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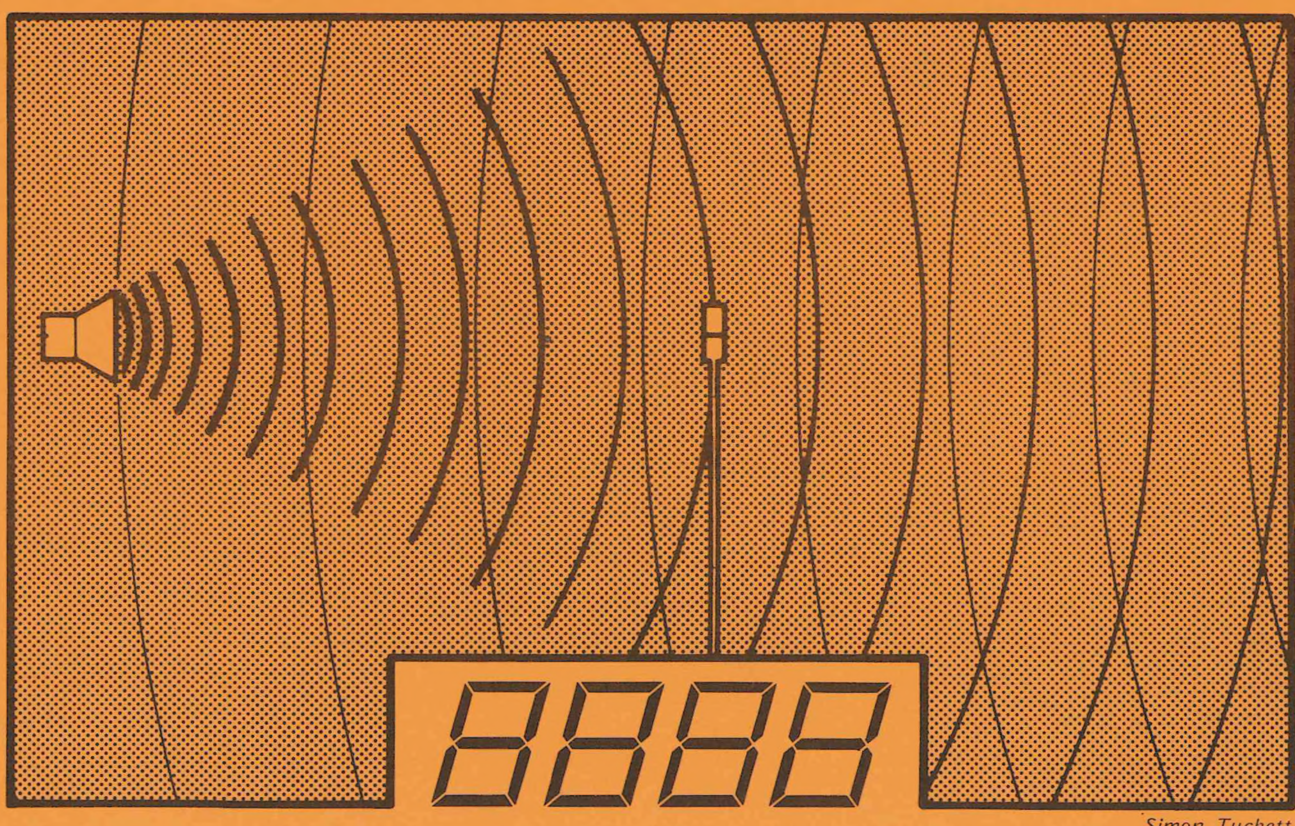

Simon Tuckett

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Articles on all aspects of Acoustics and Vibration in English and French are welcome, and should be addressed to any editor. All papers are reviewed.

Vous êtes invités à soumettre des articles sur tous les domaines de l'Acoustique et des Vibrations, en français ou en anglais. Prière de les envoyer à un des rédacteurs. Tout article est révisé.

NEWS

12th CONGRESS ON ACOUSTICS, 1986

The 12th ICA Organizing Committee will hold its first planning session on February 11, 1986 using a cross-Canada teleconferencing hook-up. Details of this session and future meeting plans will be reported in subsequent issues of Canadian Acoustics.

CAA/MOE TRAINING COURSES

The joint CAA/MOE training courses in noise control and acoustics technology will be held in Toronto during the week of March 29 to April 2, 1982. The fee for each of six training courses offered is $150.00. Information and registration is being co-ordinated by Mr. Geoff Murphy in the Noise Pollution Control Section of the Ontario Ministry of the Environment in Toronto (416) 965-1194.

INTERNATIONAL SYMPOSIUM ON TRANSPORTATION NOISE, SOUTH AFRICA

John Manuel was invited by the South African Acoustics Institute to be the key note speaker at the October 1981 international symposium on transportation noise and the annual general meeting of SAAI. At the invitation of the South African Council for Scientific and Industrial Research, he also visited the six universities where an engineering acoustics program was offered. Details of the visit and copies of papers presented at the symposium may be obtained from him (address on inside back cover).

NEW RESEARCH CONTRACTS

To Bell Northern Research Ltd., Ottawa, Ont., $233,277 for "Development of normalization of speech for automatic recognition". Awarded by the Dept. of National Defence.

To Dr. I. Reid, Halifax, N.S., $3,000 to "Develop techniques for seismic data analysis to improve velocity-depth resolution. Awarded by the Dept. of Energy, Mines and Resources.

To Autonetics Research Ass., Sooke, B.C., $5,000 for "Development and testing of sonar processing techniques". Awarded by the Dept. of Fisheries and Oceans.

To Marinav Corp., Ottawa, Ont., $19,996 for "Feasibility, requirement and economic study of vertical acoustic sweep systems for use in the Arctic". Awarded by the Dept. of Fisheries and Oceans.

To Quebec Industrial Research Centre (CRIQ), Sainte-Foy, Que., $99,180 for "Study of meat tenderization by ultrasonic shocks". Awarded by the Dept. of Agriculture.

To Tobionic Systems, Burlington, Ont., $4,477 for "Modification of survey echo sounders".

REPORT ON INTERNATIONAL CONGRESS ON ACOUSTIC INTENSITY MEASUREMENT SENLIS (FRANCE) 30 Sept.-2 Oct. 1981

Last month the first International Congress on Acoustic Intensity Measurement was held at CETIM. 40 papers were presented and at the Technical Exhibition 6 firms displayed specific instrumentation for intensity measurement. The unexpected high attendance of 230 persons from 22 countries proves the growing interest for this technique not only in the university laboratories but also in the mechanical, aerospace and automotive industries. Proceedings have already been distributed to the participants of the meeting. Further copies are available from: CETIM, Service Publication, B.P. 67, F - 60303. SENLIS (France), Price: 180 FF.

ACOUSTICS LETTERS SUPPLEMENT

This supplement contents details of events, products, new books and publications in addition to "Acoustics Letters" Contents. It is available free of charge to named individuals engaged in acoustics activities from: - Acoustics Letters Supplement, 14 Broadway, London SW1H 0BH, England.

MIT SPECIAL SUMMER PROGRAMS

The Massachusetts Institute of Technology Special Summer Program 1982 includes course 6.67s, "Speech Spectrogram Reading: An Acoustical Study of English Words and Sentences", run by Victor Zue, July 26-30, 1982, at a cost of $850. For further information contact: Director of Summer Session, Room E19-356, MIT, Cambridge, Mass., 02139, U.S.A.

1982 CALENDAR OF ACOUSTICAL EVENTS

April 1, 1982, New York City, N.Y. U.S.A.
Ultrasonic Industry Association -
13th Annual Technical Symposium
The theme of the symposium will be "Ultrasonics - Impact on Technology". For further information contact Mr. Clark Stohl, c/o Blackstone Ultrasonics Inc., 111 Allen St., Jamestown, N.Y. 14701. Tel: (716) 665-2620.

April 5-6, 1982, University of Kansas, Lawrence, Kansas U.S.A.
Annual Research Symposium on the Psychology and Acoustics of Music
These symposia bring together active researchers in psychology, physics, audiology, music, music education, music therapy and other related areas to exchange information and stimulate new research in psychology and acoustics of music. For further information contact Patricia E. Sink, Symposium Director, The University of Kansas, Art and Music Education and Music Therapy, 311 Bailey Hall, Lawrence, KS 66045, U.S.A.

May 23-27, 1982, Finlandia Hall, Helsinki Finland
16th International Congress of Audiology
The 16th International Congress of Audiology will be held in the Finlandia Hall of Helsinki 23-27 May 1982. The main topics of round table sessions are audiology past and future, cochlear mechanics, evaluation and measurement of hearing handicap, and retrocochlear hearing disorders. Contributed papers on these and other audiological subjects will also be presented. English will be the official congress language but round table sessions will feature simultaneous translation into French. An exhibition is planned in conjunction with the congress. Details are available from the Secretary-General of the Congress, Tapani Jauhiainen, Otolaryngology Department, Helsinki University Central Hospital, Haartmaninkatu 4 E, 00290 Helsinki 29, Finland.

September 13-17, 1982, University of Göttingen Fed.Rep.Germany
Joint meeting of FASE and DAGA
The joint meeting of FASE (Federation of Acoustical Societies of Europe) and DAGA (German Association for Acoustics) will take place at the University of Göttingen 13-17 September 1982. The first day of the meeting will be devoted to a colloquium on a special topic in speech research. The main areas to be covered during the congress are room and building acoustics, speech research, flow acoustics, nonlinear acoustics, and physical acoustics. The official languages are French, German, and English but no simultaneous translation is being planned. Further information is available from M.R. Schroeder, III., Physikalisches Institut, Bürgerstrasse 42-44, 3400 Göttingen, Fed. Rep. Germany.

1983 ACOUSTICS CALENDAR

May 9-13, 1983, Cincinnati, Ohio — USA

Meeting of the Acoustical Society of America
Chairman: Horst Hehmann
1928 Fullerton Drive
Cincinnati, Ohio 45240

July 13-15, 1983, Edinburgh — Scotland

INTERNOISE 83
For further information contact:
Conference Secretary, Institute of Acoustics
25 Chambers Street, Edinburgh EH1 1HU

July 19-27, 1983, Paris — France

11th ICA CONGRESS
to be preceded and followed by Satellite Symposia
in Lyon, Marseille and Toulouse
G.A.L.F. Secretariat
B.P. 40, 22301 Lannion - Cédex, France

September 1983, London — Gr.Britain

4th Conference of the British Society of Audiology
Details from B.S.A. Secretary, M.C. Martin,
105 Gower Street, London, WC1 E6AH, U.K.

October 1983, High Tatra — Czechoslovakia

22nd Acoustical Conference on Electroacoustics and Signal Recording
Details to be announced.

November 7-11, 1983, San Diego, California — USA

Meeting of the Acoustical Society of America
Chairman: Robert S. Gales, Code 5152
Naval Ocean Systems Center
San Diego, California 92152

and
INTER-NOISE 83
Will be held in Europe.

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1982 CAA Symposium on Acoustics, Toronto

Call For Papers

The 1982 meeting of the Canadian Acoustical Association will be held at the Westbury Hotel, Toronto during the week of October 18, 1982.

Abstracts on all aspects of acoustics are invited for presentation at the 21st technical symposium, October 21 and 22, 1982. Abstracts, not more than 200 words in length, should be forwarded before July 1, 1982.

Other Activities During Acoustics Week

Three 2-day seminars will be offered on October 18-19, 1982, as follows:

1) Vibration: Aspects of vibration will be examined in a multi-session program. Random exitation, analysis, case studies, experimental methods, instrumentation, control of vibration, effects on humans, standards and regulations will be covered.

2) Noise Control in Land Use Planning: The prevention of noise problems occurring in new residential developments will be the central theme of this seminar. Methods for predicting the impact of noise and methods for the control of noise in new housing subdivisions will be examined. Standards, regulations and guidelines will be evaluated.

3) Noise Control in the Workplace: The proposed Ontario occupational noise regulation and measurement guidelines will be examined in detail. Methods of controlling noise exposure in the workplace will be discussed. Case histories will be used to illustrate the implementation of the Ontario noise regulation.

Seminar Fee: $100 per person

Abstracts and seminar registrations should be forwarded to:

Convenor: John Manuel
5007-44 Charles West
Toronto, Ontario
M4Y 1R8

UN SONDAGE SUR LA PARTICIPATION DES ACOUSTICIENS FRANCOPHONES DANS LES ACTIVITÉS DE L'ACA

Le questionnaire ci-dessous a été envoyé aux membres de l'ACA habitant dans la Province du Québec en septembre 1981. En total, 54 questionnaires ont été envoyés et nous avons reçu 14 réponses (25.9%), ce qui est considéré comme un bon pourcentage de réponses dans le domaine des sondages. Une de ces réponses est nulle. Le questionnaire et les résultats sont les suivants:

Questionnaire	Résultats (% des réponses valides) Oui	Non	Pas répondu
1. D'après vous, est-ce que le niveau des activités des acousticiens francophones au sein de l'ACA est suffisant?	7.7%	84.6%	7.7%
2. Quels sont les meilleurs moyens pour assurer une participation active des acousticiens francophones?			
a) Une représentation garantie à la Direction de l'ACA?	76.9%	15.4%	7.7%
b) La création d'une ou de plusieurs chapitres locaux de l'ACA?	84.6%	7.7%	7.7%
c) L'augmentation du contenu en français dans le journal de l'ACA?	84.6%	15.4%	
d) L'encouragement des activités en français pendant la réunion annuelle de l'ACA?	76.9%	15.4%	7.7%

3. Est - ce que vous avez d'autres suggestions ou commentaires? Trois personnes ont répondu à cette question, leur réponses se résument ainsi:

- Tenir la réunion annulle de l'ACA fréquement au Québec
- Créer un chapitre local ou régional
- Créer un chapitre local.

RESULTS OF POLL ON HOW TO ENCOURAGE THE PARTICIPATION OF FRENCH-SPEAKING ACOUSTICIANS IN CAA ACTIVITIES

The questionnaire was sent to all members of CAA in Quebec (54). Responses have been received from 14 (25.9%). There was one invalid response (English-speaking person!). The questionnaire (loosely translated) and the percentage of answers based on the valid responses (13) are as follows:

	Yes	No	Not Answered
1. In your opinion, is the level of activity of the French-speaking acousticians within the CAA sufficient?	7.7%	84.6%	7.7%
2. What are the best means to ensure an active participation of French-speaking acousticians:			
a) A guaranteed representation in the directorate of the CAA;	76.9%	15.4%	7.7%
b) The formation of one or more local chapters of the CAA (in Quebec);	84.6%	7.7%	7.7%
c) The increase of the French content in the CAA journal;	84.6%	15.4%	-
d) The encouragement of activities in French during the CAA annual meeting.	76.9%	15.4%	7.7%

3. Do you have any other suggestions or comments? Three persons answered this question. Their suggestion/comment are summarized as follows:

   - More frequent CAA annual meeting in Quebec
   - Creation of a local or a regional chapter
   - Creation of a local chapter.

NOTICE OF NEXT TORONTO CHAPTER OF CAA MEETING

The next meeting of the Toronto Chapter of CAA is to be held on Monday, April 12, 1982 at 7 p.m. in the Ontario Hydro Auditorium. The topic for the meeting is "Industrial Audiometry" and invited speakers are: Dr. N. Wills, Mrs. D. Benwell and Mr. Errol Davis. For further information contact the convenors: - Dr. Sharon Abel (416) 596-3014 and Winston Sydenberg (416) 823-3200.

EXAMINATION OF THE EFFECT OF SOURCE LOCATION ON SOUND POWER MEASUREMENTS AT LOW FREQUENCIES

by

James Rennie, SNC Group, Environment Division
Montréal, Québec

Mark A. Lang, Centre for Building Studies
Concordia University, Montréal, Québec

Revision of paper D-11 presented at the 101st meeting of the Acoustical Society of America, Ottawa, Ontario, May 1981

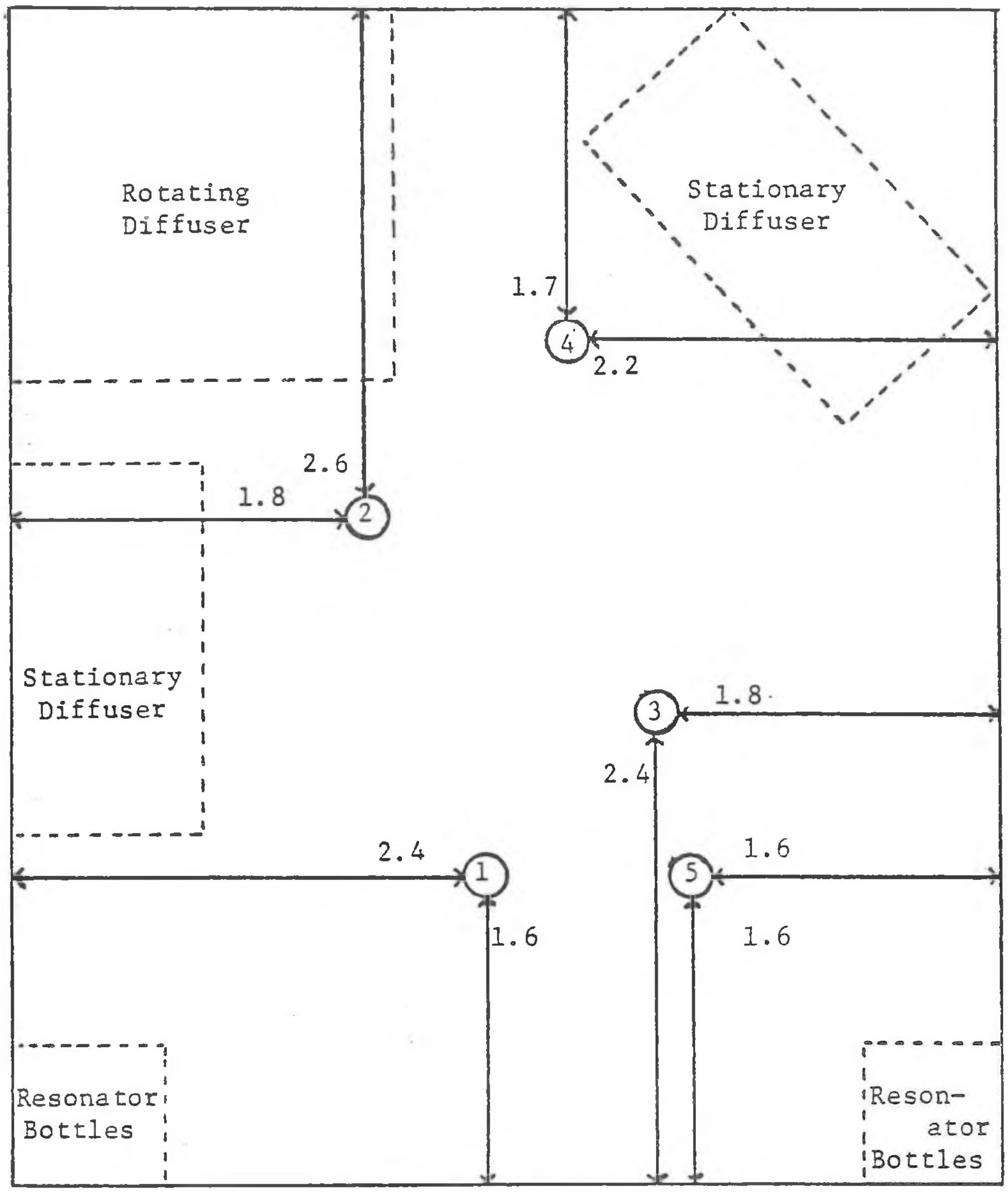
ABSTRACT

The influence of source location and room modifiers are examined with respect to the effectiveness of source position averaging in reducing measurement error for sound power measurements in reverberation rooms.

SOMMAIRE

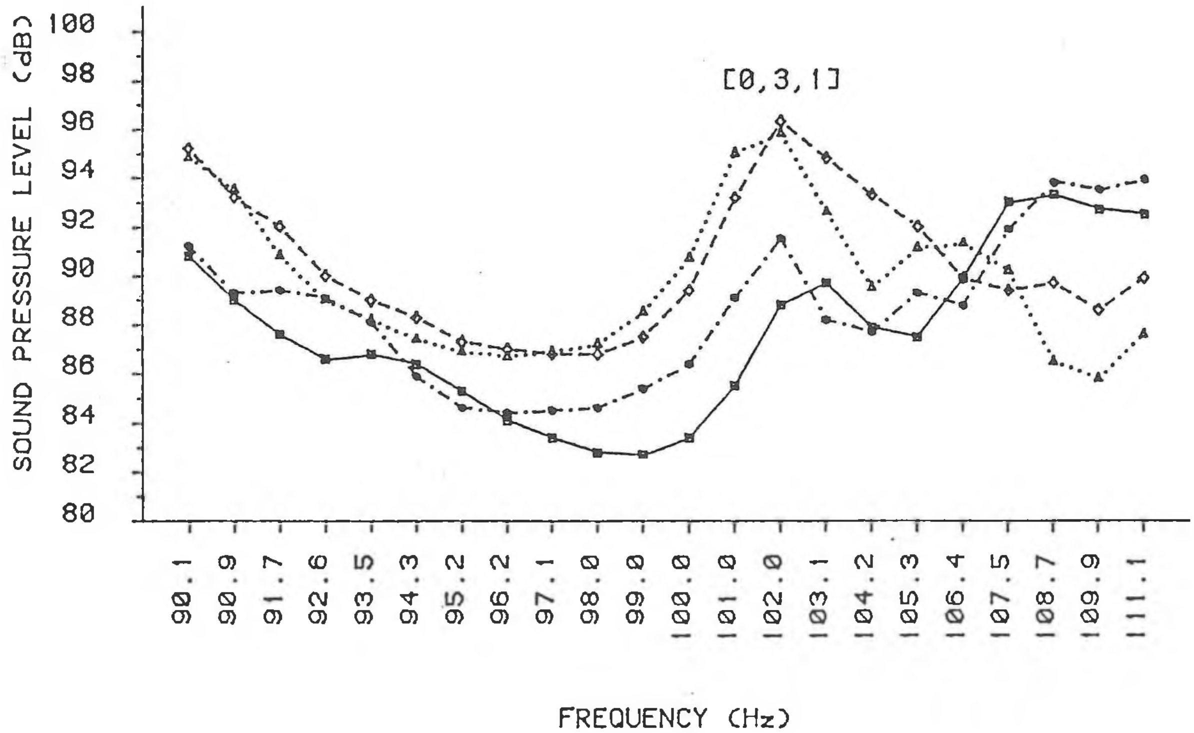
L'influence des modifications apportées à la chambre ainsi que l'incidence de la position de la source sont examinées en regard de la réduction des erreurs de mesures obtenues à partir d'une moyenne entre différentes positions de sources pour la mesure de puissance acoustique en chambre réverbérante.

It is well documented (1, 2, 3, 4, 5) that the precision with which low frequency sound power can be measured in a reverberation chamber depends not only on the effectiveness of the particular sound pressure sampling arrangement, but also on the position of the source within the room which affects the actual power radiated. In many cases, averaging of measured sound pressure levels over a number of source positions is required to determine 1/3 octave sound power levels within the limits of ANSI S1.21-1972 (6), the standard in effect at the time of this study (7).

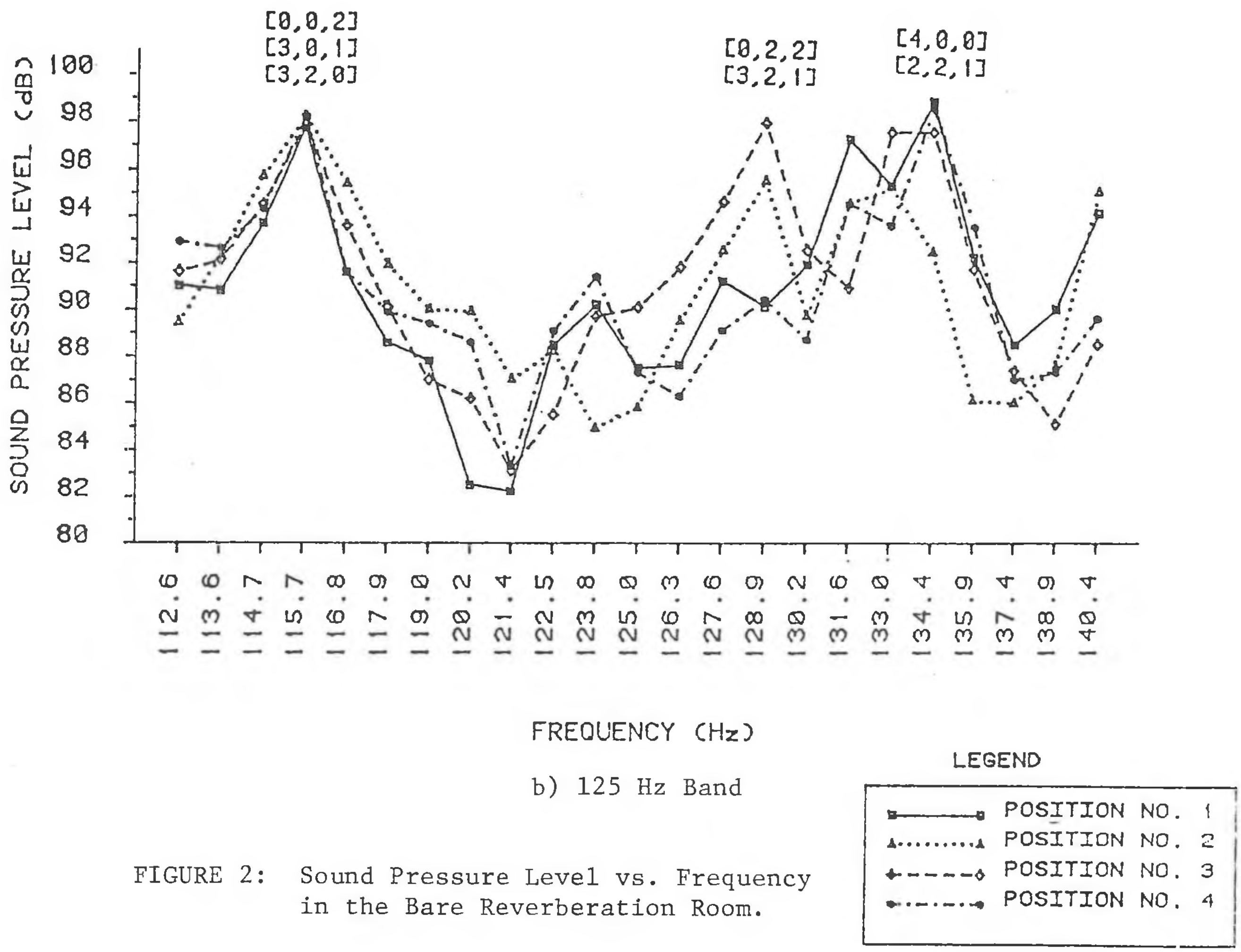


NOTE: All distances in meters

FIGURE 1: Source Position and Room Modifier Locations on 5.1 x 6.1 Meter Floor of Reverberation Room.



a) 100 Hz Band



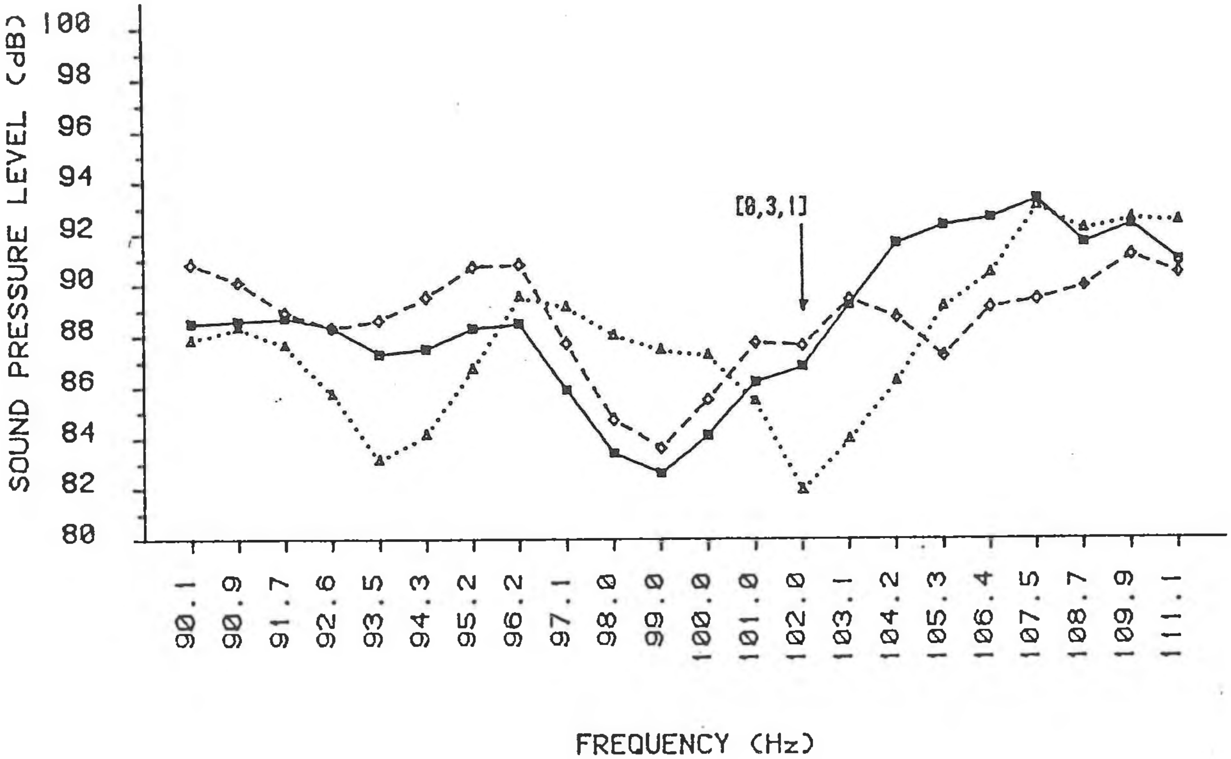
b) 125 Hz Band

FIGURE 2: Sound Pressure Level vs. Frequency in the Bare Reverberation Room.

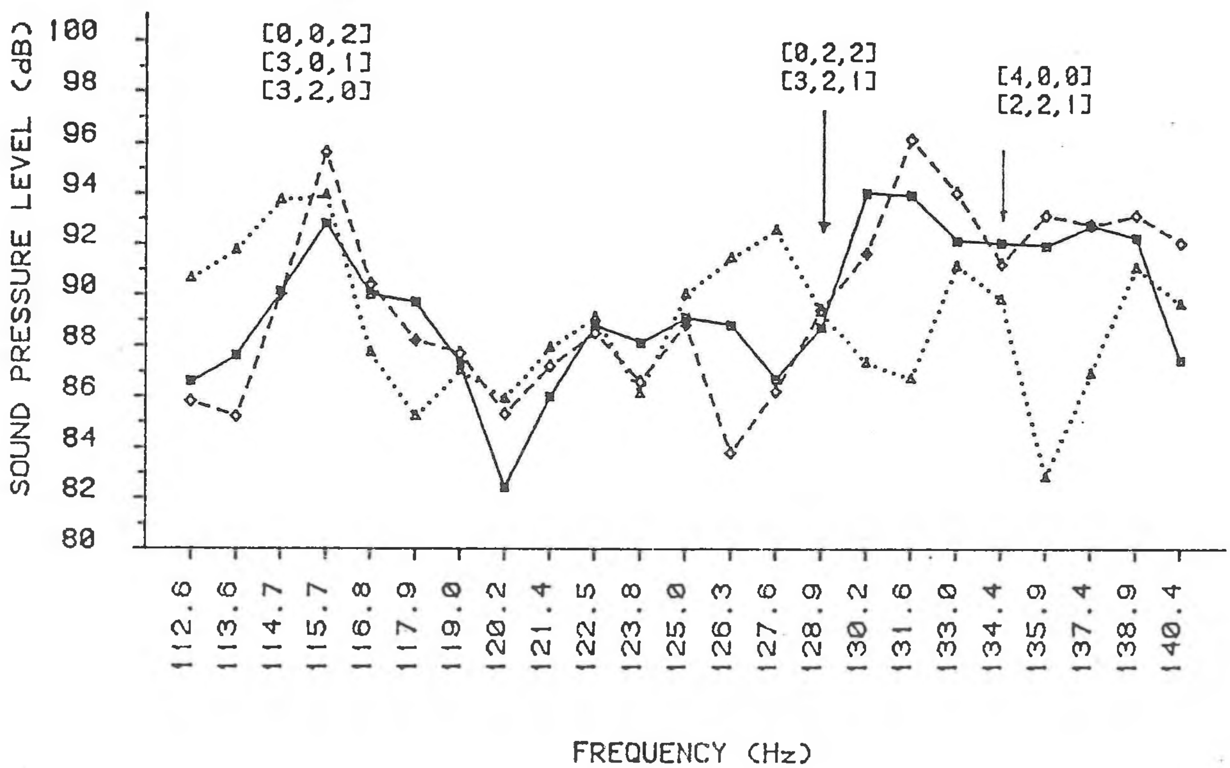
The reverberation chamber at the Centre for Building studies has a volume of 94 cubic meters and is constructed of timber framing on an isolated concrete floor. The interior walls and ceiling have an aluminum finish to provide suitably low absorption. To permit qualification of the chamber for sound power measurement of sources with discrete frequency components, it was necessary to introduce a rotating diffuser, stationary diffusers, and panel and bottle absorbers (see Figure 1). The chamber was able to qualify under ANSI S1.21 with the exception of the room volume (specified as 180 m^3 minimum) and the minimum microphone distance from the room surfaces. Room size restricted this distance to $\lambda/4$ at the lowest frequency of interest while $\lambda/2$ is recommended in the standard. The qualification procedure followed is described elsewhere (6,7).

During the qualification procedure, it was observed that averaging pressure levels over different source positions was generally ineffective in reducing measured standard deviations without the aforementioned diffusing and absorbing elements in the room, i.e. in the bare chamber. The introduction of the room modifications substantially improved source position averaging effectiveness at low frequencies. The following investigation of the measured sound pressure/frequency curves over the 100, 125, 160 and 200 Hz 1/3 octave bands illustrates the reason for the variation in averaging effectiveness.

Figure 2(a) illustrates the space-time average sound pressure levels for the 22 frequencies in the 100 Hz band specified in ANSI S1.2.-1972, for the bare chamber condition. The absorption coefficient in this frequency band is 0.06, giving a modal overlap (M) of 0.58. Modal overlap is a measure of the amount of overlap of the finite-bandwidth modes which occur in a room and can be expressed as a function of room absorption, frequency, and room volume (8). The peak observed appears to correspond to the [0, 3, 1] mode at 102 Hz. What is striking about the response curves illustrated is the remarkable similarity in shape for the four source positions. The calculated standard deviations are similar as well, ranging from 2.9 to 3.4. The similarity leads to an ineffectiveness in source position averaging the average of positions 1 (σ = 3.4) and 4 (σ = 3.1) is only 3.1 and is representative of this behavior. Where improvement is observed, the improvement is small, typically no greater than 0.4 dB of deviation. It can also be seen that for any given frequency, the measured pressure level and hence power level are highly dependent on source position, even allowing for errors resulting from the limited spatial sampling of the sound field at these frequencies (~ 2dB). Variations between source positions are as much as 8 dB.



a) 100 Hz



b) 125 Hz

LEGEND

- POSITION NO. 1
- POSITION NO. 4
- POSITION NO. 5

FIGURE 3: Sound Pressure Level vs. Frequency with Diffusers and Absorbers in Reverberation Room.

The curves for the 125 Hz band in Figure 2(b) show the same trends, especially where strong modal responses occur. The first peak at 115.7 Hz appears to correspond to the degeneracy of the [0, 0, 2] , [3, 2, 0] and [3, 0, 1] modes, and the second peak to the [4, 0, 0] and [2, 2, 1] modes. Again we see from the illustrated deviations that source position averaging is ineffective. It should also be noted that while modal overlap is increased in this band, deviations are rising for the individual source positions.

In the 160 and 200 Hz bands the pressure/frequency response curves become more uncorrelated with respect to source position, as might be expected with the rapidly increasing number of modes present. It can be shown that averaging can become much more effective at these frequencies depending on the relative correlation between source locations.

Let us now compare these results to those observed in the final room configuration, where resonator bottles, panel absorbers, and rotating and stationary diffusers were in place. From the 100 Hz band shown in Figure 3(a), it is apparent that the curves for the three source positions are not nearly as similar in shape as those for the bare room configuration. In particular, the averaging of positions 1 and 4 is now effective, reducing overall deviation to 2.5 dB. Position 5, a new position chosen because averaging over the original source positions did not permit the qualification limits to be met for all frequencies of interest, has a very low standard deviation in this band (σ = 2.0 dB).

In Figure 3(b), the previously strong modal pattern at 125 Hz that was so highly correlated over source position has disappeared, and averaging becomes generally more effective. At the higher frequencies 160 and 200 Hz averaging effectiveness is good but not clearly improved over the bare chamber conditions.

Based on the room modifications made, it appears that the combination of diffusing elements and additional absorption cause decreased correlation between source positions with consequently improved averaging effectiveness. As a means of quantifying the improvement, the equation relating total measurement variance to sound power and pressure variance can be used:

$$\sigma_t^2 = \frac{1}{N_s} (\sigma_w^2 + \sigma_p^2)$$

where Ns = number of source positions, σ_t^2 = total normalized variance, σ_w^2 = normalized variance of sound power output with source position, and σ_p^2 = normalized variance of space averaged reverberant pressure squared.

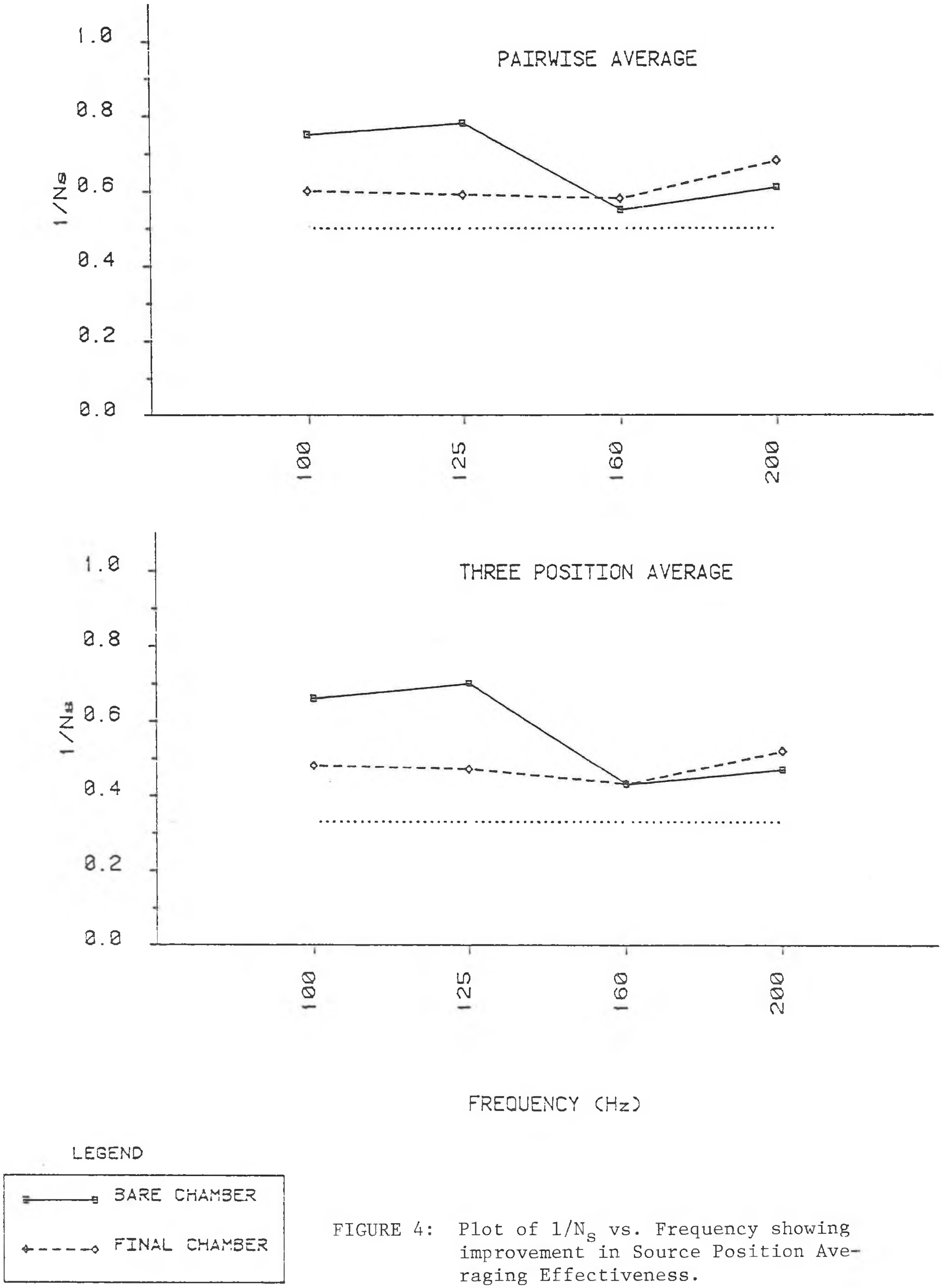


FIGURE 4: Plot of $1/N_s$ vs. Frequency showing improvement in Source Position Averaging Effectiveness.

Thus by using 2 source positions, the total variance should be reduced by a factor of 0.5, and using 3 source positions by 0.33, etc... Figure 4 shows a plot of the average values of 1/Ns for pairwise and three-way source position averages for both the bare room and final room configurations. The improvement in source position averaging effectiveness at low frequencies is clearly illustrated as the final room curves more closely approach the theoretical values.

It should be noted that a $\lambda/4$ or greater separation of source positions does not guarantee uncorrelated responses, at least for this chamber. One must be fortunate in choosing source positions that will give the desired results for reverberation chambers of this size.

REFERENCES

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6. American National Standard S1.21-1972, "Methods for the Determination of Sound Power Levels of Small Sources in Reverberation Rooms", American National Standards Institute, New York, N.Y., 1972

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Underwater Acoustics in the Canadian Archipelago.

by

R. Verrall, Defence Research Establishment Pacific

Press summary of paper presented at the Acoustical Society of America Meeting, Ottawa, May, 1981.

In order to predict the behaviour and performance of sonar equipment in the Arctic, the Defence Research Establishment Pacific in Victoria has made acoustic measurements in Arctic waters since the late 1950's. Although some of the early work was done in the Arctic Ocean proper, most of our interest has centred on the straits within the Arctic Islands.

One of the important parameters is the ambient (or naturally occurring) noise that exists in the water. The magnitude of this noise is much more variable in the Arctic than it is in other oceans. In the winter and spring, when the straits are covered with unmoving, shore-fast ice, the noise levels are very low. Most noise produced during this period is either biological in origin or is caused by the wind-driven snow. On the other hand, during periods of break-up and freeze-up the noise level is very high. Ice floes grinding together produce much noise. During the noisiest periods, the energy in the water is some 70 dB higher than it is during the very quiet spring periods.

Because of the climate, underwater research in the Arctic is much different than it is in other oceans. For example, the simple process of placing equipment into the water is slowed by the necessity of drilling through the ice. Getting the equipment back after the hole is refrozen can be even more troublesome.

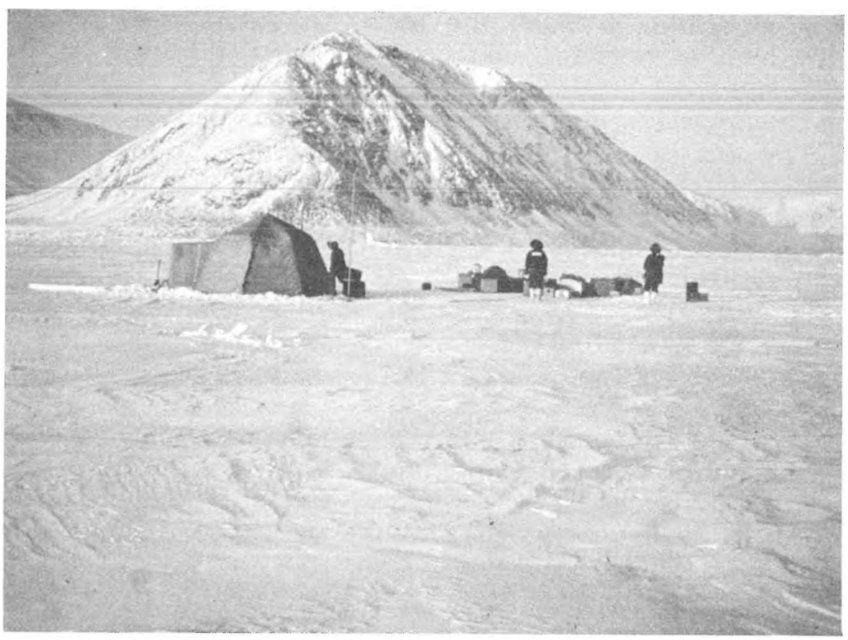
The logistics of field operations are relatively easy in the spring. We set up temporary field camps on the sea ice, and do our experiments from there. In the late summer, some of the channels are relatively ice-free, and experiments may be performed from ice breakers or ice-strengthened vessels. Other seasons of the year are more hostile. During periods of break-up and freeze-up, the ice is too heavy for a ship but not solid enough for a camp. In the winter, the perpetual darkness makes it impossible for an aircraft to land on the ice.

In order to measure year-round ambient noise in the water, we have developed Recoverable Instrument Packages (RIP). They are placed on the sea bed in the summer and are recovered the following summer after having recorded a year's worth of data.

More recently, we have cabled underwater hydrophones to shore camps in order to get 'real time' data. This will give us more control over the data processing. Two problems have been associated with this undertaking. The first is that of building quiet underwater structures. Since the noise is so very low, extemely quiet systems must be developed to measure it. The second problem is that the ice sometimes destroys cables where they cross the foreshore. We have ongoing programs to study both these problems, and we have made some progress on both fronts.

At some locations in the more northerly islands, permanent ice plugs lie locked between the sides of long, narrow fjords. Usually, a foot or two of the ice surface melts every summer, and an equivalent amount freezes to the bottom of the plug in the winter. Although it increases and decreases in thickness, the plug as a whole never breaks up. It has, therefore, the potential of being a semi-permanent platform for acoustic equipment. At one location (Nansen Sound) we have placed hydrophones through holes drilled in the plug. The acoustic signals are telemetered by a radio link to the shore. Thus, we have access to the data at the shore whenever there is enough light to land an airplane. This is particularly advantageous in the summer when it is impossible to land on the water-covered ice. A summary of the acoustic data is also sent south to the lab via satellite. As well as providing valuable information, the satellite link keeps us informed as to the health of the system.

Our future plans include making measurements from ice shelves. These bodies of floating ice are 40 to 50 metres thick - as thick as a 15 story building is high. (Ordinary first-year ice is only 2 metres thick.) We are presently developing hot water techniques to drill through these massive ice chunks.



RECOVERY OF THE ACOUSTIC REFLEX RESPONSE AS A FUNCTION OF NOISE EXPOSURE AND QUIET INTERVAL[1]

by

N.M. Lalande and R. Hétu
Ecole d'orthophonie et d'audiologie
Université de Montréal
C.P. 6128
Montréal, Qué., H3C 3J7

ABSTRACT

In order to know if the acoustic reflex does protect the inner ear from temporary threshold shift caused by the exposure to a loud industrial noise, one need to know the reflex response decay during the noise exposure, as well as the amount of recovery allowed by quiet intervals. The influence of the duration of a broadband noise, on the recovery function of the acoustic reflex has been investigated. Four noise durations and four (or occasionally five) quiet intervals were selected. Results showed that the reflex recovery is an exponential function of the quiet interval that exists between two noise exposures; after a few seconds of rest, recovery proceeds fairly rapidly, even if the continuous noise exposure lasts 12 minutes. Results also indicated that for the 6 and 12 minute exposure durations, the reflex recovery is independent of the noise duration. For the shorter durations, that is those associated with a reflex decay of 20 and 40%, the results tend to indicate that the recovery is probably independent of the percentage of decay prevailing at the end of the noise exposure.

SOMMAIRE

Pour savoir si le réflexe acoustique peut protéger l'oreille interne contre un déplacement temporaire des seuils d'audition consécutif à une exposition à un bruit industriel intense, il faut être capable de prédire la dégradation de la réponse réflexe, tout comme l'ampleur de la récupération permise par des intervalles de silence. L'influence de la durée d'un bruit à large bande de fréquences, sur la fonction de récupération du réflexe acoustique a été étudiée. Quatre durées de bruit, et quatre (ou cinq suivant le cas) intervalles de silence ont été sélectionnés. Les résultats ont montré que la récupération du réflexe est une fonction exponentielle du temps de silence compris entre deux expositions au bruit; après quelques secondes de silence, la récupération s'effectue plutôt rapidement,

[1]Presented at the 101st meeting of the Acoustical Society of America, Ottawa, Ontario, Canada (18-22 May 1981)

même si l'exposition au bruit continu est de 12 minutes. Les résultats ont également indiqué que pour des durées d'exposition de 6 et 12 minutes, la récupération du réflexe est indépendante de la durée d'exposition. Pour les durées les plus courtes, c'est-à-dire celles associées à une dégradation du réflexe de 20 et 40%, les résultats indiquent que la récupération est probablement indépendante du pourcentage d'adaptation qui prévaut à la fin de l'exposition au bruit.

1 - INTRODUCTION

In humans, the acoustic reflex (AR) refers to the contraction of the stapedius muscle (one of the two middle ear muscles) following an exposure to a moderate or a high intensity acoustic signal. When the signal is a broadband noise, a L_{pA} of 72 dB is on average sufficient to activate a minimal contraction of this muscle (Hétu and Careau, 1977), and an increase in intensity induces a larger muscle contraction over a dynamic range of approximately 50 dB (Wilson, 1979). Therefore, a large number of industrial noises are loud enough to activate an important response of the AR.

It is known that the AR changes the mobility of the ossicular chain, causing among other things, an increase in the acoustic impedance at the eardrum, and therefore, an intensity decrease in sounds transmission to the inner ear (Møller, 1972). This attenuation is greater for frequencies lower than 1000 Hz, and in humans, amounts to 15-20 dB for a full contraction of the stapedius muscle. Moreover, the attenuation offered by the AR is proportional to the graded response of the AR (Borg, 1968; Zakrisson, 1975, 1979). Therefore, the AR protects the inner ear from loud sounds.

However, to know if the AR is capable of protecting the ear against temporary threshold shifts (and possibly againts permanent threshold shifts) caused by exposure to loud industrial noises, one has to consider not only the initial response of the reflex, but also its long term response to noises of various durations and temporal characteristics. Because of the numerous difficulties involved in monitoring the AR response of a worker at his job site, one has to rely on laboratory studies.

Industrial sound environments are characterized by noises of broad frequency bands. Therefore, laboratory studies carried on the AR activated by broadband noise should help to understand and predict the AR behaviour in real life situations. Such studies have shown that an exposure to a loud continuous noise produces a rapid decay of the AR within the first 4 or 5 minutes of exposure, followed by a slower decay, and finally by a stabilization of the response at a residual amplitude. The reduced response is reached after approximately 10 minutes of exposure, and its average amplitude is near 45 to 50% of the initial response (Hétu and Careau, 1977; Lalande and Hétu, 1979). Studies have also indicated that when subjects are exposed to intermittent noises of various periods or/and duty cycles, the presence of periodic quiet intervals prevents partially or totally the degradation of the AR (Hétu and Careau, 1977; Lutman and Martin, 1978; Wiley and Karlovich, 1978; Lalande and Hétu, 1979). The partial or complete recovery of the AR response found in these studies following quiet intervals, could be related to the ON-duration of the noise, at least for ON-durations that produce a decay of the AR but has never been investigated as such. Wiley and Karlovich (1978) have indeed shown that when the ON-duration of the intermittent noise is short enough to cause no response decay, the quiet interval necessary for a complete recovery of the excitability of the AR is independent of the ON-duration of the noise. But for ON-durations long enough to produce some adaptation of the reflex, the results of Lalande and Hétu (1979) have not allowed to determine if the time interval associated with a complete recovery of the excitability depends on the ON-duration of the noise, or rather on the amount of adaptation associated with a particular ON-duration. The aim of the present study was to determine if the quiet interval necessary for a full recovery of the excitability of the acoustic reflex (AR), that is the ON-response, depends mainly (a) on the duration of the signal, or rather (b) on the amount of reflex decay associated with a particular signal duration.

2 - METHODOLOGY

2.1 Subjects

The experiment was performed on 6 young female adults. They had normal hearing thresholds and normal middle ear function. Their acoustic reflex thresholds to pink noise was obtained at a L_{pA} of 90 dB or less, and their AR decay to that noise was known, as well as the noise duration associated with a residual steady state reflex response.

2.2 Experimental conditions

Each subjects was monaurally exposed to a continuous pink noise at a L_{pA} of 105 dB, as measured in a 6 cc coupler (Brüel et Kjaer, model 4152). The right and left ear were alternately stimulated. Decay and recovery of the AR was continuously monitored in the non-exposed ear, for various durations of noise exposure and quiet interval. An otoadmittance meter with a 220 Hz probe tone (Grason Stadler, model 1720 B), and a graphic X-Y recorder equipped with a time base (Grason Stadler, model 1701) were used to record the reflex.

The experimental noise exposures are summarized in Table 1. Four noise duration conditions were selected. Two conditions involved the minimal exposure time that induced respectively 20 and 40% reflex decay for each subject. The noise duration associated with these amounts of reflex adaptation was on the average 1,25 and 2 minutes. The other two conditions were 6 and 12 minute exposures. These two durations were associated respectively with the early and the late residual steady state reflex response, which both corresponded to the same amount of reflex adaptation. This adaptation was equal to or greater than 40% for all subjects.

Recovery of the reflex response was measured after quiet intervals of (a) 1, 5, 10, 30 and 50 seconds for the two first exposure conditions, and of (b) 5, 10, 30 and 50 seconds for the 6 and 12 minutes exposures. Monitoring of reflex activity was performed during two minutes following each quiet interval. But the present results refer only to ON-response recovery.

2.3 Quantification of decay and recovery

Figure 1 illustrates how the decay and recovery of the AR was achieved. The percentage of reflex decay (%D) was obtained by the following computation:

$$\% D = 100 - (B/A \cdot 100)$$

EXPERIMENTAL CONDITIONS

reflex response under continuous exposure	noise duration - mn	quiet interval - sec
1. 20% reflex decay	≃ 1,25	1; 5; 10; 30; 50
2. 40% reflex decay	≃ 2	
3. early residual steady state reflex response	6	5; 10; 30; 50
4. late residual steady state reflex response	12	

Table 1. Values of the noise durations and quiet intervals selected for the 18 experimental conditions. The noise durations are associated either with a particular percentage of decay of the reflex response, or with the early and late residual steady state reflex response.

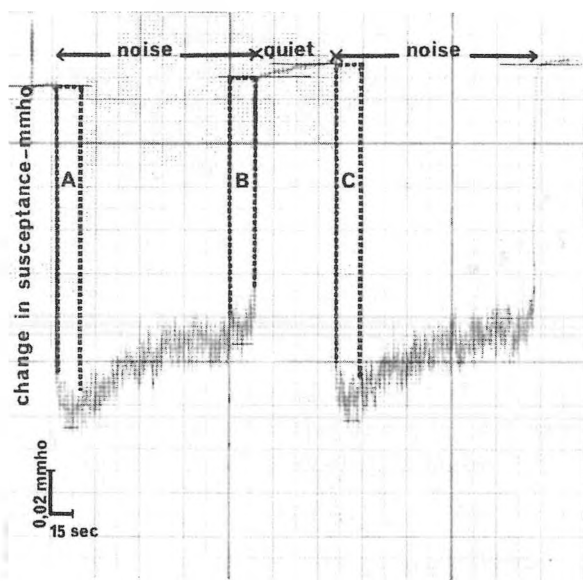


Figure 1. Recording of the acoustic reflex to a noise exposure condition, showing how the decay and the recovery of the acoustic reflex was computed.

where B is the median change in susceptance during the last 15 seconds of the initial noise exposure, and A is the median change in susceptance during the first 15 seconds of that exposure, herein defined as the initial ON-response.

The recovery of the ON-response was computed by two different methods. The first one was a measure of the amplitude of the second ON-response, relative to the initial ON-response, expressed in terms of percentage:

2nd ON-response - % = C/A . 100

where C is the median change in susceptance, of the ON-response following a quiet interval, herein defined as the second ON-response.

The second method took into account the amount of reflex decay at the beginning of the pause, therefore allowing a true measure of the recovery of the ON-response (% R). It was obtained by computing:

% R = C - B/A - B . 100

For both methods of calculation, a 100% value indicated a complete recovery of the amplitude of the initial ON-response. But the second method permitted to compare readily the recovery functions for the four exposure conditions.

2.4 Data analysis

The effect of the amount of reflex decay and of the duration of the noise, on the recovery of the AR at each quiet interval selected, were analysed using a Friedman two-way analysis of variance by ranks (Kirk, 1968). Significant effects were further analysed using the a posteriori test of Nemenyi (Kirk, 1968). Levels of confidence of 0,05 and 0,01 were adopted for all statistical analysis.

3. RESULTS AND DISCUSSION

Figure 2 shows that for very short quiet intervals, that is 1 and 5 seconds, the recovery values simply reflect the amount of response decay at the end of the exposure. When quiet intervals allowed partial or complete reflex recovery, that is for intervals of 10, 30 and 50 seconds, results show that the difference between the median recovery values for the four noise conditions is relatively small. In fact, there were no significant differences, except for the 10 seconds interval, for which there was a significant difference at the 0,05 level between the median value of the 20% condition compared to the 6 and 12 minutes conditions. The absence of significant differences could be explained by the small sample used in this study, as well as by the large intersubject variability associated with reflex recovery. However, these present results as a whole indicate that the time required for a complete or nearly complete reflex recovery is rather short, that is slightly less than one minute, for exposure durations ranging from several seconds to 12 minutes.

A better picture of the actual recovery process is obtained when the amount of reflex decay is taken into account in the computation of the reflex recovery. As shown in Figure 3, the recovery functions appear roughly similar for all exposure conditions. There were indeed no statistical significant differences across conditions except for the 5 seconds interval. In this case,

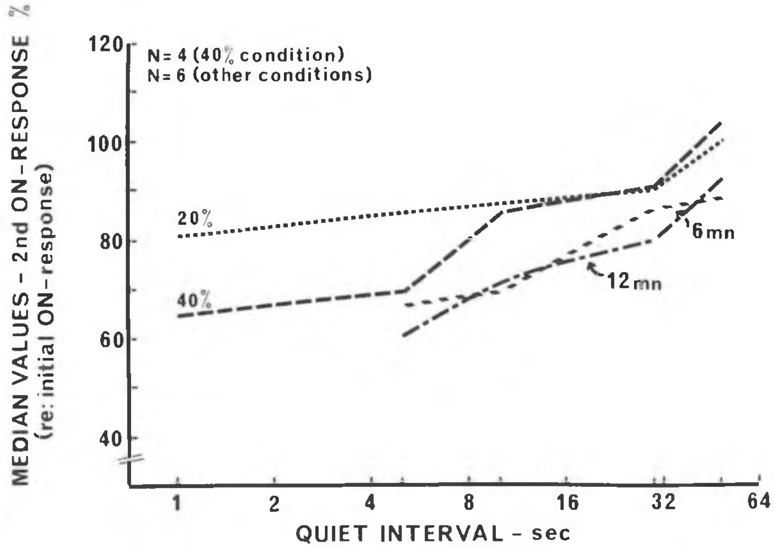


Figure 2. Median values of the second ON-response as a function of the quiet interval. The second ON-response is expressed as a percentage of the amplitude of the initial ON-response.

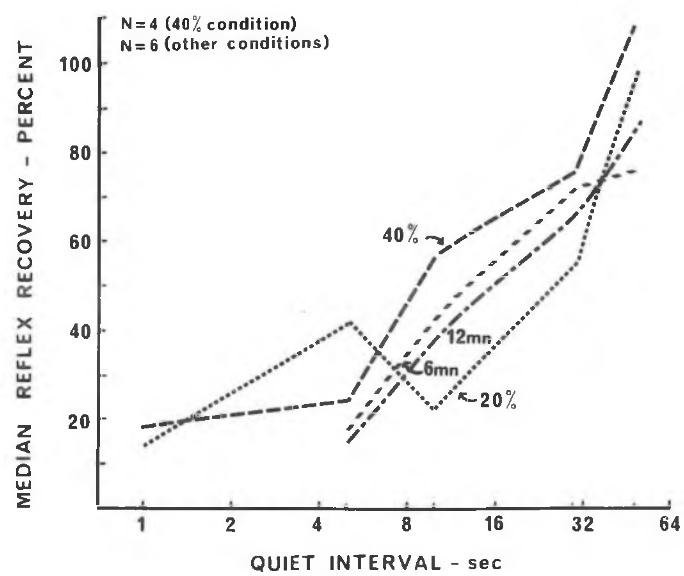


Figure 3. Median values of the percentage of recovery of the ON-response as a function of the quiet interval.

differences at the 0,05 level was found between the 20% condition compared to the 12 minutes condition. Because of this, one cannot consider that the recovery functions are perfectly identical for all conditions. Whether this difference is true or is due to the small sample used in this study, remains to be determined with a larger group of subjects. Moreover, inspection of individual data suggests that after a 50 seconds rest period, the full recovery was obtained more often when the durations of the exposure were shorter (1,25 and 2 minutes vs 6 and 12 minutes). Again, it remains to be verified with a larger group that a quiet interval of 50 seconds allows the same amount of reflex recovery for short and long noise exposures.

Nevertheless, the present results allow one to make the following two statements:

- First, recovery is an exponential function of time. After a few seconds of rest, recovery proceeds fairly rapidly even if the continuous exposure lasted several minutes. This rapid recovery of the reflex excitability has been previously reported by Borg and Ödman (1979). These authors found complete recovery in less than 10 seconds, instead of 50 seconds as obtained in this study. The spectrum and the intensity level of the stimulus are probably responsible for these differences.

- Secondly, for noise durations associated with a residual steady state response, the reflex recovery is independent of the exposure duration. A doubling of the exposure duration from 6 to 12 minutes, did not change indeed the recovery function of the ON-responses. For both durations, a recovery of 80% or more was obtained after a quiet interval slightly less than one minute. It is therefore a relatively short time compared to the exposure time to the noise. It should be mentioned however that, if the present results showed that the reflex recovery is independent of the exposure duration, this may not hold for much longer exposure durations, as suggested by the results obtained by Gerhardt et al. (1976) on chinchillas. It also may not be true for other patterns of noise exposures as suggested by Borg et al. (1979).

4. CONCLUSION

The present results indicate that for continuous exposures to a wide band noise of durations smaller than 12 minutes, the quiet interval necessary for a complete or nearly complete recovery of the excitability of the reflex response is rather short, that is, in the order of one minute. This implies that industrial noises characterized by the presence of periodic intermittency with (a) ON-durations equal to or smaller than 12 minutes, and (b) relative quiet intervals (levels lower than the acoustic reflex threshold for that noise, see Borg, 1980), probably allow the activation of the stapedius muscle at a nearly maximum contraction during the whole working day. Except for ON-durations smaller than a few seconds, for which no reflex decay takes place, the shorter is the ON-duration, the greater will be the protection offered by the reflex, since it is known that the decay of the AR to a continuous noise starts within the first 30 seconds of exposure (Lutman and Martin, 1978; Wiley and Karlovich, 1978; Wilson et al, 1978).

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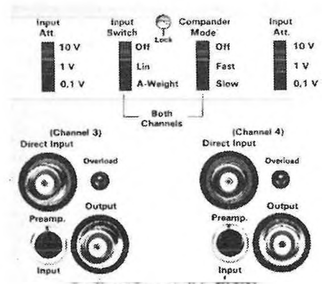

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TORONTO CHAPTER OF CAA MEETING REPORT

On January 11th, 1982, the Toronto Chapter of the C.A.A. devoted an evening meeting to Room Acoustics. Featured, were presentations by Michael Merritt of Engineered Sound Systems Limited and Aubrey Edwards of Ontario Hydro.

Michael Merritt thoroughly covered room acoustics and the Engineer's role in the design of sound reinforcement systems, going into the governing equations and, practical application (when the "correct answer" still doesn't apply). Mike also brought a P.A. System and microphones and demonstrated the effects of gain and multiple microphones on P.A. System feed back.

Aubrey Edwards detailed his research at Ontario Hydro into open office acoustics, and in particular the effectiveness of barriers, and ceiling, wall, floor and windown blind absorption using articulation index as the criterion. He then conducted a tour of the Hydro Place Offices, where his acoustic designs are in use.

Both presentations were very well received as evidenced by the many questions and comments.

The meeting was organized by John Swallow and Andy McKee. The next meeting will be held April 12, 1982 on the subject of Industrial Audiometry.

John C. Swallow
Barman Coulter Swallow Associates

ELECTRO-ACOUSTIC FACTORS DETERMINING TELEPHONE SPEECH QUALITY

by

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ABSTRACT

In this paper the electro-acoustic factors affecting the quality of telephone speech are categorized as: - listening factors such as loudness, noise, frequency response, and listener echo, - talking factors such as talker echo and sidetone, and - conversational factors such as delay. Listening factors in particular are emphasized and qualitatively interpreted in terms of acoustic pressures at the ear relative to the thresholds of hearing and pain. Typical telephone connections are then considered in which a number of the above factors are manifested; this is done in the context of a network simulation facility used by Bell-Northern Research to simulate various telephone network impairments for the purposes of objective and subjective evaluation. Grade-of-service results that relate customer opinion of transmission quality to the level of a transmission parameter are also discussed. These include listener echo and delay, and subjective-equivalence modelling of speech-correlated digital noise in terms of continuous analog noise.

SOMMAIRE

Dans ce document, les facteurs électro-acoustiques exerçant un effet sur la qualité des signaux vocaux téléphoniques transmis sont divisés en trois classes : les facteurs liés à l'écoute, comme la force des sons, le bruit, la réponse en fréquence et l'écho de la personne à l'écoute, les facteurs liés au parler, comme l'écho de la personne qui parle et l'effet local, et enfin les facteurs liés à la conversation comme le délai d'attente. L'auteur met l'accent plus particulièrement sur les facteurs liés à l'écoute ; ces derniers font l'objet d'une interprétation qualitative en termes de pression acoustique imposée à l'oreille par rapport aux seuils de l'audition et de la douleur.

Aux fins d'illustration, l'auteur examine ensuite des communications téléphoniques typiques où un certain nombre des facteurs susmentionnés se manifestent. Pour ce faire, il utilise une installation de simulation de réseau de la Société de recherches Bell-Northern pour simuler différentes réductions de la qualité de transmission pour faire une évaluation objective et subjective. L'auteur traite aussi de résultats quant au niveau du service, en établissant un rapport entre l'opinion du consommateur sur la qualité de la transmission et le niveau d'un paramètre de transmission. Ces résultats comprennent l'écho de la personne à l'écoute, le délai d'attente et l'établissement d'un modèle subjectif et équivalent du bruit de type numérique lié à la conversation en termes de bruit analogue continuel.

1.0 INTRODUCTION

In order to extend the range over which humans can communicate verbally, it is necessary to effectively open the acoustic face-to-face communication path, and bridge the resulting gap by an electrical network (i.e., the telephone network). This gives rise to an electro-acoustic communication path. The electro-acoustic components of the path correspond to the telephone sets, which on the talking side convert acoustic pressures into electrical signals to be sent into the network, and conversely on the listening side convert received electrical signals from the network into acoustic pressures.

One of the concerns of a telephone company is to have a thorough understanding of electro-acoustic factors such as attenuation, noise, distortion, echo, delay and sidetone etc., and the way in which they affect telephone speech quality[1]. This is essential in order that satisfactory service can be provided to customers on an ongoing basis as the network continually evolves as a result of new technology, economic considerations, and the need to provide new services.

It is useful to classify electro-acoustic factors according to the way in which they affect the listening, talking and conversational modes of communication, as indicated in Table 1[2]. Listening factors include continuous noise which is usually associated with the analog communication systems, and noise which is correlated with the speech signal amplitude as in digital systems; results pertaining to the subjective nature of these noise components will be presented later. Another listening factor of significance is frequency response distortion, an example of which is also discussed in terms of listener echo. Listener echo is an electrical reverberation (analogous to acoustic room reverberation) which can arise in (but is not peculiar to) telephone connections employing digital technology, if such connections are not properly engineered.

TABLE 1

CLASSIFICATION OF ELECTRO-ACOUSTIC FACTORS

- LISTENING FACTORS	: - Loudness Loss
	- Noise
	  - Continuous (Analog)
	  - Correlated (Digital)
	- Listener Echo
	- Frequency Response

- TALKING FACTORS	: - Talker Echo - Sidetone
- CONVERSATIONAL FACTORS	: - Delay - Asymmetry

A prime example of an electro-acoustic factor affecting the talking mode of communication is talker echo. If a speech signal reflected or echoed from the distant end of the connection has sufficient volume and is sufficiently delayed (due to a long connection), it can have an inhibiting effect on the talker. Thus such echo signals must be carefully controlled.

Finally, the conversational mode of communication involves a dynamic interaction between the two participants. A classical electro-acoustic factor which can

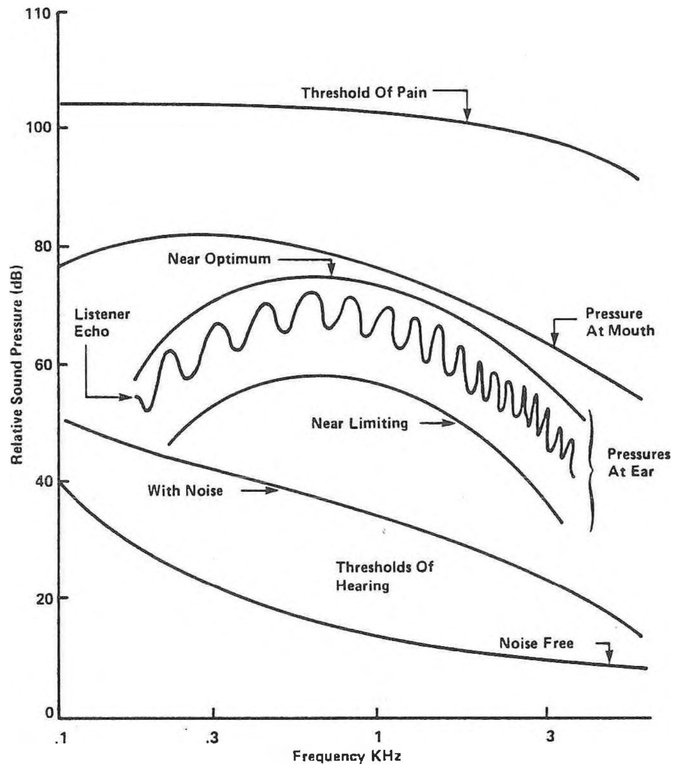


Figure 1 Putting Listening Factors In Context

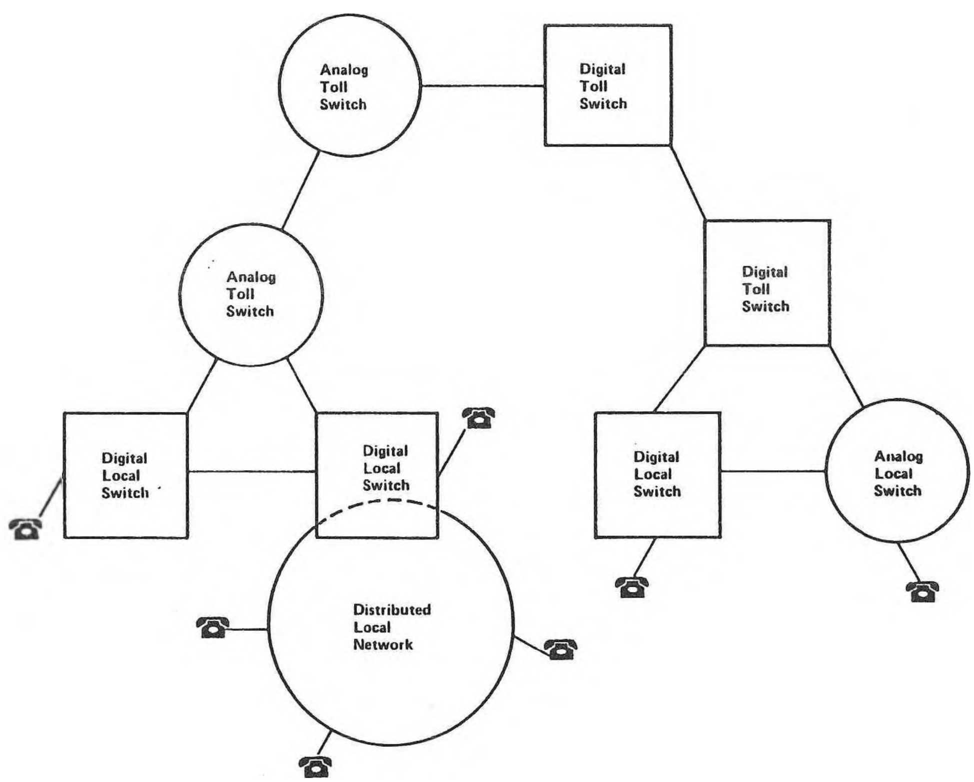


Figure 2 General Mixed Analog/Digital Network

influence this dynamic interaction is significant round trip delay; for example, on 2-hop satellite connections such delays can exceed 1 second.

Although all of the above as well as other electro-acoustic factors are important as far as telephone network performance is concerned, in order to limit the scope of this paper, only certain listening factors will be emphasized in what follows.

It is useful to interpret listening factors in terms of the way in which they are perceived at the listener's ear. This is illustrated in Figure 1 which presents relative sound pressure level at the ear versus frequency[2]. All sound pressures ordinarily fall between the threshold of pain on the high pressure side, and the noise free threshold of hearing on the low pressure side. The effect of noise generally has the undesirable effect of making low level speech signals difficult to hear and thereby effectively raises the threshold of hearing. For this basic reason the telephone network is engineered to control both received speech and noise levels.

Frequency response distortion is also evident from the two smooth convex curves in Figure 1, in that the received sound pressures do not differ by a constant amount from the transmitted pressures at all frequencies. This is largely because of imperfections in the telephone sets and the connections (called local loops) between the sets and the switching offices. The ripple response in Figure 1 is another example of frequency distortion caused by electrical reverberation, or the listener echo phenomenon referred to earlier. The subjective effect is a hollow or "rain barrel" sound, and will be an item of further discussion in this paper.

Thus Figure 1 illustrates that for satisfactory quality or grade-of- service, telephone connections should be engineered such that noise levels are suitably low, speech levels are near optimum, and frequency distortion is not excessive.

2.0 SPECIFIC LISTENING FACTORS AFFECTING LONG DISTANCE TELEPHONE NETWORK PERFORMANCE

2.1 General

Figure 2 is a simplified illustration of the telephone network showing the local and toll levels, as well as the more recent local distributed switching network. This figure indicates that the switching offices may be either analog (with electro-mechanical crosspoints) or digital (with electronic crosspoints). Similarily the connections or trunks joining the offices may be either analog or digital. The current analog/digital stage of network evolution gives rise to some specific electro-acoustic factors affecting telephone network performance, one of which is discussed in what follows.

2.2 The Nature of Analog and Digital Noise

In order to determine the performance of a mixed network, it is necessary to characterize the combination of continuous noise from the analog portion and the speech-correlated noise from the digital portion. A basic difference between these two noise components is that continuous analog noise is perceptible during both speech and silence, whereas correlated digital noise is only perceptible during speech, and only if the digital noise level is high enough.

Basic threshold of perceptibility tests for speech-correlated digital noise have been carried out by Bell-Northern Research and other organizations[2,3].

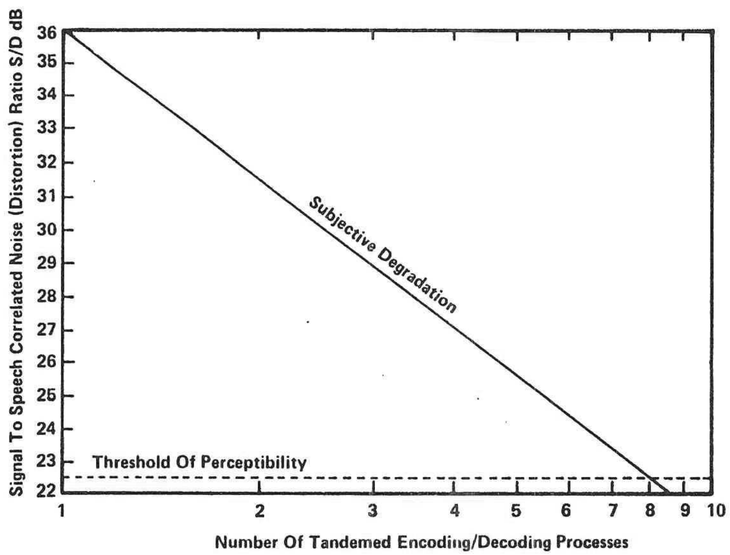


Figure 3 Degradation Of Signal To Speech Correlated Noise Ratio With Number Of Tandemed Encoding/Decoding Processes (For 64kb/s PCM With μ-Law Companding)

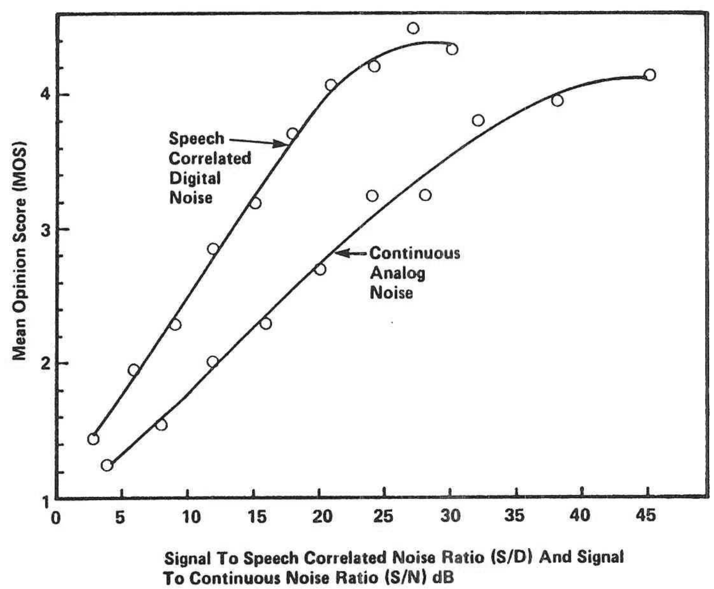


Figure 4 Subjective Test Results For Speech Correlated Digital Noise And Continuous Analog Noise

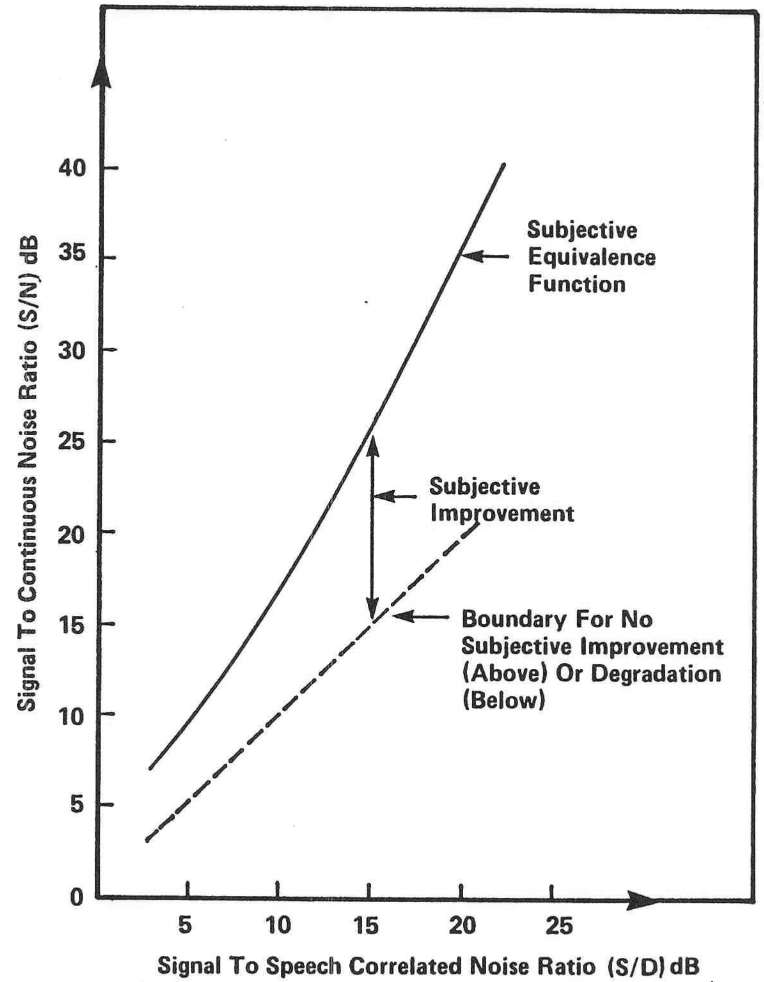


Figure 5 Signal To Continuous Noise Ratios Which Are Subjectively Equivalent To Signal To Speech Correlated Noise Ratios

These tests were conducted with subjects listening to speech plus various levels of speech correlated noise simulated by a Modulated Noise Reference Unit (MNRU)[2]. The results indicate that the perceptible threshold corresponds to a speech signal to distortion (or speech signal to speech-correlated noise) ratio (S/D) of between 21 and 24 dB.

The relevance of this to a mixed network having analog switching and digital transmission for example, is that speech signals are converted from analog to digital (A/D) and from digital to analog (D/A) by terminal equipment called channel banks[5]. In North America the encoding (A/D) and decoding (D/A) process corresponds to 64 kb/s Pulse Code Modulation (PCM) with u-law companding[8]. A single encoding/decoding process of this type produces a S/D ratio of about 36 dB, which is well above the threshold of perceptibility.

A basic question of interest to a telephone network planner is: how many tandem encoding/decoding processes of this nature can be permitted in a connection before the speech correlated digital noise becomes perceptible? Based on extensive studies at Bell Laboratories[4,6], Figure 3 indicates about eight processes, since at that number the perceived S/D ratio has degraded from 36 dB, to the perceptible threshold of about 22 or 23 dB.

Another question of interest is: how do perceptible levels of correlated digital noise compare subjectively with levels of continuous analog noise, and how should the two be combined in a mixed analog/digital network? To assist in answering this question appropriate subjective tests have been carried out by Bell-Northern Research and other organizations[3,4]. In these tests subjects were asked to rate the quality of pre-recorded segments of speech in the presence of various levels of continuous analog noise, and of speech in the presence of various levels of speech-correlated digital noise (again the speech correlated noise was simulated by an MNRU[2]).

The ratings are based on a 5 point scale similar to the one which follows, and can be obtained from the subjects by means of a question such as:

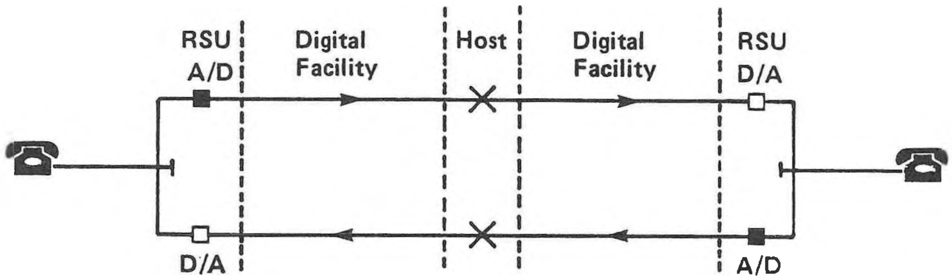
> Which of these five words comes closest to describing the quality of the connection?
>
> - Excellent (E), Good (G), Fair (F), Poor (P), Bad (B).

(The subjective testing experimental arrangement is similar to Figure 8).

Based on these tests the results shown in Figure 4 were obtained which show Mean Opinion Score (MOS) versus speech signal to distortion ratio (S/D) for speech-correlated digital noise, and speech signal to noise ratio (S/N) for continuous analog noise. (Note MOS is a weighted average rating across all test subjects for each test condition; weights of 5, 4, 3, 2 and 1 are assigned to E, G, F, P and B respectively).

Note in Figure 4 that for a given MOS, the S/N ratio for continuous noise must be higher than the S/D ratio for speech-correlated noise, to account for the subjective effect of continuous noise presence regardless of signal level or signal presence. This gives rise to the notion of levels of continuous noise which are subjectively equivalent to levels of speech-correlated noise. This subjective-equivalence can be obtained for various ratings in Figure 4, and is plotted in Figure 5.

(a)



4-Wire Path Established By An RSU–HOST–RSU Connection

(b)

(c)

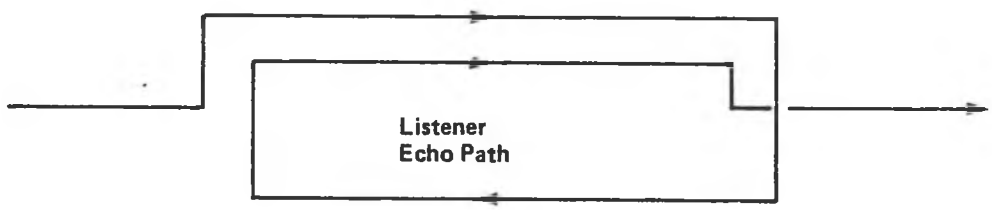


Figure 6 Local Network Connection And Associated Echo Signals

Figure 5 illustrates that equal levels of speech-correlated and continuous noise (S/D = S/N) are not perceived to be subjectively equivalent, but rather the S/N ratio must be significantly higher than the S/D ratio for the same subjective effect. Figure 5 also provides the basis for combining speech-correlated digital and continuous analog noise in a mixed analog/digital network.

3.0 SPECIFIC LISTENING FACTORS AFFECTING THE DISTRIBUTED LOCAL NETWORK

3.1 General

Brief mention of the distributed local network was made earlier in reference to Figure 2. This is a very important contemporary area in the evolution of the telephone network, since the introduction of digital transmission and switching makes it possible to provide local service more economically[7,9]. This is accomplished by converting part of the existing local network to digital transmission which can then support many customers per pair of copper wires. The digital transmission lines are terminated by a digital host switch in the central office and by smaller digital machines called remote switching units (RSU) situated near customer premises, as indicated in Figure 6(a)[10].

Connections of this nature provide highly satisfactory grade-of-service, when engineered in accordance with the considerations discussed presently.

3.2 Echo in the Local Distributed Network

Figure 6(a) illustrates the topology of the local distributed network in which customers are served by means of remote switching units connected to the digital host switch in the central office by digital lines[10,11]. An important parameter is the distance between remote switching units and the host switch, since this distance translates directly into round trip delay, which in turn can lead to a more noticeable echo impairment.

In Figure 6(a) talker A's signal will ordinarily be partially reflected by less than ideal return losses at the four to two-wire interface (at end B of the connection); this gives rise to the talker echo signal in Figure 6(b). If in addition this talker echo signal is again reflected at the four to two-wire interface at end A of the connection, then listener B will also hear an echo of talker A's signal; this gives rise to the listener echo signal in Figure 6(c).

As noted earlier listener echo in a telephone connection is analogous to acoustic room reverberation, and distorts the frequency response of the connection as suggested by the ripple response shown earlier in Figure 1. The subjective effect of this impairment is a potentially bothersome hollow sound not unlike what would be experienced by talking with one's head in a rain barrel.

In what follows emphasis will be placed on an investigation of this impairment based on both an objective laboratory simulation of the appropriate telephone network connections, followed by subjective evaluation of these connections.

From a practical standpoint such an investigation is necessary in order to be able to answer important network planning questions. For example; from a grade-of-service point of view, up to what distance can local network distribution be applied? This is relevant since longer distances correspond to longer round trip delays, which make echo signals more perceptible.

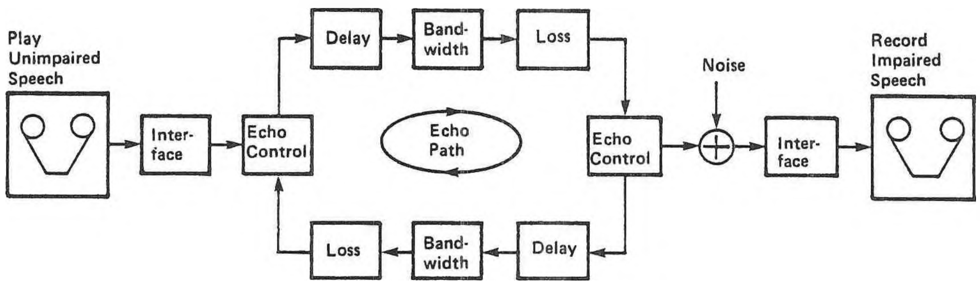


Figure 7 Preparation Of Test Conditions Using Network Simulation Facility

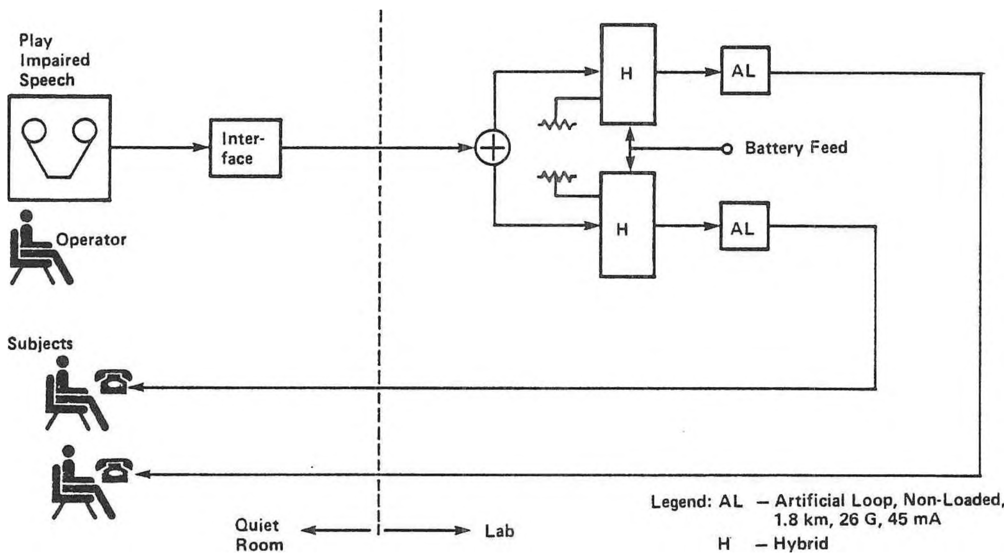


Figure 8 Arrangement For Subjective Tests

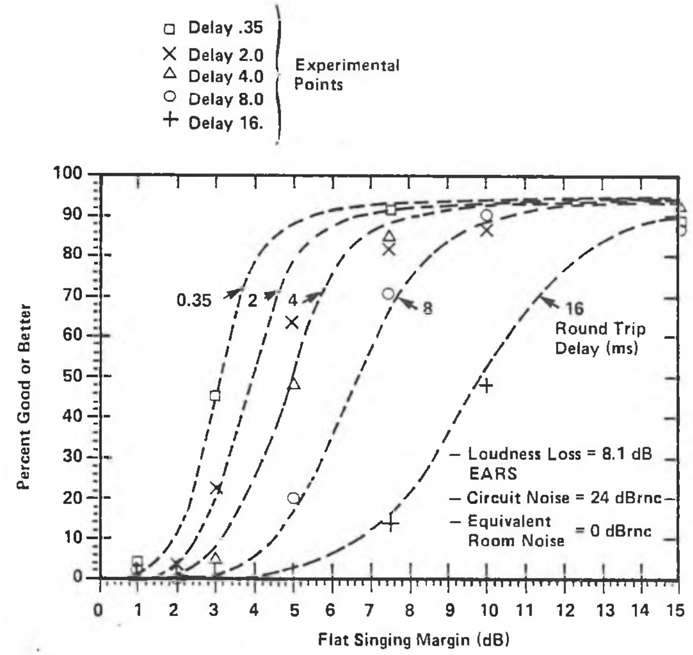


Figure 9 Loss/Noise – Listener Echo Opinion Model (% GOB)

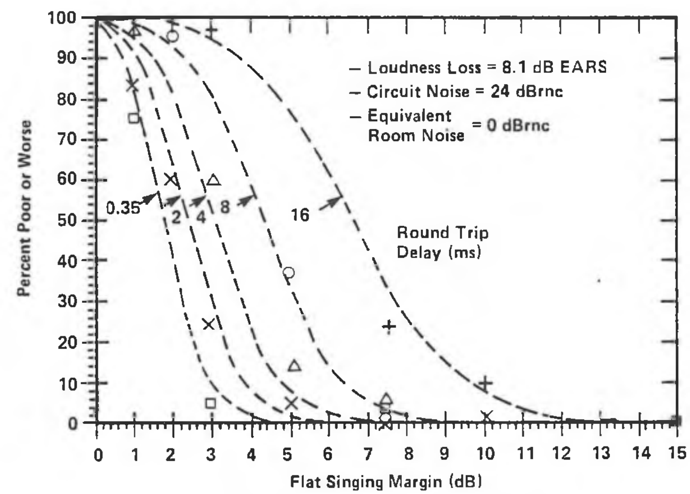


Figure 10 Loss/Noise – Listener Echo Opinion Model (% POW)

3.3 Test Methodology

Bell-Northern Research and other telecommunications organizations in other countries, have the capability of simulating virtually any telephone connection in their laboratories, for the purposes of both objective and subjective evaluation[12]. Figure 7 is one example of a connection which simulates the listener echo impairment of interest here.

The approach in Figure 7 is to play pre-recorded high quality telephone source speech into the simulation facility where various levels of echo, round trip delay and noise are imposed. The resulting impaired speech is then re-recorded and later played to and rated by test subjects; the ratings are based on the previously mentioned five-point scale and are made under controlled acoustical and electrical background noise conditions as shown in Figure 8.

3.4 Test Results

Figures 9 and 10 present the listener echo subjective test results, where on the ordinates the two grade-of-service indicators refer to the percent of subjects rating the connections good or better (% GOB), and poor or worse (%POW) respectively. In each figure the abscissa is the flat singing margin parameter which is the measure of uniform echo path loss around the four-wire path in Figure 7. (When this loss is 0 dB, the positive feedback nature of the echo path causes the circuit to oscillate or "sing"). Flat singing margin corresponds to a measure of echo attenuation. Thus a low singing margin implies high echo energy which from Figures 9 and 10 is rated very poorly; on the otherhand a high singing margin (high echo attenuation) corresponds to low echo energy which is rated much more favourably. For high enough singing margins the grade-of-service levels off at a satisfactory level governed by the background loudness loss and circuit noise conditions. The parameter in both Figures 9 and 10 is the round trip delay. Notice that grade-of-service deteriorates as the delay is increased.

Figures 9 and 10 provide the basis for answering certain network planning questions such as the one noted earlier pertaining to permissible remoting distances. For example, it is known that a high percentage of local telephone network connections (of the type illustrated earlier in Figure 6(a)) can provide singing margins of 8 to 10 dB or higher[13]. Under these circumstances Figures 9 and 10 indicate that listener echo will basically not be perceived as long as round trip delays are 4 ms or less. Thus with a knowledge of switching, processing and transmission delay performance of the connection, the above 4 ms delay figure can be translated directly into a permissible remoting distance. This indicates that distances of up to 80 km are possible between a host switch and a remote switching unit.

4.0 CONCLUSIONS

This paper has examined a number of electro-acoustic factors having a bearing on telephone network performance. Emphasis is placed on various listening factors such as continuous analog and speech-correlated digital noise, and listener echo induced frequency response effects.

In particular it is demonstrated that appropriate subjective testing methodologies and evaluations can answer planning and performance questions related to the contemporary mixed analog and digital evolutionary phase of the telephone network. For example in this paper the specific issues addressed were: tandeming of encoding/decoding processes, combining continuous analog and correlated digital noise, and determining local network remoting distances in a listener echo environment.

A thorough understanding of electro-acoustic factors affecting present day telephone speech quality will enable the network to evolve smoothly from the all analog to all digital stage of the future.

5.0 ACKNOWLEDGEMENTS

Some of the ideas included in the introductory part of this paper were suggested by P. Coverdale. The listener echo test results reported herein were contributed to by human factors consultations from J. Clegg, subjective testing support from N. Berry, as well as statistical analysis and grade-of-service modelling from M. Doughty.

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MUSICAL RULES AND PITCH JUDGMENT

by

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ABSTRACT

Observed psychoacoustic events are often incompatible with musical knowledge. The point has been made particularly in the realm of pitch theory. This paper will contend that it is not the case that musical knowledge does not apply to psychophysical tasks or that musical knowledge fails to correspond with musical practice. Rather, the judgment of pitch is the result of a complex decision involving both the identification of pitch pattern and the operation of a rule-based structural system assigning the functional values of pitches. Different tasks or task demands may assign different relative weights to the system's operators. Whatever the aesthetic merits of Western-European tonality, the experimental study of it uncovers a powerful system of pitch organization and reveals the delicate interplay between perception and cognition. Examples will be drawn from our work on interval and melody recognition.

My interest in this problem--musical rules and pitch judgment--stems not from a primary orientation towards the study of music and musical performance but from a general interest in how psychological systems pick up pattern information in their environment. That is to say, how do systems respond to periodicities or, in a more general sense, repetitive patterns in the real world? Considerable progress has been made in auditory and visual research towards the identification of pattern analyzers. In the realm of pitch perception, for example, both theory and data support the notion of periodicity detectors that respond to the temporal pattern contained by a complex waveform. However, studies of pitch memory have tended not to provide such evidence. When we examine the ability to recognize or to

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remember a sequence of tones, we encounter a number of paradoxes where findings from psychoacoustic research do not correspond with musical intuition and knowledge. The situation is surprising, because musical intuition and knowledge would seem to be an ideal source of evidence of our ability to pick up pattern invariants such as the equivalence of frequency ratio or musical intervals. Musical knowledge in our Western-European tradition includes a well-defined hierarchy of tones, chords, scales, and octaves. It suggests that the apprehension of this hierarchy is the basis of the perception of melody and of harmony. Early psychoacoustic experiments, on the other hand, gave quite a different picture of how the auditory system responds to tone sequences.

The lack of correspondence between the psychophysical mel scale and the musical scale has long been a subject of comment and debate. The mel scale does not preserve constancy of musical interval and with the mel scale it is not possible to construct harmonic music (Ward, 1970). But apart from the psychophysical scaling techniques there are other paradigms that also yield findings at odds with musical expectations derived from musical intuition, experiences or exposure--for example:

1. the absolute judgment of pitch--studies that led to the estimate of the channel capacity for pitch at about 2 bits (Miller, 1956; Pollack, 1952, 1953);

2. the delayed comparison of pitch (Bull & Cuddy, 1972; Wickelgren, 1966, 1969);

3. short-term memory for pitch (Deutsch, 1975; Massaro, 1975).

The overall picture portrayed by psychophysical studies is one of a system with exquisite sensitivity for temporal microstructure but with only a very limited capacity to preserve this information in memory--at any rate, little capacity to handle the richness of musical pattern.

When I review the problem in classes in psychology or in musical acoustics, a typical comment from the students is that the difference between psychophysical studies and musical knowledge must reflect the presence of sampling bias. That is to say, the psychophysical studies probably deal with listeners with no musical background, while musical knowledge is the product of an elite, esoteric, and possibly not entirely consistent set of theories developed by music theorists. Our experimental research has shown that sampling bias--musicians versus non-musicians--is only part of the problem. Another important and equally critical aspect of the problem involves a description of the pattern information available in the stimulus array itself (Jones, 1978). This notion was pointed out by Attneave and Olson in 1971 when they argued that even untrained listeners will respond to the logarithmic structure of a "meaningful" pitch pattern. So one task that we have set for our research is the definition of a meaningful pitch pattern.

Our experiments have looked at both the role of musical experience and the role of objective structure in determining the response to a tone sequence. The experiments tend to fall in one of two classes. The first class that I shall describe is one in which the structure of the tone sequence is apprehended or picked up across all levels of musical experience that we have studied. The levels of experience range from that of the professional musician to that of the average college undergraduate with no more than a year or two of music lessons that he or she would rather forget. In these experiments we compose short tone sequences; to describe these sequences we use the musical notions of diatonicism--membership in a scalar set of tones--and harmonic progression--lawful rules describing the order of the sequencing of harmonic units. Our typical procedure is to compose a sequence that is highly structured, highly tonal, according to musical rules, and then to generate a string of alterations of the sequence in which the rules are gradually relaxed, ignored, or violated. An example of such a set of sequences is given in Figure 1.

Figure 1 shows a set of 32 tone sequences in musical notation. They were presented in random order to two groups of 60 listeners each. One group was highly trained in music performance and theory, the other group was moderately trained. The tones were pure tones generated by a General Radio frequency synthesizer under computer control and the rate of presentation was 500 msec per tone. Each listener was asked to rate the perceived structure of the sequence on a 6-point scale, with 6 representing the upper end of the structure scale. The main points to be noted in Figure 1 are as follows (further details are available in Cuddy, Cohen & Mewhort (1981)). First, the sequences are listed in order of subjective ratings provided by the highly trained group of listeners. The sequence at the top of the left-hand column was given the highest rating of structure on the 6-point scale, followed by the sequences, in order, down the column and then the sequences, in order, down the right-hand column. For each sequence, the two numbers separated by a slash represent the mean ratings from the two separate groups of listeners. Second, the sequence accorded the highest structural rating is a diatonic sequence with the I-V-I harmonic progression, beginning and ending on the tonic, and with the leading-note-to-tonic ending. This type of construction, we have repeatedly found, leads to a perceptual judgment of a highly structured, highly organized tone sequence. Third, harmonic analyses of three professional music theorists are also given. As the sequences descend in rating of perceived structure there is an increasing divergence among the analysts.

Figure 2 shows a cross-classification of sequences derived from study of the results of the rating experiment. The rows of the figure represent levels of harmonic structure, going from a strong diatonic I-V-I progression in level 1 to multiple violations of the musical rules in level 5. The columns represent additional dimensions that we and others (e.g., Dowling, 1978) have found to be important determinants of psychological structure--contour, or the pattern of ups and downs, and excursion, or the distance in semits between the first and the last note. Annabel Cohen and I have recently reported a recognition study using these sequences (Cuddy et al., 1981). In this study each trial consisted of a standard sequence, randomly selected from the set of Figure 2, immediately followed by two transpositions of the sequence. One transposition was correct, the other contained an alteration of one note by plus or minus one semit. Here the task was to recognize the correct transposition. Other tasks that we have studied with these or similar sequences involve perceived ratings of structure, detection of a mistuning within the sequence and, finally, ratings of preference or "pleasingness" of the sequence. For all of these paradigms there is a common finding: the level of musical structure as defined by the key membership of the notes and the harmonic progression of the notes is a critical determinant of performance that holds across a variety of levels of musical training.

So far I have characterized musical structure in terms of scalar and serial rules. In a second class of experiments, the serial rules are dropped though scalar rules may be available. The absolute judgment paradigm is an example of such a case. The frequencies for the absolute judgment task may be selected without regard for correspondence on the musical keyboard, or they may be selected in accordance with a specific musical alphabet, e.g., the triad. No temporal constraints, however, are involved. The performance of the musician benefits from the presence of scalar structure in the tone-generating set, but that of the relatively untrained listener does not (Cuddy, 1971). In a current experiment we have come to the same conclusion for a task in which the listener must identify a randomly-ordered sequence of tones as being generated from either the major or minor pentatonic mode. Again we have an instance in which temporal constraints provide no information. To solve the task the listener must abstract the underlying scalar alphabet of the sequence; at least a moderate degree of training seems necessary to do this.

The conclusion to be derived from the above is that whether or not the tonal properties of a array are apprehended will depend in part upon the training (or predisposition) of the listener but also to a very large part on the nature of the task itself. Tonal structure, or what Bamberger (1978) would call the formal systematic framework of a tone set, is perceived by relatively untrained listeners provided the array contains temporal cues. Temporal cues doubtless include the order and direction of pitch movements as well as the subjective rhythmic groupings induced by a fairly rapid presentation rate. They assist the search for the tonic or fixed reference point against which individual pitches are compared and then assigned functional roles.

Exactly how temporal constraints or expectancies operate to facilitate the abstraction of tonal structure is not yet understood, nor is the question of how the trained musician operates in their absence fully answered. In the latter instance we suspect that the structures have been sufficiently internalized (in an imaginal manner) so that referents and anchors are available without need of their physical presence. But an important research issue concerns the interplay of pitch movement in real time with the abstract relations of tonality and the relative weight attached to each in the processing of melody.

There are several psychological benefits to be derived from the appreciation of tonal structures. One is that by this means our appallingly limited memory for frequency information is circumvented and overcome. Highly structured tonal sequences continually strengthen the sense of tonic, e.g., by repeatedly returning to the tonic at predictable time intervals, or by repeating the periodic structures related to the tonic. In a tonal sense, the sequence doh-mi- soh-doh is an embellishment of a single note, the low doh. Its pitch is not forgotten.

Second, in the presence of tonal structure, pitch discrimination is enhanced. We have known for a long time in psychophysics that discrimination between two frequencies is more accurate when tested with a fixed standard tone than when tested with a series of roving standard tones. Presumably the fixed standard becomes a perceptual anchor stored in some kind of permanent memory. Tonal structures make available a fixed referent system and discrimination may be performed with relation to this system. An implication for music listening is that in the presence of tonal structure one may have greater acuity for fine differences, alterations are clearer, and possibly enjoyment of a finely rendered piece is enhanced.

The aesthetic merits of Western-European tonality are of course a subject of contemporary debate. The psychological import of our research is that tonal systems reflect a powerful means of organizing pitch in memory so that patterns or regularities can be detected. Our work cannot say whether these abilities are innate or developed through exposure to everyday sounds. Research shows that there are clearly developmental trends (Gardner, Davidson & McKernan, in press; Serafine, 1980). But we do know that formal training in music is not necessary for the apprehension of tonal form if the latter is contained in the temporal properties of the objective sound pattern. For psychoacoustical research in our culture, a "meaningful" pattern may be defined as one that reduces easily to simple harmonic progressions. It is a moot point--but one of considerable relevance to contemporary composition--whether we can over-ride or default a tonal system once it is present.

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NOTES ON DEMONSTRATIONS

Example 1.

The sequences of Figure 2 are presented column by column. Note that the coherence of the first sequence of each column is gradually broken down as the musical rules are weakened, then violated.

The study employed pure tones but the effect of structure is easily demonstrated with a musical instrument.

Example 2.

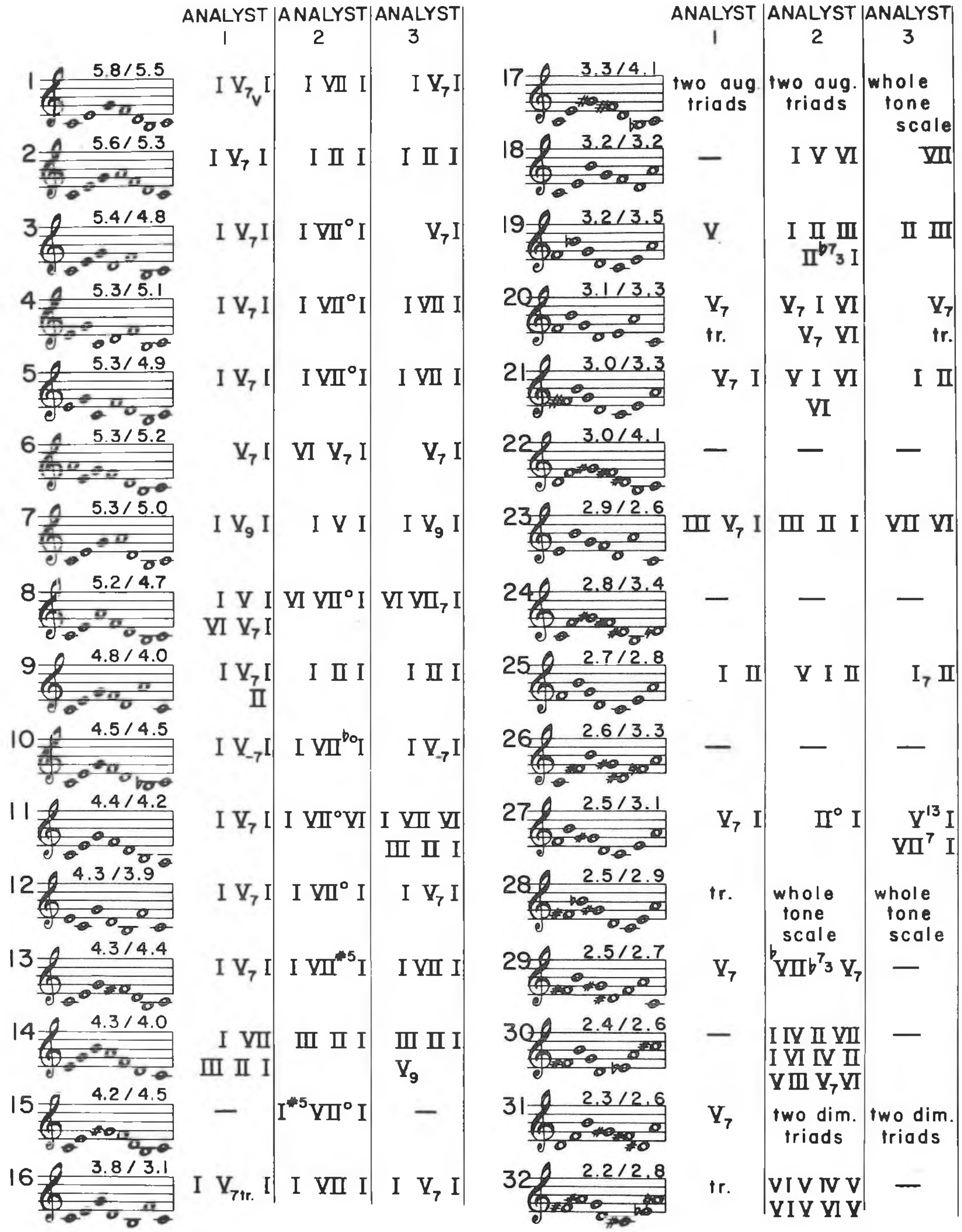
Sequences with the final note mistuned sharp by half a semit. Four sequences of low structure are presented first. They are the sequences of level S_5 of Figure 2 except that the penultimate note is raised from B flat to B. Sequences are presented across the row of level S_5. This is followed by four sequences of high structure--the sequences of level S_1 going across the top row. Note that the mistuning is more apparent with the highly structured sequences. (The alteration of sequences at level S_5 was done to produce an interval at the end identical to that contained at level S_1. The detection of mistuning therefore involves the same interval in both cases.)

Example 3.

Ten sequences are presented; first five with simple contour (three directional changes) then five with a more complex contour (five directional changes of pitch). Within each set of five there is a gradual degradation of structure, but note that structure deteriorates more rapidly than is the case for the levels of Figure 2. In the present example, the second and third sequence of each set are nominally diatonic but contain an unlikely progression VI-I. They may be considered modulating sequences, and in a recognition test proved to be as difficult as non-diatonic sequences. Such sequences provide examples of cases in which the search for a fixed referent is hampered by an unusual order of the tones (Cuddy & Lyons, unpublished manuscript available from the authors).

Note

The above examples accompanied the presentation and are available on tape. Contact the author for further details.



NOTE: tr.: tritone
— no analysis given
aug.: augmented chord
dim.: diminished chord

FIGURE 1: Thirty-two tone sequences in decreasing order of perceived structure as determined by listener ratings. Harmonic analyses of three professional music theorists are also given (from Cuddy, Cohen, and Mewhort, 1981).

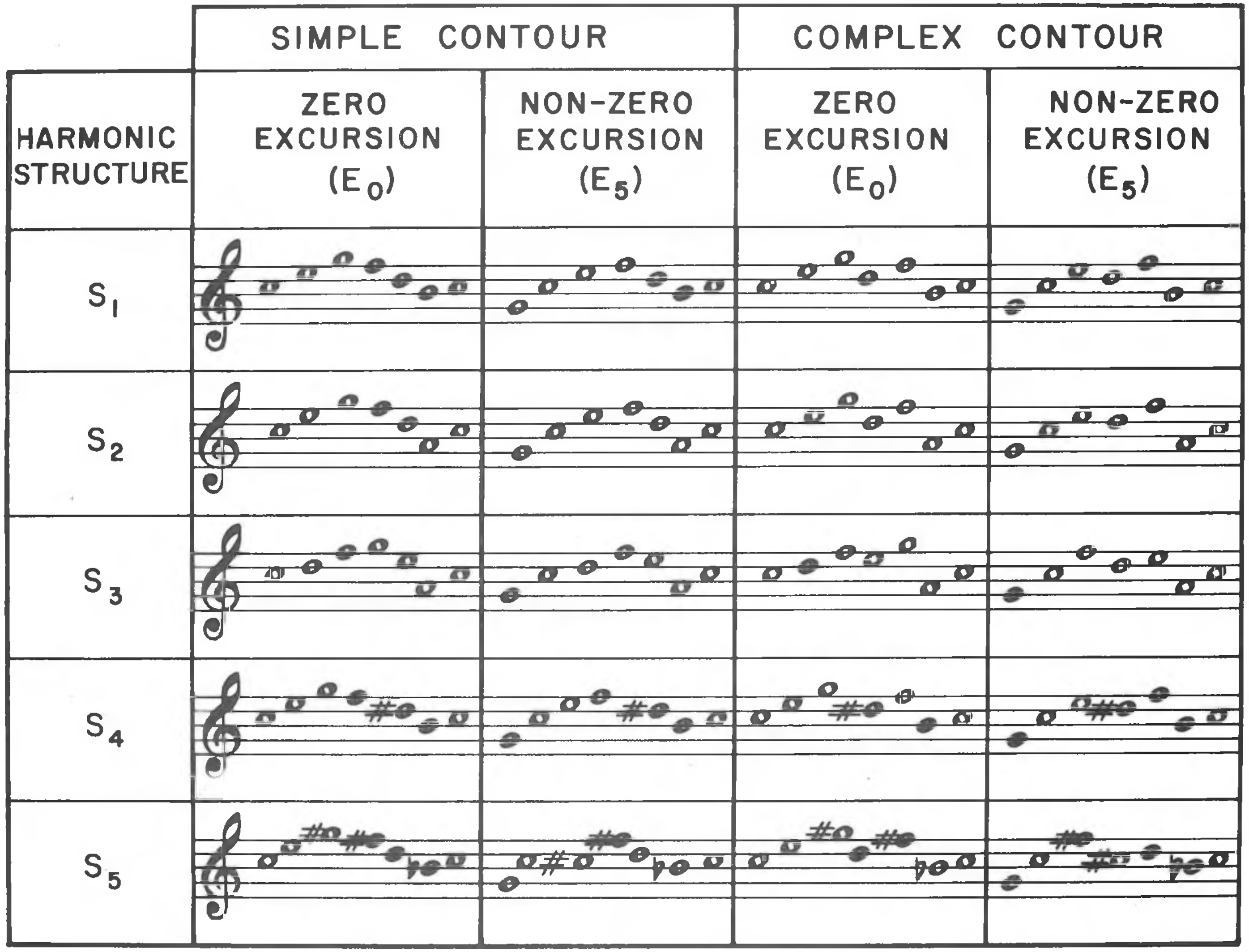


FIGURE 2: A cross classification of tone sequences according to harmonic structure, contour and excursion (from Cuddy, Cohen, and Mewhort, 1981).

"LE BRUIT ET SON CONTROLE"

JEUDI le 18 mars 1982, UNIVERSITE DE SHERBROOKE, GENIE MECANIQUE

PROGRAMME

9:15-10:30 : ACOUSTIQUE AUDIOLOGIQUE par Raymond Hétu, acousticien orthophoniste
Université de Montréal

L'oreille, risques de surdité, effets à long et court terme, effets extra-auditifs, gêne, nuisance, normes et règlements en vigueur, protecteurs auditifs.

10:30-10:45 : Pause-café

10:45-12:00 : PROPAGATION ACOUSTIQUE par Jean Nicolas, professeur-ingénieur
Université de Sherbrooke

Son, caractéristiques des ondes sonores, champs acoustiques, réflexion, absorption, transmission, diffraction, paramètres et techniques de mesure (pression, puissance, intensité), contenu fréquentiel.

13:30-15:00 : REDUCTION DU BRUIT, LES MOYENS CLASSIQUES par Jean Nicolas, professeur-ingénieur, Université de Sherbrooke

Par absorption, par transmission, par diffraction, par filtres réactifs et dissipatifs, réduction des impacts et de l'énergie irradiée par les structures, réduction des vibrations. Démonstrations expérimentales.

15:00-15:15 : Pause-café

15:15-16:00 : EXEMPLES DE REDUCTION DU BRUIT par Yvan Champoux, ingénieur "Control Acoustik"

- Les moteurs électriques
- Problèmes posés par les enceintes acoustiques

16:00-17:00 : VISITE DES LABORATOIRES DE L'UNIVERSITE DE SHERBROOKE
Démonstrations expérimentales

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Dans le cadre d'échange de publications avec la France, nous recevons deux publications: le bulletin du CIDB (Centre d'Information et de Documentation sur le Bruit) - Neuilly-sur-Seine,,et "Recherches/Acoustique" du Centre National d'Etudes des Télécommunications (CNET) - Lanion.

Le bulletin du CIDB est publié tous les deux mois. Une partie de ce bulletin est consacrée aux documents sélectionnés et analysés par des spécialistes dans plusieurs domaines de l'acoustique, notamment: Acoustique théorique, Législation-Politique-Economie, Mesure du bruit et appareillages, Bruit urbain - Bruit des véhicules, Bruit dans le bâtiment, Bruit d'avions, Bruit industriel, Aspect médical, Isolation et prévention et Divers. Nous avons reçu jusqu'à maintenant six bulletins (no. 17 à no. 23).

En ce qui concerne la publication "Recherches/Acoustique" du CNET, on y trouve des articles de recherches publiés par la Division Traitement du Signal de Parole et Services. La table des matières du dernier volume reçu (vol. VI 1979/1981) est la suivante:

Si vous êtes intéressé à consulter une de ces publications, veuillez vous adresser à:

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Aviation Noise and Vancouver International Airport	Transport Canada Civil Aviation Planning 739 West Hastings Street Vancouver, British Columbia V6C 1A2 A.D

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D.A. Benwell and M.H. Repacholi Noise Hazard and Control	Information Directorate Department of National Health and Welfare Brooke Claxton Building, 5th Floor Ottawa, Ontario, K1A 0K9 79-EHD-29, March 1979 A.D
D.A. Benwell Calibration and Evaluation of Audiometers in Environmental Health Directorate Noise Chamber	Information Directorate Department of National Health and Welfare Brooke Claxton Building, 5th Floor Ottawa, Ontario, K1A 0K9 80-EHD-64 B.D
D.A. Benwell and M.H. Repacholi Safety Code-23. Guidelines for the Safe Use of Ultrasound. Part I. Medical and Paramedical Applications.	Information Directorate Department of National Health and Welfare Brooke Claxton Building, 5th Floor Ottawa, Ontario, K1A 0K9 80-EHD-59 A.D
D.A. Benwell and M.H. Repacholi Safety Code-24. Guidelines for the Safe Use of Ultrasound. Part II. Industrial and Commercial Applications.	Information Directorate Department of National Health and Welfare Brooke Claxton Building, 5th Floor Ottawa, Ontario, K1A 0K9. 80-EHD-60 A.D.

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