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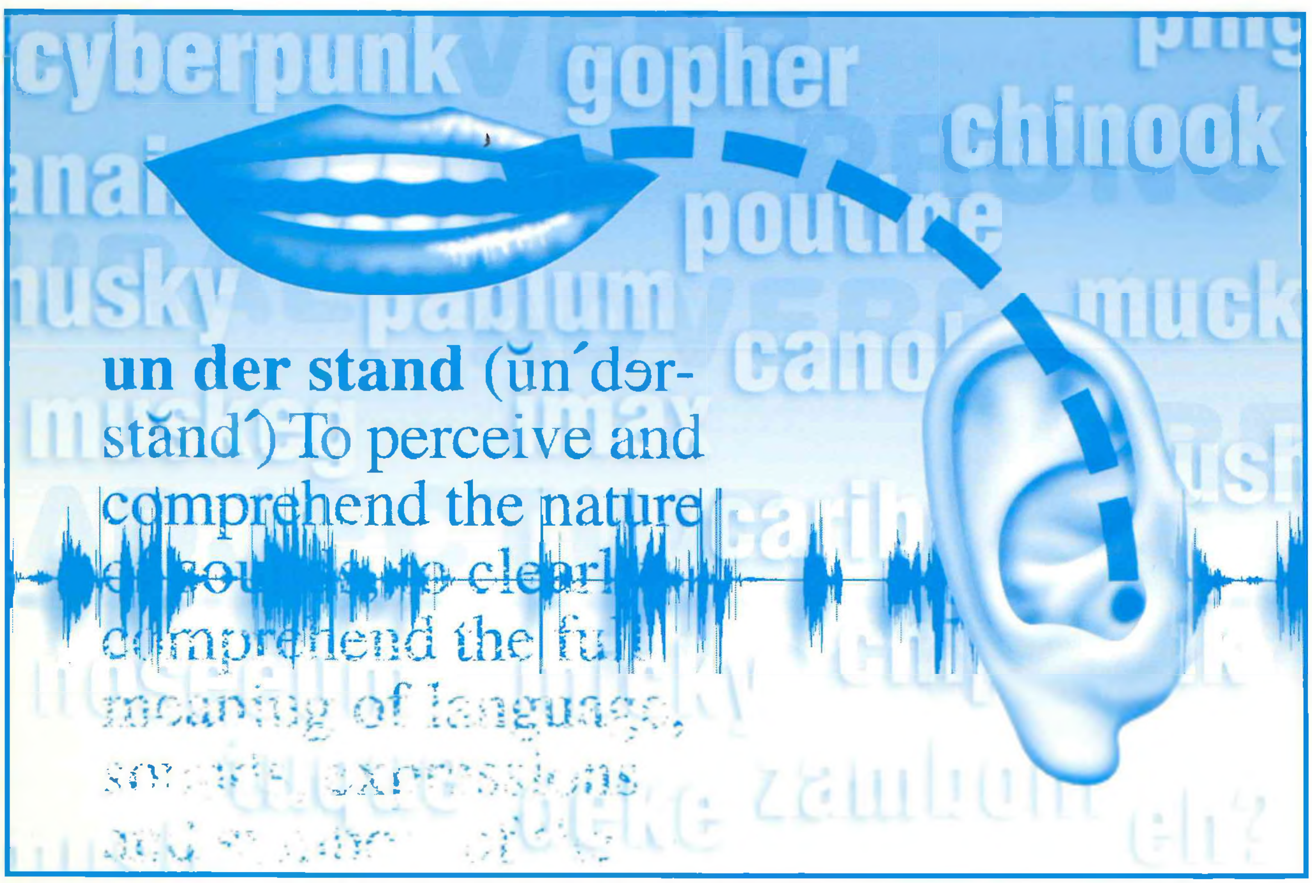

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THE CANADIAN ACOUSTICAL ASSOCIATION
P.O. BOX 1351, STATION "F"
TORONTO, ONTARIO M4Y 2V9

L'ASSOCIATION CANADIENNE D'ACOUSTIQUE
C.P. 1351, SUCCURSALE "F"
TORONTO, ONTARIO M4Y 2V9

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EDITORIAL

Vous trouverez, dans ce numéro, des articles portant sur l'électroacoustique et l'intelligibilité de la parole. Des informations ainsi que l'appel de communications pour la Semaine canadienne d'acoustique 1996 qui se tiendra à Calgary en octobre sont aussi présentés dans ce numéro. Prenez en note les dates limites pour la soumission des résumés et des sommaires qui paraîtront dans les actes. **Veuillez prendre note que les sommaires seront limités à une page cette année.**

Vous remarquerez que l'équipe éditoriale a été modifiée récemment. Jim Désormeaux quitte le poste de Rédacteur associé/Informations après un certain nombre d'années. Francine Desharnais du DREA prendra la relève. Je suis certain que tous les lecteurs se joignent à moi pour remercier Jim pour son travail consciencieux.

Même s'il est trop tôt pour tirer des conclusions claires, il y a des indices à l'effet que le Comité éditorial de *l'Acoustique Canadienne* (du moins quelques membres) commence à porter fruits. Alors que seulement six papiers ont été soumis en 1995, j'en ai déjà reçu cinq cette année.

Les lecteurs savent sans doute qu'on m'a demandé de former et de présider un comité qui vise à considérer les options dans le but de proposer un prix de l'ACA à la mémoire de Raymond Hétu. Le comité a considéré plusieurs options et désire obtenir les commentaires des membres avant de formuler une proposition au Conseil d'administration en mai. Nous vous invitons à prendre connaissance de la discussion apparaissant à la page 25 pour plus de détails.

J'aimerais rappeler aux annonceurs - tel que précisé dans le dernier numéro - que les tarifs de publicité ont été haussés pour couvrir l'augmentation des coûts.

This issue presents papers on electroacoustics and speech intelligibility.

Also published is information regarding - and the call for papers for - Acoustics Week in Canada 1996 to be held in Calgary in October. Note the deadline dates for the submission of abstracts and proceedings summaries. **Note also that proceedings summaries are this year limited to one page in length.**

This issue also sees a change in the editorial staff. Jim Desormeaux leaves the post of News Editor after a number of years in that position. Francine Desharnais of the DREA picks up the torch. I'm sure all readers will join me in thanking Jim for his dedicated work.

While it's too early to draw firm conclusions, there is evidence that we are starting to see the fruits of the labours of (at least some members of) the *Canadian Acoustics* Editorial Board. Whereas a total of six papers were submitted to your journal in 1995, I've already received five this year!

Readers may know that I have been asked to form and chair a committee to consider options for establishing a prize in memory of Raymond Hétu. The committee has considered several options and would like to hear comments from members before making a proposal to the Board of Directors in May. I would appreciate if members could refer to the discussion on page 34 for more details.

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Canadian Acoustics / Acoustique Canadienne 24(1) 3-11 (1996) **Research article / Article de recherche**

DEVELOPMENT, EVALUATION AND SCORING OF A NONSENSE WORD TEST SUITABLE FOR USE WITH SPEAKERS OF CANADIAN ENGLISH

Margaret F. Cheesman and Donald G. Jamieson
Hearing Health Care Research Unit
Department of Communicative Disorders
The University of Western Ontario
London, ON N6G 1H1 CANADA

SUMMARY

Hearing researchers and clinicians frequently need to estimate the overall accuracy of consonant identification for a listener, over time or in various listening conditions, and to know how frequently specific types of consonant confusion errors are made in each condition. The present paper summarizes the development of a closed-set nonsense word test that provides both a general measure of listeners' abilities to identify consonant sounds, and an indication of the types of confusion errors that listeners make. The acoustical characteristics of test items and statistics of performance measures are summarized and two different scoring procedures are evaluated. The test, termed the University of Western Ontario Distinctive Features Differences test (UWODFD), is comprised of high-quality digital recordings of 21 items spoken by four native speakers of Canadian English; two male and two female. All items occur in a fixed, word-medial context. All aspects of testing, including presentation of stimuli, recording of subject responses, and the scoring and presentation of results, are under computer control. The test can be administered relatively quickly, it has been found to be appropriately sensitive to changes in listening conditions and has been used successfully with listeners from a variety of linguistic backgrounds.

SOMMAIRE

Les chercheurs et les practiciens orthophonistes ont souvent besoin d'estimer la totale exactitude de l'idenfication des consonnes, selon un laps de temps précis ou dans des conditions d'écoute variées, et de savoir à quelle fréquence des erreurs de confusion des types spécifiques de consonnes, se produisent dans chaque condition. Cet article résume le développement d'un test sur un ensemble délimité de mots insignificants qui implique à la fois une évaluation générale des capacités de l'auditeur à identifier des sons consonantiques, et une indication des types d'erreurs de confusion que font les auditeurs. Les caractéristiques des tests de mots et les statistiques des degrés de performance sont condensées et deux différentes procédures de scores sont évaluées. Le test, intitulé de Test Différetiel des Traits Distinctifs de l'Université de Western Ontario [UWODFD], est effectué à partir d'enregistrements digitaux de haute qualité de 21 mots énoncés par 4 anglophones canadiens; deux masculins et deux féminins. Tous les phonèmes apparaissent dans un contexte déterminé, en position médiane du mot. Tous les aspects du test, y compris la présentation des stimuli, l'enregistrement des réponses du sujet, le score et la présentation des résultats, sont sous contrôle informatique. Le test peut être administré relativement vite, il s'est avéré suffisamment sensible [de façon appropriée] aux changements des conditions d'écoute et a été utilisé avec succès auprès d'auditeurs de diverses provenances.

1. INTRODUCTION

The measurement of the ability of a listener to perceive spoken language is a fundamental need for many audiologists and hearing researchers. As examples, such measurements are important in medico-legal applications requiring a measurement of the speech-related hearing disability experienced by a listener, in rehabilitation research applications requiring quantification of the benefit that a specific hearing aid provides to a given listener, in the development of improved speech compression, synthesis, and coding systems, and in the evaluation of spoken language learning by English as a second language students. Suitable speech intelligibility tests must be sensitive (i.e., yielding different results for different listening conditions), valid (i.e., yielding results that are related to "real-world" performance), reliable (i.e., yielding results that are highly reproducible), and feasible (i.e., able to be used easily by subjects and clinicians working under typical circumstances).

No single test is likely to meet all needs, so that a battery of tests is normally required for use in research studies. One important component of such a battery is a measure of listeners' abilities to understand speech based purely on the acoustic information provided to them -- i.e., for which performance was not strongly influenced by higher-level cognitive ability. A test should be computer-controlled, with high-quality recordings of speech in the dialect of the local subject pool. For hearing researchers at the University of Western Ontario, subjects are typically native speakers of central Canadian English. Researchers in this group required a test that provided both an overall measure of intelligibility and a diagnostic measure with which to characterize the specific pattern of confusion errors made by listeners. Unable to find an existing test that met these criteria, a new test was developed which drew from the assets of several existing tests. This paper describes the development and evaluation of this test.

1.1. Objectives and Test Specifications

Five characteristics of the test materials were determined to be essential characteristics of the new test: (1) the target sounds should be representative of all consonant sounds; (2) target sounds should be presented in intervocalic position, to approximate the contextual cues to consonant identity which are available in "running" speech; (3) speech tokens should be obtained from at least four talkers, two men and two women, the accents of all speakers being appropriate for the typical UWO listener; (4) high-quality digitized acoustic signals should be used; (5) all speech tokens should be free of idiosyncracies and anomalies in pronunciation and intonation, and free of apparent accent to central Canadian English-speaking listeners.

Four characteristics of the test implementation were determined to be key: (1) the test was to be automated with stimulus selection, stimulus presentation, presentation of response alternatives, response recording, response scoring, and presentation of results to be under computer control; (2) the administration of a complete form of the test was to require not longer than five minutes, under typical testing situations; (3) the test was to be suitable for use with all adult subjects, and by most adult patients typically seen in clinical situations; and (4) a final characteristic of the test was the ability to analyze the test results in a variety of ways, including overall percentage of correct responses, confusion matrices and feature scoring.

2. SELECTIVE LITERATURE REVIEW

Analytic consonant perception tests have received increasing attention for use in audiological habilitation, particularly as potential tools for hearing aid evaluation (e.g., Jamieson, Brennan & Cornelisse, 1995). Test results can be summarized in the same way as the tests traditionally used as part of an audiological assessment, when they are scored in terms of the overall percent correct word identification or as the signal-to-noise level required to obtain some fixed level of performance. However, in addition, analytic speech perception test responses can be examined with respect to the pattern of errors that occur, i.e. they can provide analytical insight into the nature of the perceptual confusion.. Either confusion matrices (Dubno & Levitt, 1981; Miller & Nicely, 1955; Gordon-Salant, 1987) or feature-based scoring (Feeney & Franks, 1982; Danhauer & Singh, 1975; Miller & Nicely, 1955) can be used to quantify the pattern of response errors.

A number of tests have been developed to address objectives similar to those outlined above. Phoneme-based tests introduced over the past 20 years include the CUNY Nonsense Syllable test (Levitt & Resnick, 1978; Resnick, Dubno, Hoffnung & Levitt, 1975; CUNY-NST), the Modified Rhyme test (House, Williams, Hecker & Kryter, 1965; MRT), the Diagnostic Rhyme test (Voiers, 1983; DRT), and the Four Alternative Auditory Features test (Foster & Haggard, 1987; FAAF). Such tests restrict the set of response alternatives available to the listener on any particular trial to a subset of the complete consonant set. The choice of alternatives is based on the *a priori* probability of errors and/or restriction to confusions along a particular (feature) dimension.

As an example, the CUNY-NST tests initial and final consonant positions separately, within three vowel environments, /i/, /a/, and /u/. It contains 62 items, grouped into seven subtests. Each subtest is designed to measure consonant identification with a focus on a particular feature, within a syllable-initial or syllable-final position, and in one of the three vowel contexts. However, because the testing format involves a restricted set of speech stimuli and possible responses, subjects' confusion errors are restricted to those stimuli contained in the specific distractor set. In many instances, it is of interest to determine which errors subjects will make, when the range of these errors is not constrained through the *a priori* selection of the stimulus and response sets.

2.1. Feature-Based Testing

Some speech testing procedures offer the advantage of permitting feature-based scoring procedures. Feature-based scoring procedures measure performance in terms of a set of acoustic, phonetic, or perceptual features, rather than merely in terms of the proportion of complete consonant targets that are identified correctly. A feature approach has appeal for both clinical and research applications because it may be more sensitive to small differences in listening conditions than is whole item scoring (Feeney & Franks, 1982). Feature-based testing may therefore permit more efficient assessments of speech perception ability than testing based on whole items.

A similar argument has been made by Boothroyd (1968), who proposed scoring word lists on a phoneme-by-phoneme basis to increase the sensitivity of tests based on word lists. Efficiency is particularly important, because testing is costly and testing time is often severely restricted, such as when speech perception ability is assessed as part of hearing aid evaluation research, or in clinical applications. Historically, the routine application of feature-based scoring procedures has been precluded by the relatively complex scoring methods required. However, the widespread availability of computer-assisted testing protocols in audiological facilities has reduced such considerations.

2.2. The Distinctive Feature Differences Test (DFD)

Feeney and Franks (1982) developed a closed-set consonant recognition task that was designed to be scored on the basis of a set of distinctive feature confusions rather than whole phoneme recognition. This Distinctive Feature Difference (DFD) test was formed from 13 target consonants (/b,t,d,f,dz,k,p,s,ʃ,tʃ,θ,δ,v/) presented in an /ʌCIl/ context (e.g., "abil"). These 13 consonants were chosen because they were the consonants frequently perceived in error by hearing-impaired listeners, when presented in word-initial or word-final positions (Owens & Schubert, 1968). Because target consonants occurred in syllable-medial position in the DFD test, contextual cues to consonant identity were preserved in adjacent portions of the vowel-consonant-vowel syllable (VCV) as would be expected to occur for many consonants in continuous speech.

Feeney and Franks (1982) reported that feature-based scoring of the DFD test increased the reliability of the speech discrimination scores, because the number of scoreable units in the test could be increased without changing the amount of time required to complete the task. However, reliability coefficients were not reported for their test. Moreover, the DFD test was not automated, complicating administration, data collection, and scoring.

3. PRESENT WORK

The present study describes the development of a DFD test that is automated and consists of high quality digital recordings of all test items. Whole-item scoring for this test has been compared with feature-based scoring and both scoring procedures have been used in a variety of applications.

The test set includes a larger set of test items than Feeney and Franks' (1982) DFD test. For this test, designated the University of Western Ontario DFD (UWODFD), the set of consonant targets was increased to include most single English intervocalic consonants. The UWODFD is essentially an "open-set" test, because it includes most of the single consonants that can occur in the given context. The larger set of consonants allows listeners to make a broad range of perceptual errors and increases the range of perceptual confusions that can occur and the variety of alternative scoring schemes that can be used. To further increase generalizability, four different talkers, two men and two women, were used so the test includes a range of voices and speaking styles. All talkers were native speakers of central Canadian English, thereby increasing the appropriateness of the test for use with an anglophone Canadian subject or client population.

4. STIMULUS PREPARATION

4.1. Test items

Initial target test items were nonsense words of the form /ʌCIl/ in which **C** was one of the 22 consonants /b, tʃ, d, f, g, h, j, k, l, m, n, p, r, s, ʃ, t, θ, δ, v, w, y, z/ spoken by one of four talkers. The talkers were two male and two female young adults. All were native speakers of central Canadian English.

4.2. Recordings

To obtain the initial set of tokens, each talker was instructed to utter each target token within the carrier phrase "Point to the word /ʌCIl/". Several tokens of each word were digitized using the carrier phrase, while minimizing variation in the peak levels of the phrase across tokens. All recordings were made with the talker seated in a double-walled, IAC, sound-attenuating room, using a Shure unidirectional microphone coupled to a Shure M267 mixer. The output signal from the mixer was low-pass filtered at 8.0 kHz (Kemo VBF 25MD) and sampled to disk (16-bit recording at 20 kHz via an Ariel DSP-16 A/D card), using the Computerized Speech Research Environment (CSRE) software (Avaaz Innovations, 1995; Jamieson, Ramji, Kheirallah & Nearey, 1992). The test tokens were then edited from the carrier phrase.

4.3. Item selection

A series of behavioural tests was prepared that presented the speech tokens together with a list of the full set of response alternatives displayed on the computer screen (see below). Individual listeners then performed a sequence of tasks to identify speech sounds to be included in the final test protocol. This approach identified speech tokens that met the following criteria: (1) tokens were readily identifiable as the target sounds when presented in quiet to normally-hearing listeners; (2) tokens were rated as good exemplars of the target category; and (3) tokens were determined to be free of idiosyncracies such as atypical pitch contours, loudness differences, or pronunciation irregularities. Tokens that failed to meet all three criteria were deleted from the candidate set.

5. INSTRUMENTATION

Prior to statistical measurement of the long-term spectrum of the stimuli and their subsequent use in the perceptual tests reported here, the 84 digitized stimuli were converted to 12-bit samples to enable the tests to be undertaken with the equipment described below. Stimulus presentation was controlled with a DT-2801A D/A converter and low-pass filtered at 8.0 kHz. Signal level was controlled using a TTE PA-2 programmable attenuator and an Amcron D-75 amplifier.

For behavioural testing, the stimuli were presented monaurally to listeners via TDH-49 earphones. Listeners were tested individually while seated in an IAC double-walled sound-attenuating booth. The masking noise was generated by a TTE white noise generator and shaped to the ⅓-octave band L(eq, 5 min) of the 84 stimuli with two Industrial Research Products DG-4017 equalizers applied in series. The full-band long-term L(eq, 5 min) of the speech-shaped noise was 70 dB(A).

6. PILOT TESTING

Pilot testing with 16 normal-hearing young adult listeners was used to select the final test stimuli from the multiple recordings of each test item. During this testing, subjects were given a list of all recorded test items and were asked to identify each medial consonant when presented at 70 dB SPL. The final test items selected were highly intelligible under such optimal listening conditions, being identified with 95% accuracy or better, and were free of apparent idiosyncracies such as unusual intonation contours or syllable durations that might serve as cues to the identity of the consonant after repeated presentations of the test items.

The nonsense word /ʌθɪl/ was originally included in the set of test items. The results of the pilot identification testing indicated that, despite repeated attempts to obtain highly recognizable test tokens, /θ/ tokens were confused very often with /f/ tokens by the normal hearing listeners in quiet. Furthermore, inclusion of both the voiced and voiceless alveo-dental fricatives /θ/ and /ð/, for which English has no orthographic distinction, required some level of phonetic training and sophistication for the listeners and resulted in response errors that may have not accurately reflected perceptual errors. This is one limitation of a set of test materials that includes a wide variety of possible consonantal responses; the test format must provide unambiguous response items that are constrained by common orthographic practise. Elimination of /ʌθɪl/ resulted in a set of 21 response alternatives that could be unambiguously described using standard English spelling.

7. STATISTICAL DESCRIPTION OF STIMULI

Statistical descriptions of the long-term spectrum of the 84 stimuli (4 talkers x 21 consonants) were obtained through the sound delivery system using a Bruel and Kjaer 2231 sound level meter, statistical module BZ-7101, and a 1625 filter set using ⅓-octave settings. All measurements were made in a 6-cm^3 coupler. Statistical analyses of 5 minute samples of the continuous output (no silent gaps) of the 84 stimuli were made in ⅓-octave bands from 125 to 8000 Hz. The band pressure levels which were exceeded in 1%, 10%, 50%, 90%, and 99% of the 125 ms measurement intervals, and the L(eq) (Earshen, 1986), were measured when the overall level of the speech was adjusted to 70 dB(A).

The distribution of the ⅓-octave long-term speech levels is shown in Figure 1. The spectrum is dominated by the repeated high-intensity portions of the test stimuli, that is, the initial vowel and the second syllable. The dynamic range of the speech spectrum, computed as the difference between the band pressure levels exceeded in 99 and 1% of the measurement intervals, varies from 25.5 dB in the ⅓-octave bands centred at 315 and increases with increasing frequency, to a maximum of 40.5 dB in the 3150 Hz band.

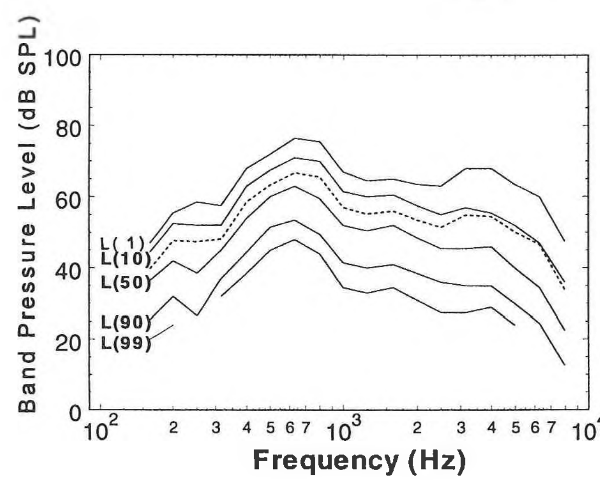

Figure 1. Distribution of the ⅓-octave long-term speech levels for 84 items contained in the UWODFD test. Dashed line is L(eq, 5 min).

To examine the spectrum of the target consonant in isolation from the surrounding context, the target consonants were edited from the test stimuli using a wave-form editor (Jamieson et al., 1992). Formant transitions were included with the consonants. The distribution of these excised consonants is presented in Figure 2. The influence of the adjacent vowels remained visible, however, the dynamic range of the consonant-only portion of the speech materials is narrower than for the entire nonsense word, particularly in the higher frequency regions.

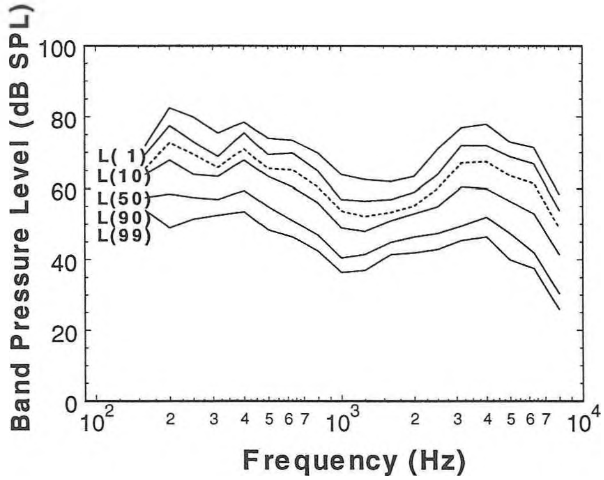

Figure 2. Distribution of the ⅓-octave long-term speech levels of the target consonants only. Dashed line is L(eq, 5 min).

8. BEHAVIOURAL TESTING

8.1. Subjects

Subjects were twenty young adult (age range 20-34 years) staff and students at the University of Western Ontario. All had pure-tone thresholds better than or equal to 20 dB HL (ANSI, 1989) from 250-8000 Hz in the test ear.

8.2. Procedures

No carrier phrase was used during nonsense word presentation. Within the test, stimulus presentation was blocked according to talker and within each talker block, the order of stimulus presentations was randomized without replacement. The listener's task was to choose which consonant was heard from a set of 21 possible responses displayed on a video monitor. The response alternatives were represented on the screen as **b, ch, d, f, g, h, j, k, l, m, n, p, r, s, sh, t, th, v, w, y,** and **z** [1]. Listeners selected one of these response alternatives prior to presentation of the next test item. The complete test of 84 stimuli was used in each speech-in-noise and filtering condition.

8.3.1. Performance-intensity functions. Performance on the test was measured in the presence of a 70 dB(A) noise that was shaped to the ⅓-octave L(eq) of the stimuli (cf. Figure 1 - dashed line). Thirteen signal-to-noise ratios (SNR) ranging from +4 to -20 dB in 2-dB steps were used. Following an initial test in quiet with the speech at 70 dB(A), the test was repeated 13 times, with the order of the SNR for each test randomized for each listener.

Six listeners also completed the test using an audiometer-generated speech-shaped noise masker (Grason Stadtler GSI-16) at eight SNR ranging from -15 to +15 dB and in quiet. The overall speech level was 75 dB SPL.

8.3.2. Filtered speech functions. Fifteen different filtering conditions for the speech stimuli were used: low-pass filtering at 250, 380, 550, 800, 1300, 2300, and 3500 Hz and high-pass filtering at 300, 550, 800, 1300, 2250, 3500, and 5500 Hz and a broadband condition (125 - 8000 Hz). Broadband speech-shaped noise was used in all conditions. The SNR for the equivalent broadband condition was fixed at +4 dB. Following an initial test in the broadband condition, the filtering conditions were completed in a randomized order.

9. RESULTS AND DISCUSSION

9.1. Performance-intensity functions

The mean performance scores as a function of SNR for the broadband listening conditions are shown in Figure 3. The slope of the performance-intensity function is very shallow, averaging 3%/dB in the SNR range from -20 to 0 dB. French and Steinberg (1947) obtained slopes of approximately 5%/dB for their nonsense syllable task and Duggirala, Studebaker, Pavlovic and Sherbecoe (1988) reported slopes of 5.74%/dB for the diagnostic rhyme test. The shallow slope obtained here with the UWODFD may be enhanced by the noise being matched to the combined spectra of the four talkers, rather than to each of the individual talkers (Studebaker, Pavlovic & Sherbecoe, 1987).

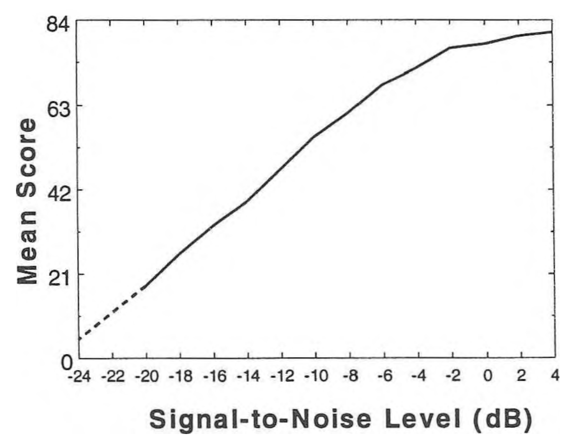

Figure 3. Mean performance scores on the UWODFD test, as a function of the signal to noise level for the broadband listening conditions.

In an independent test with speech shaped noise generated by an GSI-16 audiometer, the mean slope of the linear portion of the performance-intensity functions for 6 subjects, tested from -15 dB SNR to +15dB SNR, was 3.1%/dB. Thus, the very shallow function for the UWODFD test appears to be a property of the test itself rather than reflecting the specific noise used as a masker

Unlike conversational speech, where higher-level cognitive factors combine with the available acoustic information to produce very steep performance-intensity functions, shallow performance-intensity functions are expected for nonsense syllables. Such a shallow performance-intensity function has a significant advantage for applications where performance differences need to be measured over a wide range of listening conditions.

9.2. Filtered speech functions

The results of the filtered speech conditions are displayed in Figure 4, where the mean score for each of the four blocks (talkers) of the test is shown as a function of cut-off frequency. The crossover frequencies for the high- and low-pass conditions are slightly higher for the female talkers than for the males. The crossover frequency for the test taken as a whole is 2170 Hz, which is higher than that reported by French and Steinberg (1947) for nonsense syllables spoken by male and female talkers, and higher than other reports for nonsense syllables using male voices (Dubno & Dirks, 1989; Duggirala, Studebaker, Pavlovic & Sherbecoe, 1988).

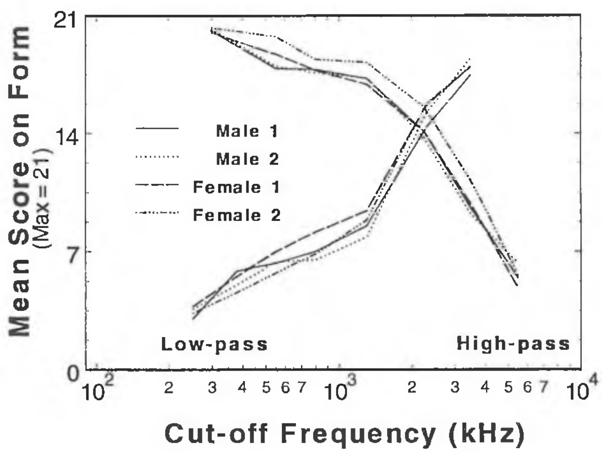

Figure 4. Performance as a function of filter cut-off frequency, for each of four talkers.

9.3. Applicability of conventional Articulation Index weights

Cheesman, Appleyard and Lawrence (1992) reported a series of studies designed to determine whether or not the Articulation Index (ANSI, 1969) frequency-importance weights for nonsense syllables could be applied directly to the UWODFD materials, without modification. ANSI Articulation Index weights did not result in accurate prediction of performance on the UWODFD test, with the fit being particularly poor for the filtering conditions. The dependence of the Articulation Index on a 30-dB dynamic range, which underestimates the dynamic range of the UWODFD materials particularly at higher frequencies (cf., Fig 1), combined with the high cross-over frequency for the UWODFD materials likely contribute to the poor predictive power of the Articulation Index for these materials.

9.4. Comparison of alternative approaches to scoring

The UWODFD test can also be scored using any of a variety of scoring systems based on phonetic feature descriptions of the signals. Feeney and Franks (1982) suggested using a seven-feature scoring system of Voice, Continuant, Strident, High, Back, Anterior, and Coronal for their DFD test. The extension of the stimulus set from 13 consonants to 21 consonants for the UWODFD test required additional feature scoring assignments. The results obtained when the data displayed in Figure 3 are scored using this system, are plotted in Figure 5.

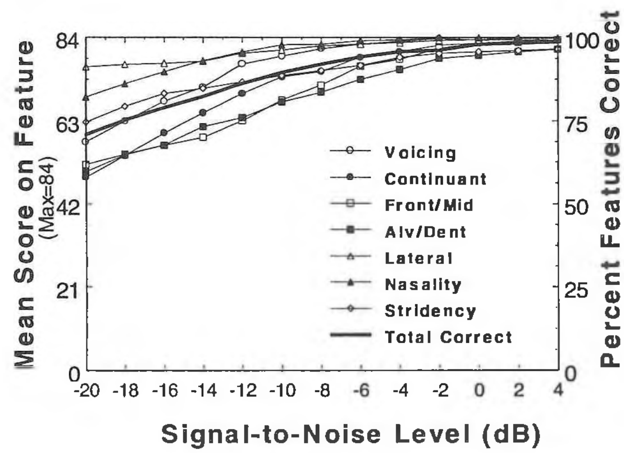

Figure 5. Performance-intensity functions for the data displayed in Figure 3, plotted as the number of correctly-identified features. A separate performance-intensity function is plotted for each of the features analysed (left axis). The total number of correct features (expressed as a percentage) is shown by the solid line (right axis).

Differences in the slope and form of the functions from feature to feature are clear. For example, some features have very low error rates, so they do not contribute to the aggregate curve; for other features, the performance-intensity curve is steeper, indicating that listeners are sensitive to the feature only over a very narrow SNR region.

This seven-feature analysis differs dramatically from the three-feature analysis provided by Cheesman, Lawrence, and Appleyard (1992) as shown in Figure 6. The score for the manner feature is similar to the whole item test score (cf. Figure 3) in the three-feature system. Because the seven-feature system breaks place and manner characteristics into several features each, there are fewer errors on any single place or manner-related features. This results in shallower performance-intensity functions for both the individual feature functions and for the function of total features correct.

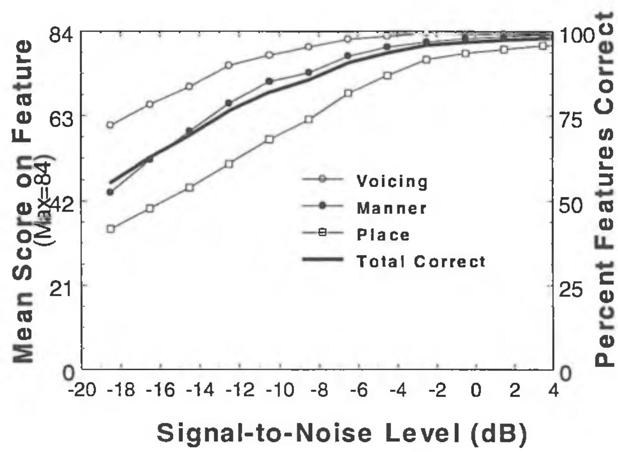

Figure 6. Performance-intensity functions for the three-feature scoring system used by Cheesman, Lawrence, and Appleyard (1992).

9.5. Possible advantages of feature-based scoring

According to Feeney and Franks (1982) and other authors, estimates of subject's performance on a speech intelligibility test such as the DFD are more reliable if data are scored in terms of specific feature errors rather than in terms simply of entire items being correct or incorrect. For example, using whole item scoring, a response of /d/ for /b/ and a response of /t/ for /b/ are equally severe errors. However, in a feature-based scoring approach, the /t/ response is more severe, as /t/ differs from /b/ in both Place of Articulation and Voicing, whereas /d/ differs from /b/, only in Place of Articulation (i.e., Voicing is reported correctly).

Another consideration is that the reliability of the test cannot be predicted readily from the number of test items when items in a test such as the distinctive feature difference test are scored on a feature-by-feature basis. This is because the binomial distribution is unlikely to approximate the test score distribution, because the individual features are not independent and errors are therefore correlated across features. For example, a relatively simple feature scoring system is one in which only place, manner and voicing features are scored as correct or incorrect. If the test item /ʌmɪl/ is presented and the manner feature is correctly perceived (as nasal) then the voicing feature will also be correctly identified, because voiceless nasals are not included in the response set of English consonants.

The data obtained from the filtering conditions can be used to evaluate the proposal that feature-based scoring increases reliability. Reliability coefficients were calculated both when the test was scored on a whole item basis and when the test was scored in terms of the percentage of features correctly identified. The procedures outlined by Winer (1962, **p.** 124) for estimating the reliability of measurements using an analysis of variance model were used.

The estimate of reliability obtained for a single measurement was .52 for both the whole-item and feature scoring approaches. The reliability of the average of the 15 measurements (equivalent to Spearman-Brown reliability, Winer, 1962) was .94 for both scoring methods. Thus, the estimated reliability of the measurements did not differ for the two procedures.

A similar pattern of results obtained with the noise-masked data, for which individual test administration reliability was .37 and .33 for whole-item and feature scoring, respectively, and .89 and .87 for the average of the 14 listening conditions, for whole-item and feature scoring, respectively.

Although these measures do not directly address test-retest reliability under identical listening conditions, they do indicate that, from a test reliability perspective, feature scoring using this seven-feature set does not provide an advantage over the whole-item scoring procedure, despite increasing the number of "scoreable units". This is in contrast to Feeney and Franks' (1982) hypothesis.

Notwithstanding this failure of feature scoring to improve the reliability of speech intelligibility estimates, feature-based scoring may offer an important analytical advantage over traditional speech perception measures. As one example, Jamieson, et al. (1995) used a three-feature (Place, Manner, and Voicing) scoring approach to examine the effects of applying a noise reduction scheme that used a "voicing detector" to toggle the estimate of the background noise provided to the processor. This analysis showed that voicing confusion errors did not increase when the noise reduction scheme was applied. Such a conclusion would not have been possible from consideration of whole-item test results alone and requires the analytic feature approach made possible by the DFD.

10. APPLICATIONS

This modified version of the DFD test has received extensive use in a variety of research projects undertaken by members of Western's Hearing Health Care Research Unit over the past several years. A frequent application has been evaluation of the benefit provided to individual hearing aid users by alternative hearing aid systems. This is a challenging task, requiring high test sensitivity, as the incremental benefit of switching a listener from one carefully-fitted hearing aid to another hearing aid with similar processing characteristics may be relatively small. Jamieson and Cornelisse (1992) used these speech test materials successfully in their evaluation of the differences in listener performance when hearing aid users were fitted with K-amp and linear hearing aids. Jamieson and

Brennan (1992) and Jamieson, Brennan and Cornelisse (1995) used the test successfully to measure the benefit provided to listeners by an adaptive noise reduction filtering system designed for use in future generations of digital hearing aids.

These test materials have also been used as a basic tool to evaluate overall speech intelligibility performance by individual listeners. As one example, Cheesman, Armitage and Marshall (1994) used the UWODFD to measure the speech perception abilities of younger and older Canadians, in a study examining the relation between speech perception ability and growth of masking. Yu and Jamieson (1994) used the UWODFD to quantify changes in the ability of native speakers of the Korean language to identify English-language consonants, following extended exposure to the English language after immigrating to Canada, and throughout the course of a structured program of English-language training.

11. CONCLUSIONS

The studies reviewed here have established that the UWODFD is an appropriate test for a variety of applications requiring measurement of listeners' abilities to identify English language consonants based primarily on acoustic information. The test has been shown to be appropriate for use with subjects from several different educational and cultural backgrounds, it can be administered and scored quickly, it is sensitive and has high reliability. For these reasons, it may prove useful for inclusion as part of a battery of tests for the measurement of spoken language perception. While there is no evidence that feature-based scoring increases the reliability of an overall measure of speech intelligibility performance, such scoring provides a level of analysis not available in conventional approaches.

ACKNOWLEDGEMENTS

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NOTES

1. We have since modified our response display screen to provide full orthographic representations of the nonsense words (e.g., abil, achil, adil, afil).

Stimuli are available from the first author at the address listed above or via e-mail at cheesman@uwovax.uwo.ca

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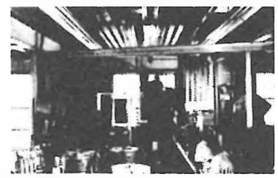

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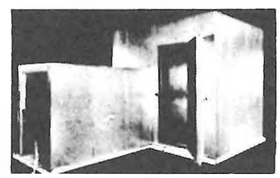

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Canadian Acoustics / Acoustique Canadienne 24(1) 13-22 (1996) ***Review article / Article de revue***

EQUALIZATION IN SOUND REINFORCEMENT: PSYCHOACOUSTICS, METHODS, AND ISSUES

Russell Ovans

Jason Sound Industries, 1709 Welch St., N. Vancouver, BC V7P 3G9
and
School of Computing Science, Simon Fraser University, Burnaby, BC V5A 1S6
ovans@cs.sfu.ca

SUMMARY

The practise of electroacoustic sound system equalization demands an understanding of psychoacoustics, room acoustics measurement, and subjective room acoustics. In this paper we undertake a review and synthesis of the literature pertaining to the relevant psychoacoustic and room acoustic phenomena, and relate it to a number of issues regarding the current methods of large venue sound system equalization.

SOMMAIRE

L'utilisation de systèmes équalisateurs de son électroaccoustique exige la compréhension de la psycho-accoustique, des mesures de l'accoustique de salle et de l'accoustique subjective de salle. Nous présentons dans cet articles une étude de synthèse de la litérature relative aux phénomènes de la psycho-accoustique et de l'accoustique de salle et relions cet étude à de nombreux problèmes qui se rapportent aux méthodes courantes pour les systèmes équalisateurs de son pour grande salle.

1. INTRODUCTION

Large venue sound reproduction systems consist of a network of signal processing equipment through which an original source signal is routed on its journey to multiple speakers at various locations within a room. As part of this network, equalizers are often used to modify the frequency spectrum of the source signal before it is fed to the speakers, in an effort to compensate for unevenness in the frequency response of both the speakers and the venue. In touring systems, how the audio engineer chooses to do the time-constrained and complex task of adjusting the equalizers is as much a black art as it is a science.

Our main objective in writing this paper is to review and hopefully synthesize much of the research relating to equalization as it pertains to electroacoustic sound reinforcement technology in large venues. While equalization relates to the way very large loudspeaker systems function in any acoustical environment (including outdoors), the discourse is restricted to enclosed spaces. The practise of room equalization requires an understanding of the science of room acoustics and the psychoacoustic considerations of mapping objective, quantifiable measurements to subjective listener preference. In this respect, equalization is closely related to the field of subjective room acoustics.

In this paper we first consider the relevant psychoacoustic (Section 2) and room acoustic phenomena (Section 3) before introducing a number of issues regarding the practise of sound system equalization (Section 4). Though our particular perspective is touring systems and audio engineering, it is our belief that the principles discussed are also of interest to acoustical consultants for fixed installations.

2. PSYCHOACOUSTIC PRELIMINARIES

Psychoacoustics is the specific branch of psychophysics concerned with the relationship between the objective, physical, and quantifiable properties of sound stimuli in the environment and the subjective, psychological, and qualitative responses they evoke [Rasch82a]. There are two important psychoacoustic issues with respect to equalization: the frequency response and critical bandwidth of the ear.

2.1 Frequency Response of the Ear

Pitch is the perceptual correlate of frequency. However, this correlation is not linear: as the number of cycles per second increases linearly, our perceived sense of pitch increases only logarithmically. Alternatively, as our sense of pitch increases linearly, the frequency increases exponentially. For example, the doubling of frequency with every octave represents an exponential growth in frequency as our pitch impression grows linearly.

The range of human hearing is well known to be 20 - 20,000 Hz. However, perception is not equally sensitive at all frequencies; i.e., the ear does not exhibit flat frequency response. Fletcher and Munson's famous curves of equal loudness [Fletcher33] illustrate quite clearly that the ear's sensitivity to loudness is frequency dependent. The main characteristic of the F-M curves is decreased sensitivity at low and high frequencies, but as intensity increases, sensitivity flattens out. At any level, maximum sensitivity occurs at about 3 kHz, corresponding to the resonant frequency of the ear canal [Houtsma87].

The F-M curves were determined using a small set of pure tones in an anechoic space with the sound source directly in front of the test subjects. The listeners had one ear blocked with cotton balls soaked in Vaseline. The F-M results are therefore a measure of the monaural perception of pure tones in a free field with an on-axis sound source.

However, none of the succeeding studies are in agreement with the F-M curves [Holman78]. In particular, the ISO adopted curves for free field listening are parallel at all levels above 400 Hz. So while the ear is not flat at high frequencies, it does exhibit the same response regardless of level. Moreover, the frequency response of the ear depends on the sound field [Holman78] and the position of the sound source [Fletcher53]. According to ISO standard 454, in a diffuse field, the ear is +3 dB more sensitive at 1 kHz, -2 dB at 2.5 kHz, and +4 at 10 kHz. Staffeldt and Rasmussen have shown these numbers to be an approximation of the directional sensitivity of the ear to high frequencies, due to the diffraction caused by the head, torso, and ears. Perceptual sensitivity is a function of distance from the sound source and the room size [Staffeldt82]. The distance from the source changes the diffraction caused by the head and external ear. The size of the room influences the amount of diffusion.

Particularly important is the diffraction due to the pinnae, or outer ear flaps. At its simplest, the pinna acts as a low-pass filter for sounds from behind the head, which provides a cue for distinguishing front from back for high frequency sounds. Research in the 1970s produced convincing evidence that additional localization cues are provided by the reflections of the incident sound off the intricate ridges and depressions of the pinna. These reflections introduce short time delays that are manifest as high-Q notches in the frequency response starting at approximately 6 kHz [Rodgers81]. Because of the geometry of the pinna, as a sound source is raised in elevation the first prominent notch in the frequency response occurs at a higher and higher frequency. Kendall and Martens later asserted that we use these head-related transfer functions as a mechanism for localization on the vertical and front/back planes [Kendall84].

In summary, the frequency response of the ear is dynamic, depending on the listening environment, loudness, and position of the sound source.

2.2 Critical Bandwidth

The basilar membrane – the main sensing mechanism of the ear – is a 35 mm long spiral coil that bulges at a frequency dependent location in response to sound stimuli. The *critical bandwidth* for a given frequency is the smallest band of frequencies around it that will activate the same part of the basilar membrane [Truax78]. Perceptually, the critical bandwidth is the ear's resolution of discrimination; i.e., its resolving power for *simultaneous* tones.

Plots of the size of critical bandwidth as a function of centre frequency indicate that the bandwidths lie between 1/3-octave and 1/6-octave for frequencies above 400 Hz [Houtsma87]. Below 400 Hz the bandwidth is more or less constant at a rather staggering 100 Hz. 24 critical bands traverse the length of the cochlea and therefore define the range of hearing. However, critical bands are different than 1/3-octave analyzers in that "the set of critical band filters is continuous; that is, no matter where you might choose to set the signal generator dial, there is a critical band centered on that frequency" [Everest89, p. 32].

An understanding of critical bandwidth is important to the practise of equalization as it is often (erroneously) cited as a psychoacoustic basis for choosing a particular measurement resolution. Critical bandwidths are more directly relevant to theories of consonance and dissonance (i.e., the subjective agreeability or disagreeability of simultaneous sounds). Two simultaneous pure tones within a critical bandwidth of each other, but not of the exact same frequency, are perceived as dissonant. The two tones result in beats if close together, roughness if further apart, until finally breaking into separate distinguishable tones once they differ by the limit of frequency discrimination. Consonance results only once the tones cross the critical difference and henceforth differ by at least the critical bandwidth. Sounds with spectral content that cross critical bandwidths are perceived as louder than sounds that do not, even if the two sounds have equal rectangular area of sound intensity (defined by intensity per Hz).

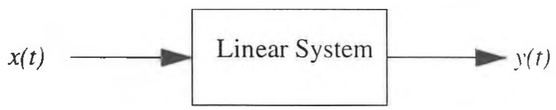
3. MEASUREMENT AND EVALUATION

There is nothing quite as upsetting as viewing one's first attempt at measuring the 'frequency response' of a room [Everest89, p. 205].

The three core objective parameters of sound quality for a room are reverberation time, frequency response, and the impulse response. Reverberation time and frequency response are both derivable from the impulse response. In order to understand the impulse response, the room first must be understood as a linear system.

3.1 Linear Systems

A linear system is a mathematical abstraction used to describe any system where the relationship between the output and input is governed by a linear differential equation with constant coefficients:



where $x(t)$ and $y(t)$ are time domain representations of signals; for our purposes, these functions represent time-varying sound pressures.

The other predominant way to represent a signal is the frequency domain, whereby a signal is described by the presence of energy at certain frequencies. The two domains are duals: equivalent information is contained in each. Transforms are mathematical tools that enable the movement from one domain to the other.

The Fourier transform is a method of converting between the time and frequency domains. Named for the French mathematician, it is based on his famous theory which states that any periodic time-varying signal can be expressed as the sum of an infinite series of sine and cosine terms each with a specific amplitude and phase. If $x(t)$ is a signal, its Fourier transform $X(f)$ is a function that maps frequency onto a complex number $A + Bi$. The amplitude of the signal content at frequency f is the magnitude of the complex number (i.e., the square root of $A^2 + B^2$) whereas the phase (i.e., its relative alignment) is given by the argument ($\theta = \operatorname{atan}(B/A)$). The frequency domain therefore consists of both an amplitude response and phase response. By convention, the term *frequency response* refers to the amplitude response only.

In practise, one is restricted to discrete time signals obtained with a particular sampling interval. The fast Fourier transform (FFT) is the name for a class of algorithms that quickly compute the Fourier transform of a discrete time signal.

Every linear system is completely described by either its impulse response, or its amplitude and phase response. The impulse response, $h(t)$, is a system's output to the delta function. The delta function, $\delta(t)$, is defined by:

$$\delta(0) = 1$$
$$t \neq 0 \Rightarrow \delta(t) = 0$$

The output $y(t)$ of a linear system to an arbitrary input $x(t)$ is the convolution (*) of the input and the impulse response:

$$y(t) = x(t) * h(t)$$

Convolution works as follows. Each $x(t_i)$ can be thought of as a scaled and delayed version of the delta function. Each $x(t_i)$ therefore produces a scaled, delayed version of $h(t)$. The output $y(t_i)$ is the sum of the scaled, delayed versions of $h(t)$ as generated by each $x(t_j)$, $j \leq i$.

Convolution in the time domain is equivalent to multiplication in the frequency domain:

$$y(t) = x(t) * h(t)$$

$$Y(f) = X(f)H(f)$$

where $X(f)$, $Y(f)$, and $H(f)$ represent $x(t)$, $y(t)$, and $h(t)$ respectively in the frequency domain. The transfer function of a system is its ratio of output to input expressed in either the time domain or the frequency domain. The impulse response is the time domain representation of the transfer function. The frequency domain representation is given by:

$$H(f) = \frac{Y(f)}{X(f)}$$

i.e., the ratio of output to input.

If $h(t) = \delta(t)$, then $y(t) = x(t)$ and $H(f) = 1$, for all f. That is to say, a system with perfectly flat frequency response and no gain or attenuation has an impulse response equal to the delta function.

3.2 Interpreting the Impulse Response of a Room

A room's effect on sound can be modeled as a linear system. Since the impulse response completely defines a linear system, all characteristics of interest are derivable from it, including in the case of a room, reverberation time and frequency response. The traditional method of obtaining a room impulse response is to excite the room with an impulse and to record the decaying sound pressure. A recent paper by

Norcross and Bradley compares four competing approaches to obtaining the impulse response of a room, each of which is shown to produce similar results [Norcross94].

The true impulse response h of a room is an oscillating signal of amplitude vs. time. In acoustics literature, the impulse response is often instead illustrated as the squared impulse response p^2, which is a plot of the square of the amplitude vs. time. The energy-time curve (ETC) is another non-negative real-valued alternative to the impulse response. The ETC and its calculation are described in [Duncan88].

Given an impulse response, the frequency and phase response is contained within its Fourier transform. The reverberation time (RT_{60}) is the length of time it takes the impulse response to attenuate 60 dB. $RT_{60}(\lambda)$ is the reverberation time when the room is excited not with an impulse (which has equal energy per frequency), but with a pure tone of wavelength λ.

In applying linear systems theory to room acoustics, an important consideration is that every pair of source-receiver locations defines a different transfer function. The room as a whole does not possess a single impulse response, but rather defines one for each pair of possible locations. RT_{60} is therefore a function of not only frequency, but also location. "When reverberation time for a given frequency is reported, it is usually the average of multiple observations of each of several positions in the room. This is the pragmatic way of admitting that the reverberatory conditions differ from place to place in the room" [Everest89, p. 207]. Only in a perfectly diffuse sound field is RT_{60} the same at each frequency and location. However, "one still talks of 'a concert hall with RT_{60} of 1.8 s,' as if true for all frequencies and true for all source-receiver combinations in the hall" [Barman93]. The sensitivity to position can be striking: changes in distance as little as 10 cm of either the source or receiver can result in statistically significant changes in measured values of the early decay time (EDT, the first 10 dB of decay) [Bradley89]. Another controversial and inconclusive study [Barman93] reported differences of over 0.5 s at different measurement locations, and 0.6 s at different frequencies for the same location in a large room.

Interpretation of the impulse response and reverberation time is most closely associated with the field of subjective room acoustics. Subjective room acoustics is the psychoacoustic study of perception in enclosed spaces with the goal of determining the important quantitative variables in the design of concert halls and auditoria. According to Rasch and Plomp in the introduction to their excellent survey of the field, subjective room acoustics is the study of the perceptual effects of the indirect sound field, which is responsible for what is loosely called the *acoustics* of a room or hall [Rasch82b]. The indirect sound field is comprised of sound that arrives after one or more reflections. Indirect sound is further classified as either early reflections if it arrives within 50 ms of the direct sound, or reverberant sound otherwise. Depending on the context, early reflections are often counted as part of the direct sound. The indirect sound has three effects [Rasch82b]:

1. it adds sound energy resulting in a perceived increase in loudness;

2. it arrives later than the direct sound, thus reducing definition as it masks the preceding direct sound; and,

3. it arrives from other directions than the direct sound, resulting in a perceived spaciousness.

Although reverberation time is now considered inadequate as a single objective descriptor of room quality [Bradley90], it was once fashionable to consider the question, what is the optimal reverberation time? A thorough theoretical examination of this question is given in [Mankovsky71] where a number of plots of optimal reverberation time versus room size are shown to be inconclusive. Certainly a high reverberation time is a problem for speech intelligibility as it means the masking of new information by old. But higher reverberation times (1.5 to 2.1 s) are acceptable and even desirable for music, particularly romantic classical music [Rasch82b]. The lesson to note is that flat frequency response is not optimal for the reproduction of music.

In current thinking, the ratio of direct to indirect sound is considered more important than reverberation time in predicting the quality of a room's acoustics, particularly with respect to speech intelligibility. As such a number of measures of room quality have been proposed that are based on the ratio of two integrals of the impulse response (see for example [Bradley90, Davis87]). These ratios of integrals usually differ only in how much of the early reflections count towards the direct sound. Citing the Haas effect [Haas72], many descriptors count early reflections that arrive within 50-80 ms as part of the direct sound. These newer ratios of direct to indirect sound are important, Bradley and Halliwell claim, because they relate well to the subjective assessments of the acoustical characteristics of halls [Bradley89].

However, methods of prediction based entirely on the time domain (i.e., RT_{60} and direct/indirect ratios) suffer two serious shortcomings due to the following limitations of the impulse response as a representation device.

1. It provides no clues as to the directionality of the reflections. The lack of information concerning the directivity of the reflections is critical since "after it was established that early reflections are subjectively important, the direction of arrival of these reflections was next found to be important"

[Bradley90, p. 17]. In particular, early lateral energy (first reflections from the side walls) is considered of fundamental import [Barron71, Schroeder84]. In summarizing the various ratios of energy as a predictor of room quality, Rasch and Plomp note the importance of sound coming from the sides and from the rear later than 40 ms and earlier than 80 ms after the direct sound [Rasch82b]. Because it arrives earlier than 80 ms, it functions as direct sound and improves clarity. Since it arrives from the sides and back after 40 ms, it increases the sense of spaciousness. Since auditorium design is generally a trade-off between clarity (for speech) and spaciousness (for music), they conclude that these reflections are potentially very important. In an attempt to measure directional characteristics of reverberation, Abdou and Guy developed a PC-based measurement system that employs six microphones arranged in cartesian coordinates [Abdou93]. Their system captures the temporal arrival, direction, and magnitude of reflections and plots this information as a series of intensity vectors in time.

2. It provides no immediate clue as to the frequency response. While it is true that frequency domain information is contained within the impulse response, it is not obvious what it is simply from inspection. In other words, the time and frequency domain representations share the same information, but their respective representations are more amenable to the extraction of different information. For example, Toole has criticized the impulse response because it is inferior to the frequency domain for the identification of audible resonances [Toole86a]. Given that the goal of the direct/indirect ratios is to move towards understanding what a desirable transfer function for a room is, one wonders why work in subjective room acoustics seems universally restricted to the time domain. Since the amplitude response is basically a picture of the relative RT_{60} along the frequency axis, it seems strange to dismiss it as a tool. Conversely, audio engineers working with electroacoustic sound reinforcement systems operate exclusively with frequency domain representations.

There are, however, many valuable guidelines to be learned here from the work in subjective room acoustics. In particular, the importance of directivity of reverberation. Moreover, it would be instructive and interesting to consider both the impulse *and* frequency response measurements of halls judged to be excellent.

4. EQUALIZATION

Equalization is the purposeful alteration of a signal to add and/or remove spectral content. From an engineering perspective, equalization is the deconvolution or inversion of the transfer function of a linear system. This is based on the assumption that all artifacts of the intervening linear system are unwanted.

With sound reinforcement systems the common practise is to alter the frequency spectrum of a signal using a 1/3-octave equalizer, which is a collection of 30 independent bandpass filters that each can boost or cut the signal by approximately 12-dB at their centre frequency. This compensation, applied just before the amplifiers in the audio chain, is to correct for aberrations in the response due to the interaction of the loudspeakers and room. The goal of this compensation is twofold [Davis87]:

1. to ensure a specified tonal response at each listener's ears; and,

2. to maximize overall acoustic gain by reducing peaks in the frequency response that can cause the system to enter a feedback loop.

Implicit in point 1 is that the specified tonal response will result in improved sound quality. To that end, equalization of the frequency (amplitude) response is considered the single most important method of improving the listener preference rating of a loudspeaker [Fortier94].

ISO standard 2969 describes a recommended method for equalization: excite the room with pink noise, measure the frequency response using a 1/3-octave real-time analyzer (RTA), and adjust the equalizer until a certain curve is realized on the RTA. The frequency response of a venue as viewed on a 1/3-octave RTA is called the house curve. Pink noise is used because it has equal energy per octave. White noise, because it has equal energy per Hz, exhibits a +3-dB/octave rise in energy with increasing frequency and is therefore less suitable when using constant percentage bandwidth filters such as those used in 1/3-octave analyzers.

This ISO standard raises many issues and questions:

1. What is the ideal house curve?
2. Do you equalize based on measurement of the direct, indirect, or total sound?
3. Is a single measurement point adequate?
4. Should the audience be present?
5. What about time domain equalization?
6. Is 1/3-octave resolution enough?

In the remainder of this section we consider each of these questions in turn.

4.1 The Ideal House Curve

The immediate question arises: what is the ideal house curve? Toole has presented convincing evidence that in anechoic conditions, listeners prefer loudspeakers with the smoothest and flattest frequency response, both on- and off-axis [Toole86b]. Conversely, the Athena project has suggested that in a typical small listening room flat frequency response is **not** the optimal transfer function [Fortier94], but project participants have not revealed what they believe it to be.

Given this inconsistent state of affairs, let us first consider what equalization should attempt to correct. Bucklein has examined the effect a nonuniform frequency response has on speech intelligibility over telephone lines [Bucklein81]. The result of his study, which also held for music and white noise, was that peaks in the transfer function are clearly more disturbing than corresponding valleys. Satisfactory intelligibility requires that narrow peaks must be avoided, while several small valleys, even if these are deep, are tolerable. Test subjects perceived no difference in the source material if the transfer function contained a single 5-dB valley an octave wide. The narrower a valley becomes, the greater its depth must be to remain audible; e.g., a 20-dB dip with bandwidth $\Delta f / f = 0.2$ was judged inaudible at all frequencies measured (NB: $\Delta f / f = 0.23$ for 1/3-octave). The subjective judgement is also worth note: the listeners reported that the audible valleys do not alter the sound quality as much as equally large peaks, which can appear "very unpleasant." If peaks are unavoidable, two narrow peaks are better than one wide one, and the farther apart, the better. Note the consistency with critical bandwidth theory, which predicts that wide peaks that cross critical bandwidths will be perceived as louder than narrow peaks that do not. A number of widely spaced peaks is better than a single wide peak in terms of intelligibility (and corresponding tonal colouration).

Current guidance – as espoused in for example [Davis87] – is that one should measure the house curve with a flat response free-field microphone placed about 30 m from the source. The equalizer should be adjusted so that the house curve is flat up until about 1 kHz where a roll-off down to -10 dB at 10 kHz should begin. What explanation is there for this high-frequency attenuation? Papers by Schulein [Schulein75] and Staffeldt and Rasmussen [Staffeldt82] address this question. Taken together, these two papers are crucial in understanding the psychoacoustic considerations of equalization.

Schulein considers the question of the high-frequency roll-off: why is it that a flat house curve, obtained by exciting a sound reinforcement system with pink noise and viewing on a 1/3-octave analyzer, sounds too bright? Through an ingenious experiment, Schulein deduced two causes: the increased sensitivity of the human auditory system to high-frequency diffuse sound as opposed to near-field frontal sound; and, the roll-off in diffuse-field sensitivity versus free-field sensitivity in commercially available measurement microphones. "Due to the polar characteristics of the human listener, a lower sound pressure level is required for equal loudness at high frequencies for a diffuse sound field than for a frontal sound field" [Schulein75, p. B-47]. He cites the shape of the head as the culprit, and suggests the design of microphones that mimic the resulting directional pattern. Microphones embedded in dummy heads would seem a more expedient alternative.

Schulein's view is reinforced by the work of Staffeldt and Rasmussen [Staffeldt82]. The important points are as follows. If a human equalizes two loudspeakers such that they sound equally loud at all frequencies, and one of the loudspeakers is in the distance such that it produces a reverberant field, and the other is within the critical distance such that it mimics a free field – the distant loudspeaker producing a diffuse field will have its high frequencies attenuated. Equivalently, if you replace the human with a microphone and do equalization such that the measured frequency response of both loudspeakers is flat, the diffuse field loudspeaker will sound brighter. However, if you embed that microphone inside the ear of a dummy head and repeat the process, the perceived brightness disappears. It is not adequate to use an omnidirectional microphone with flat free field sensitivity and flat diffuse field sensitivity; the dummy head is necessary. In fact it is psychoacoustically invalid to use an omnidirectional microphone: as noted before, the human ear is not uniform in directivity at high frequencies. For example, the ear is +10-dB more sensitive at 6400 Hz to sound 90-degrees off-axis than it is to sound on-axis [Fletcher53]. In a diffuse field, where sound is entering the ear from all directions, this sensitivity is stimulated.

Moreover, high frequency response is dependent on distance to the source (regardless of whether the listener is inside or outside the critical distance): an equalized loudspeaker in anechoic conditions will sound different with distance. This is because the free-field response of the external ear depends on the distance between the head and the loudspeaker:

...It is concluded that the high-frequency attenuation necessary for a distant loudspeaker when compared with a nearby loudspeaker is largely determined by **[BOTH]** *the free-field and diffuse-field diffraction phenomena at the head and the external ear* [Staffeldt82, p. 642].

As a caveat, Staffeldt and Rasmussen warn that these results may not be generalizable to large venues.

4.2 Direct vs. Indirect Sound

Modern sound reinforcement systems for large spaces are almost always comprised of multiple loudspeakers. The use of multiple source positions impacts the perceived reverberation of an enclosed space.

The critical distance is the point at which the intensity of the direct field is the same as that of the reverberant field. Beyond the critical distance, the ratio of direct to indirect sound steadily degrades. Ideally then every seat in the audience should be within the critical distance. Unfortunately, for all venues of any significant size, most of the audience is beyond the critical distance. That is, most people are listening to the reverberant sound field more than the direct field. For example, an omnidirectional sound source in a large concert hall (volume = 27,000 m^3, $RT_{60} = 2.2$ s) has a critical distance of 11 m [Plomp73]. Since intelligibility and clarity are proportional to the ratio of direct to indirect sound [Rasch82b], sound quality can degrade significantly beyond the critical distance.

Employing multiple loudspeakers is not the solution. In fact, multiple loudspeakers is part of the problem. Every loudspeaker that is added contributes to the indirect field and therefore degrades the direct/indirect ratio. The optimal direct/indirect ratio is obtained with a single radiating point.

O'Keefe has looked at the problem of critical distance and multiple loudspeakers in very large reverberant spaces. "...The fundamental dilemma associated with very large rooms: increasing the number of speakers means that some people will be exposed to better direct and early sound. For people located elsewhere in the room these same loudspeakers will introduce detrimental late sound" [O'Keefe94, p. 71]. For the Galleria in Toronto, a 90,000 m^3 space, O'Keefe separately calculated and measured the direct, early, and late sound and plotted their intensity as a function of distance from sound source. The direct and early sound levels decayed linearly with similar slope; the late sound was virtually constant. The point at which the direct and reverberant lines cross is of course the critical distance, which he found to be 7 m. He empirically noted that beyond 7 m speech intelligibility decreased significantly. This was for a single loudspeaker: "the important difference between a single loudspeaker system and a distributed system with several loudspeakers is that the distant loudspeakers generate sound that a listener will interpret as late or detrimental" [O'Keefe94, p. 72]. As loudspeakers are added, the critical distance drops. With 16 loudspeakers, their location and spacing became insignificant for listeners more than 2 or 3 m from the nearest speaker, leading O'Keefe to conclude that no matter where one stood in the room, there must be a loudspeaker within 3 m. A grim conclusion to say the least.

This begs the question, why do concert sound systems rely on massive arrays of speakers? The conventional wisdom is that the best way to battle the critical distance problem is by increasing the intensity of the direct sound.[1] A large semi-circular array of loudspeakers is believed to deliver a higher ratio of direct to indirect sound to all portions of the audience, though this is theoretically a dubious claim.

In large reverberant spaces where most listening locations are subjected to a poor direct/indirect ratio, a question to consider is what do you equalize: the direct, indirect, or total sound? Meyer claims that "it is known that the ear generally perceives early reflections as the 'frequency response' of the space" [Meyer, p. 3]. Meyer's technique is to use correlation with the excitation source to reject reverberation in his measurements. Truncation of the impulse response also provides a means of considering just the direct sound [Genereux90]. But does it make psychoacoustic sense to reject the predominant sound field?

Cabot [Cabot88] has noted that the pink noise RTA technique results in a measurement of the steady-state room response, or the integral of the direct and indirect sound. Consistent with [Staffeldt82], Cabot finds that "... a flat sounding system will usually not be flat in the direct field response or in the steady state response," which leads him to conclude that "it is therefore as incorrect to equalize the steady state response as it is to equalize the direct sound" [Cabot88, p. 392]. Missing from his analysis, unfortunately, is an explanation due to the high-frequency directional sensitivity of the ear.

Toole and Olive found that a loudspeaker is judged favourably when on- and off-axis response are both flat, whereby the direct and indirect sound match [Toole88]. While they did not make the connection, the Haas effect [Haas72] is likely responsible. In the case of matching early reflections, the Haas effect holds and the direct sound is reinforced [Rodgers81]. If, however, the frequency response of the early reflections differs significantly from that of the direct sound, the Haas effect is defeated and the early reflections will be perceived as annoyingly audible, helping to further degrade the direct/indirect ratio.

This suggests that one might consider equalizing the indirect sound such that it matches the direct sound. (Unfortunately, this is complicated by the fact that you can't change one without affecting the other.) Sliding the FFT time window to the latter portion of the impulse response permits the measurement of the frequency response of the indirect sound field. Alternatively, assuming a flat free field response (a reasonable assumption for most loudspeakers) one could improve the likelihood of matching direct and indirect frequency response through the addition of off-axis full-range loudspeakers. This is the concept behind the design of Bose loudspeakers, and the by-product of any semi-circular array configuration.

4.3 Single vs. Multiple Measurement Points

As noted above, the transfer function of a room differs from location to location. Plomp and Steeneken attempted to cod-

1. This is only part of the answer. A significant factor is that the artist thinks it looks cool. Another reason is that it has to sound like a rock concert regardless of where you are sitting in the audience. For large, highly reverberant spaces, there is only one way to ensure that: **volume**.

ify the amount of fluctuation one can expect in a reverberant space [Plomp73]. In their paper, they show that location dependence is caused by the variability in the amplitudes and phases of the individual harmonics of a complex tone in a diffuse field. The variabilities in amplitude (SPL) as a function of location are derived theoretically to have a standard deviation of 5.57 dB for pure tones in a diffuse field; phase differences are random (0 to 2π) and found to be negligible for complex tones with fundamental frequencies above 100 Hz [Plomp73]. Since timbre is correlated to the relative amplitude of the harmonics of a complex tone, timbre differs from location to location as a function of this variance in SPL. Their empirical study supports this theoretical variation.

Worse, the problem of location dependent frequency response variation can actually be exacerbated by equalization. Elliott and Nelson have empirically shown that optimizing for a single location within a small room is detrimental to all other points within the room [Elliott89]. This phenomenon may or may not generalize to the case of large enclosures, but the anecdotal evidence suggests that it does.The burden of averaging multiple measurement points seems to be the answer. For touring systems, however, there is generally not enough time nor the proper equipment to undertake multiple simultaneous measurements. Moreover, the complexity of adjusting multiple equalizers while simultaneously considering multiple room response curves would quickly result in cognitive overload for the audio engineer. It is for these reasons that the best seat is usually the one next to the front-of-house mixing console.

4.4 Audience Presence

Ideally, the audience should be present before a room is equalized. The audience has two significant effects on a room. First, the audience increases the absorption characteristics and thus affects the reverberation time and the frequency response of the room. Second, the audience increases the temperature of the air, creating temperature gradients that change room mode interactions and thus affect the resulting frequency response.

This is not to suggest that equalization of an empty hall is pointless. The presence or absence of an audience does not greatly affect resonances and echoes that are a function of the dimensions of the hall. Some have even suggested that the audience is not truly significant [Beranek62]. We will only add that equalization without an audience present is better than no equalization at all.

4.5 Time Domain Equalization

Time domain equalization is equalization specified by the impulse response of an arbitrary filter, providing user control over both the amplitude and phase response. Traditional equalizers affect both the amplitude and phase response, but provide user control over only the amplitude response. The digital delay is a degenerate case of time domain equalization consisting of a delta function capable of being offset in time.

The most obvious shortcoming of frequency (amplitude) equalization is that it is not a complete substitute for time domain equalization. "Equalization in the frequency domain effectively only equalizes the minimum phase part of the response due to the presence of all-pass phase components in the very complex room response" [Fortier94, p. 60]. A thorough and theoretical treatment of this phenomenon is Neely and Allen's seminal paper on the invertibility of room impulse responses [Neely79]. [Elliott89] is one of a class of papers dealing with the use of time domain room equalization in the form of adaptive digital filters. An adaptive digital filter is a digital filter that changes in response to an error signal, either continuously, or is "programmed" once from measured data. A series of automatic equalization schemes based on adaptive digital filters have appeared in the literature supporting optimization at a single point with a single filter [Genereux90], multiple points with a single filter [Elliott89], and multiple points with multiple filters [Munshi92]. Equalizing multiple loudspeakers with different filters is the only approach capable of completely inverting the impulse response of a room [Munshi92].

In multi-loudspeaker systems, nulls will occur in the polar pattern as a result of waveform interference. While dips in the amplitude response are correctable in the frequency domain, nulls are correctable only in the time domain [Reams94]. As an example, Davis and Davis cite the comb filter effects produced by misaligned speakers, which cannot be detected with a 1/3-octave analyzer or corrected with a 1/3-octave equalizer [Davis87]. While the audibility of these nulls is questionable, these comb filters can be mistaken for the high-frequency notches used as localization cues [Rodgers81]. The problem is easily solved by aligning the loudspeaker wavefronts via the introduction of a delay.

However, the general consensus is that phase equalization is a distant second to amplitude equalization in importance (e.g., [Toole86a]). Equalization in the time domain improves phase effects that many listeners are insensitive to [Fortier94].

4.6 Measurement Resolution

Many have cited the critical bandwidth of the ear as support for the use of 1/3-octave analyzers and equalizers (e.g., [Schulein75]). If you consider the definition of critical bandwidth put forward by Davis and Davis, it seems particularly apt: the bandwidth within which the human ear cannot detect spectrum shape when listening to complex sounds [Davis87]. In other words, correcting for anomalies in the spectrum at a resolution finer than 1/3-octave is pointless. This is a reasonable first approximation, but is unfortunately incorrect. Critical bandwidths define regions of dissonance, within which it is still possible to detect spectrum shape.

Meyer suggests that high-resolution DFT analysis techniques are by far preferable to 1/3-octave [Meyer84]. Toole and Olive concur: "with any measurement it is clearly important that there be adequate frequency resolution to reveal the presence of high-Q resonances.... The popular 1/3-octave measurements are useful only to reveal gross features in the frequency domain" [Toole88, p. 140].

5. CONCLUSION

In this paper we have looked at the practical aspects and psychoacoustic considerations of room measurement and equalization. Hopefully some questions have been answered, but many ambiguities remain. If this survey is to function as a springboard for further research, one might consider the following questions as a guide.

1. What is the most appropriate method for measuring the frequency response of a sound reinforcement system? Some research has suggested that microphones embedded in dummy heads provide a more reliable measure of perceived frequency response, and that high resolution DFT analysis is preferred to a 1/3-octave RTA. But, how many measurements at how many locations? How should multiple measurements be averaged? Should the source material be pink noise or impulses? Pink noise techniques implicitly measure the steady-state total sound energy. Impulses permit the separation of direct and indirect sound. Which is best?

2. It seems that the choice of target transfer function is dependent on the measurement technique. Many researchers have attempted to explain away these inconsistencies as a function of directional reverberation in rooms and the directional sensitivity of both ears and microphones. Given an understanding of these interactions, is there an ideal transfer function for which equalization should strive? We have pointed out the necessity to eliminate peaks in the frequency response for two reasons: to increase overall gain without causing feedback, and to avoid tonal colouration. But how flat is too flat?

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[Truax78] Truax, Barry ed. *Handbook for Acoustic Ecology.* A. R. C. Publications, 1978.

Acoustic and elastic wave scattering using boundary elements, by J.J. do Rego Silva

This book deals with the numerical aspects of the Boundary Element Method (B.E.M.) in acoustic and elastic wave scattering problems. The book is mainly devoted to the theoretical and computational aspects of formulations involving hypersingular integrals. A particular attention is devoted to the hypersingular formulations used to solve the non-uniqueness problem. The material in the book is divided into five chapters.

The book starts off with a general discussion of the advantageous of the B.E.M. for solving radiation and scattering problems in acoustics and elastodynamics.

Chapter 2 presents a quick tour of the theory and numerical implementation of the boundary element method in acoustics and elastodynamics. The computational implementation using both triangular and quadrilateral continuous and discontinuous elements is presented. The presentation is brief. It is for readers who are familiar with the techniques. The main formulas and procedures are presented without thorough discussion. However, references are used generously. Several examples are presented to illustrate the accuracy of the implementation.

Chapter 3 discusses the hypersingular formulations in the boundary element method. The author demonstrates the use of such formulations for solving the non-uniqueness problem, scattering by thin bodies and finally Stokes flow in ducts. Special attention is devoted to the smoothness conditions for the existence of the corresponding hypersingular integrals. The core of this chapter is spent in detailing an algorithm for evaluating the Hadamard finite part together with its numerical implementation using triangular and quadrilateral discontinuous isoparametric elements.

Chapter 4 presents Panich's formulation for solving the non-uniqueness problems in exterior acoustic problems. The main contributions of this chapter are twofold. First, a detailed description of the implementation of this formulation using higher order isoparametric continuous elements is presented. Second, a convincing discussion on the smoothness requirement for the existence of the corresponding hypersingular integral is given. Comparison with Burton and Miller's approach is presented to demonstrate the superiority of Panich's formulation.

Finally, chapter 5 extends Panich's work in acoustics to solve the non-uniqueness problem in elastodynamics. Once more, both the smoothness requirements and the numerical implementation are discussed.

The style of the book resembles journal papers. Actually, the book seems like a collection of research papers. For each topic discussed the author presents, briefly, the integral formulation, the computational implementation with the theoretical justification, the integration scheme used and validation examples. The author takes a great deal of time justifying the material presented in each chapter as original. In my opinion, part of this claim is not always true.

A particular omission of this book is a discussion on the variational boundary element method (V.B.E.M) and how it solves the hypersingular problem elegantly. Furthermore, one major shortcoming of this book, as with the majority of books in this area, is that the examples chosen deal with classical academic problems. Typical industrial applications together with comparisons with either experiments or other techniques would have been welcomed.

As a whole the presentation of the material discussed is quite advanced. The book assumes the reader to be familiar with several practical and theoretical aspects of the B.E.M. applied to acoustics and elastodynamics. Furthermore, the style of the book leads to a number of repetitions between the different chapters.

This book is definitely not intended to be used for a course. Its style makes it difficult to read for students or researchers coming to the field for the first time. However, viewed as a whole, the book contains a valuable source of information about hypersingular boundary integral formulations in acoustics and elastodynamics. It will be appreciated by advanced readers; I certainly appreciated it.

Reviewed by: Noureddine Atalla, Université de Sherbrooke

[This book - ISBN 1-5625-2217-5 - is available from Computational Mechanics Inc. at a price of US$69]

Computational Acoustics and its Environmental Applications, C. A. Brebbia, ed.

There are too many conferences these days. It continues to amaze me the number of conferences it is possible to organize in such a small field as acoustics. The proceedings of the numerous conferences are useful snapshots of what is going on in the field. But the papers they contain certainly don't provide the detailed information that a journal article or book does. There is also an increase in the number of conferences organized by companies of acoustical products. In these cases one must always ask what the objective of the conference is, and to what extent the information contained in the proceedings is intended to diffuse scientific information or to sell a product or service. Another problem with conferences and proceedings is that the information presented may not be unique - it may have (probably has?) been reported elsewhere too.

This book is the proceedings of a conference on computational acoustics organized by Computational Mechanics. The publicized application is to 'environmental applications'; to my mind this is more using the trendy term 'environment' to make the conference attractive than providing information on what's inside.

What is inside is, in general, a very interesting collection of 34 papers on the application of state-of-the-art computational techniques to acoustical problems. The book's sections are: industrial noise and vibration; aero- and hydro-acoustics; ambient noise problems; underwater acoustics; building acoustics; noise in the marine environment; sound absorption (sic) materials; numerical and computational techniques; noise and vibration of soils and foundations.

Topics covered vary from analysis of thin-plate sound transmission using boundary-integral equations to wave propagation in acoustic horns through modal decomposition to outdoor sound propagation to method-of-image and ray-tracing prediction of room acoustics to transfer-matrix and finite-element prediction of composite materials.

The book is very nicely produced. The average paper length of 9 pages allows the papers to provide useful information on their topic.

Should you buy this book (conference proceedings). Yes, if you want a nice summary of modern computational methods in use today, and if you have the money to purchase the proceedings of every acoustical conference that takes place.

There is a lot of interesting reading here. Some of it you'll have seen elsewhere. But it's certainly worth a look.

Reviewed by: Murray Hodgson, University of British Columbia.

[This book - ISBN 1562522329 - is available from Computational Mechanics Inc. at the price of US $121]

Prix de l'ACA à la mémoire de Raymond Hétu

A la dernière réunion du Conseil d'administration tenue à Québec, j'ai accepté de former et de présider un comité qui vise à étudier la possibilité d'établir un prix de l'ACA à la mémoire de Raymond Hétu. Sharon Abel, Blaise Gosselin et Chantal Laroche font aussi partie de ce comité. Nous avons considéré plusieurs options pour ce prix - chacune présentant des avantages et des désavantages (particulièrement financiers). Le comité invite les membres de l'ACA à transmettre leurs commentaires à l'égard de ces options ou de toute autre option. Je vous serais reconnaissant de me transmettre vos commentaires avant le 1er avril. Le comité fera une synthèse des commentaires sous forme de proposition qui sera discutée lors de la prochaine rencontre des Directeurs en mai. Les options considérées par le comité sont les suivantes:

Option 1 - Établir un nouveau prix annuel pour les étudiants gradués - d'un montant d'environ 500$ - pour l'étudiant(e) canadien(ne) qui mènera le meilleur projet de recherche poursuivant les principaux intérêts de recherche de Raymond Hétu - à savoir, élucider les problèmes concrets vécus par les travailleurs en milieu industriel, et améliorer leur sort (autrement que par des mesures de contrôle du bruit puisque le prix Eckel pour le contrôle du bruit existe déjà). Un prix de 500$ demanderait un investissement de 5000-8000$. L'ACA n'a pas de surplus d'argent à engager en ce moment. Le nouveau prix pourrait alors être financé de plusieurs autres façons: augmenter les frais d'adhésion des membres, solliciter des dons auprès des membres, réduire le montant d'un ou de plusieurs prix déjà offerts (par exemple, les prix étudiants pour les trois meilleures présentations étudiantes ou les prix des Directeurs);

Option 2 - Renommer un des prix déjà offerts (par exemple, un ou tous les prix pour les présentations étudiantes ou les prix des Directeurs) à la mémoire de Raymond Hétu. Ceci n'aurait évidemment aucune conséquence financière. Les conditions pour renommer le prix pourraient être modifiées afin de prendre en considération les priorités de Raymond Hétu discutées plus haut.

Murray Hodgson
Courrier élec.: hodgson@mech.ubc.ca
Téléphone: (604) 822-3073
Télécopieur: (604) 822-9588

CALL FOR PAPERS
Acoustics Week in Canada 1996

SYMPOSIUM, October 9 - 11

This years CAA conference will deal with noise control programs within the Environment, Society and Industry. Presentations covering acoustics within theses areas are solicited. A number of special technical sessions on particular themes have already been created. The list of the special sessions is as follows:

Psychoacoustics	Physiological Acoustics
Automatic Speech Recognition	Speech Perception
Speech Production	Occupational Hearing Loss & Hearing Protection
Architectural Acoustics	Musical Acoustics
HVAC	Airport (transportation) Noise
Legislation/Environment Noise	Underwater Acoustics & Sound Propagation
Industrial Noise Control	Active Noise Control
Vibration Control	

Submitted abstracts will be incorporated into the program by assigning them to the existing sessions or creating new sessions when necessary.

To submit an abstract:

- Send an abstract of 250 words maximum to the technical program chair **before 1 May 1996.** This deadline will be strictly enforced. The abstract should be prepared in accordance with the instructions enclosed in this issue of **Canadian Acoustics.**

- A notification of acceptance will be sent to the authors **by 15 May 1996** with a registration form.

- A **one-page** summary paper, prepared in accordance with the enclosed instructions, will be sent to the technical program chairman **by 1 July 1996**. This deadline will be strictly enforced. The summary papers will be published in the proceedings issue of **Canadian Acoustics.**

Address the abstracts and summary papers to:

Dr. Elzbieta Slawinski
Psychology Dept.
University of Calgary
2500 University Drive NW
Calgary AB T2N 1N4
Tel. (403)220-5205, Fax. (403)282-8249
e-mail: eslawins@acs.ucalgary.ca

Registration fee: the registration fee for the Symposium and the completed registration form must be sent with the summary paper.

Summary of dates:

1 May 1996	Deadline for receipt of abstracts.
15 May 1996	Notification of acceptance.
1 July 1996	Deadline for receipt of summary paper, registration form and registration fee.
9 - 11 October 1996	Symposium.

Student competition: student participation to the Symposium is strongly encouraged. Monetary awards will be given to the three best presented papers. Students must signify their intention to compete by submitting the "**Annual Student Presentation Award**" form in this issue, to be enclosed with the abstract.

APPEL DE COMMUNICATIONS
Semaine canadienne d'acoustique 1996

SYMPOSIUM, 9 - 11 octobre

Cette année, la theme pour Semaine Canadienne d'Acoustique 1996 est Environnement, Societée, et l'Industrie. Des présentations sont sollicitées sur tous les domaines de l'acoustique et des vibrations. Un nombre de session techniques protant sur la thème sont déjà planifiées. En voici la liste:

Psycho-acoustique	Physio-acoustique
Audition	Parole
Audiologie	Contrôle du Bruit en Milieu de Travail
Acoustique Architecturale	Acoustique Musicale
HVAC	Contrôle du Bruit de l'Aeroport et des Aeroplanes
Reglements et Bruit Environmental	Acoustique Sous-marine
Contrôle du Bruit Industrial	Contrôle Actif du Bruit
Contrôle du Vibration	

Les présentations soumises seront réparties dans les sessions précédentes ou dans d'autres sessions si besoin est.

Pour soumettre une présentation:

- Envoyer un résumé de 250 mots maximum au responsable technique **avant le 1 mai 1996.** Cette échéance devra être scrupuleusement respectée. Les résumés devront être préparés en suivant les instructions incluses dans ce numéro d'**Acoustique canadienne.**

- Une notification d'acceptation du résumé sera encoyée aux auteurs **avant le 15 mai 1996** avec un formulaire d'inscription au Symposium.

- Un sommaire de une-page, préparé suivant les instructions incluses dans ce numéro d'**Acoustique canadienne,** devra être envoyé au responsable technique **avant le 1 juillet 1996**. Cette échéance devra être scrupuleusement respectée. Les sommaires seront publiés dans les actes du Symposium.

Veuillez faire parvenir les résumés et les sommaires à:

Dr. Elzbieta Slawinski
Psychology Dept.
University of Calgary
2500 University Drive NW
Calgary AB T2N 1N4
Tel. (403)220-5205, Fax. (403)282-8249
e-mail: eslawins@acs.ucalgary.ca

Frais d'inscription: les frais d'inscription au Symposium et le formulaire d'inscription dûment complété devront être expédiés avec le sommaire.

Résumé des dates importantes:

1 mai 1996	Date limite de réception des résumés.
15 mai 1996	Notification d'acceptation.
1 juillet 1996	Date limite de réception du sommaire, du formulaire d'inscription et des frais d'inscription.
9 - 11 octobre 1996	Symposium.

Concours étudiants: la participation des étudiants au Symposium est fortement encouragée. Des prix en argent seront décernés pour les trois meilleures communications. Les étudiants doivent indiquer leur intention de participer en complétant le formaire " **Prix annuels relatifs aux communications étudiantes** " qui figure dans le présent numéro et en le joignant au résumé.

Instructions for the Preparation of Abstracts

1) Duplicate copies of an abstract are required for each meeting paper; one copy should be an original. Send the copies to the Technical Program Chairperson, in time to be received by the deadline. Either English or French may be used. A cover letter is not necessary. **2)** Limit the abstract to 300 words, including title and first author's name and address; names and addresses of coauthors are not counted. Display formulas set apart from the text are counted as 40 words. Do not use the forms "I" and "we"; use passive voice instead. **3)** Title of abstract and names and addresses of authors should be set apart from the abstract. Text of abstract should be one single, indented paragraph. The entire abstract should be typed double spaced on one side of 8 1/2 x 11 in. or A4 paper. **4)** Be sure that the mailing address of the author to receive the acceptance notice is complete on the abstract, to insure timely deliveries. **5)** Do not use footnotes. Use square brackets to cite references or acknowledgements. **6)** Underline nothing except what you wish to be italicized. **7)** If the letter l is used as a symbol in a formula, loop the letter l by hand and write "lc ell" in the margin of the abstract. Do not intersperse the capital letter O with numbers where it might be confused with zero, but if unavoidable, write "capital oh" in the margin. Identify phonetic symbols by appropriate marginal remarks. **8)** At the bottom of an abstract give the following information: **a)** If the paper is part of a special session, indicate the session; **b)** Name the area of acoustics most appropriate to the subject matter; **c)** Telephone and fax numbers, including area code, of the author to be contacted for information. Non-Canadian Authors should include country; **d)** If more than one author, name the one to receive the acceptance notice; **e)** Overhead projectors and 35mm slide projectors will be available at all sessions. Describe on the abstract itself any special equipment needed.

Instructions pour la Préparation des Résumés de Conférences

1) Deux copies du résumé sont requises pour chaque papier soumis; une des copies doit être un original. Envoyer les copies au Président du Comité technique, suffisamment à l'avance pour qu'elles soient reçues avant la date de tombée. L'anglais ou le français peut être utilisé. Une lettre de présentation n'est pas requise. **2)** Limiter le résumé à 300 mots, incluant le titre, le nom et l'adresse du premier auteur; les noms et les adresses des co-auteurs ne sont pas comptabilisés. Les formules en retrait du texte comptent pour 40 mots. Ne pas utiliser la forme "je" ou "nous"; utiliser plutôt la forme passive. **3)** Le titre du résumé, les noms et les adresses des auteurs doivent être séparés du texte. Le texte du résumé doit être présenté en un seul paragraphe. Le résumé entier doit être dactylographié à double interlignes sur une face d'une page 8 1/2 x 11 pouce ou du papier A4. **4)** S'assurer que l'adresse postale complète de l'auteur qui doit recevoir l'avis d'acceptation est inscrite sur le résumé afin d'assurer une livraison rapide. **5)** Ne pas utiliser les notes de bas de page. Utiliser les crochets pour les références et les remerciements. **6)** Ne souligner que ce qui doit être en italique. **7)** Si la lettre l est utilisée comme symbole dans une formule, encercler la lettre l à la main et écrire "lc ell" dans la marge du résumé. Ne pas introduire la lettre majuscule O dans les chiffres lorsqu'elle peut être confondue avec zéro, mais se cela n'est pas possible, écrire "O majuscule" dans la marge. Identifier les symboles phonétiques à l'aide de remarques appropriées dans la marge. **8)** A la fin du résumé, fournir les informations suivantes: **a)** Si la communication fait partie d'une session spéciale, indiquer laquelle; **b)** Identifier le domaine de l'acoustique le plus approprié à votre sujet; **c)** Les numéros de téléphone et de télécopieur, incluant le code régional, de l'auteur avec qui l'on doit communiquer pour information. Les auteurs étrangers doivent indiquer leur pays; **d)** S'il y a plus d'un auteur, mentionner le nom de celui qui doit recevoir l'avis d'acceptation; **e)** Des projecteurs à acétates et à diapositives seront disponibles dans chaque session. Indiquer les besoins spéciaux, si nécessaire.

Instructions pour la Préparation des Articles à être Publiés dans le Cahier des Actes du Congrès

Général - Soumettre un article prêt-à-copier d'un maximum de deux pages présenté en deux colonnes. Ne pas inclure de sommaire. Tout le texte en caractères Times-Roman. Disposer les figures dans le haut ou le bas des pages si possible. Lister les références dans un format logique à la fin du texte. Envoyer l'article au président du Programme Technique avant la date de tombée . Le format optimal peut être obtenu de deux façons:

Méthode directe - Imprimer directement sur deux feuilles 8.5" x 11" en respectant des marge de 3/4" dans le haut et sur les côtés et un minimum de 1" dans le bas. Titre en 12pt, caractères gras, en simple interligne (12pt), centrés sur la page. Le reste du texte en 9pt en 0.75 (9pt) interligne, dans un format en deux colones, avec une largeur de colonnes de 3.4" et une séparation de 1/4". Noms des auteurs et adresses centrés sur la page avec les noms en caractères gras. Les titres de sections en caractères gras.

Méthode indirecte - Dactylographier ou imprimer comme suit, réduire au trois-quart (s.v.p., s'assurer de bonnes photocopies) et assembler l'article sur un maximum de deux pages 8.5" x 11" avec les côtés et un minimum de 1" dans le bas. Titre en 16pt avec 1.33 (16pt) interligne, centré sur la page. Le reste du texte en 12pt avec simple (12pt) interligne. Noms et adresses des auteurs centrés sur la page avec les noms en caractères gras. Titres des sections en caractères gras. Imprimer les colonnes de texte sur quatre feuilles 8.5" x 14" avec une largeur de colonnes de 4.5", une longueur maximum de 12.25", en laissant de la place pour le titre, les noms et les adresses sur la première page.

Instructions for Preparation of Articles to be Published in the Conference Proceedings Issue

General - Submit the camera-ready article on a maximum of two pages in two-column format. Do not include an abstract. All text in Times-Roman font. Place figures at the top and/or bottom of the pages, if possible. List references in any consistent format at the end. Send to the Chairperson of the Technical Programme by the deadline. The optimum format can be obtained in two ways:

Direct method - Print directly on two sheets of 8.5" x 11" paper with margins of 3/4" top and sides, and 1" minimum at the bottom. Title in 12pt bold with single (12pt) spacing, centred on the page. All other text in 9pt with 0.75 (9pt) line spacing, in two-column format, with column width of 3.4" and separation of 1/4". Authors' names and addresses centred on the page with the names in bold type. Section headings in bold type.

Indirect method - Type or print as follows, reduce to three-quarters size (please ensure good copies) and assemble article on a maximum of two 8.5" x 11" pages with margins of 3/4" top and sides, and 1" minimum at the bottom. Title in 16pt bold type with 1.33 (16pt) line spacing, centred on the page. All other text in 12pt with single (12pt) line spacing. Authors' names and addresses centred on the page with the names in bold type. Section headings in bold type. Print individual text columns on four sheets of 8.5" x 14" paper with a column width of 4.5", a maximum length of 12.25", and leaving room for the title and names and addresses on the first page.

ANNUAL STUDENT PRESENTATION AWARDS

The Canadian Acoustical Association makes awards to students whose papers are presented at the CAA Annual Symposium. Students contemplating presenting papers at the Symposium should apply for these awards with the submission of their abstract.

RULES

1. These awards are presented annually to authors of outstanding student papers that are presented during the technical sessions at Acoustics Week in Canada.
2. In total, three awards of $500.00 are presented.
3. Presentations are judged on the following merits:
   - i) The way the subject is presented;
   - ii) The explanation of the relevance of the subject;
   - iii) The explanation of the methodology/theory;
   - iv) The presentation and analysis of results;
   - v) The consistency of the conclusions with theory and results.
4. Each presentation is judged independently by at least three judges.
5. The applicant must be:
   - i) a full-time graduate student at the time of application;
   - ii) the first author of the paper;
   - iii) a member of the CAA;
   - iv) registered at the meeting.
6. To apply for the award, the student must send this application simultaneously with the abstract. Multiple authors are permitted, but only the first author may receive an award.

PRIX ANNUELS RELATIFS AUX COMMUNICATIONS ETUDIANTES

L'Association Canadienne d'Acoustique décerne des prix aux étudiant(e)s qui présenteront une communication au congrès annuel de l'ACA. Les étudiant(e)s qui considèrent présenter un papier doivent s'inscrire à ce concours au moment où ils (elles) soummettent leur résumé.

REGLEMENTS

1. Ces prix sont décernés annuellement aux auteurs de communications exceptionelles presentées par des étudiants lors des sessions techniques de la Semaine Canadienne d'Acoustique.
2. Au total, trois prix de 500$ sont remis.
3. Les présentations sont jugées selon les critères suivants:
   - i) La façon dont le sujet est présenté;
   - ii) Les explications relatives à l'importance du sujet;
   - iii) L'explication de la méthodologie;
   - iv) La présentation et l'analyse des résultats;
   - v) La consistence des conclusions avec la théorie et les résultats.
4. Chaque présentation est evaluée séparément par au moins trois juges.
5. Le candidat doit être:
   - i) un étudiant à temps plein de niveau gradué au moment de l'inscription;
   - ii) le premier auteur du papier;
   - iii) un membre de l'ACA;
   - iv) un participant au congrès.
6. Afin de s'inscrire au concours, l'étudiant doit envoyer ce formulaire d'inscription en même temps que son résumé. Plusieurs auteurs sont permis, mais seul le premier auteur peut recevoir le prix.

APPLICATION FOR STUDENT PRESENTATION AWARD AT ACOUSTICS WEEK IN CANADA

FORMULAIRE D'INSCRIPTION POUR LES PRIX DECERNES AUX ETUDIANTS LORS DE LA SEMAINE CANADIENNE D'ACOUSTIQUE

NAME OF THE STUDENT:____________________NOM DE L'ETUDIANT

SOCIAL INSURANCE NUMBER:____________________NUMERO D'ASSURANCE SOCIALE

TITLE OF PAPER:____________________TITRE DU PAPIER

UNIVERSITY/COLLEGE:____________________UNIVERSITE/COLLEGE

NAME, TITLE OF SUPERVISOR:____________________NOM ET TITRE DU SUPERVISEUR

STATEMENT BY THE SUPERVISOR: The undersigned affirms that the above-named student is a full-time student and the paper to be presented is the student's original work.

Signature:____________________

DECLARATION DU SUPERVISEUR: Le sous-signé affirme que l'étudiant(e) mentionné(e) ci-haut est inscrit(e) à temps plein et que la communication qu'il (elle) présentera est le fruit de son propre travail.

Date:____________________

APPLICATION FOR STUDENT TRAVEL SUBSIDY TO ACOUSTICS WEEK IN CANADA

Travel subsidies are available to students presenting papers at Acoustics Week in Canada if they live at least 150 km from the conference venue, if the subsidy is needed, if supporting receipts are submitted, and if they publish a summary of their paper in the proceedings issue of *Canadian Acoustics*.

I wish to apply for a CAA Travel Subsidy: _____yes _____no.

STATEMENT BY THE SUPERVISOR: The undersigned affirms that the CAA Travel Subsidy, combined with other travel funds that the above-named student may receive to attend the meeting will not exceed his/her travel costs.

Signature:____________________

FORMULAIRE DE DEMANDE DE REMBOURSEMENT POUR FRAIS DE DEPLACEMENT A LA SEMAINE CANADIENNE D'ACOUSTIQUE

Un remboursement de frais de déplacement est offert aux étudiants qui présentent une communication lors de la Semaine Canadienne d'Acoustique, s'ils demeurent à plus de 150 km du site du congrès, si le remboursement est nécessaire, si les reçus à l'appui sont soumis et s'ils publient un résumé dans les Actes du Congrès.

Je désire demander un remboursement: _______oui _______non.

DECLARATION DU SUPERVISEUR: Le sous-signé affirme que le remboursement, jumelé à d'autres fonds que l'étudiant(e) ci-haut mentionné(e) peut recevoir ne dépasseront pas ses coûts réels de voyage.

Date:____________________

NEWS

CONFERENCES

The following list of conferences was mainly provided by the Acoustical Society of America:

39th meeting of the Acoustic Emission Working Group (AEWG): 25-28 March 1996, California. Contact: Harold L. Dunegan, Program Chair, Dunegan Engineering Consultants, Inc., P. O. Box 1749, San Juan Capistrano, CA 92963, Tel.: 714-661-8105.

Spring Meeting of the Acoustical Society of Japan: 26-28 March 1996, Tokyo, Japan. Contact: Acoustical Society of Japan, Ikeda Building, 2-7-7 Yoyuogi, Shibuya-ku, Tokyo 151 Japan; Fax: +81 33379 1456.

FORUM ACUSTICUM (European Congress on Acoustics): 1-4 April 1996, Antwerp, Belgium. Contact: A. Dancers, ISL, P.O. Box 34, 68301 Saint Louis, Cedex, France.

Innovations in Noise Control for the Energy Industry: 14-16 April 1996, Banff, Alberta. Contact: David DeGagne, Alberta Energy and Utilities Board, 640 - 5 Avenue SW, Calgary AB T2P 3G4. Tel: (403) 297-8311; Fax: (403) 297-7336.

Catgut Acoustical Society, Inc. and the Michigan Violinmakers Association Joint Conference on Stringed Instruments: Violins--Guitars: 26-28 April 1996, Ann Arbor, MI. Contact: Catgut Acoustical Society, 112 Essex Avenue, Montclair, NJ 07042, Tel.: 201-774-4029; FAX: 201-744-9197.

2nd AIAA/CEAS Aeroacoustics Conference (17th AIAA Aeroacoustics Conference): 6-8 May 1996, State College PA. Contact: AIAA Customer Service Center, Tel.: 202-646-7400; via World Wide Web: http://cac.psu.edu/~lnl/aiaa96.html.

131st Meeting Acoustical Society of America: 13-17 May 1996, Indianapolis, IN. Contact: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; FAX: 516-576-2377; E-mail: asa@aip.org, WWW: http://asa.aip.org.

4th Speech Production Seminar: 21-24 May 1996, Autrans, France. Contact: ICP-INPG, 46 avenue Felix Viallet, 38031 Grenoble, cedex 01, France; Fax: +33 76 57 47 10; E-mail: etrwspm@icp.grenet.fr.

International Symposium on Acoustic Remote Sensing of the Atmosphere and Oceans: 27-31 May 1996, Moscow, Russia. Contact: Secretariat ISARS'96, 3 Pyzevsky Line, Moscow, 109017 Russia; Fax: +7 095 233 1652; E-mail: postmaster@iaph.msk.su.

Noise and Planning '96: 28-31 May 1996, Pisa, Italy. Contact: G. Lombardi, via Bragadino 2, 20144 Milano, Italy; Fax: +39 2 480 18833.

XXIemes Journées d'Etude sur la Parole: 10-15 June 1996, Avignon, France. Contact: J. Gourret, Laboratoire d'Informatique, Faculte des Sciences, 33 rue Louis Pasteur, 84000 Avignon, France; Fax: +33 90 27 00 95; E-mail: jep96@univ-avignon.fr.

Nordic Acoustical Meeting: 12-14 June 1996, Helsinki, Finland. Contact: NAM, Helsinki University of Technology, Acoustics Laboratory, Otakaari 5A, 02150 Espoo, Finland; Fax: +358 460224; E-Mail: nam96@hut.fi.

24th Annual Meeting: 12-14 June 1996, Italian Acoustical Association, A.I.A., Trento, Italy. Contact: Mrs. A. Giacomazzi, Provincia Autonoma di Trento, via Mantova 16, 38100 Trento, Italy; Fax: +39 461-236574.

CONFÉRENCES

La liste de conférences ci-jointe a été offerte en majeure partie par l'Acoustical Society of America:

39e rencontre du Groupe de travail sur les émissions acoustiques: 25-28 mars 1996, Californie. Renseignements: Harold L. Dunegan, Program Chair, Dunegan Engineering Consultants, Inc., P. O. Box 1749, San Juan Capistrano, CA 92963, Tel.: 714-661-8105.

Rencontre de printemps de la Société acoustique du Japon: 26-28 mars 1996, Tokyo, Japon. Renseignements: Acoustical Society of Japan, Ikeda Building, 2-7-7 Yoyuogi, Shibuya-ku, Tokyo 151 Japan; Fax: +81 33379 1456.

FORUM ACUSTICUM (Congrès européen sur l'acoustique): 1-4 avril 1996, Antwerp, Belgique. Renseignements: A. Dancers, ISL, P.O. Box 34, 68301 Saint Louis, Cedex, France.

Innovations dans le contrôle du bruit pour l'industrie de l'énergie: 14-16 avril 1996, Banff, Alberta. Renseignements: David DeGagne, Alberta Energy and Utilities Board, 640 - 5 Avenue SW, Calgary AB T2P 3G4. Tel: (403) 297-8311; Fax: (403) 297-7336.

Conférence conjointe de la Société d'acoustique Catgut Inc. et de l'Association des fabricants de violons du Michigan sur les instruments à corde: violons et guitares: 26-28 avril 1996, Ann Arbor, Michigan. Renseignements: Catgut Acoustical Society, 112 Essex Avenue, Montclair, NJ 07042, Tel.: 201-774-4029; FAX: 201-744-9197.

2e conférence d'aéro-acoustique de l'AIAA/CEAS (17e conférence d'aéro-acoustique de l'AIAA: 6-8 mai 1996, State College, Pennsylvania. Renseignements: AIAA Customer Service Center, Tel.: 202-646-7400; via World Wide Web: http://cac.psu.edu/~lnl/aiaa96.html.

131e rencontre de l'Acoustical Society of America: 13-17 mai 1996, Indianapolis, Indiana. Renseignements: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; FAX: 516-576-2377; E-mail: asa@aip.org, WWW: http://asa.aip.org.

4e séminaire sur la synthèse de la parole: 21-24 mai 1996, Autrans, France. Renseignements: ICP-INPG, 46 avenue Félix Viallet, 38031 Grenoble, cedex 01, France; Fax: +33 76 57 47 10; E-mail: etrwspm@icp.grenet.fr.

Symposium international sur le télésondage de l'atmosphère et des océans: 27-31 mai 1996, Moscou, Russie. Renseignements: Secretariat ISARS'96, 3 Pyzevsky Line, Moscow, 109017 Russia; Fax: +7 095 233 1652; E-mail: postmaster@iaph.msk.su.

Bruit et planification '96: 28-31 mai 1996, Pise, Italie. Renseignements: G. Lombardi, via Bragadino 2, 20144 Milano, Italy; Fax: +39 2 480 18833.

XXIèmes Journées d'Etude sur la Parole: 10-15 juin 1996, Avignon, France. Renseignements: J. Gourret, Laboratoire d'Informatique, Faculté des Sciences, 33 rue Louis Pasteur, 84000 Avignon, France; Fax: +33 90 27 00 95; E-mail: jep96@univ-avignon.fr.

Nordic Acoustical Meeting: 12-14 juin 1996, Helsinki, Finlande. Renseignements: NAM, Helsinki University of Technology, Acoustics Laboratory, Otakaari 5A, 02150 Espoo, Finland; Fax: +358 460224; E-Mail: nam96@hut.fi.

13th International Congress on Audiology: 16-20 June 1996, Bari. Contact: Audiology & Otology Center, University of Bari, 70124 Bari, Italy; Fax: +39 80 5562171.

14th International Symposium on Nonlinear Acoustics (ISNA): 17-21 June 1996, Nanjing. Contact: Ronjue, Wei, Nanjng University, Institute of Acoustics, Nanjng 210008, China; Fax: +86 25 330 2728.

International Symposium on Cardiovascular Imaging: 24-26 June 1996, Leiden. Contact: Heymeriks & Van Ginneken, P.O. Box 4334, 3006AH Rotterdam, The Netherlands; Fax: +31 10 414 7988.

3rd European Conference on Underwater Acoustics: 24-28 June 1996, Heraklion. Contact: J. S. Papadakis, Foundation for Research and Technology, P.O. Box 1527, Heraklion 711 10, Crete, Greece; Fax: +30 81 238868.

4th International Congress on Sound and Vibration: 24-28 June 1996, St. Petersburg. Contact: M. J. Crocker, Mechanical Engineering Department, Auburn University, Auburn, AL 36849; Fax: +1-334-844-3306.

Fifth Meeting European Society of Sonochemistry: 7-11 July 1996, Cambridge, U.K. Contact: T.J. Mason, School of NES, Conventry University, Priory St., Coventry CV1 5FB, U.K., Fax: +44 1203 838282.

ESCA Workshop on Auditory Basis of Speech Perception: 15-19 July 1996, Keele, U.K. Contact: ESCA Workshop, Department of Communication and Neuroscience, Keele University, Keele, Staffordshire, ST5 5BG, U.K.; Fax: +44 1 782 583055; E-mail: cob03@keele.ac.uk.

Arrays and Beam Forming in SONAR: 25-28 July 1996, Bristol, U.K. Contact: Institute of Acoustics, 5 Holywell Hill, St Albans, Herts AL1 1EU, UK; Fax: +44 1 727 850553; E-mail: acoustics@clus1.ulcc.ac.uk.

Inter-Noise '96: 30 July - 2 August 1996, Liverpool. Contact: Institute of Acoustics, P.O. Box 320, St. Albans AL1 1PZ U.K.

19th International Congress of Theoretical and Applied Mechanics: 25-31 August 1996, Kyoto, Japan. Contact: Eiichi Watanabe, Civil Engineering Department, Kyoto University, Sakyo-ku, Kyoto 606-01, Japan; E-mail: ictam@strsum1.kuciv.kyoto-u.ac.jp; Fax: +81 75 752 5296.

Noise and Vibration Engineering Conference: 18-20 September 1996, Leuven, Belgium. Contact: L. Notre, K. U. Leuven PMA, Celestiknenlaan 300B, 3001 Heverlee, Belgium; Fax: +32 16 32 29 87; E-mail: lieve.notre@mech.kuleuven.ac.be.

XLIII Seminar on Acoustics: 16-21 September 1996, Ustron-Beskidy Mts., Poland. Contact: Institute of Acoustics, Silesian Technical University, Krzywoustego 2, 44-100 Gliwice, Poland.

FASE Symposium "Transport Noise": 23-25 September 1996, St. Petersburg, Russia. Contact: J. Thoen, FASE Secretariat, K. U. Leuven~ATF, Celestiknenlaan 200D, 3001 Leuven, Belgium; Fax: +32 16 32 79 84; E-mail: jan.thoen@fys.kuleuven.ac.be.

33rd Conference on Acoustics "Building and Architectural Acoustics": 23-25 September 1996, Prague, Czech Republic. Contact: CsAS Technicka 2, 166 27 Praha 6, Czech Republic; Fax: +42 2 311 1786.

Noise-Con 96: 29 September - 2 October 1996, Bellevue, WA. Contact: Noise-Con 96 Conference Secretariat, Engineering Professional Programs, 3201 Fremont Avenue North, XD-51, Seattle, WA 98103, Tel.: 206-543-5539; FAX: 206-543-2352; E-mail: uw-ept@engr.washington.edu.

Centennial meeting of the American Academy of Otolaryngology--Head and Neck Surgery: 29 September - 3 October 1996, Washington, DC. Contact: American Academy of Otolaryngology--Head and Neck Surgery, One Prince St., Alexandria, VA 22314. Tel.: 703-836-4444; FAX: 703-683-5100.

24e rencontre annuelle de l'Association italienne d'acoustique, AIA: 12-14 juin 1996, Trento, Italie. Renseignements: Mrs. A. Giacomazzi, Provincia Autonoma di Trento, via Mantova 16, 38100 Trento, Italy; Fax: +39 461-236574.

13e congrès international d'audiologie: 16-20 juin 1996, Bari, Italie. Renseignements: Audiology & Otology Center, University of Bari, 70124 Bari, Italy; Fax: +39 80 5562171.

14e Symposium international sur l'acoustique non-linéaire (ISNA): 17-21 juin 1996, Nanjing, Chine. Renseignements: Ronjue, Wei, Nanjng University, Institute of Acoustics, Nanjing 210008, China; Fax: +86 25 330 2728.

Symposium international d'imagerie cardio-vasculaire: 24-26 juin 1996. Leiden. Renseignements: Heymeriks & Van Ginneken, P.O. Box 4334, 3006AH Rotterdam, The Netherlands; Fax: +31 10 414 7988.

3e conférence européenne d'acoustique sous-marine: 24-28 juin 1996, Heraklion, Grèce. Renseignements: J. S. Papadakis, Foundation for Research and Technology, P.O. Box 1527, Heraklion 711 10, Crete, Greece; Fax: +30 81 238868.

4e congrès international de Sons et Vibrations: 24-28 juin 1996, St. Petersburg. Renseignements: M. J. Crocker, Mechanical Engineering Department, Auburn University, Auburn, AL 36849; Fax: +1-334-844-3306.

5e rencontre de la Société européenne de sonochimie: 7-11 juillet 1996, Cambridge, Royaume-Uni. Renseignements: T.J. Mason, School of NES, Conventry University, Priory St., Coventry CV1 5FB, U.K., Fax: +44 1203 838282.

ESCA Workshop sur les bases auditives de la perception de la parole: 15-19 juillet 1996, Keele, Royaume-Uni. Renseignements: ESCA Workshop, Department of Communication and Neuroscience, Keele University, Keele, Staffordshire, ST5 5BG, U.K.; Fax: +44 1 782 583055; E-mail: cob03@keele.ac.uk.

Réseaux d'antennes et formage du faisceau en SONAR: 25-28 juillet 1996, Bristol, Royaume-Uni. Renseignements: Institute of Acoustics, 5 Holywell Hill, St Albans, Herts AL1 1EU, UK; Fax: +44 1 727 850553; E-mail: acoustics@clus1.ulcc.ac.uk.

Inter-Noise '96: 30 juillet - 2 août 1996, Liverpool, Royaume-Uni. Renseignements: Institute of Acoustics, P.O. Box 320, St. Albans AL1 1PZ U.K.

19e congrès international de mécanique théorique et appliquée: 25-31 août 1996, Kyoto, Japon. Renseignements: Eiichi Watanabe, Civil Engineering Department, Kyoto University, Sakyo-ku, Kyoto 606-01, Japan; E-mail: ictam@strsum1.kuciv.kyoto-u.ac.jp; Fax: +81 75 752 5296.

Conférence de l'ingénierie du bruit et des vibrations: 18-20 septembre 1996, Leuven, Belgique. Renseignements: L. Notre, K. U. Leuven PMA, Celestiknenlaan 300B, 3001 Heverlee, Belgium; Fax: +32 16 32 29 87; E-mail: lieve.notre@mech.kuleuven.ac.be.

43e séminaire d'acoustique: 16-21 septembre 1996, Ustron-Beskidy Mts., Pologne. Renseignements: Institute of Acoustics, Silesian Technical University, Krzywoustego 2, 44-100 Gliwice, Poland.

Symposium du FASE sur le bruit des transports: 23-25 septembre 1996, St. Petersburg, Russie. Renseignements: J. Thoen, FASE Secretariat, K. U. Leuven~ATF, Celestiknenlaan 200D, 3001 Leuven, Belgium; Fax: +32 16 32 79 84; E-mail: jan.thoen@fys.kuleuven.ac.be.

Fourth International Conference on Spoken Language Processing: 3-6 October 1996, Philadelphia, PA. Contact: ICSLP 96, Applied Science & Engineering Laboratories, A.I. duPont Institute, P. O. Box 269, Wilmington, DE 19899, Tel.: 302-651-6830; TDD: 302-651-6834; FAX: 302-651-6895; E-mail: ISCLP96@asel.udel.edu; WWW: http://www.asel.udel.edu/speech/icslp/html.

Autumn Conference--Speech and Hearing: 24-27 October 1996, Windmere, U.K. Contact: Institute of Acoustics, P.O. Box 320, St. Albans, AL1 1PZ, U.K.

Acoustics Week in Canada 1996: 7-11 October 1996, Calgary, Canada. Contact: Dr. E. Slawinski, Department of Psychology, University of Calgary, 2500 University Drive NW, Calgary, AB, T2N 1N4. Tel.: 403-220-5205; FAX: 403-282-8249; E-mail: eslawins@acs.ucalgary.ca.

Third Joint Meeting of the Acoustical Society of America and the Acoustical Society of Japan: 2-6 December 1996, Honolulu, HI. Contact: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; FAX: 516-576-2377; E-mail: asa@aip.org, WWW: http://asa.aip.org.

14th World Conference on Non-Destructive Testing: 8-13 December 1996, New Delhi. Contact: B. Jaj, Metallurgy and Materials Group, Indira Gandhi Centre for Atomic Research, Kalpakkam 603102, India; E-mail: dmg@igcar.iitm.emet.in.

International Symposium on Simulation, Visualization and Auralization for Acoustic Research and Education: 2-4 April 1997, Tokyo, Japan. Contact: M. Morimoto, Faculty of Engineering, Kobe University, Rokko, Nada, Kobe 657, Japan; Fax: +81 78 881 2508.

International Conference on Acoustics, Speech, and Signal Processing ICASSP 97: 21-24 April 1997, Munich, Germany. Contact: H. Fastl, Lehrstuhl fur Mensch-Maschine-Kommunikation, Technische Universitat Munchen, 80290 Mnchen, Germany; Fax: +49 89 2105 8535; E-mail: fas@mmk.e-tchnik.tu.muenchen.de.

133rd Meeting of the Acoustical Society of America: 16-20 June 1997, State College, PA. Contact: Acoustical Society of America, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; WWW: http//asa.aip.org.

Internoise 97: 25-27 August 1997, Budapest, Hungary. Contact: OPAKFI, Fo. u. 68, 1027 Budapest, Hungary; Fax: +36 1 202 0452.

5th European Conference on Speech Communication and Technology: 22-25 September 1997, Patras Greece. Contact: G. Kokkinakis, Department of Electrical and Computer Engineering, University of Patras, 26110 Rion-Patras, Greece; Fax: +30 61 991 855, E-mail: gkokkin@wcl.ee.upatras.gr.

1997 IEEE Ultrasonics Symposium: 7-10 October 1997, Toronto, Canada. Contact: S. Foster, Department of Medical Biophysics, Sunnybrook Health Science Ctr., 2075 Bayview Avenue, Toronto, Ontario M4N 3M5, Canada; E-mail: stuart@owl.sunnybrook.utoronto.ca

134th Meeting of the Acoustical Society of America: 1-5 December 1997, San Diego, CA. Contact: Acoustical Society of America, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; WWW: http//asa.aip.org.

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33e conférence d'acoustique "Acoustique architecturale et de bâtiment": 23-25 septembre 1996, Prague, République Tchèque. Renseignements: CsAS Technicka 2, 166 27 Praha 6, Czech Republic; Fax: +42 2 311 1786.

Noise-Con 96: 29 septembre - 2 octobre 1996, Bellevue, WA. Renseignements: Noise-Con 96 Conference Secretariat, Engineering Professional Programs, 3201 Fremont Avenue North, XD-51, Seattle, WA 98103, Tel.: 206-543-5539; FAX: 206-543-2352; E-mail: uw-ept@engr.washington.edu.

Rencontre centenaire de l'American Academy of Otolaryngology -- chirurgie de la tête et du cou: 29 septembre - 3 octobre 1996, Washington, DC. Renseignements: American Academy of Otolaryngology--Head and Neck Surgery, One Prince St., Alexandria, VA 22314; Tel.: 703-836-4444; FAX: 703-683-5100.

4e conférence internationale sur le traitement de la langue parlée: 3-6 octobre 1996, Philadelphia, PA. Renseignements: ICSLP 96, Applied Science & Engineering Laboratories, A.I. duPont Institute, P. O. Box 269, Wilmington, DE 19899, Tel.: 302-651-6830; TDD: 302-651-6834; FAX: 302-651-6895; E-mail: ISCLP96@asel.udel.edu; WWW: http://www.asel.udel.edu/speech/icslp/html.

Conférence d'automne - parole et audition: 24-27 octobre 1996, Windmere, Royaume-Uni. Renseignements: Institute of Acoustics, P.O. Box 320, St. Albans, AL1 1PZ, U.K.

Semaine canadienne d'acoustique 1996: 7-11 octobre 1996, Calgary, Alberta, Canada. Renseignements: Dr. E. Slawinski, Department of Psychology, University of Calgary, 2500 University Drive NW, Calgary, AB, T2N 1N4. Tel.: 403-220-5205; FAX: 403-282-8249; E-mail: eslawins@acs.ucalgary.ca.

3e rencontre conjointe de l'Acoustical Society of America et de l'Acoustical Society of Japan: 2-6 décembre 1996, Honolulu, HI. Renseignements: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; FAX: 516-576-2377; E-mail: asa@aip.org, WWW: http://asa.aip.org.

14e conférence mondiale sur les tests non-destructifs: 8-13 décembre 1996, New Delhi, Inde. Renseignements: B. Jaj, Metallurgy and Materials Group, Indira Gandhi Centre for Atomic Research, Kalpakkam 603102, India; E-mail: dmg@igcar.iitm.emet.in.

Symposium international sur la simulation, visualisation et l'auzalisation pour la recherche et l'éducation en acoustique: 2-4 avril 1997, Tokyo, Japon. Renseignements: M. Morimoto, Faculty of Engineering, Kobe University, Rokko, Nada, Kobe 657, Japan; Fax: +81 78 881 2508.

Conférence internationale sur l'acoustique, la parole et le traitement de signal ICASSP 97: 21-24 avril 1997, Munich, Allemagne. Renseignements: H. Fastl, Lehrstuhl fur Mensch-Maschine-Kommunikation, Technische Universitat Munchen, 80290 Mnchen, Germany; Fax: +49 89 2105 8535; E-mail: fas@mmk.e-tchnik.tu.muenchen.de.

133e rencontre de l'Acoustical Society of America: 16-20 juin 1997, State College, Pennsylvanie. Renseignements: Acoustical Society of America, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; WWW: http//asa.aip.org.

Internoise 97: 25-27 août 1997, Budapest, Hongrie. Renseignements: OPAKFI, Fo. u. 68, 1027 Budapest, Hungary; Fax: +36 1 202 0452.

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CAA Prize in Memory of Raymond Hétu

At the last Board of Directors' meeting in Quebec City I accepted the task of forming and chairing a committee to look into the possibility of establishing a CAA prize in memory of Raymond Hétu. Sharon Abel, Blaise Gosselin and Chantal Laroche have joined me on this committee. We have considered various options for this prize - each with advantages and disadvantages (particularly financial). The committee invites comments from CAA members regarding these and other options. Please communicate your comments to me by April 1. The committee will synthesize the comments into a proposal to be tabled at the next Directors' meeting in May. The options considered by the committee are as follows:

Option 1 - Establish a new annual graduate student award - in the amount of about $500 - for the Canadian student undertaking the best research project furthering Raymond Hétu's main research objective - to elucidate the real-world problems experienced by people in occupational settings, and to improve that experience (by means other than engineered noise control, since the CAA Eckel Student Prize for Noise Control exists). A prize of $500 would require an investment of $5000-8000. The CAA does not have surplus money to commit at this time. Thus, the new prize could be financed in one of several ways: increase membership fees; solicit donations from members; reduce the amount of one or more existing prizes (for example, the three student-presentation prizes, or the Directors' awards);

Option 2 - Rename an existing prize (for example, one or all of the three student-presentation prizes or of the Directors' awards) in memory of Raymond Hétu. This would clearly have no financial implications. The conditions of the renamed prize could be modified to reflect Raymond Hétu's priorities discussed above.

Murray Hodgson *Tel: (604) 822-3073*
E-mail: hodgson@mech.ubc.ca *Fax: (604) 822-9588*

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5e conférence européenne de la communication et la technologie de la parole: 22-25 septembre 1997, Patras Grèce. Renseignements: G. Kokkinakis, Department of Electrical and Computer Engineering, University of Patras, 26110 Rion-Patras, Greece; Fax: +30 61 991 855, E-mail: gkokkin@wcl.ee.upatras.gr.

Symposium de 1997 de l'IEEE sur les ultrasons: 7-10 octobre 1997, Toronto, Canada Renseignements: S. Foster, Department of Medical Biophysics, Sunnybrook Health Science Ctr., 2075 Bayview Avenue, Toronto, Ontario M4N 3M5, Canada; E-mail: stuart@owl.sunnybrook.utoronto.ca

134e rencontre de l'Acoustical Society of America: 1-5 décembre 1997, San Diego, Californie. Renseignements: Acoustical Society of America, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; WWW: http//asa.aip.org.

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The Canadian Acoustical Association
l'Association Canadienne d'Acoustique

PRIZE ANNOUNCEMENT

A number of prizes, whose general objectives are described below, are offered by the Canadian Acoustical Association. As to the first four prizes, applicants must submit an application form and supporting documentation to the prize coordinator before the end of February of the year the award is to be made. Applications are reviewed by subcommittees named by the President and Board of Directors of the Association. Decisions are final and cannot be appealed. The Association reserves the right not to make the awards in any given year. Applicants must be members of the Canadian Acoustical Association. Preference will be given to citizens and permanent residents of Canada. Potential applicants can obtain full details, eligibility conditions and application forms from the appropriate prize coordinator.

Edgar and Millicent Shaw Postdoctoral Prize in Acoustics

This prize is made to a highly qualified candidate holding a Ph.D. degree or the equivalent, who has completed all formal academic and research training and who wishes to acquire up to two years supervised research training in an established setting. The proposed research must be related to some area of acoustics, psychoacoustics, speech communication or noise. The research must be carried out in a setting other than the one in which the Ph.D. degree was earned. The prize is for $3000 for full-time research for twelve months, and may be renewed for a second year. Coordinator: Sharon Abel, Mount Sinai Hospital, 600 University Avenue, Toronto, ON M5G 1X6. Past recipients are:

1990	*Li Cheng*	*Université de Sherbrooke*
1993	*Roland Woodcock*	*University of British Columbia*
1994	*John Osler*	*Defense Research Establishment Atlantic*
1995	*Jing-Fang Li*	*University of British Columbia*

Alexander Graham Bell Graduate Student Prize In Speech Communication and Behavioural Acoustics

The prize is made to a graduate student enrolled at a Canadian academic institution and conducting research in the field of speech communication or behavioural acoustics. It consists of an $800 cash prize to be awarded annually. Coordinator: Don Jamieson, Department of Communicative Disorders, University of Western Ontario, London, ON N6G 1H1. Past recipients are:

1990	*Bradley Frankland*	*Dalhousie University*
1991	*Steven D. Turnbull*	*University of New Brunswick*
	Fangxin Chen	*University of Alberta*
	Leonard E. Cornelisse	*University of Western Ontario*
1993	*Aloknath De*	*McGill University*
1994	*Michael Lantz*	*Queen's University*
1995	*Kristina Greenwood*	*University of Western Ontario*

Fessenden Student Prize in Underwater Acoustics

The prize is made to a graduate student enrolled at a Canadian university and conducting research in underwater acoustics or in a branch of science closely connected to underwater acoustics. It consists of $500 cash prize to be awarded annually. Coordinator: David Chapman, DREA, PO Box 1012, Dartmouth, NS B2Y 3Z7.

1992	*Daniela Dilorio*	*University of Victoria*
1993	*Douglas J. Wilson*	*Memorial University*
1994	*Craig L. McNeil*	*University of Victoria*

Eckel Student Prize in Noise Control

The prize is made to a graduate student enrolled at a Canadian academic institution pursuing studies in any discipline of acoustics and conducting research related to the advancement of the practice of noise control. It consists of a $500 cash prize to be awarded annually. The prize was inaugurated in 1991. Coordinator: Murray Hodgson, Occupational Hygiene Programme, University of British Columbia, 2206 East Mall, Vancouver, BC V6T 1Z3.

1994	*Todd Busch*	*University of British Columbia*
1995	*Raymond Panneton*	*Université de Sherbrooke*

Directors' Awards

Three awards are made annually to the authors of the best papers published in *Canadian Acoustics*. All papers reporting new results as well as review and tutorial papers are eligible; technical notes are not. The first award, for $500, is made to a graduate student author. The second and third awards, each for $250, are made to professional authors under 30 years of age and 30 years of age or older, respectively. Coordinator: Blaise Gosselin, Hydro Québec, 16e étage, 75 boul. René Lévesque ouest, Montréal, QC H2Z 1A4.

Student Presentation Awards

Three awards of $500 each are made annually to the undergraduate or graduate students making the best presentations during the technical sessions of Acoustics Week in Canada. Application must be made at the time of submission of the abstract. Coordinator: Alberto Behar, 45 Meadowcliffe Drive, Scarborough, ON M1M 2X8.

The Canadian Acoustical Association
l'Association Canadienne d'Acoustique

ANNONCE DE PRIX

Plusieurs prix, dont les objectifs généraux sont décrits ci-dessous, sont décernés par l'Association Canadienne d'Acoustique. Pour les quatre premiers prix, les candidats doivent soumettre un formulaire de demande ainsi que la documentation associée au coordonnateur de prix avant le dernier jour de février de l'année durant laquelle le prix sera décerné. Toutes les demandes seront analysées par des sous-comités nommés par le président et la chambre des directeurs de l'Association. Les décisions seront finales et sans appel. L'Association se réserve le droit de ne pas décerner les prix une année donnée. Les candidats doivent être membres de l'Association. La préférence sera donnée aux citoyens et aux résidents permanents du Canada. Les candidats potentiels peuvent se procurer de plus amples détails sur les prix, leurs conditions d'éligibilité, ainsi que des formulaires de demande auprès du coordonnateur de prix.

PRIX POST-DOCTORAL EDGAR ET MILLICENT SHAW EN ACOUSTIQUE

Ce prix est attribué à un(e) candidat(e) hautement qualifié(e) et détenteur(rice) d'un doctorat ou l'équivalent, qui a complèté(e) ses études et sa formation de chercheur, et qui désire acquérir jusqu'à deux années de formation supervisée de recherche dans un établissement reconnu. Le thème de recherche proposée doit être relié à un domaine de l'acoustique, de la psycho-acoustique, de la communication verbale ou du bruit. La recherche doit être menée dans un autre milieu que celui où le candidat a obtenu son doctorat. Le prix est de $3000 pour une recherche plein temps de 12 mois avec possibilité de renouvellement pour une deuxième année. Coordonnatrice: Sharon Abel, Mount Sinai Hospital, 600 University Avenue, Toronto, ON M5G 1X6. Les récipiendaires antérieur(e)s sont:

1990	*Li Cheng*	*Université de Sherbrooke*
1993	*Roland Woodcock*	*University of British Columbia*
1994	*John Osler*	*Defense Research Establishment Atlantic*
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1990	*Bradley Frankland*	*Dalhousie University*
1991	*Steven D. Turnbull*	*University of New Brunswick*
	Fangxin Chen	*University of Alberta*
	Leonard E. Cornelisse	*University of Western Ontario*
1993	*Aloknath De*	*McGill University*
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1994	*Todd Busch*	*University of British Columbia*
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