-cochlear bundle, where the cochlear nerves meet, has provided a basis for theories of a contralateral feedback mechanism actively controlling cochlear micromechanics (Fex, 1965; Mountain, 1980; Siegel, 1982; Brown, 1983; Bonfils, 1986; Kim, 1986).

Because of their cochlear origin and susceptibility to changes in the micromechanical behaviour of the basilar membrane in the cochlea, oto-acoustic emissions can be used to monitor any efferent induced changes in the cochlea. For example, Mountain (1980) used distortion product emissions in guinea pigs

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to show a change in cochlear nonlinearity caused by electrical stimulation of the crossed olivo-cochlear bundle. He introduced two tones (f_1 and f_2), and measured the distortion product $2f_1 - f_2$.

Stimulus Frequency Emissions (SFE)

There is considerable evidence supporting the existence of sounds generated within a cochlea at the same frequency as a tone of long duration stimulating that cochlea (Kemp, 1979; Kemp & Chum, 1980; Wilson, 1980, Whitehead, 1986). The ipsilateral cochlea acts as an active source of sound at the same frequency as the stimulus, adding vectorially to the sound field present in the ear canal. Because the emitted sound amplitude is generally less than the stimulating tone amplitude while the frequencies of the signals are identical, special methods must be used to separate SFE from the stimulus, based on phase variation and nonlinear amplitude saturation of the emission. This detection problem, separating stimulus from emission, is similar to the problem of detecting a contralateral effect.

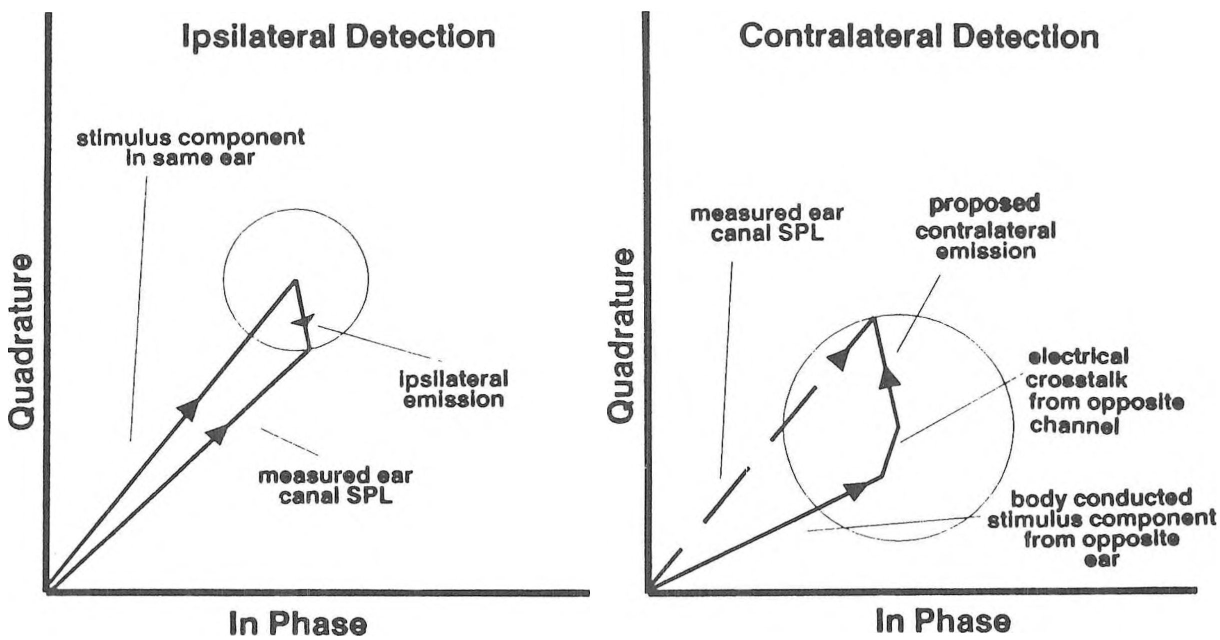


Figure 1. The problem in detecting Stimulus Frequency Emission (SFE) phenomena is separation of the stimulus component from the physiologically generated component in the measured signal.

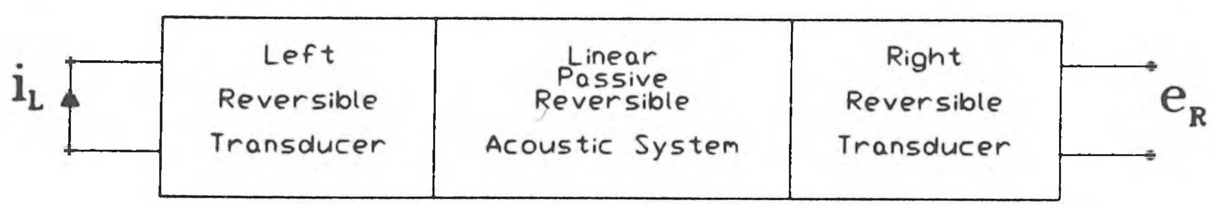
Direct measurement of any contralateral active emission is probably impossible because an unknown portion of the stimulus will be transmitted between the ears through body conduction (See Figure 1). In the frequency range of interest here (250 to 4,000 Hz) the attenuation of the body conducted signal is between 40 and 70 dB. At a given frequency the stimulus and the body conducted signals will, however, have a fixed phase and amplitude relationship. The activation threshold of efferent neurons to acoustic stimuli in animals has been shown to vary, with most units having thresholds above 40 dB SPL, and many with thresholds between 60-70 dB SPL (Fex, 1965). Thus, the most effective stimuli levels will generally be detected to some degree in the opposite ear. As an example, Buno (1978) indicated that contralateral acoustic stimuli levels above 30 dB SPL may have been more effective in influencing cat auditory nerve activity, but the resulting body conduction masked primary neuron responses in the opposite ear. A method is thus required for separating the acoustically transmitted contralateral sound from any possible emission activity generated by the contralateral ear at the stimulus frequency.

Acoustic Reciprocity

The reciprocity theorem states simply that at two terminals of any passive, linear, reversible system (as represented in Figure 2):

$$i_L/e_R = i_R/e_L \qquad (1)$$

where i_L is a current source on the left side, and e_R the open-circuit voltage on the right side. The right side of Equation (1) represents similar parameters when the current source and open circuit have been transposed.



$$\frac{i_L}{e_R} = \frac{i_R}{e_L}$$

Figure 2. Reciprocity principle demonstrated in a linear, passive, reversible system.

In the emission detection method presented here, **i** represents the driving current entering the terminals of a linear reversible transducer used as a sound source, and **e** is the open circuit voltage appearing at the transducer terminals when the same transducer is used as a microphone. Any acoustic system which contains non-reversible, active or nonlinear elements may not comply with (1). Because SFE are the result of active sources within the cochlea, they can be indicated by deviations from the identity (1). It is important to realize, however, that violation of (1) due to nonlinear or active elements, when $\mathbf{i}_L$ and $\mathbf{i}_R$ are the same, requires asymmetry with respect to the two system terminals. Reciprocity thus detects bilateral asymmetry in the active and nonlinear properties of the hearing system. This can be a benefit of the method as symmetrical sources of nonlinearity (such as similar middle ear reflexes) do not affect the results.

Application of the Reciprocity Method

We can use the reciprocity method to measure the acoustic transfer function from one electroacoustic transducer in one ear to a similar transducer in the other. By comparing measurements made both ways, we can show any difference from a passive, linear, reversible system.

The reciprocity based method presented here detects deviation from the passive, linear case, when one ear is stimulated as compared to the case when the other ear is stimulated. Thus, the result from two ears is a single curve showing deviation from passive linearity as a function of frequency. If there was no asymmetrical nonlinearity (a unilateral acoustic stapedius reflex is an example of this type), the curve would indicate the difference in the source activity between the ears. A curve that was not significantly different than zero would be evidence that no contralateral SFE can be detected at those stimuli levels. The curve is ideal for correlating with differences in other frequency varying aural properties, for example, spontaneous oto-acoustic emissions (SOAE). These SOAE are sinusoidal vibrations produced within the cochlea, and transmitted outward towards the ear drum. They generate measurable tones in the ear canal, with frequencies of 1 - 3 kHz and levels of -20 to +20 dB SPL.

Subjects with SOAE in one ear were used to validate asymmetries detected with the reciprocity technique. The relationship between SOAE and ipsilateral SFE was studied by Zwicker & Schloth (1984), who indicated that the two types of emissions are similar in nature and may be closely related. An SFE maximum was found at the SOAE frequency, and spacing in the SFE maxima closely resembles the spacing of SOAE in subjects with multiple emissions. Thus, it was assumed here that contralateral emission activity, if present, may have a local maximum at frequencies close to the SOAE in that ear.

If SFE are evoked in the contralateral ear as a result of the reciprocity test stimulus, they could be due to either body conducted sound, efferent neural activity, or a combination of both. This method will not differentiate between these factors; other methods must be used to investigate which factor is most important.

METHODS

1. Two identical insert probes were constructed, each containing a linear reversible acoustic transducer (Knowles subminiature magnetic speaker) and fitted with flexible rubber ear tip for sealing within the outer ear canal. A computer driven switching circuit allowed either the open circuit voltage (Microphone operation) or driving current (loudspeaker operation) from each probe to be monitored on a separate channel of the Bruel & Kjaer 2032 dual channel signal analyzer (Figure 3).

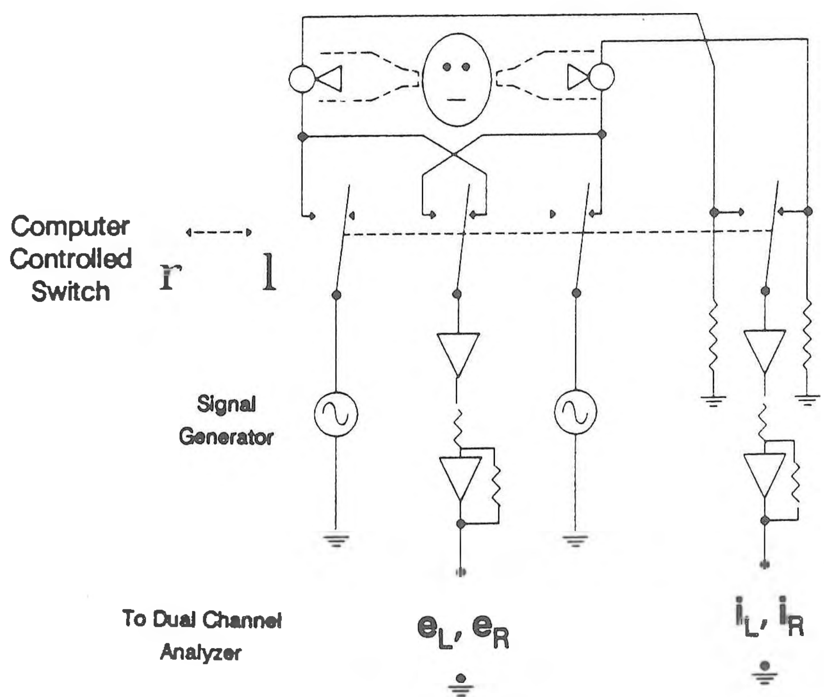


Figure 3. Schematic of reciprocity measurement apparatus.

2. The spontaneous emission activity in each ear was recorded by simultaneously performing spectral analysis on the signal from each probe when used as a microphone. The spectra were found using a Fast Fourier Transform technique. Spontaneous emissions are noted as narrow band frequency spikes >5 dB above the ear canal noise spectrum. A second probe with miniature electret microphones was also used, providing increased sensitivity to -20dB SPL (re 20 μPa)

3. The sound source probe had its driving voltage adjusted until the current input matched that determined in a previous calibration run. This probe's current (i_L) and the opposite probe's open circuit voltage (e_R) were then simultaneously recorded by computer with the ratio (i_L/e_R) calculated for several averages.

4. A rest period (10 seconds of silence) allowed the stimulated ear to recover from the effects of the tonal stimulus. This was intended to reduce any masking, adaptation, or suppression effects.

5. The circuit is then automatically reversed with a tone generated in the former microphone probe and picked up in the probe which was previously the source. The new ratio (i_R/e_L) is similarly calculated; in a passive linear system (i_L/e_R)-$(i_R/e_L) = 0$.

6. Steps 3,4 and 5 were repeated for all the desired stimulus frequencies.

Stimulus Amplitude and Frequency

The stimulus frequency range was chosen as 1000-2000 Hz, as SOAE occur most often within this range. The upper frequency limit was set at 2000 Hz as a large increase was noted in interaural attenuation beyond this range. The stimulus current driving the loudspeaker at each frequency was selected so as to give relatively uniform levels of body conducted sound in the contralateral ear, across the stimulus frequency range. The current i is thus a function of probe acoustic load impedance and inter-aural attenuation and varies with frequency. For this reason, the SPL in the stimulated ear was not held constant. At a specific frequency, however, the SPL in the stimulated ears will be nearly the same. Any difference in stimulated ear SPL is a result of asymmetry in aural acoustic load between the right and left ears, resulting from different ear canal geometry, for example. To account more accurately for amplitude nonlinearity the stimuli could be adjusted to give the same SPL in each ear. The first experiments were simplified by assuming such differences would be small.

The reciprocal nature of the system was verified with the probes coupled by an air filled tube. This test was repeated with the cavity filled with sufficient attenuation to simulate conditions between the ears. Deviation from passive linearity in this case is shown in Figure 4. Ideally, these data should be zero for all frequencies.

Electrical Noise and Artifacts

The background noise for each microphone measurement was measured at the time of spontaneous emission detection. This allows a measurement to be discarded if it falls "beneath the noise floor". This noise includes a component resulting from slight electrical cross-talk between the channels, present even after scrupulous electrical grounding and shielding. This component was recorded by monitoring the amplitude and phase of the signal present in the microphone probe during stimulation while that probe was removed from the ear canal. This small component was later subtracted vectorially from the total signal measured with the microphone probe in the ear.

The unavoidable random (thermal) noise generated within the probe microphones, and also within the head and auditory system, will affect the measurement. This effect on the results is again limited to the degree of asymmetry in those factors.

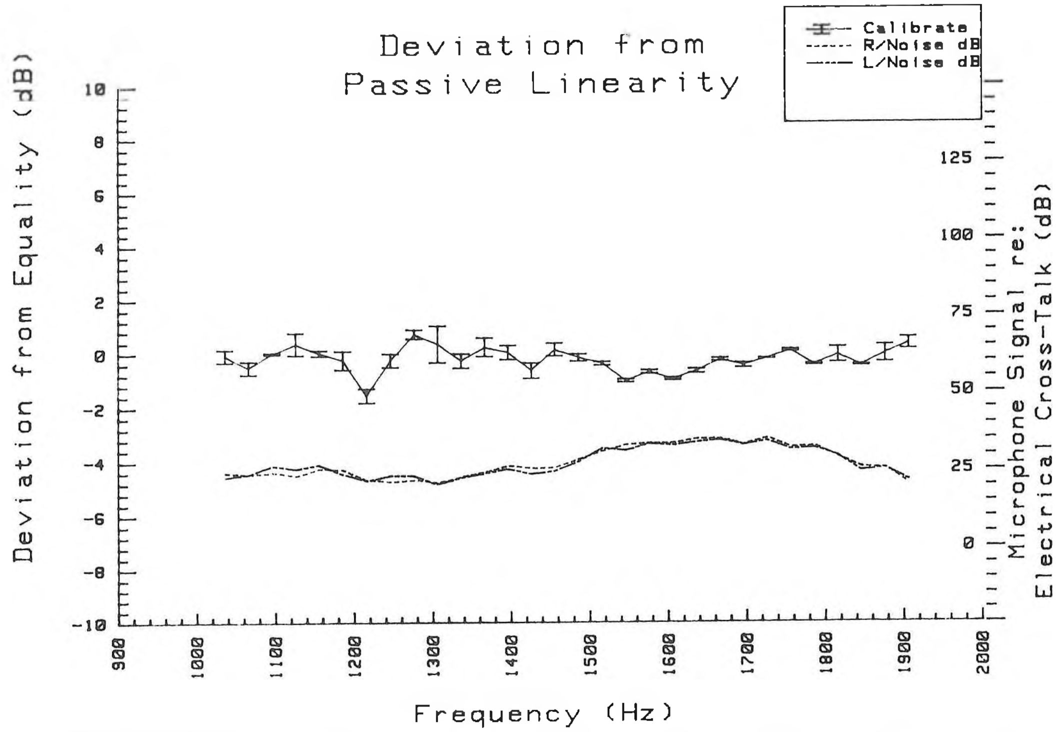


Figure 4. Deviation from ideal reciprocity in a passive linear acoustic system. Calibration measurement in a tube coupler containing attenuating foam. Upper curve indicates deviation and refers to left axis, lower curves indicate microphone probe signal relative to electrical noise and refer to the right axis.
Note: Calibration with sound source signals 25 dB lower than those used in human measurements.

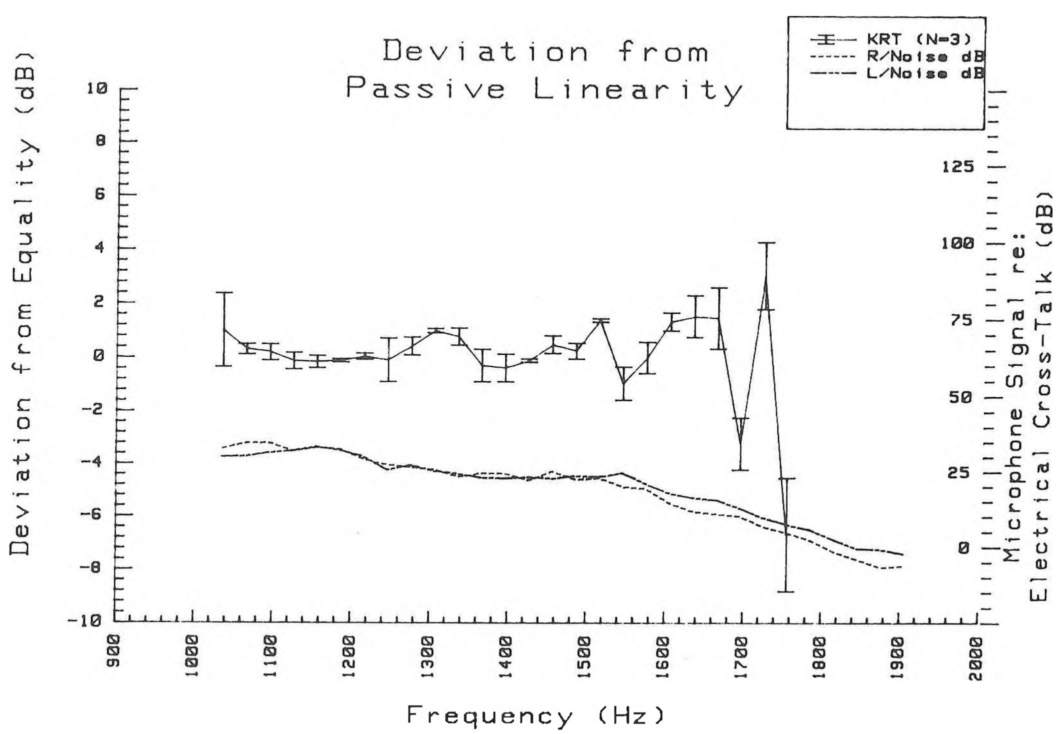


Figure 5. Deviation from passive linearity in one non-SOAE subject with three measurements performed at each frequency. Lower curves indicate microphone probe signal relative to electrical cross-talk noise and refer to right axis. Frequencies at which points on the lower curves are below 5 dB are excluded from the upper deviation curve.

RESULTS

The experiment was performed on 10 normally hearing subjects. Of these subjects, four had no history of SOAE, four had SOAE activity at the time of measurement, and two had a history of SOAE but no activity at the time of measurement. Results of a measurement on one non-SOAE subject are shown in Figure 5. Each data point is the average of three reciprocity measurements at that frequency. Compared with the calibration run (Figure 4), there is little significant deviation from passive linearity below 1600 Hz. The measurements above 1600 Hz do show deviation. However, even though the source amplitude was increased over this range, the levels of body conducted sound at these frequencies was significantly less than that at lower frequencies. The ratio of microphone signal amplitude to electrical cross-talk is indicated on a separate axes; it is possible that deviations at higher frequencies are due to poorer response of the measurement system at lower signal levels.

The results from subjects with SOAE are not obviously or systematically different from those of non-SOAE subjects. In most cases, passive linearity is best attained at those frequencies (< 1500 Hz) where interaural acoustic transmission is greatest. In this range, the stimulus SPL in the contralateral ear was approximately 40 dB. Ipsilateral SFE with a stimulus of this level can be expected at levels 30 dB below the stimuli (Kemp & Chum, 1980). This would contribute very little to deviation from the passive linear case; any deviation found would likely be due to nonlinearity.

The results do not obviously indicate contralateral stimulus frequency emissions at levels significantly greater than a body conducted component of 40 dB SPL. Apparent asymmetry at lower stimulus levels and different frequency ranges suggest further study with more sensitive reversible transducers is necessary before a separate contralateral mechanism can be rejected. Indeed, it a greater deviation at lower stimulus levels should be investigated. Such deviations are compatible with a saturating emission mechanism. Further improvements in signal-to-noise ratios may be necessary through better ageraging techniques before this can be achieved, however.

While the experiments presented here do not reflect significant finds, the important result of this paper is the establishment of a theoretically valid and novel method for separating body conducted sound from active emissions, which can be extended in further experiments.

RECOMMENDATIONS

The method can be used to compare the difference in contralaterally stimulated ear canal sound levels to the difference in SFE activity measured using conventional methods. Thus, one can investigate whether the body conducted component of the contralateral stimulus creates SFE in the same way as stimuli sent into the canal of the emitting ear. Because of the different mechanisms through which body conducted stimuli enter the cochlea, it can be expected that the SFE generated by such sounds may have very different properties than those produced by "forward travelling" stimuli entering the oval window from the middle ear. The method may thus provide new ways for investigating latency characteristics of the conventional SFE mechanism.

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1989, 17 (4): 23-35

CSRE: A SPEECH RESEARCH ENVIRONMENT

Donald G. Jamieson[1], Terrance M. Nearey[2], and Ketan Ramji[1]
[1]Speech Communication Laboratory
Hearing Health Care Research Unit
Department of Communicative Disorders
University of Western Ontario
London, Ontario
and
[2]Department of Linguistics
University of Alberta
Edmonton, Alberta.

ABSTRACT

CSRE (The Canadian Speech Research Environment) is designed as a comprehensive, integrated, inexpensive, micro-computer-based workstation to support speech research. The focus of the project has been to develop the basic functionality required by speech researchers, using mass-produced, widely-available hardware. In its present form, CSRE includes: 1) waveform editors to capture, cut, paste, and measure signals; 2) facilities to produce colour spectrograms and waterfall displays from sampled data using a variety of spectral analysis techniques; 3) a parametric digital speech synthesizer, based on that developed by Klatt (1980);888 4) and an experiment generator, developed to speed the programming of experimental sequences. The system requires an IBM/AT compatible computer (typically, Zenith Model 386 or Model 248, with math coprocessor and EGA or VGA graphics), a data acquisition system (e.g., Data Translation DT2801-A or ARIEL DSP-16, plus microphone, preamplifier and filters), and a Microsoft-compatible mouse.

RESUMÉ

Le CSRE, "Canadian Speech Research Environment" est un système complet et peu coûteux d'étude de la parole à partir d'un micro-ordinateur. Nous avons d'abord cherché à intérgrer les équipements de base disponibles sur le marché et accessibles aux chercheurs. Dans son état actuel, le CSRE comprend trois programmes d'analyse de formes d'ondes acoustiques, deux programmes d'analyse spectrale et un synthétiseur de parole fondé sur le programme de Klatt (1980). Le système recquiert un ordinateur IBM/AT (ou un ordinateur équivalent, tel que le Zénith modèle 386 ou 248), un dispositif d'acquisition de données (Data Translation DT2801-A ou ARIEL-DSP16), un microphone, un pré-amplificateur et des filtres ainsi qu'une souris Microsoft.

CSRE is designed as a comprehensive, integrated, inexpensive, micro-computer-based workstation to support speech research. Such research requires five basic facilities: 1) a facility to record, store, and play back high quality natural speech signals; 2) a facility to edit stored natural speech signals -- to eliminate unwanted portions of the signal and to concatenate (paste together) two different signals; 3) a facility to analyze and measure speech signals -- to accurately measure the duration and amplitude of different parts of the speech signal and to accurately measure the frequency and amplitude of critical portions of the speech signal; 4) a facility to parametrically synthesize speech signals; and 5) a facility to control speech output -- as for experiments or assessment procedures.

Our initial focus has been to implement these basic functions which speech researchers require, using mass-produced, widely-available hardware. Subsequently, we have sought to develop these functions within an interactive environment, in which researchers can move readily between alternative applications, and to improve the ratio of the user's productive "investigative" time to "clerical" time.

The system we have developed requires an IBM/AT compatible computer (typically, Zenith Model 386 or Model 248, with math coprocessor (Intel 80x87) and EGA (16 colors) or VGA graphics), a data acquisition system (e.g., Data Translation DT2801-A or ARIEL DSP-16, plus microphone, preamplifier and filters), and a Microsoft-compatible mouse. The components of the software package have now been tested extensively and the system is presently being distributed to all interested university-based researchers. The most important functions available within the package are described in the following sections.

COMPONENT PROGRAMS OF THE CSRE PACKAGE

I. WAVEFORM EDITORS

I.A. CAPTURE (the Waveform Capture Editor) is used to capture (digitize and edit) a portion of a speech signal. This signal may be recorded directly from microphone or tape input, using CAPTURE, or taken from a disk file. Mouse-driven menu commands are used to select functions -- for example to display a signal, to isolate the desired portion of the displayed signal, to play the selected signal, to cut the selected signal and save it to disk, and so forth (See Figure 1). The capture functions permit the user to display, in the window at the top of the screen, the entire signal which has been previously sampled or digitized by CAPTURE. The signal display includes time in milliseconds on the horizontal axis, and amplitude as coded by the ADC on the vertical axis, the name of the file being displayed, and the location of the starting and ending data points. Alternatively, the users may sample (record, digitize) a sound from an external source. After selecting this function, sampling will begin (and end) on the user's command (by pressing the mouse button to begin and end). At this point, the full signal may be "spoken"; alternatively, a window may be positioned on any portion of the signal in the top viewport and that part played back.

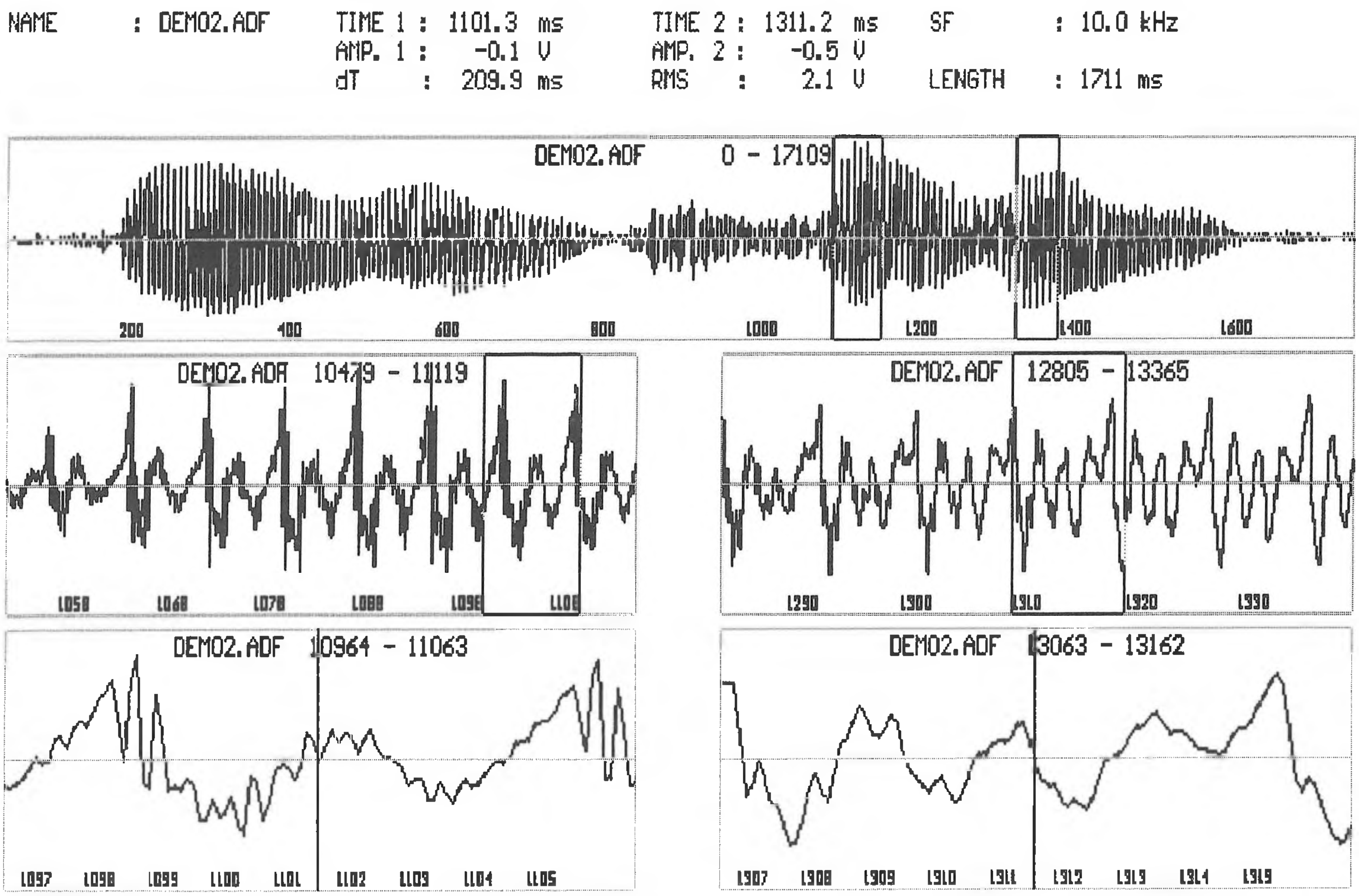
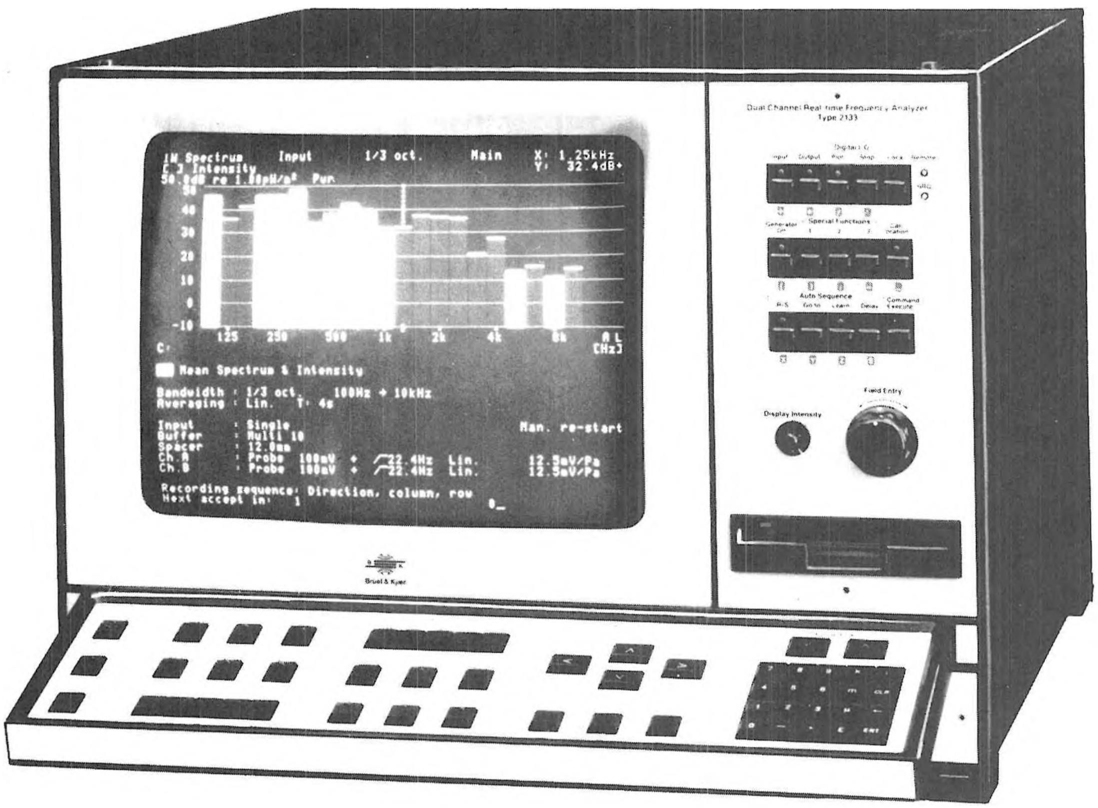


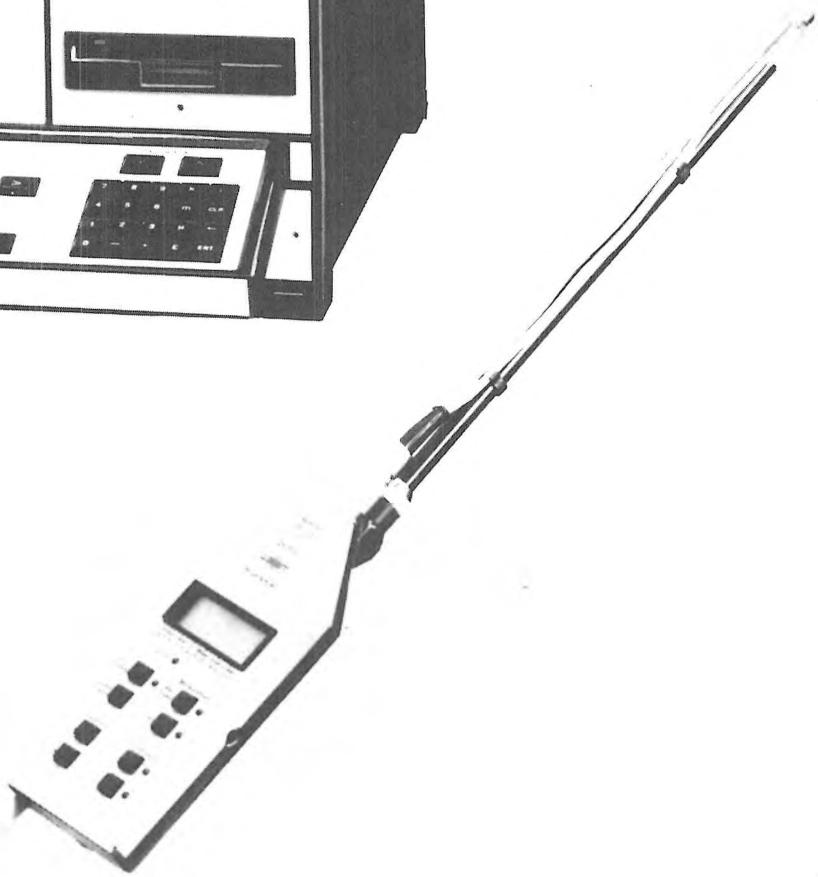
Figure 1. An example of the format of the CAPTURE screen when a signal is being displayed. Signals are initially displayed in the uppermost window, with the initial and final portions of the selected signal subsequently displayed at increasingly higher resolution within the lower four windows.

CAPTURE is designed specifically to capture (cut, or isolate) part of a signal. To do so, the user must indicate, precisely, both the beginning and the end of the signal to be captured (cut), by placing markers (cursors) within the primary signal -- i.e., the signal in the top window. A series of windows are used to indicate precisely where the **beginning** and **end** of the captured potion of the signal should be (middle windows) and to "fine-tune" both the beginning and the endpoints of the selected signal in preparation for cutting the signal (lower windows). As the user "tunes" the positions of the markers, current time/amplitude values are displayed in real time. When both markers have been specified, the display will also indicate the time difference and the RMS value in voltage of all points between these markers. To listen to the signal between the markers (what the user has "captured"), there is also an option to play back this portion. Finally, a **save** option permits the user to save the edited signal as a named disk file.

THE COMPLETE ACOU

Our state-of-the-art instruments do far more than measure the intensity of sound in any given situation. With this system, Brüel & Kjaer





The dual-channel real-time analyzer Type 2133: It has extensive data processing features along with built-in storage capabilities. One of two such analyzers created by Brüel & Kjaer, its measurements range from 1/3-octave dual-channel real-time analysis to 11 kHz (1/1-octave to 22 kHz).

I.B. PASTE allows user to concatenate portions of two digitized signals. Both original signals are displayed in the top and bottom windows respectively and both signals can also be played back. The user can zoom into a portion of the signal in the top viewport to indicate the approximate desired point at which the initial part of the signal should end. This magnified signal will appear in the middle window (left). The user can then fine tune the initial segment using a mouse-controlled marker. Similarly, the magnified final portion of the signal from the bottom window will also appear in the middle window (right) with a mouse-controlled marker used to fine tune the final segment. (See Figure 2). The concatenation will be such that all points from the beginning of the signal in the top window to the marker in the middle-left window will be pasted **in front** of all points beginning at the marker in the middle-right window to the end of the signal in the bottom window. Both individual signals and the concatenated result can be heard individually using the **playback** options. Finally, the **save** option permits the user to save the resulting concatenated signal to a named disk file.

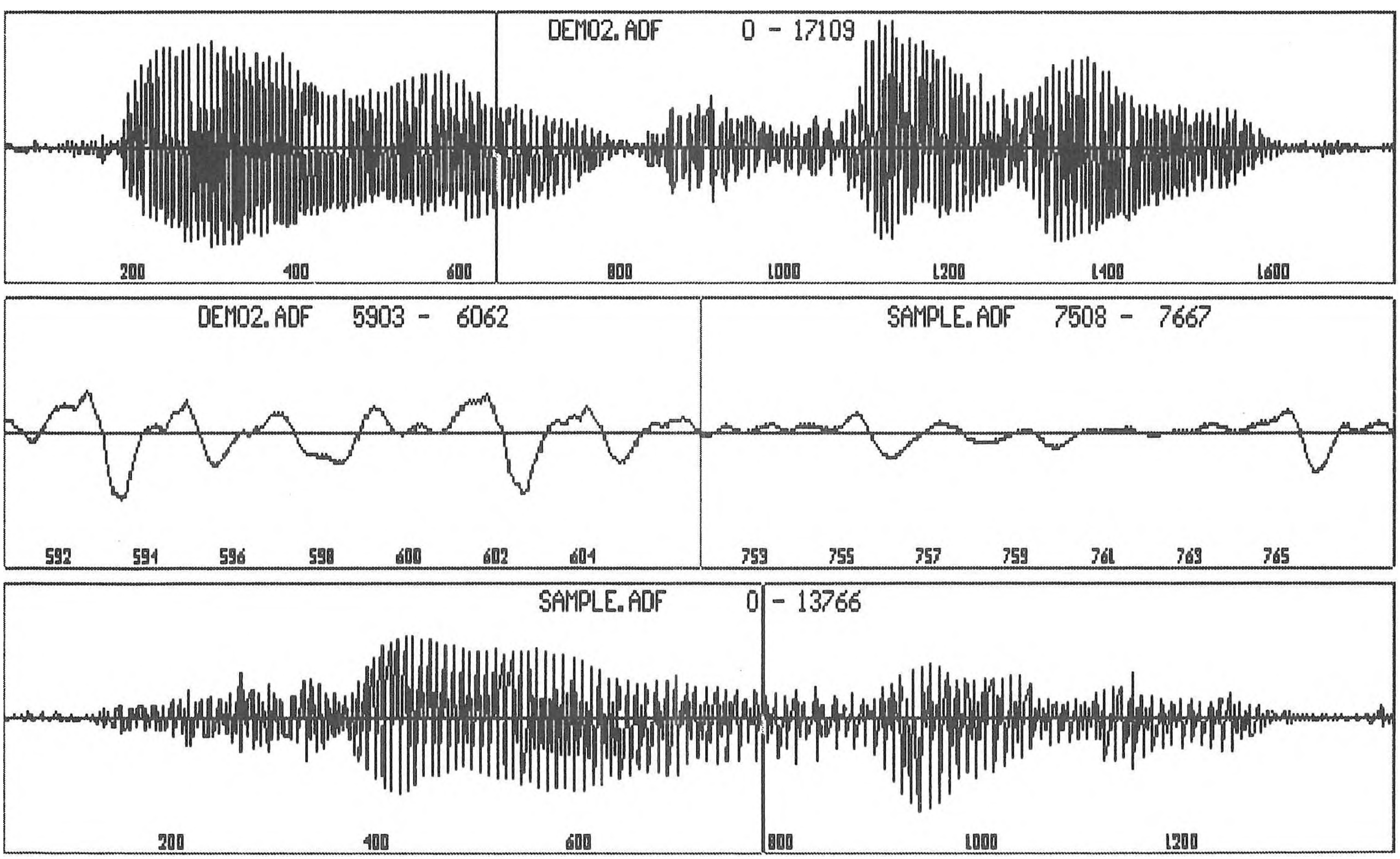


Figure 2. An example of the format of the PASTE screen used to concatenate two signals to form a third signal. The viewports in the middle of the display indicate the exact point at which the initial signal (from the top viewport) will be joined with the final signal (from the bottom viewport).

II. SPECTRAL ANALYSIS PROGRAMS

II.A. COLOR SPECTROGRAM. The Spectrogram program produces a perceptually-compelling time/frequency/amplitude display of a digitized signal, with time on horizontal axis, frequency on the vertical axis and amplitude as one of 16 possible colors to code the amplitude of the signal. The dB range represented by each color is shown as part of the display, and the range of dB values displayed can be altered by the user. The user is also offered several alternative mappings from intensity to color: a rainbow scale, a temperature (heat) scale, and a brightness scale. For users having a suitable monochrome monitor, a true gray scale is also available, together with a reverse gray scale (brightness). Inexpensive, high-quality, colour hardcopy output is straightforward, using an HP PaintJet printer, with a print utility such as Pizzaz Plus.

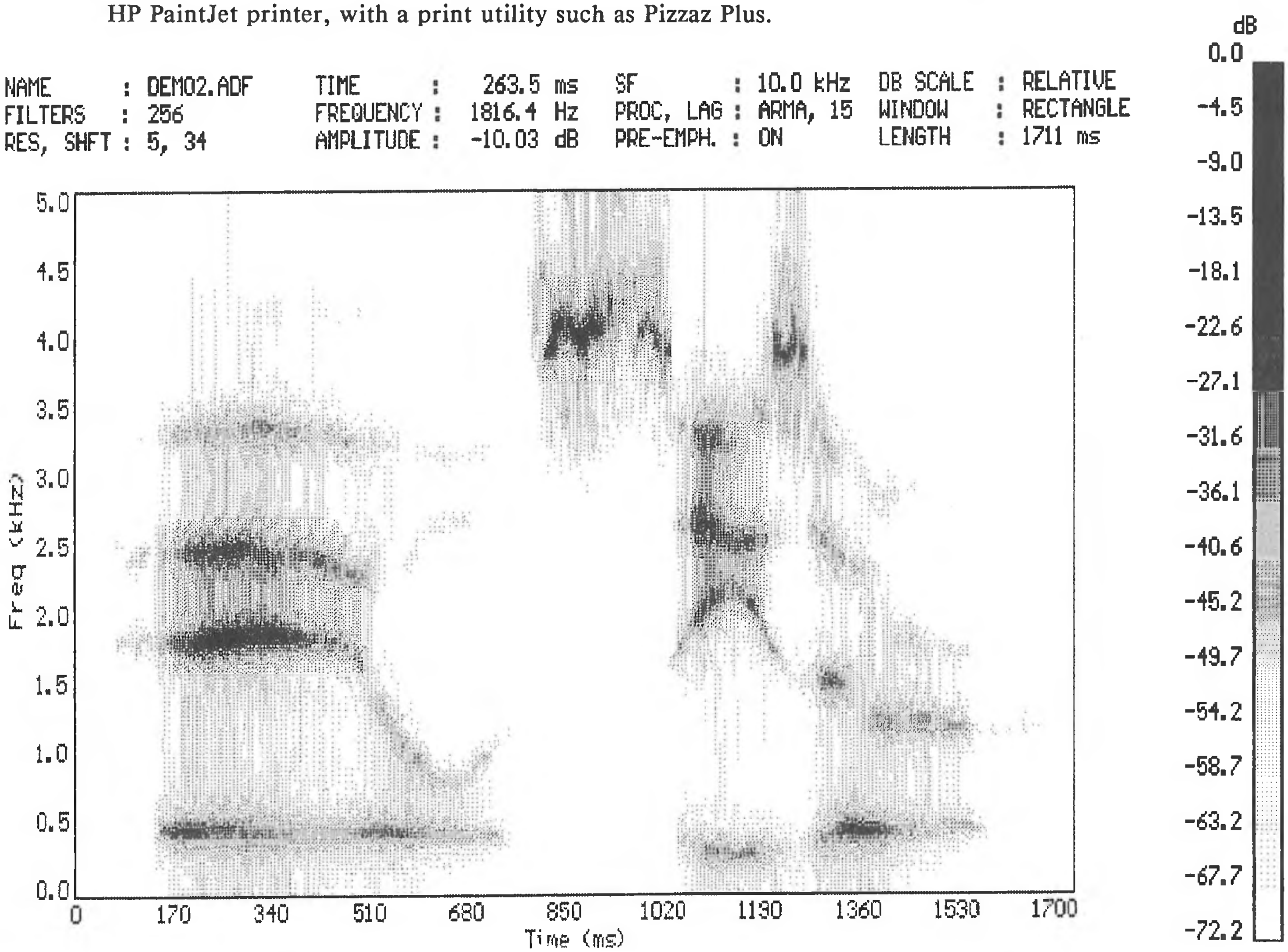


Figure 3. An example of the format of the SPECT screen showing the results of a computed spectrogram in true gray-scale mode, with direct measurement from the display. The program provides a time/frequency/amplitude display of a digitized signal, with time on the horizontal axis, frequency on the vertical axis and amplitude coded either as one of 16 possible colors or, in monochrome mode, as the relative brightness of the display. The dB range represented by each color (gray-level) is shown to the right of the display, and can be altered by the user. The user is also offered several alternative mappings of intensity, and a large number of analysis options (see text for details). After a spectrogram has been computed and displayed, the mouse can be used to define and play a portion of the signal, and/or to obtain a direct readout of the time, frequency, and amplitude coordinates of any desired point on the display.

For any desired sampled signal, the power spectrum density is calculated and a spectrogram can be generated under menu control (See Figure 3). Each parameter has a default value, but users can override this default to specify: 1) the desired frequency resolution (number of filters); 2) the desired temporal resolution; 3) the lower and upper frequency boundaries of the spectrogram (in Hz); 4) the sampling frequency of the signal; 5) the type of analysis procedure to be used (fast-Fourier Transform (FFT), Auto-regressive (AR), or modified AutoCovariance methods); 6) the color scale to be used; 7) the range in dB (subset of colors) to be displayed; 8) whether the intensity of the spectrogram is to be scaled **relative** to the within-signal maximum value, or **independent** of signal intensity -- with scaling relative to the maximum signal which could be recorded using a 12 (or 16) bit analog-to-digital converter; and 9) whether or not the signal is to be preemphasized.

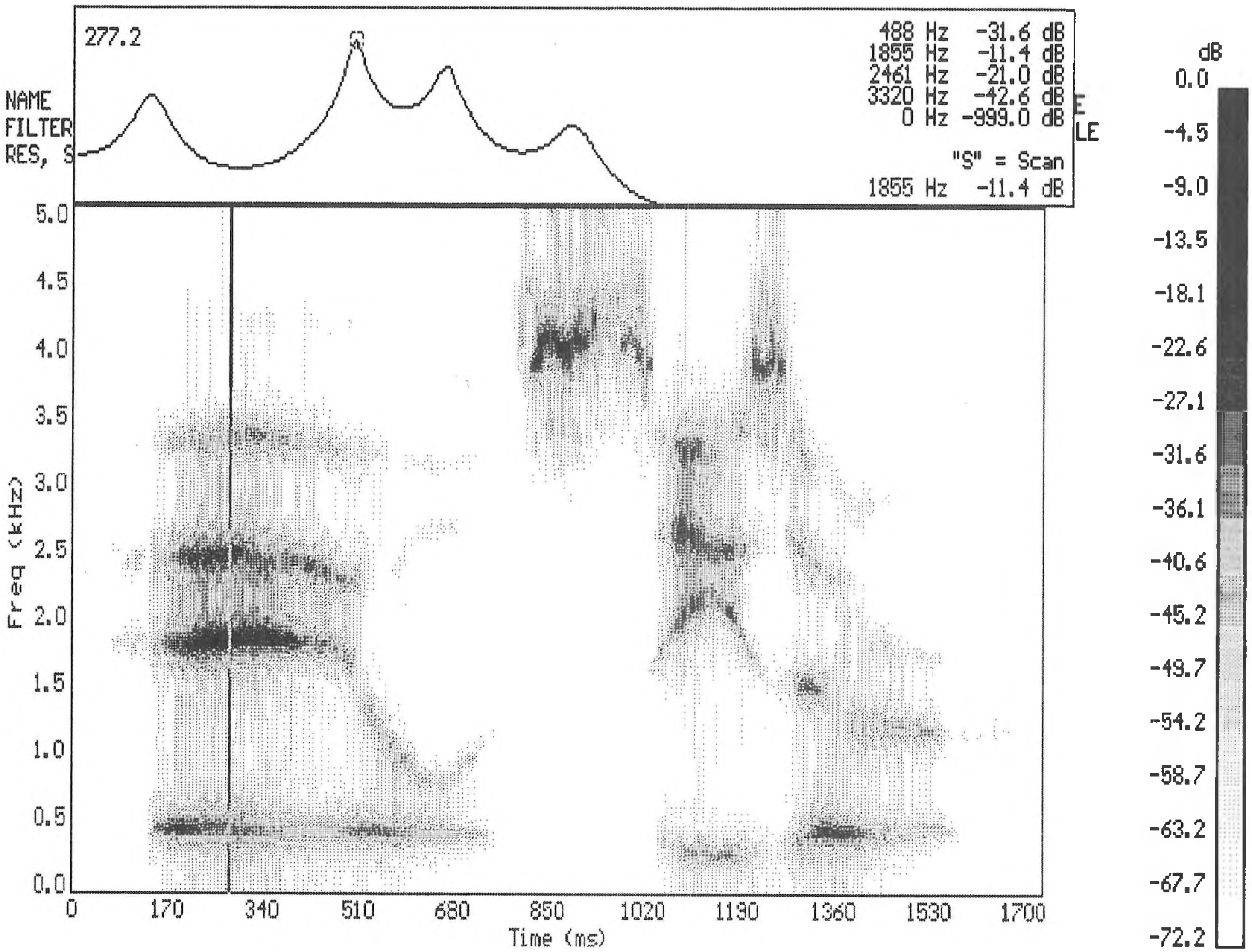


Figure 4. An example of the screen format of SPECT as the user selects a single "spectral time slice" to be displayed just above the actual spectrogram display. The frequencies and amplitudes of the major peaks for this slice are automatically extracted and listed. SPECT also provides the option to "scan" the slice for frequency/amplitude values along the time slice.

Users may also compute a second color spectrogram, using a different signal or different settings with the same signal, and store this on a separate graphics page, toggling back and forth between the two displays, as desired. Once a spectrogram has been displayed, users may move a cursor under mouse control to define and play a region of the signal to which they wish to listen, or to obtain a direct readout of the time, frequency, and amplitude coordinates of any desired point (See Figure 3). Users may also select a "spectral slice" (amplitude-by-frequency display) at a particular time point in the signal, for display in conjunction with the spectrogram (See Figure 4). When such a spectral slice is displayed, the major peaks from this spectral slice are automatically extracted and listed; in addition, a mouse-or key-controlled cursor can be used to obtain amplitude values for any desired frequency.

II.B. WATERFALL DISPLAY. A waterfall display (time by frequency by amplitude in 3 dimensions) is another way to display a power spectrum density resulting from the spectral analysis of a digitized signal. The result is displayed as a series of "stacked" time slices (See Figure 5).

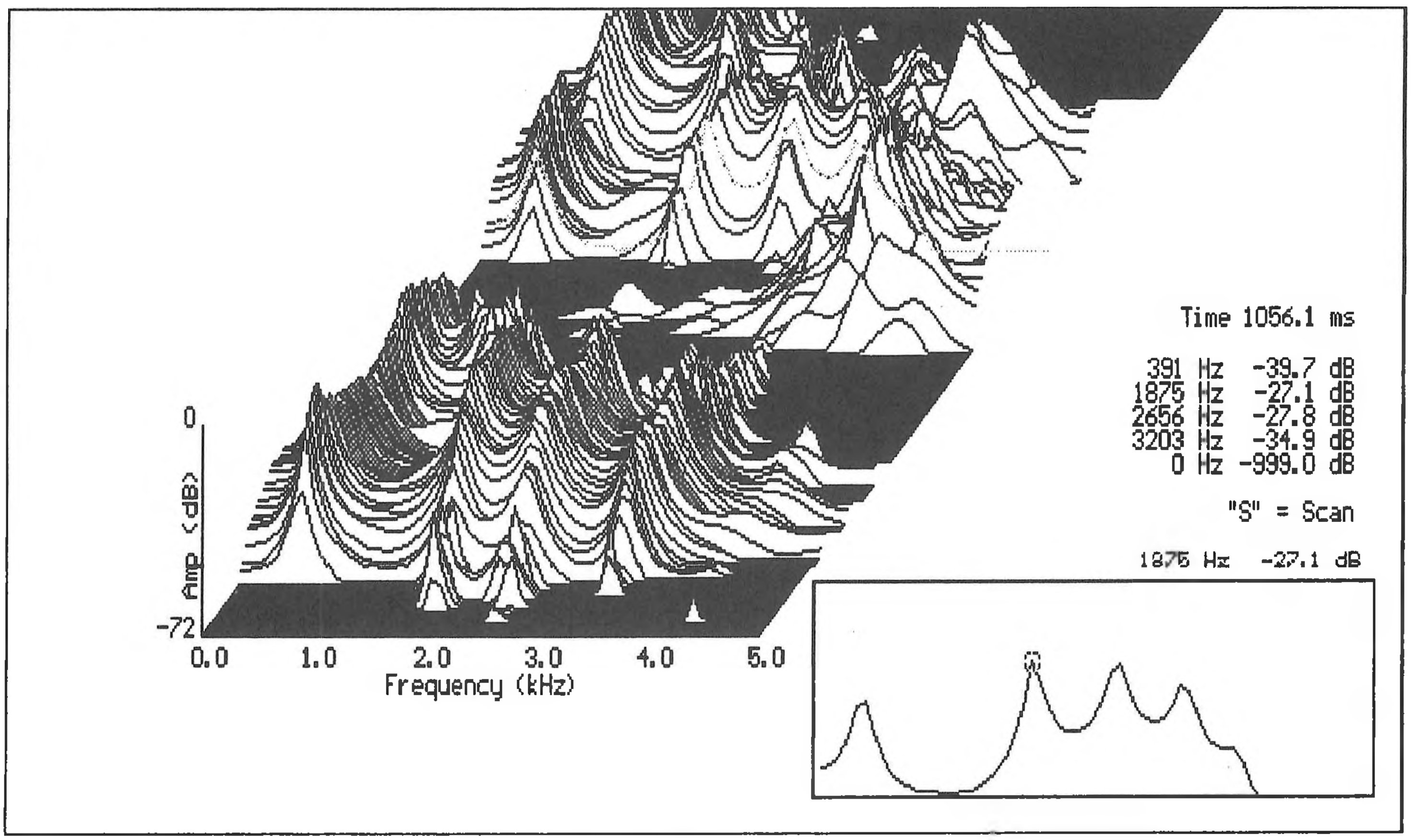


Figure 5. An example of a waterfall display (time-by-amplitude-by-frequency) in three dimensions. The display shows the results of a spectral calculation as a series of "stacked" time slices. Any time slice in the display can be selected using the mouse or cursor keys; once a time slice has been selected, a marker can be moved along this time slice to obtain instantaneous frequency and amplitude values for any desired portion of the signal.

The program also allows one to select any time slice from the display with mouse control. Upon this selection, one can also run a special marker along this time slice to obtain instantaneous frequency and amplitude values for any desired portion of the signal.

III. PARAMETRIC SPEECH SYNTHESIZER

SYNTH is a parametric speech synthesizer which generates **audio data** files from the specifications of 40 control parameters -- the 39 parameters of the KLATT80 synthesis, plus parameter 40, CORRSW, added by Kewley Port. In this implementation, the synthesizer is controlled by an INTERPolated file, which specifies the values of each parameter, updated every 5 ms (See Figure 6). The INTERPolated files are generated from KNOT files, which define "skeletonized" parameter tracks, specifying "piecewise linear" functions of the desired parameter tracks. (These KNOT files are fed to an interpolator to generate the complete files, prior to synthesis.)

CSRE Knot Editor

==

Klatt Parameter Information

Fundamental freq of voicing				
f0_	V	0	500	0
Name	Type	Min	Max	Def

Options [G]

G)et Parameter
P)rocess
S)ave
E)xit

Editing [Q]

A)dd
C)hange
D)elete
Q)uit <esc>

Time	Value

#	Time	Value	#	Time	Value
1	0	100	16		
2	50	125	17		
3	100	120	18		
4	200	110	19		
5			20		
6			21		
7			22		
8			23		
9			24		
10			25		
11			26		
12			27		
13			28		
14			29		
15			30		

File Name

KNOTBASE.KFE

Figure 6. An example of a SYNTH screen, showing initial input to the parametric speech synthesizer. Using SYNTH, the 39 control parameters for the synthesizer are defined at each of the critical "change" points within the signal. Subsequently, these "skeletonized" parameter tracks, specifying "piecewise linear" functions of the desired parameter tracks, are INTERPolated to produce a file which specifies the values of each parameter, updated every 5 ms. The INTERPolated files are then used to control the digital synthesizer.

IV. CONTINUUM GENERATOR

Many studies using synthesized speech involve, at one stage or another, an exploration of the perceptual effects of systematically varying the synthesis parameters. Typically, stimuli which bear such a parametric relationship to each other are conceptualized in terms of a "continuum" in which adjacent stimuli differ by a step in one (or more) parameter(s). Using conventional procedures, the creation of such a parametrically-varied continuum is slow and requires a high degree of attention, if errors are to be avoided; moreover the task is distinctly uninspiring. To facilitate such explorations, we have therefore developed a "Continuum Generator", in the form of a high-level "programming language".

In its simplest application, the Continuum Generator allows a researcher to specify a range of values which a variable parameter is to assume -- starting value, number of steps, and size of steps -- and then to synthesize all of the speech signals which would result from the combination of each of the possible values of the variable parameter, in turn, together with a specified set of fixed values for all of the other parameters which are required to control the synthesizer. In this way, the Continuum Generator operates much like a "do" or "for" loop in a conventional high-level programming language such as Basic or Fortran. By extrapolation, nested loops can also be generated, resulting in the full set of synthesized speech signals produced by the factorial combination of two (or more) sets of variable parameters.

Using these procedures, we have been able to quickly generate large numbers of speech signals for use in speech perception experiments, with a minimum of effort, and with the details of the synthesis parameters controlled automatically (and without error). Indeed, some experiments using these procedures involve several hundred parametrically-synthesized signals -- offering new opportunities to explore combinations of parameters which would previously have been impossible because of the time constraints of traditional synthesis techniques. One application in which the Continuum Generator has proved to be of particular value involves the study of interactions between parameters; another valuable application involves the identification of combinations of parameters which result in very-high quality synthesized speech.

IV. EXPERIMENT GENERATOR

The Experiment Generator is a menu driven program which permits an experimenter to implement an audio experiment by filling a form specifying values for the predictable parameters of audio stimulus file names, number of times each signal is to be presented per block, rate of presentation, time permitted for response, inter-trial interval, and so forth. The program then produces an experiment specification file which is read and implemented by the Experiment Controller to carry out the experiment as specified. The response timing within the experiment is controlled by a clock which is preprogrammed to produce a resolution of 1 ms. The parameters such as stimulus and block lists which make up the experiment can be generated/edited through the same menu driven system.

V. FUTURE PLANS

To date, CSRE has been used in four ways: 1) to edit natural speech tokens to produce standardized tests of speech intelligibility for hearing impaired listeners (e.g., Jamieson, DellOrletta, and Ramji, 1988); 2) to prepare synthetic speech signals for experiments on the perception of native-language contrasts (Cheesman, 1989) and non-native speech sounds, during second language training (Jamieson and Morosan, 1989); 3) to analyze the temporal and spectral properties of English and Mandarin Chinese fricative speech sounds; and 4) to analyze lung and airway sounds associated with different respiratory tract disorders (Slawinska, Jamieson, McMillan, and Wake, 1987).

In our current phase of the project, we are expanding the range of application of CSRE in these areas, adding new features such as F0 extraction, formant tracking, and parameterization for burst and fricative sounds, and investigating a range of applications in the area of the disorders of human communication. In addition to expanding the range of applications of the CSRE software, our future work will focus on three topics: 1) the development of higher-level tools to support speech research; 2) the implementation of specific signal- and graphics-processing procedures, using more powerful processors; 3) the comparative study of alternative processing and display procedures, including, where required, the development of new algorithms and procedures.

ACKNOWLEDGEMENT AND AVAILABILITY

This paper was delivered as an invited talk at the 1988 annual meeting of the Canadian Acoustical Association. The contributions of Terry Baxter, Meg Cheesman, Alex Kania, Whitney Allsop, Phil Lieberman, and John Mertus, and the support of the Natural Sciences and Engineering Research Council of Canada, Bell Northern Research, and the University Research Incentive Fund are gratefully acknowledged.

CSRE is being developed and distributed on a nonprofit, cooperative/shareware basis. Researchers who are interested in using CSRE can contribute to the project in one of two ways. First, they can contribute programs and/or utilities which they have written, which will increase the usefulness of the package, and can be adapted to form an integrated package, working together with the other parts of CSRE, as part of subsequent releases of the package. Second, researchers can make a financial contribution (presently $250) in return for the CSRE software. The funds obtained in this manner will be directed to the further development of the package. Interested researchers should contact Dr. D.G. Jamieson at the indicated address, or by e-mail (JAMIESON@UWOVAX.UWO.CA)

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Cheesman, M.F. Acoustical factors underlying speech perception. Doctoral dissertation, University of Minnesota, 1989.

Kania, A., Jamieson, D.G., Nearey, T.M., and Ramji, K.V. (1988). A comparison of alternative procedures for digital spectral analysis of speech. **Journal of the Acoustical Society of America, 84, S1.**

Jamieson, D.G. and Morosan, D. (1989). Evaluation of a technique for training new speech contrasts: Generalization across voices, but not word-position or task. **Journal of Speech and Hearing Research, 32,** 501-511.

Jamieson, D.G. and Nearey, T.M. (1988). CSRE : A Speech Research Environment. **Abstracts of the Association for Research In Otolaryngology, 11,** 228.

Jamieson, D.G., Seewald, R.C., and Raftery, E. (1988). A general-purpose hearing aid prescription, simulation, and testing system. **Canadian Acoustical Association,** annual meeting, Toronto, October, 1988.

Slawinski, E.B., Jamieson, D.G., McMillan, D.D., Wake, C.H. and Hambly, M.D. (1987). Acoustical analysis of respiration sounds in neonates. **Proceedings of the Canadian Acoustical Association,** 85-91.

Yaphe, M., Raftery, E., and Jamieson, D.G.. (1989). A general-purpose facility for adaptive testing in psychoacoustics. **Behavioral Research Methods, Instrumentation, and Computers,** 21, 275-280.

COMMENTAIRES CONCERNANT L'EDITION REVISEE DU VOLUME, "NOISE AND VIBRATION CONTROL", L.L. Beranek, ed.

Le volume "Noise and Vibration Control" de Leo L. Beranek qui n'était plus disponible depuis quelques années a été réédité en 1988. Ce volume traite des ondes sonores et de leur propagation (dans les locaux et à l'extérieur), de la mesure et l'analyse des sources de bruit et de vibration, des matériaux et des techniques utilisés pour contrôler le bruit et les vibrations, des effets du bruit et des critères de bruit et vibrations en vigueur aux Etats-Unis. La nouvelle édition diffère quelque peu de l'édition précédente (1971). Les principaux changements concernent la mesure de l'intensité acoustique, les critères de bruit et vibration dans l'environnement, les édifices et les véhicules, et l'ajout ou la mise à jour de plusieurs références citées à la fin des chapitres.

Ce volume est un bon ouvrage de référence pour le praticien en acoustique car il touche la plupart des aspects de base. Il compte également plusieurs exemples/exercises utiles pour la compréhension. Il convient aussi au néophyte de l'acoustique puisque les sujets sont présentés dans un ordre logique allant de la base fondamentale jusqu'à l'application. Finalement, la consultation de cet ouvrage est facile, rapide et efficace grâce à son indexe extrèmement détaillé.

Cette nouvelle édition comporte au chapitre 6 une nouvelle section intitulée "Measurement of Sound Intensity". Une telle section était nécessaire suite à l'évolution des techniques de mesure depuis l'édition précédente. Le sujet est cependant traité de façon très sommaire (un peu plus d'une page). Il aurait été souhaitable d'approfondir d'avantage car l'acousticien contemporain se doit de bien connaître les techniques de mesure de l'intensité acoustique et leurs limitations.

Le contenu du chapitre 9 s'intitulant "Sound in Large Rooms" n'a guère été modifié. Il ne présente pas les nouveaux développements concernant le calcul de la propagation du son dans les locaux.

Le chapitre 18 qui traite des critères de bruit et de vibration a été complètement ré-écrit. On y a mis beaucoup d'emphase sur les critères de bruit dans les bâtiments. On y décrit d'ailleurs un nouveau critère (critère NCB) développé par l'auteur. Contrairement aux critères NC et PNC, il tient compte de l'allure du spectre lors de l'évaluation du bruit. Cette section traite également du design des bureaux à aire ouverte en utilisant des valeurs empiriques simples pour les calculs des performances.

La partie traitant des critères de bruit environnemental (sections 18.6 à 18.8) est décevante. Le sujet n'est pas bien introduit et les objectifs de la section sont difficiles à cerner. Les critères de bruit décrits sont essentiellement américains et ne s'appliquent donc pas nécessairement au Canada. On ne traite pas des descripteurs EPN dB et NEF qui sont ceux utilisés au Canada pour l'évaluation du bruit et de la nuisance des aéronefs.

Il est difficile d'approfondir tous les domaines de l'acoustique en un seul volume. Toutefois, il aurait été avantageux de traiter l'acoustique environnementale plus en profondeur. L'analyse statistique du bruit par exemple ne fait l'objet que d'un seul paragraphe. Pourtant elle est très souvent utilisée pour caractériser l'environnement sonore des sites. De plus, on ne fait pas mention de l'existance de logiciels officiels pour le calcul du bruit le long des routes et en bordure des aéroports. Ces logiciels sont utilisés couramment par les organismes gouvernmentaux et les consultants.

La mesure en chantier des performances acoustiques des cloisons n'est également pas traitée. Il aurait été intéressant d'introduire les descripteurs normalisés NIC, NNIC et FIIC utilisés pour évaluer des performances d'insonorisation en chantier.

En conclusion, mentionnons que même si les changements apportés dans la nouvelle édition sont décevants et que certains sujets pratiques ne sont pas traités suffisamment en profondeur, le volume "Noise and Vibration Control" demeure un très bon ouvrage de révérence et facile à consulter.

Blaise Gosselin, ing.
Ingénieur en chef, Gestion du bruit des aéroports
Groupe de gestion des aéroports
Transport Canada, Ottawa

[Nota - ceux et celles qui envisagent l'achat de ce volume seront intéressés à savoir qu'il est en train d'être entièrement ré-écrit et republié - Ed.]

REVIEW OF THE REVISED EDITION OF NOISE AND VIBRATION CONTROL, L.L. Beranek, ed.

The volume, "Noise and Vibration Control" by Leo L. Beranek, which has not been available for several years, was revised in 1988. This text deals with acoustic waves and their propagation (in rooms and outdoors), the measurement and analysis of sources of noise and of vibration, acoustic materials, techniques for controlling noise and vibration, the effects of noise, and noise and vibration criteria in effect in the United States. The new edition differs somewhat from the previous edition (1971). The main changes relate to the measurement of acoustic intensity, noise and vibration criteria in the environment, buildings and vehicles, and the addition and up-dating of various references cited at the end of chapters.

This volume is a good reference work for the acoustic practitioner since it discusses the main fundamental concepts. It also contains many examples/exercises which aid comprehension. It is useful to the new-comer to acoustics, since the subjects are presented in a logical order from fundamental concept to application. Finally, the book is easy, quick and efficient to consult, thanks to its detailed index.

The new edition contains, in Chapter 6, a new section entitled, "Measurement of Sound Intensity". Such a section is necessary in the light of the evolution of measurement techniques since the appearance of the previous edition. The subject is, however, dealt with in a very summary fashion (a little more than one page). A more detailed treatment would have been welcome since the contemporary acoustician must have a good knowledge of acoustic intensity techniques and their limitations.

The content of Chapter 9 entitled, "Sound in Large Rooms" has hardly been modified. As a result, it does not present new developments concerning the prediction of sound propagation in rooms.

Chapter 18, which deals with noise and vibration criteria, has been completely re-written. Much emphasis has been put on noise criteria for buildings. Moreover, a new criterion (NCB), developed by the author, is described. Contrary to the case of the NC and PNC criteria, it takes the spectrum shape into account in evaluating the noise. This section also discusses the design of open-plan offices using simple empirical values in calculating the performance.

The sections on environmental noise criteria (18.6 and 18.8) are disappointing. The subject is not well introduced and the section's objectives are difficult to understand. The described noise criteria are essentially American and don't necessarily apply to Canada. The descriptors EPN dB and NEF, used in Canada in the evaluation of aircraft noise and annoyance, are not discussed.

It is difficult to cover in detail all the branches of acoustics in one volume. Nonetheless, it would have been useful to cover environmental acoustics in more detail. For example, the statistical analysis of noise is the subject of only a single paragraph. Yet it is very often used to characterize the acoustic environment of sites. In addition, nowhere is the existence of official software for calculating the noise along roads and near airports mentioned. Such software is often employed by government organizations and consultants.

Also not mentioned is on-site measurement of the acoustic performance of partitions. It would have been interesting to introduce the normalized descriptors NIC, NNIC and FIIC, used to evaluate noise reduction in the field.

In conclusion, it should be mentioned that even if the changes made in the new edition are disappointing, and certain practical subjects are not dealt with in sufficient depth, the volume, "Noise and Vibration Control" remains a very good, interesting, and easy-to-consult reference work.

BLAISE GOSSELIN, P.Eng.
Chief Engineer, Airport Noise Management
Airport Management Group
Transport Canada
Ottawa

[Editor's Note: readers considering purchasing this book might be interested to know that it is now in the process of being completely rewritten, to be published again next year.]

News / Informations

Conferences / Congrès

International Congress on Recent Developments in Air & Structure Borne Sound and Vibration, Auburn University, Alabama ,March 6-8, 1990. Contact: P.K. Raju, Dept. of Mechanical Engr, 210 Ross Hall, Auburn University, AL 36849-3541, U.S.A.. Tel.: (205) 844-3301.

Premier Congrès Français d'Acoustique, Lyon, France, 10-13 avril 1990.

IEEE 1990 Iternational Conference on Acoustics, Speech, and Signal Processing, Alberquerque, New Mexico, April 3-6, 1990. Contact: Prof. D. Etter, EECE Department, The University of New Mexico, Albuquerque, NM 87131. Tel.: (505) 277-6564.

Mechanics and Biophysics of Hearing, University of Wisconsin, Madison, June 24-29, 1990. Contact: Mechanics of Hearing 1990, Department of Neurophysiology, University of Wisconsin-Madison, 283 Sciences Building, Madison, WI 53706 USA. TEL.: (608) 263-7357.

XVIIIèmes Journées d'Etudes sur la Parole, Montréal, 28-31 mai, 1990. Contacter: Danièle Archambault, XVIIIèmes JEP, Université de Montréal- Dept. de linguistique, Laboratoire de phonétique, C.P. 6128, Montréal, Canada, H3C 3J7. Tél.: (514) 343-7041.

Fourth International Symposium on the Effects of Noise on Hearing, Beaume, France, May 28-30, 1990. Contact: Dr. A. Dancer, Groupe Pysiologie, I.S.L., BP 34, 68301 St-Louis, France.

Symposium on Physical Acoustics; Fundamental and Applications, Kortrijk, Belgium, June 19-23, 1990. Contact: Prof. O. Leroy, Katholieke Universiteit Leuven Campus Kortrijk, E. Sabbelaan, B-8500 Kortrijk, Belgium. Tel.: (056) 21 79 31.

INTER-NOISE 90, Gothenburg, Sweden, August 13-15, 1990. Contact: Tor Kihlman, Department of Applied Acoustcs, Chalmers University Technology, S-412 96 Gothenburg, Sweden. Tel.: (046) 31 72 22 11.

12th International Symposium on Nonlinear Acoustics, Austin, Texas, August 27-31, 1990. Contact: Prof. M. F. Hamilton, Dept. Mechanical Eng., University of Texas at Austin, Austin TX 78712-1063, USA.

New Products

From Brüel & Kjaer Canada

A new acoustics module allows the accurate and extensive measurement of all aspects of sound within an enclosed space, making it an ideal instrument to use in the design and operation of concert halls and auditoria. Known as the "BZ 7109", the software module is an advanced version of an earlier processor module, the BZ 7104. Both were designed for use with the highly sophisticated sound level meter produced by the firm, known as the Type 2231.

A tiny yet powerful probe microphone recenlty introduced to the Canadian market will enable users to test for sound quality in previously inaccessible spots. Known as the type 4182, the new instrument is ideal for research work in difficult to reach places, harsh environments or in places so close to the source of sound that traditional technology will not suffice. The tip of its 100 mm stiff probe tube is so tough it can be exposed to temperatures of more than 700 degrees celsius, making it ideal for measuring sound levels in the hot gasses inside automobile mufflers or sound pressure measurements inside industrial chimney stacks.

For further information: Andrew McKee, Brüel & Kjaer Canada Ltd, 90 Leacock Road, Pointe Claire, Que. H9R 1H1. (514) 695-8225.

From Scantek, Inc.

Scantek, Inc. announces the Rion NA-29 Sound Level Meter/Octave band Analyzer. This hand-held real time analyzer weighs less than 2.5 pounds and measures just 8.7 x 4.7 x 1.6 inches. The NA-29 displays overall level, levels of octave bands centered at 31.5 through 8 000 Hz or level vs time. Measurment modes include Lp, Lmax, Ln, Leq, Lae. Sampling periods for Leq and Lae are selectable from 1 ms to 24 hours. Up to 1500 levels or octave spectra can be stored and later recalled for display or readout.

Scantek, Inc. annouces the ENM (Environmental Noise Model) . ENM is an easy-to-use computer program that rivals any that is used commercially or publicly in the world. The program allows the user to input sound power level data (in octave- or 1/3rd-octave bands), 3-dimensional source and receiver coordinates, and directivity for several source types, among others. Data can be read directly from a spectrum analyzer and graphed or manipulated in a spreadsheet format.

For further information: Richard J. Peppin, P.E., President (301) 279-9308.,

From Kay Elemetrics Corp.

Kay Elemetrics Corp has produced a video tape demonstration on the new DSP Sonagraph, model 5500 speech analysis workstation. Some of the Sona-Graph's standard features mentioned on the tape include: Real-time analysis, display and acquisition at sampling rates up to 81 kHz, Dual channel analysis and display, High resolution graphics (color and grey scale), Scrolling to review stored signal, High speed computer interface and Continuous grey scale printing. Additional options include: Voice analysis program and LPC Program.
For further information: Robert McClurkin, M.S. CCC SP/L ar 1-800-289-5297 or write to Kay Elemetrics Corp, 12 Maple Avenue, Pine Brook, New Jersey 07058.

CAA DIRECTORS AWARD WINNERS - 1988-89

Dr. Jos J. Eggermont, Department of Psychology, University of Calgary. For his paper titled

"White Noise Analysis of Non Linear Systems with Application to the Auditory System".

Mile Sylvie A. Bilodeau, Groupe d'acoustique de l'université de Montréal. Pour son article intitulé

"Prédiction des conditions sonores dans les salles de classe au moyen des caractéristiques physiques de l'environnement".

Prize Winners for the Best Presentation at the Annual CAA Conference - 1989

Mlle M. J. Ross, Groupe d'acoustique de l'université de Sherbrooke. Pour sa communication intitulée

"Non-destructive Measurement of Porous Material Propagation Constants".

M. A. Berry, Groupe d'acoustique de l'université de Sherbrooke. Pour sa communication intitulée

"Vibro-acoustic Behaviour of Semi-complex Structures".

M. Daniel Ouellet, Groupe d'acoustique de l'université de Sherbrooke. Pour sa communication intitulée

"Integral Equation Method for Acoustic Response of Bounded volumes with Impedance Discontinuities".

Congratulations / Félicitations

CAC/ISO TC43 and CAC/ISO TC43/SC1

Liaison Report to CSA Z107 Committee on Acoustics and Noise Control

Annual Meeting - October 17th, 1989, Halifax

The Canadian Advisory Committee to International Standards Organization Technical Committee's TC43 "Acoustics" and TC43/SC1 "Noise", successfully hosted Plenary Sessions of these Committees in Toronto, October 11-14, 1988, together with their associated Working Group Meetings. The meetings were held in the Downtown Toronto Board of Trade immediately following Plenary Sessions of IEC TC29 "Electro-acoustics", and the Canadian Acoustical Association's Annual Acoustics Week. All participants were encouraged to stay at the Westbury Hotel, and there was a significant opportunity for exchange between Canadian and foreign acousticians.

A meeting of the Canadian Advisory Committee was held on July 27, 1988, in Ottawa, to formulate the Canadian position for the International meetings. The Canadian delegation consisted of Dee Morison (Head of Delegation), Alberto Behar, Bob Johnston, Leslie Kende, George Krishnappa, Joe Piercy, Cameron Sherry and George Wong. The report of the meeting is published in Canadian Acoustics, 17(2), April 1989, pp.47-49.

The next Plenary Sessions of ISO TC43 and ISO/TC43/SC1 are scheduled for May 14-18, 1990, in the Netherlands. There will be a meeting of the Canadian Advisory Committee to these groups in the latter half of March 1990 to prepare the Canadian position for these international meetings.

Canadian working group members have actively participated in meetings throughout the year. These activities are being reported in the CAA Seminar in a paper on October 19th and are written up in the proceedings.

A detailed summary of documents reviewed by the CAC from October 1987 to October 1988 is attached. It is hoped to publish this in Canadian Acoustics.

Prepared by:
D.A. Morison
October 10, 1989

Canadian Advisory Committee to
ISO Technical Committee 43 "Acoustics",
and ISO Technical Committee 43/SC1 "Noise"

CAC/ISO/TC43 and CAC/ISO/TC43/SC1

Documents received for Canadian Comment
October 1988 - September 30, 1989

Number	Title	Action
43/1 N644	ISO/TC43/SC1/WG30 "Frequency Weighting 'A' for noise measurements". Convenor's report and revised draft.	Disapproved. Significantly modification required (G. Wong).
43/1 N643 (70/5 N411)	ISO/DP 6798 "Acoustics - Test code for the measurement of airborne noise emitted by reciprocating internal combustion engines - Engineering method and survey method".	Approved with comments.
43/1/ISO DIS 9053	"Acoustics - Materials for acoustical applications - Determination of airflow resistance".	Approved with substantial comments (D. Quirt, Y. Champoux).
ISO/TC72/ WG11 N20	ISO/DP 9902 "Textile Machinery - Acoustics - Determination of sound pressure levels and sound power levels emitted by textile machines - Engineering and survey method".	
ISO/TC43 N793	Annual Report of ISO/TC43 "Acoustics" and its 2 sub-committees, SC1 "Noise" and SC2 "Building Acoustics" for 1988.	Approved with minor correction.
43 N796	Proposal for minor revision of ISO 4869: 1981 "Acoustics - Measurement of sound attenuation of hearing protectors - subjective method.	Approved (A. Behar).

43/1 N655	Second ISO/DP 9613 "Acoustics - Attenuation of sound during propagation outdoors - Part 1: Calculation of the absorption of sound by the atmosphere".	Approved (J. Piercy).
43/1 N658	First ISO/DP 10302 "Acoustics - Method for the measurement of air-borne noise emitted by small air-moving devices".	Approved (D. Quirt).

FIVE YEAR REVIEW OF THE FOLLOWING 9 STANDARDS: -

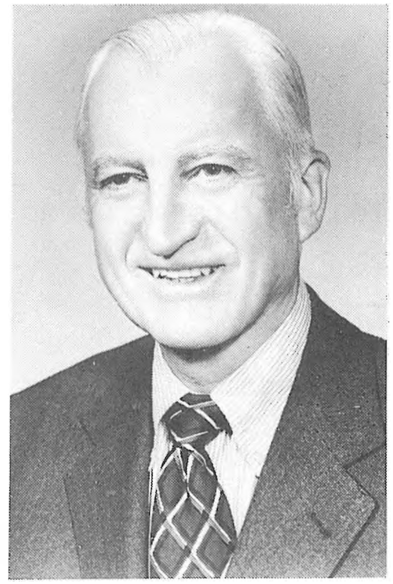
1) ISO 3822-2	"Acoustics - Laboratory tests on noise emission from appliances and equipment used in water supply installations - Part 2: Mounting and operating conditions for draw-off tops".	Reconfirmation recommended.
2) ISO 3822-3	As above. "Part 3: Mounting and operating conditions for in-line values and appliances".	Reconfirmation recommended.
3) ISO 4871	"Acoustics - noise labelling of machinery and equipment".	Abstain.
4) ISO 5135	"Acoustics - Determination of sound power levels of noise from air terminal devices, high/low velocity /pressure assemblies, dumpers and values by measurement in a reverberation room".	Reconfirmation recommended.
5) ISO 7029	"Acoustics - Threshold of hearing by air conduction as a function of age and sex for otologically normal persons".	
6) ISO 7182	"Acoustics - Measurement at the operators position of airborne noise emitted by chain saws".	
7) ISO 131	"Acoustics - Expression of physical and subjective magnitudes of sound or noise in air".	Approved.
8) ISO 2249	"Acoustics - Description and measurement of physical properties of sonic booms".	?

9) ISO 3746	"Acoustics - Determination of sound power levels of noise sources - survey method.	Under Revision.

PROPOSAL FOR A NEW WORK ITEM

N794	Sound absorbers - Rating of measurement results.	Approved and will participate (D. Quirt).

OLIVER C. ECKEL



It is with deep sorrow that the Canadian Acoustical Association has learned of the death of Oliver C. Eckel, a recognized authority in acoustic engineering and mechanical design, and the founder of Eckel Industries, Inc. Mr. Eckel, who was 84, passed away on April 11, 1989 in Naples, Florida, following a brief illness.

In the early days of his career, Oliver C. Eckel was involved with numerous mechanical design and development programs. In 1939, he became the New England distributor for Owens Corning Fiberglas Corp. and established his own fabrication and warehouse operation in Cambridge, Massachusetts, now known as Eckel Industries, Inc. Prior to and during World II, his company was responsible for the design and manufacture of insulated metal jackets for vapor compression salt water distillation systems.

Starting in 1940, Mr. Eckel cooperated with Leo Beranek and his research team at the Cruft Laboratory, Harvard University in the design and fabrication of anechoic wedges for the first anechoic chamber in the U.S. for testing communications equipment and for behavioral and psychological studies. As a result of his work, Oliver C. Eckel was granted numerous patents on anechoic wedge design and construction, including one for a spring tensioned floor construction. Many of these patents are still being successfully used in chambers engineered by Eckel Industries.

In conjunction with his work with gas turbine and aircraft manufacturers, Mr. Eckel developed and patented a special panel system for the quieting of jet engine test calls. Further innovative work in the acoustics field included the creation of unique demountable modular panels for noise control/noise isolation, special facilities, and double-acting insulated doors.

After his retirement from Eckel Industries, Inc. in 1978, Mr. Eckel established a R&D laboratory devoted to wind turbine design. He continued his engineering development of a shrouded wind turbine up to the time of his death.

A native of New York and former resident of Weston, Mass., Oliver C. Eckel received his mechanical engineering degree, with top honors, from New York University's College of Engineering in 1926. He was a Fellow of the Acoustical Society of America, a member of the Institute of Noise Control Engineering, a life member of ASHRAE, a member of the Academy of Applied Science and was a Registered Professional Engineer. He is survived by his second wife, Helen, a son by his first marriage, Alan, who is President, Eckel Industries, Inc. and 12 grandchildren.

The Eckel family and Eckel Industries of Morrisburg, Ontario will establish a Scholarship as a memorial to Mr. Oliver C. Eckel. The Canadian Acoustical Association will be involved with the administration of this scholarship.

THE CAA ALEXANDER GRAHAM BELL GRADUATE STUDENT PRIZE IN SPEECH COMMUNICATION AND BEHAVIORAL ACOUSTICS

Award

The award consists of an $800 cash prize to be awarded annually.

Eligibility

The candidate must be a graduate student enrolled at a Canadian University pursuing studies in speech communication or behavioral acoustics. Preference will be given to Canadian citizens and landed immigrants.

The candidate must be a member or a student member of CAA and may only receive the prize once.

Application Procedure

Applicants must submit a written proposal for the research work to be done during their graduate program. A letter from the student's supervisor must accompany the application stating that the work to be undertaken will be the independant work of the student.

Selection Process

All applications will undergo a review by a Subcommittee named by the President and Board of Directors of the Canadian Acoustical Association.

Deadline

Applicants must submit the following form and a letter from their academic supervisor to the Executive Secretary of CAA by February 28.

The successful candidate will be notified in writing before May 30. The winner of the prize will also be announced in the spring issue of Canadian Acoustics.

CANADIAN ACOUSTICAL ASSOCIATION

ALEXANDER GRAHAM BELL GRADUATE STUDENT PRIZE IN SPEECH COMMUNICATION AND BEHAVIORAL ACOUSTICS.

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4. CITIZENSHIP:

( ) Canadian Citizen

( ) Other:______________________

5. ACADEMIC INSTITUTION:

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6. ACADEMIC SUPERVISOR (NAME):

---

7. TITLE OF RESEARCH PROPOSAL:

---

8. DATE FOR COMMENCEMENT OF PROGRAM

---

9. DEGREES AND DIPLOMAS HELD:

Degree	Institution	Discipline	Date

---

10. PROPOSED RESEARCH: (Please attach a description of the research proposal to be carried out as part of the graduate program. This proposal should not exceed 500 words. The usual elements of a well developed research proposal should be present a statement of the problem or topic of Investigation, its significance (scholarly and/or practical), the approach and research methods, sources and resources required).

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11. ATTACHMENT: Letter from academic supervisor stating that the applicant is enrolled in a graduate program and that the work described in the proposal will be carried out independantly by the student.

( ) enclosed

( ) to follow under separate cover

DATE:

SIGNATURE OF APPLICANT:

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EDITOR / REDACTEUR

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