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Volume 2
CONTRIBUTED PAPERS
FOREWORD

The 10th International Congress on Acoustics, held under the auspices of the International Commission on Acoustics (ICA), is jointly sponsored by the International Union of Pure and Applied Physics of UNESCO and the Australian Acoustical Society. The Congress theme is "Acoustics in the 1980s" and the series of invited and contributed papers present an opportunity to examine current activities and new developments in all branches of acoustics.

Volumes II and III of the Congress publications include over 600 abstracts of the contributed papers published as submitted by the authors. The ten invited lectures suggested by the ICA are published in full version in Volume I.

The full version of the papers presented at the Satellite Symposia on "Engineering for Noise Control" in Adelaide and "Basic Causes of Noise Deafness" in Perth are also published in separate volumes.

In accordance with the directions of the ICA, one page only was allotted to each abstract. During the Congress registered participants will be able to buy copies of the full version of any papers provided by the authors.

J.A. ROSE
Chairman
Congress Executive Committee
CONTENTS

PART ONE

A. SPEECH COMMUNICATION
B. PHYSIOLOGICAL AND PSYCHOLOGICAL ACOUSTICS
C. NOISE: ITS EFFECTS AND CONTROL
D. SHOCK AND VIBRATION

PART TWO

E. ARCHITECTURAL AND BUILDING ACOUSTICS
F. BIOACOUSTICS
G. ULTRASONICS, QUANTUM ACOUSTICS AND PHYSICAL EFFECTS OF SOUND
H. UNDERWATER SOUND
I. PHYSICAL ACOUSTICS
J. AEROACOUSTICS, ATMOSPHERIC SOUND
K. MUSIC AND MUSICAL INSTRUMENTS
L. TRANSDUCTION: ACOUSTICAL DEVICES FOR THE GENERATION AND REPRODUCTION OF SOUND
M. ACOUSTICAL MEASUREMENTS AND INSTRUMENTATION: SIGNAL PROCESSING: STATISTICAL METHODS IN ACOUSTICS
N. OTHER ACOUSTIC TOPICS
A. SPEECH COMMUNICATION
DATA ON THE GLOTTAL VOICE SOURCE BEHAVIOUR IN VOWEL PRODUCTION

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The relationships between waveform and spectrum of the glottal volume velocity voice source have recently been clarified in certain respects (Sundberg & Gauffin 1979; Fant 1979). Two waveform characteristics have been found to determine important spectral characteristics: (a) the maximum amplitude, \( U_g \), determines the amplitude of the source spectrum fundamental; and (b) the maximum value of the derivative, \( \text{Max} \, \frac{dU_g}{dt} \), determines the amplitudes of the source spectrum overtones. The main purpose of the present investigation was to explore the phonatory variability in singers and untrained voices by studying how \( U_g \) and \( \text{Max} \, \frac{dU_g}{dt} \) are influenced by changes of vocal effort and pitch, and how \( \text{Max} \, \frac{dU_g}{dt} \) correlates with spectral amplitude in various frequency bands under conditions of varied vocal effort. Sustained vowel sounds produced at several pitches and degrees of vocal effort are studied by means of inverse filtering and spectral analysis. The results are relevant to speech and singing synthesis and contribute to a phonatory understanding of long time average spectrum data of different voices.

References:


A CORRELATION ANALYSIS OF EMG ACTIVITY AND THE MOVEMENT OF SELECTED SPEECH ORGANS

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For the study of the temporal aspects of the speech production process, it is necessary to investigate the pattern of the motor control signals and the dynamic characteristics of the speech organs which act in response to the control signals. The present study is an attempt to analyze the relationship between the movements of selected speech organs recorded by our x-ray microbeam system and the EMG activity of the related speech muscles. In this paper, the results of an analysis of velar movement will be presented as an example, in reference to the pattern of the EMG activity of the levator palatini muscle.

DATA RECORDING AND ANALYSIS

The subject read a list of test words containing several types of nasal sounds in Japanese. For recording velar movement, a strip of thin film with a lead pellet attached to its end was passed through a nostril, placing the pellet on the velum. The pellet movement was recorded by the x-ray microbeam system at a rate of 190 frames/sec. Hooked-wire electrodes were inserted into the levator palatini muscle and the EMG signal was recorded together with the frame pulses of the x-ray tracking. The EMG signal was then digitized and absolute values were taken and integrated over each time frame. The Y-coordinate of the pellet was selected to represent the movement of the velum. It was assumed that EMG signal at a given time can be given as

\[ \hat{E}_i = c_0 + c_1 y_i + c_2 \dot{y}_i + c_3 \ddot{y}_i \]

where the subscript \( i \) denotes the \( i \)-th time sample. The coefficients \( c_i \)'s which give the best approximation were determined by minimizing the error

\[ \text{Err} = \sum (\hat{E}_i - \hat{E}_i)^2 \]

where \( E_i \) is the time sample of the observed EMG.

RESULTS

The characteristic constants of the linear second order system were calculated from the estimated values of \( c_i \)'s. The value of the damping factor was found to be close to 1, indicating that the second order system is nearly critically damped. The characteristic time constant was approximately 70msec, and was considerably greater than that generally accepted for the vowel to vowel transition. In view of the estimated value of the time constant, the pattern of the velar movement for /m/ can be taken as ballistic (impulse response-like), because the duration of the suppression of the levator activity is relatively short. The nasalization of the vowel between the nasal sounds (in such a word as /memee/) can be interpreted as the result of a mechano-inertial effect. Similar analysis pertaining to the other speech organs are now being considered.

A digital technique known as linear predictive coding has been used to generate a sequence of gain versus frequency spectral envelopes for the vocal tract during continuous speech, with emphasis on the spectral range up to 3000 Hz. Changing vocal tract shapes during speech causes changes in the resonant frequencies (known as formants). The centre frequencies of the formants have been tracked as a function of time during a selected sample of continuous speech for a group of six fluent and seven cerebral palsied adults. Comparison of the formant tracks for the two groups has indicated a statistically significant reduction of the spectral range for the second formant (F2) in the group of cerebral palsied speakers.
ELECTROMYOGRAPHIC STUDY OF DYSARTHRIC SPEECH IN CEREBRAL PALSY

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Electromyogram (EMG) recordings from eighteen muscles controlling movement of lips, tongue and jaw have been obtained from seven cerebral palsied dysarthric speakers and six normal fluent speakers during fifty repetitions of a selected passage of speech. Comparisons have been made between subject groups in respect of patterns of EMG activity in speech articulator muscles. This unique data allows a number of hypotheses concerning the neuromuscular basis of speech defects in cerebral palsied subjects to be tested. Results indicate that dysarthric speech in cerebral palsy cannot be attributed to release of infantile reflexes or to spasticity in lip and tongue muscles resulting from hypersensitive tonic stretch reflexes. Reproducibility of abnormal patterns of EMG activity from one repetition of the sentence to the next suggests that lack of intelligibility is not caused by involuntary writhing contractions and facial grimacing commonly observed in athetotic cerebral palsied subjects. Worthy of further consideration is the speculation that disruption of speech in cerebral palsy is a consequence of an inability to plan and communicate appropriate motor commands to speech musculature because of disruption of sensory-motor integration skill.
Electromyographic recordings were obtained from eighteen muscles controlling the lips, tongue and jaw in six normal and seven cerebral palsied subjects. Up to fourteen muscles were recorded simultaneously. As a preliminary step to the investigation of muscle activity in speech, the functions of individual muscles were examined during performance of voluntary (non-speech) gestures which were designed to activate specific muscles. Isolated muscle function proved virtually impossible to elicit. In most gestures a pattern of muscle activity emerged; significant differences in such patterns of activity were observed between normal and cerebral palsied subjects.
THE CONVERSION OF TELETEXT INTO SPEECH

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Research at Cambridge University Engineering Department on speech synthesis has been concerned with providing speech output from a computer-controlled water supply network (1,2). In this system, termed Speech Synthesis from Concept (SSC), data extracted from the database is converted into a sequence of words with pitch and timing information using a pre-defined knowledge of the information domain and a knowledge of simple English syntax.

This work is being extended to cover the conversion of teletext information directly into speech. Teletext is the generic name used to describe a class of systems whereby information is transmitted in digitally coded form, superimposed upon the conventional television signal. A special decoder is then used to display this information on a normal television receiver.

The aim of the research is to eventually provide a spoken version of teletext for the blind and also allow its access over the telephone and radio networks. Teletext information has a number of constraints which makes this aim feasible, the most important being that a relatively fixed and constrained set of sentence structures are employed.

A problem encountered in converting teletext into speech is that graphics characters can also be displayed interspersed within the text. Also entire pages can consist of graphics characters alone. To remove this problem we are initially concentrating on a limited number of pages that carry only textual information. Once a page has been selected, a parse is performed for each sentence on the page. This extracts the syntactic structure of each sentence and is used as the input to the pitch and timing rules. At present we are limited to a fixed vocabulary of words but we will eventually implement a phoneme synthesis program which will remove this constraint and also allow the extension to the conversion of viewdata transmissions.

REFERENCES


ACKNOWLEDGEMENT

This project is supported by a research grant from the Science Research Council, London, who also provide support for one of the authors (RM).
FLEXIBLE SPEECH SYNTHESIZER CONTROLLED BY A MINICOMPUTER

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For a text-to-speech system a flexible speech synthesizer was required. Flexibility should incorporate the possibility to select each of different sets of parameters describing the speech production process. On the one hand the vocal tract area function derived from either an articulatory model or physiological data should be usable, with the possibility of adding a nasal tract to the existing synthesizer. On the other hand the reflection coefficients obtained by a linear prediction analysis of the speech signal should be a possible input to the synthesizer.

The underlying concept for the implementation of the hardware synthesizer was described in [1]. Furthermore the excitation signal for the synthesizer should be arbitrary and changeable in period and amplitude.

![Diagram of the speech synthesizer](image)

Figure 1 shows a block diagram of the implemented speech synthesizer. The voiced excitation source of the generator part is a 256x16 bit RAM, in which an arbitrary discrete time signal - as generated in a master HP MX minicomputer - can be stored. The digital filter is implemented as a multiplexed two-port adaptor, connecting unit-elements of different characteristic resistance [2]. Due to this multiplexing facility the order of the filter can be varied between four and seventeen. To test their influence on the quality of the synthesized speech more variations of parameters are possible. The radiation load and/or the coupling of the excitation source to the filter can be changed. The sampling frequency can be varied from 4 to 17 kHz. The sensitivity of the synthesized speech to all these parameters is being investigated, with the aim of improving naturalness.

LPC vocoder without pitch extractor

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Speech quality of an analysis synthesis system like a vocoder greatly depends on the excitation source of synthesizer. Many efforts have been made on the detection of fundamental frequency Fo and on wave shape of excitation source. However, it is serious problem to detect the precise Fo from speech signal of practical communication circuit. In order to avoid the trouble caused by Fo detection, a constant pitch is sometimes employed in a vocoder of private communication circuit though it gives unnatural sound.

It is well known that amplitude of speech wave is well correlated to Fo. And it is considered that Fo is more closely related to amplitude of glottal wave than to that of speech wave, because the amplitude of glottal wave has not been affected by the resonance characteristics of vocal tract. Fortunately, the amplitude of glottal source is obtained in an LPC vocoder as gain G.

This paper proposes an LPC vocoder without pitch extraction. Impulse train generated from G excites a synthesizer. The outline of original LPC vocoder is as follows: 12 order of k parameter, sampling frequency; 10 kHz, frame period; 10 ms, window length; 25.6 ms, and pitch detection; modified SIPT (window length; 40 ms).

Analysis is carried out on monosyllables, words, phrases uttered by a male, and sentences by 4 males. It is found that Fo is well represented by a linear function of A (=20logG) as shown in Eq.(1). The correlation coefficient r between Fo and A is in the range from 0.7 to 0.95 at monosyllables, and from 0.3 to 0.8 at words or phrases. r of sentences spoken monotonously is as high as 0.8, but by an announcer is almost 0. However, if their relation analyzed word by word or phrase by phrase, considerably high r is obtained.

As regression analysis shows that a is from 0.6 to 2.3 and b from 70 to 100, pulse frequency Go for synthesis is calculated by giving a=1.3 and b=80 in Eq.(1). Now LPC vocoder is excited by k parameters, G, Go and V/UV. All parameters are not quantized and linearly interpolated every 1/Go.

Synthetic sound is heard quite natural in case of monosyllables and words. Synthetic sentences are heard more natural than that by constant pitch, but sometimes, it gives unnatural and exotic intonation. There are some reasons to produce unnatural sound; A changes with change of articulation, Fo is inproportion to A in case of interrogative sentence, and contour of Fo is basically different from contour of A.

This vocoder has many advantages such as simplification of analyzer, trouble free from pitch extraction, saving bit rate for Fo transmission. It will be concluded that present status of LPC vocoder without pitch extraction is acceptable in a private communication for the purpose of transmission of message though this vocoder can not transmit the individuality. It will be a powerful system in a noisy environment. Synthesis by rule of Fo contour from A and information of unvoiced sound will contribute to get more natural sound.
Mid-band speech transmission system combined with ADM and SPAC

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A speech Processing system named SPAC (Speech Processing system by use of AutoCorrelation function) had been developed. SPAC can reduce the noise level duperposed on speech signal and compress or expand the speech spectrum. Fully utilizing these functions of SPAC, Three types of mid-bit-rate transmission system whose bit rate is 8 or 10 kbps are proposed. These are (B) ZEM (clipped speech) + SPAC, (D) ADM + SPAC and (E) SPAC + SPAC. SPAC is used for noise reduction in systems (B) and (D). System (E) employs SPAC as a band-compression and -expansion system (compression ratio: 3). (A) ZEM and (B) ADM are also tested as reference systems.

In this experiment, accumulation time for calculation of short-time autocorrelation function is fixed at 20 ms. The reduction rate of band-limited white gaussian noise by SPAC attains about 16 db. If this noise is processed after clipping (ZEM), the reduction rate reaches as high as 20 db. If a sinusoidal wave embedded in noise (S/N=0 db) is processed by SPAC, S/N is improved by 13 db.

Sentences uttered by four males were processed by all systems. Processed sentences were presented to a pair comparison test. The result is transformed into Thurston psychological scale (Case V) and shown in Fig.1. Speech of (D) was more natural than others.

A syllable articulation test is made on systems (A), (B), (C) and (D) of 8 kbps. The result is shown in Table. Subjective S/N (multiplicative noise) of systems (C) and (D) are evaluated as a function of bit rate. The result is drawn in Fig.2. It is indicated that (D) of x kbps is almost equivalent to (C) of 2x kbps if x is less than 10.

As SPAC, ZEM and ADM are tolerable for considerable bit error, it will be conclusively said that ADM and ZEM combined with SPAC are applicable in a transmission circuit with poor quality or a private communication link. Further experiments are required on the selection of more sophisticated ADM, and the optimization of the system of ADM + SPAC.

Physiological and Acoustic Correlates of Vowel Perception
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A good deal of speculation and some experimentation have been directed at the nature of the presumed complex interrelationships among the articulation, acoustics and perception of speech. We have described some relationships between articulation and its resultant acoustic output in terms of the differing strategies used by two groups of speakers in producing a subset of English vowels (Bell-Berti et al., 1979). We have also described the relationship between the differing articulatory strategies of our subjects and their vowel perception in terms of the degree of phoneme boundary lability in a labeling task which employed both equal probability and anchoring conditions (Bell-Berti et al., in press).

In the present study we examine the remaining, undescribed relationship between the differing acoustic outputs of our subjects and the manner in which those outputs are perceived. Two experiments were performed. In the first, our subjects participated in a vowel identification task designed to test their sensitivity to an acoustic variable which differed from that of the original (anchoring) perception test. The variable chosen was derived from the acoustic and articulatory differences found between the two groups of subjects. In the second experiment we determined whether professionally trained phoneticians were able to discriminate the vowel productions of the two groups of speakers.

The results of these experiments provide further support for the existence, and allow for a more detailed description, of the complex interrelationships generally assumed to obtain among the articulatory, acoustic and perceptual dimensions of speech.

Dynamic articulatory and acoustic properties of lingual fricatives in English.

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The analysis of fricatives in speech production presents certain difficulties both in its acoustic and articulatory aspects. The acoustic spectrum is complex because of the presence of a radical constriction and associated excitation within the supralaryngeal vocal tract, and precise measurement of dynamic articulatory behaviour, particularly tongue activity during fricative production, has a number of technical problems associated with it (Hardcastle, 1974).

Much of the recent literature on fricative production in natural speech is focussed on either articulatory (e.g. Subtelny et al. (1972), Wolf et al. (1976)) or acoustic aspects (e.g. Jassem (1965), Heinz and Stevens (1961)). In an attempt to integrate both these aspects, the present study is an analysis of the acoustic consequences of dynamic lingual articulation for a selected group of English fricatives. Electropalatography and aerodynamic measurements have been employed to estimate the properties of the fricative stricture in relation to accompanying airflow. The spectral characteristics of the acoustic signal have been determined using conventional spectrographic and FFT analyses.

References


On Delimiting the Vowel
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A number of studies have demonstrated the perceptual importance of vowel duration in cueing (1) vowel identity, (2) lexical stress, (3) syntactic boundaries, and (4) final-consonant voicing class. In most cases the critical duration manipulated has been that of the steady-state vowel.

The studies reported here, which employed both synthetic and natural speech stimuli, indicate that the duration of a vowel which serves as a cue in a variety of perceptual tasks includes acoustic information (such as formant transitions) which also convey information about consonant identity. The extent to which a segment of the acoustic signal supplies information about both consonant identity and vowel duration appears to be positively correlated with the degree of coarticulation between consonant and vowel.
CONTEXT EFFECTS IN PHONETIC AND NON-PHONETIC JUDGMENTS OF VOWELS

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The perception of a sound stimulus is influenced by its context. The present study investigates the relationship between the magnitude of the context effect and the time interval (ISI) as well as the temporal order of the context and the target stimuli in the case of stationary vowels.

METHOD

Context effects were measured on two stimulus continua, one phonetic (/u/ vs. /a/) and the other non-phonetic (high vs. low in pitch). Stimuli on the first continuum were generated by varying the first formant frequency from 420 Hz to 600 Hz in seven equal steps, while their fundamental frequency was held constant at 132 Hz. Stimuli on the second continuum were generated by varying the fundamental frequency from 114 Hz to 150 Hz at seven equal steps, while their first formant frequency was held constant at 510 Hz. Frequencies of the higher formants were held constant at 1250, 2750, 3500, and 4500 Hz for all the stimuli. The subjects were four female adults. Category boundaries of individual subjects were first determined by using a randomized sequence of the seven target stimuli. Each target stimulus was then paired with (i.e., preceded or followed by) a context stimulus, and the pairs were randomized. The shift in the category boundary, measured individually for each subject, was used as an index for the magnitude of the context effect. The measurement was made at ISI's of 0.1, 0.2, and 0.5 sec.

RESULTS AND DISCUSSION

The ratio of the shift of category boundary (Δμ) to the standard deviation (σ) of the context-free identification curve can be used for comparing context effects on the two stimulus continua. As shown in Fig. 1, the context effect is generally greater in phonetic judgments than in non-phonetic judgments. Furthermore, the effect of backward context is more contrastive than that of forward context in phonetic judgments, but is less contrastive in non-phonetic judgments. The results can be interpreted as the combination of two contextual influences with different decay rates, possibly associated with pre-categorical and categorical short-term storage systems, respectively.

REFERENCES

ON MINIMAL DURATION OF ISOLATED SPEECH SOUNDS

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We are interested to know what the minimal duration of the initial segment of the speech sound is for its reliable recognition. For the purpose of this investigation the natural isolated speech sounds of the Serbo Croatian language were used to prepare the experimental stimuli. Each speech sound was employed to construct six stimuli whose lengths were one sixth, two sixth, three sixth, four sixth, five sixth and six sixth of the total speech sound duration respectively. The auditory stimuli were presented to the subject whose task was to write down what he believed to have heard. The recognition of stimuli was judged as reliable when 90% or more of responses to that particular stimulus were correct.

The results are summarized in Table 1. Each Serbo Croatian speech sound as written in IPA is associated with the critical length (expressed in milliseconds) of the initial segment that was reliably recognized. The numbers were rounded off to the next zero or five milliseconds.

Table I
Minimal duration of initial segment required for reliable recognition of isolated speech sounds

<table>
<thead>
<tr>
<th>sound in IPA</th>
<th>i</th>
<th>E</th>
<th>a</th>
<th>o</th>
<th>u</th>
<th>p</th>
<th>b</th>
<th>t</th>
<th>d</th>
<th>k</th>
<th>g</th>
<th>ts</th>
<th>tij</th>
<th>tS</th>
</tr>
</thead>
<tbody>
<tr>
<td>duration in ms.</td>
<td>30</td>
<td>20</td>
<td>25</td>
<td>55</td>
<td>30</td>
<td>30</td>
<td>132</td>
<td>55</td>
<td>155</td>
<td>50</td>
<td>120</td>
<td>55</td>
<td>75</td>
<td>35</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>dji</th>
<th>dž</th>
<th>f</th>
<th>v</th>
<th>s</th>
<th>z</th>
<th>š</th>
<th>x</th>
<th>j</th>
<th>r</th>
<th>m</th>
<th>n</th>
<th>nj</th>
<th>l</th>
<th>lj</th>
</tr>
</thead>
<tbody>
<tr>
<td>125</td>
<td>155</td>
<td>145</td>
<td>55</td>
<td>80</td>
<td>30</td>
<td>150</td>
<td>85</td>
<td>120</td>
<td>145</td>
<td>75</td>
<td>65</td>
<td>150</td>
<td>140</td>
<td>70</td>
</tr>
</tbody>
</table>

The results suggest that on the average the lengths of the initial segment required for the reliable recognition is directly proportional to the degree to which the particular speech sound wave is quasi-stationary (within its total length), and to which the articulatory movements can be predicted from the early portion of the speech sound signal.
An Electronic Speech Support System for Laryngectomees

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We shall describe the design principles of a hands-free, variable-pitch, controlled-volume, unobtrusive speech support system for laryngectomees. The scheme is aimed to make best use of residual articulatory faculties of laryngectomees while heavily relying on available digital signal processing techniques and devices. The following novel features characterize our system now under development:

(1) The vocal tract is excited by a miniature, low-power driver affixed inside the oral cavity; the ensuing sound vibrations are sensed by a likewise intraoral miniature acoustic sensor.
(2) Control and information signals to and from the mouth are transmitted by electromagnetic coupling.
(3) Distortions arising from the non-conventional propagation pattern in the tract are adaptively compensated for by a subsequent signal processor.
(4) The actual speech sounds are radiated, after amplification, from a concealed loudspeaker worn on the user's chest.
(5) Intonation and dynamics are regulated by the speaker via the rate of exhalation, which is sensed by a miniature device positioned near the stoma.

The presentation will concentrate on several experimental and theoretical results which provide the basis for the operation of the system:

(a) Illustrations of the nature of power spectral distortion resulting from arbitrary placing of source and sensor in the vocal tract, as assessed by computer simulation and by measurements on a multi-segment acoustic tube laboratory model of the vocal tract.
(b) Statistical dependence between instantaneous pitch and intensity in natural speech.
(c) Breath control of Fo and intensity of synthesized speech.
(d) Effects of the fine structure of Fo variations on the naturalness of partially or entirely synthetic speech.

This project is supported by Grant # PDT-113 of the American Cancer Society.
Study of Speech Perception is an area of multi-disciplinary interest and the use of the Speech segmenter in experiments dealing with speech perception is fairly of recent origin. In the present investigation speech perception of vowels is studied. Fifteen token words in /h-d/ frame were recorded in a random order and presented to 4 subjects for identification of the sounds heard. The data was analysed and discussed. Results of the present study suggest that length is a cue of varying importance in the identification of certain Australian English sounds. There were a fewer correct identifications made than expected. Results involving the gross identifications of /a/S (short) heard as /l/V (long) or vice versa and errors involved with /i/ and /i/.
DISCRIMINATION OF JAPANESE SYLLABLES BY SPANISH LISTENERS
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In a previous work we have pointed out the spectral and phonetic coincidence between the five Japanese and Spanish vowels /i, e, a, u, o/. We have also anticipated a close phonetic approximation between syllables of both languages (Guirao, 1978). In an attempt to further inquire on this problem, we have extended our study to consonantal phonemes common to both systems, presented in CV, CVV, V and VV syllables. Similarities and differences were determined by using perceptual tests.

Perceptual Tests
One hundred Japanese syllables pronounced by a Japanese talker were recorded and presented to ten Spanish listeners. Seventy syllables represented nineteen consonants with vowel (CV); twenty-one combined seven consonants with /ia/, /iu/, /io/ (CVV); the five vowels were presented alone as V syllables; finally, three syllables were a combination of /i/ with vowels /a, o, u/, and together with syllable /wa/ constituted a final group VV. Subjects were instructed to listen to each sound and then to write the syllable using their own alphabetic symbols.

Vowels and the consonantal portion of each syllable were grouped in Table I. This preliminary classification follows a criterion used before for Spanish phonemes (Guirao, 1979).

<table>
<thead>
<tr>
<th>PERIODIC SOUNDS</th>
<th>NOISF.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Energy Concentration</td>
<td>Bursts</td>
</tr>
<tr>
<td>High</td>
<td>/p/ 97.5</td>
</tr>
<tr>
<td>Low</td>
<td>/t/ 80.0</td>
</tr>
<tr>
<td>/s/</td>
<td>/k/ 88.7</td>
</tr>
<tr>
<td>/r/</td>
<td>/b/ 95.0</td>
</tr>
<tr>
<td>/l/</td>
<td>/d/ 100.0</td>
</tr>
<tr>
<td>/g/</td>
<td>/y/ 77.5</td>
</tr>
</tbody>
</table>

A Remarkable Similarity
As it is shown in the Table, listeners gave fairly accurate percentages of identification. Sounds labelled as periodic and bursts of noise were in close coincidence with the data obtained for Spanish sounds of similar spectrum (Massone & Guirao, 1980). Three of the five friction-noise sounds /s/, /x/ and /s/ gave also a high score of recognition. The other two obtained lower rates. Two of the three phonemes non-existent in Spanish were also discriminated; percentages were /s/ 77.5 and /ts/ 70. This sound was interpreted as a fusion of /t/ with /s/. The sound /z/ gave a 50% of recognition. In general, Spanish listeners assimilated quite easily the Japanese phonemes to their own linguistic system.

Other than vowel and syllabic systems, both languages have a similar syllabic rhythm. A possible influence of this factor on syllabic phonetic similarity is suggested.

References
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THE RELATIONSHIP BETWEEN THE SPECTRUM OF THE SIGNAL AND MASKER, AND ERRORS MADE IN A SPEECH INTELLIGIBILITY TEST IN SPEECH SPECTRUM NOISE.

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In the course of evaluating a multi-band compression-expansion hearing aid system, consonant-vowel and vowel-consonant type nonsense syllables imbedded in speech spectrum shaped noise were presented to a number of hearing impaired subjects. The nonsense syllables were presented to each subject under several different experimental conditions involving variations in the computer processing applied to the signal. The nonsense syllables were an Australian voice version of the test developed by Resnick & Levitt (1978). The test consisted of seven lists of nonsense syllables. Within each list items differed from each other only in the initial or final consonant. All 55 items in the test were presented to each subject on each trial. A closed response set, forced choice format was used in which the subject was required to select the item heard from among all of the items in that particular list.

None of the non-linear amplification patterns tested resulted in a significant improvement in intelligibility scores over the best linear system, but in the course of analysing the results a confusion analysis was performed and this revealed some interesting trends. The short term spectra of the signals were utilized to try to isolate the reasons for the errors which occurred. Short term spectra were measured using a 50ms time window centred on the consonant in question. The resulting spectra were classified in several ways: (i) High intensity versus medium intensity versus low intensity. (ii) Long duration versus short duration. (iii) Medium frequency emphasis versus high frequency emphasis versus flat spectral shape.

An explanation of the perceptual confusions could be made on the following basis: (i) High intensity consonants were always perceived correctly. (ii) Medium intensity consonants were mostly perceived correctly. Incorrect responses were always confusions between consonants of the same duration. (iii) Low intensity consonants were perceived poorly. Stimuli incorrectly chosen tended to have spectral and temporal properties similar to that of the masker.

Thus it appears that when the energy in the consonant was not clearly heard, a section of the masker was able to impersonate the perceptually missing consonant. If this was the mechanism, then it is apparent that the properties of the masker will play an important role in the conclusions drawn from intelligibility tests of this type.
PERCEPTUAL DIFFERENCES AMONG SPANISH CONSONANTS
Laboratorio de Investigaciones Sensoriales, CONICET

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The present work was planned to compare the perceptual differences among Spanish consonants in terms of their psychoacoustical features as described in a previous paper (Guirao, 1979). Perceptual differences were determined by identification tests with the stimuli and responses cast into a confusion matrix.

Ten subjects were instructed to identify the initial portion of CV syllables and the final portion of VC syllables paired with the five Spanish vowels. Stimuli were presented under Forward (played forward) and Backward (played backward) conditions. The sample included all the Spanish consonants. When the data were processed, these sounds were divided in three groups: periodic sounds: /n, l, m, n, r/; bands of noise: /s, f, x, ,rr, t /; and bursts of noise: /p, t, k, b, d, g/.

Forward Condition
Perceptual confusions were not significant under Forward condition. Almost all sounds were identified in the initial position, while in the final position rates were somewhat lower (87%).

Backward Condition
Perceptual confusions were observed for periodic sounds and bursts of noise (30% and 59% of misidentification, respectively). Bands of noise, expect /t / (94%), nearly the same identification score as in Forward condition (96%). VC syllables presented higher scores of identification than CV syllables.

Both experimental conditions showed that bursts of noise and /n, r, t / were not heard as mirror-image acoustic patterns. The rest of the sounds were identified as perceptually invariable. These results were corroborated by performing other tests with VCV syllables. The recognition of /b, d, g, r/ was more frequently correct in stimuli of this syllabic type.

Conclusion
In general, on the basis of perceptual differences, sounds classified as periodic and as bands of noise show a perceptual differentiation. These sounds are isomorphic with the continuous category. Bursts of noise correlate with perceptual confusion and are isomorphic with the non continuous category.

Reference
The Intelligibility and Acceptability of Distinctive Features
Tennessee State University, Department of Communication
Barach, Carol M. and Stewart, James Monroe

3500 Centennial Blvd. Nashville, TN. 37203 U.S.A.

PURPOSE

The study investigates the severity of articulation and its applicability to a general theory of distinctive features. Three questions are addressed. Is it possible to judge the severity of misarticulations by analyzing the erred features? Are distinctive features equally salient with respect to judgments of severity of misarticulations? Does the acceptability of misarticulations correlate with the degree of intelligibility of articulation for each erred feature.

METHOD

Three distinctive features, voice, strident, and coronal (Chomsky & Halle, 1968), yielded ten English phonemes, which were minimally contrastive with each other. The phonemes were embedded in the initial and medial word positions. Words were manipulated such that their minimally contrastive (substitutions) phonemes replaced the target (correct) phonemes, yielding misarticulated words with minimal-feature distinctions.

Subjects were required to make acceptability judgments on one set of 68 stimuli and intelligibility judgments on a second set. Both types of judgments were based upon a seven-point, equal-appearing interval scale.

RESULTS

In relation to the specific questions investigated, it is possible to judge the severity of articulation by analyzing the erred features, and distinctive features are not equally salient with respect to misarticulations. More detailed results and the answer to the third question are in progress.

The results support the existence of distinctive features; the groups studied reflected similar judgments with respect to the features; and the features in the initial and medial positions were correlated but manifested different saliences.
PERCEPTUAL EFFECTS OF ACOUSTIC PARAMETERS OF VOICED STOP CONSONANTS

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INTRODUCTION

It is known that acoustic cues for the perception of stop consonants are $F_1$, $F_2$, $F_3$ onset frequencies ($F_1^0$, $F_2^0$, $F_3^0$) and their transitions. The present study was conducted to investigate the effects of the transition duration ($T_j$) and the $F_0$ contour on the perception of voiced stop consonants /b,d,g/.

EXPERIMENTS AND RESULTS

CV syllables for listening tests were generated using the digital simulation of a terminal analog synthesizer with a sampling rate of 10kHz. The trajectories of the first four formants for a typical stimulus and five kinds of $F_0$ contours are shown in Fig.1. Subjects were 3 male adults.

EXPERIMENT 1 The effect of $T_j$ was investigated on the perception of /b,d,g/. Stimuli with parameters of $F_2^0$, $F_3^0$ and $T_j$ were used for the identification tests. Following vowels were /a,i,u,e,o/ in Japanese. The $F_0$ contour was constant at 130Hz. The subjects' judgements were /b,d,g or others/, and the identification rates of /b/, /d/ and /g/ were obtained on the $F_2^0$-$F_3^0$ plane corresponding to each $T_j$ (30, 40, 50, 60ms). The result shows that $T_j$ giving the largest region of 80% contour were about 30ms for /b/, 40ms for /d/ and 60ms for /g/, but there is a little difference due to following vowels, especially /i/.

EXPERIMENT 2 The effect of $T_j$ was confirmed by the perception of stimuli near /da-ga/ boundary. $F_2^0$ of stimuli was fixed at 1800Hz and $F_3^0$ was selected at 2200, 2300 and 2400Hz. Subjects judged /da/ or /ga/ by the forced choice. The result shown in Fig.2 represents the perceptual effect of $T_j$ in which the shorter $T_j$ give the perception of /da/, and the longer $T_j$ give that of /ga/.

EXPERIMENT 3 The perceptual effect of the $F_0$ contour was investigated on identifying /da/ and /ga/. $F_2^0$ and $F_3^0$ of stimuli were fixed at 1800Hz and 2400Hz near the /da-ga/ boundary, respectively. $T_j$ was fixed at 50ms. Five kinds of $F_0$ contours were chosen as in Fig.1. Subjects judged /da/ or /ga/ by the forced choice. The result shown in Fig.3 represents that the rising $F_0$ contours give the perception of /da/, and the falling $F_0$ contours give that of /ga/.

Fig.1. Trajectories of the first four formants (upper) and $F_0$ contours (lower)

Fig.2. Judgements of /da/, /ga/ due to the transition duration

Fig.3. Judgements of /da/, /ga/ due to the $F_0$ contours.
The Audible Condition of Peak Harmonic Components in Vowel-like Sounds

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INTRODUCTION
Following +DL (plus side Difference Limen) and -DL (minus side DL) of the harmonic components (HCs) of Japanese vowels [1], the audible condition of the peak HC(s) in major formant area is obtained in terms of the area parameter (α_i, i+1) and the frequency spacing between two peak HCs (ΔF_i, i+1) (i=1, ..., 4).

METHOD
Ten vowel-like sounds with appropriate formant frequencies F_s and their bandwidths B_s(i=1, ..., 4) were synthesized by 32 component harmonic synthesizer whose fundamental frequency f_0 is 125 Hz. Five of them correspond to 100% identification points and the other five to 75% points on F_1-F_2 diagram. They are called prototype sounds (PSs). For each PS, three kinds of comparison sounds (CSs) were synthesized by rejecting one, two or three adjacent HC(s) in major formant areas from the frequency spectrum of PS, P(f) (see Fig.1). Subjects are asked whether there is noticeable change or not before and after the rejection of the HC(s). If the probability of the noticeable change is more than 50%, the HC(s) are defined to have NCR (Noticeable Change for Rejection); in detail, NCR(1), NCR(2) and NCR(3), respectively, corresponding to cases (a), (b) and (c) in Fig.1. From each PS, 6 to 8 derived sounds (DSs) are made by gradually increasing the formula, B_s = B_s^0 + 50·j Hz, where B_s^0 is formant bandwidth of PS, j=1, ..., 8. For all the DSs, the same experiments were done.

RESULT AND DISCUSSION
The relation between the existence of HCs with NCR and the bandwidth parameter j is shown for four formant areas in Fig.2. For example, a circle is shown for abscissa j=1, ordinate i=2. It means that the peak HC in F_2 area has NCR(1). We can see that the number of HCs with NCRs decreases as B_s increases. Only 2 or 3 HCs which are located in formant area have NCRs. One example is shown in Fig.3, in which four regions are shown. The meaning is as follows; if an HC has (α_{12}, ΔF_{12}) in the region NCR(1), the HC has NCR(1). The same thing can be said as to NCR(2) and etc. α_{12} is the area of hatched region in Fig.1 and ΔF_{12} is the frequency difference between F_2 and F_1 (see Fig.1).

REFERENCE
THE SYLLABLE AS AN AUDITORY SIGNAL
Laboratorio de Investigaciones Sensoriales, CONICET
GUIRAO, Miguelina

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This work represents an attempt to classify V and VC syllables in terms of their acoustic spectra and auditory dimensions. The speech signals were taken from the Spanish language which has a relatively stable phonetic structure. The sample consisted of the five Spanish vowels and all consonant sounds paired in CV syllables with vowel /a/.

The acoustical classification is based on data presented in previous works (2,3) also by other authors (5,6,7). The psychoacoustical categories were: loud/soft, large/small, compact/diffuse, acute/grave, rough/smooth, clear/confuse, continuant/non continuant, long/short. The proposed scheme for the psychoacoustical attributes was checked by perceptual (4) and confusion tests (8).

Psychoacoustical categories were first divided in two classes: complex tones (periodic spectral distribution), designated class I; non-periodic sounds (noise), designated class II. Class I was divided into subclasses according to the frequency distribution: high amplitude formants /i, e, a, o, u/ and low amplitude formants /n, l, m, r, p/. Class II was subdivided in terms of duration: bands of noise (relatively invariant in time), and bursts of noise (transients). Bands of noise /s, f, x, ñ, rr, t/ were further subdivided in keeping with their duration: /d, b, g/ and their respective shorter transients /t, p, k/.

CV syllables represent the 56% and V syllables 4.5% of all possible combinations in Spanish language. Sounds labelled as periodic syllable nucleus or "carrier sounds" are about 68% of the Spanish phonemes; the rest of the sounds are represented by bands of noise, 12% and bursts of noise, 20% (1).

These figures suggest that the phonemic structure of the Spanish speech depends on a dynamic sequence of both transitional and invariant acoustical signals. Bursts of noise are highly dependent on the periodicity of complex sounds with which they are coupled, eg. /ka/, while bands of noise (aperiodic signals) remain relatively independent of the periodic signal.

This type of scheme brings the possibility of re-examining the classification of phonemes, giving emphasis to inter and intragroup hyperacoustic acoustic and perceptual contrast. It therefore permits the consideration of the syllable as a sound unit of speech.

References

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A Unified Subjective Measure of Speech Quality for Digital Coders

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The ultimate performance measure for evaluating a voice communication system is the subjective quality of the received speech. However, no adequate single uniform performance criterion for various digital waveform coders has yet been developed. This work takes a utilitarian approach in attempting to satisfy the urgent requirement for a practical measurement method. A subjective speech-to-noise-ratio SNR_{subj} was tested as a single absolute measure of overall speech quality in this study. At the same time, the relationship between the SNR_{subj} and several objective measures was investigated.

EXPERIMENTAL PROCEDURE

High quality digital recordings of two sentences by each of two speakers (one male and one female) were processed by three types of coders: logPCM(μ=225), ADM with one bit memory /1/ and ADPCM with variable or fixed third order predictor (-V or -F)/2/. Nine different combinations of the coders and transmission rates comprised the test signals. The test signal is compared with reference signals corrupted by varying amount of multiplicative white noise /3/. The SNR_{subj} of the test signal is defined as the SNR of that reference signal which, on the average, is equally preferred by groups of listeners. A total of 392 pairs of signals were randomized and presented to 11 naive listeners for preference testing.

RESULTS AND DISCUSSION

The SNR_{subj} of 9 coder configurations for the test data pooled over 4 speech samples and 11 listeners are shown in Table 1. An analysis of variance shows that none of two main experimental factors, speech sample and listeners, introduces any statistically significant variation in the SNR_{subj} estimate. An arbitrary selection of 5 or 6 listeners out of 11 yields a stable SNR_{subj} estimate. On the other hand, the SNR_{subj} for adaptive coders show some variations depending on the speech sample. We define a segmental speech-to-granular-noise ratio SNR_{seg} as the ratio of the speech energy to that component of the noise energy which shows a sign change between adjacent samples. SNR_{seg} and SNR_{seg} predict SNR_{subj} with RMS errors of 5.2, 4.0 and 2.6 dB respectively. Thus SNR_{seg} is the best predictor of quality among those tested.

The proposed method is expected to be a useful tool for the establishment of engineering criteria on speech quality. These experimental results provide a useful data base for monitoring and comparing newly developed coders. Stability of the ratings obtained on the different occasions should be studied further.

Table 1. Subjective and objective quality measures

<table>
<thead>
<tr>
<th>Coder Bit-Rate (kb/s)</th>
<th>SNR_{subj} (dB)</th>
<th>SNR_{seg} (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>PCM 64</td>
<td>39.42</td>
<td>39.23</td>
</tr>
<tr>
<td>ADPCM-V 32</td>
<td>34.84</td>
<td>33.58</td>
</tr>
<tr>
<td>PCM 56</td>
<td>32.42</td>
<td>33.43</td>
</tr>
<tr>
<td>ADPCM-F 32</td>
<td>31.08</td>
<td>30.88</td>
</tr>
<tr>
<td>ADM 32</td>
<td>26.52</td>
<td>22.67</td>
</tr>
<tr>
<td>PCM 48</td>
<td>24.97</td>
<td>27.46</td>
</tr>
<tr>
<td>PCM 40</td>
<td>19.31</td>
<td>21.43</td>
</tr>
<tr>
<td>ADM 16</td>
<td>16.57</td>
<td>18.43</td>
</tr>
<tr>
<td>PCM 32</td>
<td>13.37</td>
<td>16.43</td>
</tr>
</tbody>
</table>

REFERENCES


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** also with Bell-Northern Research Ltd.
To evaluate the performance of a new synthesis by rule system, comparative intelligibility tests were conducted with the synthesised speech and natural speech using three types of test material. These were; /h-d/ frame vowels, CV syllable consonants, and a PB word list (CVC structure). The materials were presented in quiet and noise masked listening conditions, to upwards of twenty subjects in each case. Measures were incorporated in the experimental design to minimise effects caused by learning, fatigue and similar listener artifacts.

Considerable differences were evident in the intelligibility characteristics of the synthetic speech with the three types of test material. Synthetic /h-d/ frame vowels were as intelligible as their natural counterparts in quiet listening conditions (approaching 100%), and equally or more resistant to degradation by noise masking. Synthetic CV syllable consonants showed an intelligibility level of 82% in quiet compared to 96% for their natural counterparts. Synthetic PB words showed similar although somewhat better results with 86% intelligibility in quiet compared to approaching 100% for their natural counterparts. Both showed markedly greater vulnerability to degradation by masking noise than did natural speech.

Phonetic error patterns for the performance of both speech types with each of the test materials are presented. Comparisons between the two, noting essential similarities and differences, are made with the aid of data analysis procedures intended to illustrate the nature of these patterns in a phonetically meaningful way.

Some tentative conclusions are offered about the implications of the results of this investigation both for the synthesis by rule system under test, and for terminal analogue synthesised speech and its use in phonetic research.
Intelligibility of Frequency Compressed Speech processed by PARCOR · Speech Analysis-Synthesis Method
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The high frequency components of speech sounds carry more of the intelligibility than the low frequency ones. So persons who have some type of high frequency response loss show a greater intelligibility loss than those with a low frequency loss. Several techniques of frequency compression for transposing high frequency components of speech into the usable lower frequency range have been proposed to provide intelligible speech for this type of deaf person. It appears, however, that additional distortions of speech spectrum have been derived from these techniques. In this study, a new technique of frequency compression based on the PARCOR speech analysis-synthesis method have been studied.

Speech signal was lowpass filtered to 5 KHz, sampled at 10 KHz and digitized in 10 bits. Then PARCOR speech analysis was made every 5 msec. on Hamming windowed speech signal of 20 msec. duration time. The order of analysis was 12 and the fundamental frequency was derived from detecting the peak of the autocorrelation of the residual wave. The voiced/unvoiced judgement was performed taking both values of the peak height of the autocorrelation of the residual wave and of the PARCOR coefficient of first order into consideration.

The frequency compression of speech was achieved by the reduction of the sampling frequency in PARCOR speech synthesis processing. To hold the original duration time of speech signal, the number of speech samples per frame interval was reduced in accordance with the reduction of sampling frequency. As for the fundamental frequency of synthesized speech, two kinds of test conditions, one was as same as the original speech and another was reduced in accordance with the reduction of sampling frequency, were used. The frequency compression ratios of synthesized speech were varied from 0.8 to 0.4. Synthesized speech was lowpass filtered at several cutoff frequencies around 1 KHz to simulate the high frequency response loss and used for the articulation test. Test materials were 100 Japanese monosyllables and the test crew was with normal hearing.

Results showed that (1) sound articulation scores were not only unaffected by the frequency compression ratio, but also improved slightly in case of the lowpass filtered speech of around 1 KHz, although sound articulation scores were deteriorated in accordance with the reduction of frequency compression ratio in the case of 5 KHz frequency band speech, (2) the reduction of fundamental frequency of synthesized speech in accordance with the frequency compression ratio brought about considerable improvement of sound articulation score in lowpass filtered speech.

It is concluded that the frequency compressed speech with the frequency compressed ratio of 0.7 to 0.5 accompanied by the reduction of fundamental frequency may be available for high frequency acuity impaired persons.
A laboratory speech processor has been developed for a multiple-channel cochlear implant prosthesis. The speech processor accepts the speech waveform as an input and produces a pattern of electrical stimulus data as output. The electrical stimulus data are transmitted to the implanted receiver-stimulator by a transmitter which is external to the speech processor.

Four speech signal parameters were estimated every 20 ms in the parameter estimation section of the speech processor. These parameters included the fundamental frequency (F0), a low frequency energy measure (A0), the second formant frequency (F2) and its amplitude (A2).

The speech parameter estimates (F0, A0, F2, A2) were transformed to electrical stimulus parameters in the encoding section. In the present speech processor, only one electrode was activated in any 20 ms time frame. For a given F2 estimate an electrode was selected according to a predetermined F2-to-electrode transformation map based on the results of psychophysical tests and correlated with the tonotopical organisation of the cochlea. The subband of lowest F2 estimates was assigned to the electrode with the dullest sensation, while the subband of highest F2 was assigned to the electrode with the sharpest sensation. The current level for the single-electrode pulse train was determined from A2. A 20 ms speech segment was classified as voiced if A0 exceeds a pre-selected threshold, and unvoiced otherwise. For unvoiced speech segments, a constant low pulse rate for electrical stimulation was used. For voiced speech segments, the pulse rate was proportional to F0, and was higher than the pulse rate used for the unvoiced segments.

Vowel and consonant confusion studies were conducted with two cochlear implant patients. The test vowels included /i/, /u/, /y/, /I/, /e/ and /a/. The test consonants were /p/, /k/, /f/, /r/, /s/, /z/, /l/, /n/, /w/, /j/, /t/. The test materials were presented live by a female and a male speaker without lip-reading being involved. The mean percentage correct scores across patients and speakers were 77% for the six vowels and 35% for the ten consonants. Analysis of the vowel confusion data showed that the patients were able to make use of the duration and the electrode assignment of the electrical stimuli as cues for vowel identification. For consonant identification, the results showed that the overall percentage correct for voiced/unvoiced classification was 70%, while the overall percentage correct for transition classification was 67.5%.
DEVELOPMENTAL DIFFERENCES IN IDENTIFYING CV SYLLABLES

Lois L. Elliott, Israel Raz, Karen Zucker, Cheryl Longinotti

This study examined the ability of three groups of subjects (normally-hearing 6- and 10-year olds and adults) to label: (1) digitized natural speech tokens (/ba/-/da/-/ga/); (2) the same natural speech tokens with the initial burst excised; (3) members of a 13-stimulus continuum of four-formant synthesized /ba/-/da/-/ga/ stimuli with appropriate 5-msec bursts; and (4) the same synthesized continuum without bursts. An AX (same-different) task with reinforcement for correct responses was also performed on the latter two types of stimuli.

Developmental differences were observed even though all subjects scored essentially 100% correct on a standard test of speech discrimination (WIPI Test for children and W-22 for adults). The role of the noise bursts for synthesized stimuli was not consistent. However, strong developmental effects were observed for boundaries between adjacent perceptual "categories" (synthesized syllables) for both the adaptive labeling procedure (Raz, I. and Wightman, F.L., J. Acoust. Soc. of Amer. 64, 1978, S1-19) and for the AX task. The developmental differences for the tasks employing synthesized CV syllables, to some extent, parallel performance on the tasks employing digitized natural speech stimuli. It is concluded that tasks such as these can provide a much more "fine-grained" analysis of the patient's speech discrimination skills than traditional tests of speech discrimination. (Work supported in part by grants from: the Bureau of Education of the Handicapped; the National Institute of Neurological and Communicative Disorders and Stroke, and the National Science Foundation.)

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Three different speech discrimination tests are in various stages of development for use with deaf students at NTID. Six-choice numeral, and vowel, initial-and final-consonant tests have successfully categorized a pilot group into five meaningful groups for rehabilitation management. These two tests have subsequently been combined into a single 50-item test and used in categorizing the 1979/80 entering class.

Paralleling this effort are a series of tests combining the Kalikow, Stevens and Elliott (J. Acoust. Soc. Am. 61, 1337, 1977) SPINIKER concept with the Modified Rhyme Test format. There are four types of this SPINIKER test where the low-predictable sentence foils are a 4-choice vowel, initial consonant, final consonant, or numeral items. The high-predictable sentence foils include the designers' key word and three others generated by NTID students who filled in the missing last (key) word on a printed text. Typical examples of word (sentence) pairs which appear on alternate lists are:

Please wipe your feet on the mat (mat rug grass rag)
Tom wants to know about the mat (mat met might mate)
X plus Y equals ninety (9 19 90 99)
Nine times ten equals ninety (9 19 90 99)

Each of these four types are designed in three equivalent forms for use in a visual mode (no sound), a visual-auditory mode, and an auditory mode.

A third type of test designed for use in hearing aid fitting, but as yet unrecorded, combines a multiple choice format for the first few very common words and an open response set for the final word. The first words include choices among the most common and important words in any language, the interrogatives, verbal auxiliaries, personal pronouns, (possessive) adjectives, and/or articles. Four equivalent verb sentences for alternate lists and the answer format follow:

A1 When might we throw? 1. Where can I
B1 What can I drink? When may you
C1 How must you choose? What might it
D1 Where may it break? How must we

Four equivalent noun sentences for alternate lists and sample words for the answer format follow:

All How are the houses? 1. Where is my/our/your
Bll Where were their bags? Which is his/her/their
Cll When is your test? When was a/the
Dll Which was that boy? How were that/these

The basic difficulty of the tests is determined by the familiarity of the last word.
The purpose of this paper is to describe the results of an auditory feedback technique for stimulating infants and young children to vocalize. The assumption is that an audition-production cycle in early life contributes to the interest or motivation required for later verbal communication.

Some pilot attempts with normal subjects were to determine a procedure. Six subjects, 9-12 months of age, were the source of data. Vocalizations were recorded and feedback through a loudspeaker at a delay time. Responses were recorded. All recorded vocalizations were passed through a power level recorder on to rotating paper via an inked stylus. Measurements were made by a millimeter rule. The subjects responded with an increased amount of vocalization during feedback conditions. A set of measures were obtained from additional subjects, age 9-14 months. Ten were selected for study. Five who were living at home with their natural parents were stimulated to increase significantly their amount of vocalization, but five who lived with foster parents did not.

With twenty normal subjects, 9-14 months, a graduate student made repeated recordings and measurements of vocalizations during conditions of no-feedback and of feedback. The infants were recorded in their own homes and in a playpen with a microphone suspended directly above. Subjects produced significantly increased vocalization during and immediately following feedback exposure. With thirty additional subjects, several delay periods and custom-made earphones were employed. Again, vocalizations increased significantly during and following feedback exposure. There were also interesting changes in range of pitch under feedback exposure. The writer, assisted by graduate students, studied the same 30 subjects. Measures of loudness of the subjects' vocalizations were an average of the three highest peaks of amplitude for each segment of a recording. A significantly different amplitude measure was obtained for vocalizations during and immediately following feedback exposure.

The writer applied the technique to nineteen young hearing impaired children, 3-8 years old. By amplification of returned vocalization, the subjects increased significantly in their vocalizations. All were attending special schools for auditory training in Yugoslavia, Belgium or France.

It would appear that audition and vocalization in early life are importantly interlaced when vocalization is both the stimulus and the response. A technique in the manner of delayed auditory feedback may be a useful method of intervention.
The assessment of the computer-vowel-trainer: methodology and results
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The Computer-Vowel-Trainer (CVT) developed at Cambridge University uses a linear prediction to estimate and display the shape of the speaker's vocal tract during an utterance (1). A portable version of the CVT has been experimentally implemented in a unit for the deaf and a microprocessor version is now also available.

The assessment of the CVT has been carried out over a period of three years, with a group of 8 profoundly deaf children, aged 4 to 12 years, and their performance has been compared to that of a matched control group, which followed the same teaching programme using traditional methods. Intelligibility scores were obtained by presenting the children's utterances, tape-recorded before training and at different intervals of time after training, to panels of naive listeners, who performed an identification and a forced-choice task.

Two main results will be discussed. The first regards the learning process using the CVT and shows that improvement in speech production is time-locked. Prolonged training on the same phoneme is ineffective, while after a pause in the training of a particular phoneme the output is better. This finding is relevant for the structure of future teaching programmes. The second major result concerns the retrieval of the encoded articulatory information. Deaf children who during training were given visual feedback by the CVT remember the correct articulation of the learned phonemes after longer periods of non-practice than the control group. This suggests that meaningful visual feedback plays an important role in the long-term retention of articulatory gestures.

Acknowledgement
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The population of persons in the United States deafened in adulthood is estimated at approximately 1% with one in four over the age of 65 developing a hearing disorder. Because language and speech has developed essentially normally for this population, speech therapy, as opposed to speech conservation, has not been regarded as essential in the rehabilitation process. Histories and voice recordings for over 60 cases of severe to profound hearing losses indicate specific voice and speech production problems over which the patient expresses anxiety. These problems, methods of subjective and objective analysis, and specific therapeutic techniques are briefly discussed. Four representative histories and recordings are presented. Therapy results with 10 cases show voice production problems can be alleviated through the use of direct therapy, use of prosthetic devices, and behavior modification techniques. Speech production problems may require rather extensive therapy with guarded results dependent upon the existence of speech problems prior to hearing loss.
ACOUSTICAL FEATURES OF FUNDAMENTAL FREQUENCY CONTOURS OF JAPANESE SENTENCES

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A model has been presented by the authors for the analytic representation and
quantitative characterization of contours of the voice fundamental frequency
($F_0$-contours) of isolated words in Japanese and in English.\textsuperscript{1-3} Comparatively
little is known, however, about the characteristics of $F_0$-contours of sen-
tences. The present paper describes a model constructed toward such characteri-
zation of sentence $F_0$-contours, and presents some new findings.

A MODEL FOR THE SENTENCE $F_0$-CONTOUR

The model is based on the formulation of the process whereby the logarithmic
fundamental frequency is controlled in proportion to the sum of two kinds of
components, generated respectively by the phrase and accent commands. These
commands are assumed to be step functions, but their effects appear on the
$F_0$-contour after being smoothed by various neural, myoelectric, mechanical and
aerodynamic mechanisms in the control process of vocal cord vibration. It is
assumed that these smoothing characteristics are essentially identical to
those adopted in the word model,\textsuperscript{1} but the numbers of phrase and accent com-
mands are not limited to one, and their amplitudes may not be the same even
within a sentence. The model allows one to determine, by the method of Anal-
ysis-by-Synthesis, characteristic parameters of a given sentence $F_0$-contour.

ANALYSIS OF $F_0$-CONTOURS OF DECLARATIVE SENTENCES

The analysis was made on $F_0$-contours extracted from utterances of various
declarative sentences ranging from 8 to 24 morae in length. The two panels in
Fig. 1 show results for the same utterance obtained with different number of
phrase commands — one in (a) and two in (b). The close agreement between
the model (solid line) and the measured $F_0$-contour (○) in (b) supports the as-
sumption for two phrase commands, even though the utterance was produced in
one breath group. Results of analysis of a number of utterances also reveal
that the shape of the phrase component is not appreciably influenced by sen-
tence length, but longer sentences tend to possess two phrase commands. The
amplitudes of various accent commands within an utterance are found to fall
roughly into two groups, and suggest the binary nature of prominence at
least as far as the present utterance samples are concerned.

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Fig. 1. Example of results of Analysis-
by-Synthesis of a sentence $F_0$-contour.
A COMPARISON OF SPECTROGRAMS AND VOCAL TRACT AREA FUNCTIONS FOR DISPLAYING SPEECH

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Using a linear prediction analysis of speech it is possible to derive an estimate of vocal tract shape, at least for voiced speech. The possibility thus arises for displaying speech in the form of articulatory estimates as a function of time. A comparison of an intensity-modulated picture of such estimates with the standard spectrogram is made. The comparison is made on the basis of speaker-to-speaker consistency and variations within a class of utterances from one speaker.
VERSTANDLICHKEIT GEFLÜSTERTER KUNSTLICHER WÖRTER

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Einen sinnvollen alternativen Ansatz zur automatischen Erkennung natürlicher Sprache, die bis heute grundlegende theoretische Mängel und technische Unzulänglichkeiten nicht überwunden hat, stellt die Verwendung konstruierter Sprachen restringierten Phoneminventars dar. Beschränkungen auf der phonematischen Ebene erlauben eine erhebliche Vereinfachung der Entscheidungsstrategien in Spracherkennungsverfahren bei gleichzeitig hoher Freizügigkeit für die Kombination der gewählten Phoneme zu künstlichen Wörtern. Der Prozeß der Auswahl des optimalen Phoneminventars ist zunächst durch die akustische Struktur der Sprachlaute bestimmt, er läßt sich sodann durch objektive (am Spracherkennungsgerät) und subjektive (Ermittlung der Verständlichkeit an VPP) Tests in Extremsituationen (Signaldestruktion) überprüfen.

Es wird ein Verfahren zur subjektiven Überprüfung eines Kodes aus drei Vokalen (i, o, a) und drei Konsonanten (s, t, n) auf der Grundlage geflüsterter künstlicher Wörter vorgestellt, das eine Ableitung der Ergebnisse (Verständlichkeitsniveaus) aus den Lauten selbst, ihrer Umgebung und der Wortprosodie gestattet.
VOICE SOURCE DYNAMICS
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Much less information has been accumulated on human vocal source characteristics than on vocal tract filter functions. The imbalance is especially apparent with respect to temporal properties. A project has been initiated to study successive events of glottal flow and vocal tract excitation within a voice fundamental period as well as within a whole utterance. The flow model adopted has a smoothly rising branch, \( U = U_0 \left( \frac{1}{2} - \cos 2\pi F_g t \right) \), and a falling branch \( U = U_0 \left[ K \cos \left( 2\pi F_g t - \pi \right) - K + 1 \right] \) which terminates abruptly to zero value with a slope of \( U_3' = -U_0 \frac{2\pi F_g (2K - 1)^{1/2}}{K > 0.5} \). \( K \) is a steepness factor. The three parameters \( U_0, F_g \), and \( K \) may be substituted by \( U_0', F_g' \), and \( U_3' \), the latter parameter carrying the formant amplitude proportionality of the main excitation. These parameters may be determined experimentally from inverse filtering. Earlier reports, Fant 1979a, 1979b) gave a detailed theoretical analysis and experimental validation of time and frequency domain properties of the model including the temporal distribution of excitation and losses within a voice fundamental period. The study of Fant and Liljencrants (1979) gave a perceptual basis for defining mean glottal bandwidths.

A continuation of this project is the study of voice source parameters as a function of time in connected speech. The main source induced variations in formant amplitudes are related to \( U_3' \) via \( K \) and \( F_g \). Once voicing has been initiated, \( U_0' \) varies less than \( U_3' \) and determines a low frequency rather stable "carrier" below \( 2F_g \) which at weak voice efforts \( K < 1 \) appears as a "glottal formant".

Variations of the voice source parameters with voice onset and offset, subglottal pressure, supra-glottal narrowing and degrees of vocal cord abduction and adduction are discussed. As pointed out by Rothenberg et al (1974) glottal articulations account for a range of useful voicing between the extremes of abduction (aspiration) and adduction (glottalization). Leaky voicing and abduction produce excessive formant damping, e.g. in "F1-cutback". Source parameter correlates of linguistically defined prosodic elements are exemplified.

References
Some Remarks on the Optimization of Articulatory Parameters from Speech Waves

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A signal model for low bit-rate coding of speech based directly on the physics of speech production has recently proposed (J.L. Flanagan et al.; Signal Models for Low Bit-Rate Coding of Speech). The parameters are estimated by an adaptive procedure that minimizes spectral differences between a natural human speech and the synthetic output of the dynamic speech synthesizer. An effort to improve the accuracy of the estimation of parameters and to save the computation time is currently in progress. This paper describes some results of a part of the effort.

For the optimization of articulatory parameters we use the real spectral envelope that connects the peaks of the fine structure in the short-time log amplitude spectrum. The conventional spectral envelope such as those cepstrally smoothed or derived from LPC is not suitable for better estimation, because they are still dependent upon details of the source characteristics even for the same area function of the vocal tract.

The dynamic behavior of the vocal cord/vocal tract model in the window duration is incorporated into the adaptation loop. This is essential to the optimization of the parameters for the consonants and also phonemic boundaries where cord vibration is not steady.

In the original dynamic speech synthesizer, the wave equation in the lossy vocal tract with yielding walls is approximated as a set of difference equations by the bi-linear transform. The computation of this formulation is fairly time-consuming. For computational economy, the wave equation with all losses and wall impedances of the vocal tract is approximated by wave matrices and is incorporated with the vocal cord model. The relevant quantities can now be computed explicitly. Both the formulations for the wave equation are compared with respect to accuracy and computation speed.
VOCAL TRACT AREA FUNCTION RECOVERY WITH RESISTIVE LIPS TERMINATION
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One of the aims of speech analysis is to accurately recover the shape of the vocal tract from the speech waveform. The need for tractable mathematics has forced existing models of speech production to differ from the real situation, and so make accurate area function recovery difficult.

This paper looks at the addition of the following constraints which provide a more realistic model and enable improved area function recovery.

(1) At the lips we require a resistive termination.
(2) We impose physical constraints on the geometry of the vocal tract.

The analysis procedure initially depends on identification of an inverse filter by conventional means. Then the constraints are incorporated in finding a corresponding lossless transmission line model, which has a matched glottal source. This step requires the solution of a set of simultaneous non-linear equations. We have found the use of a colour display very helpful in understanding the behaviour of the non-linear solution algorithms, in stability, and in ambiguity aspects. Constraining the geometry of the vocal tract reduces the problems of a non-unique solution, which are encountered when assuming a non-zero radiation impedance.

Initial analysis was carried out with synthetic speech waveforms generated from a computer simulation of a lossless transmission line, with matched glottal source and general lips radiation impedance. Results of this initial analysis are presented and show that improved area function recovery is obtained, as compared with linear prediction methods. In addition we will present the results of the new analysis method as applied to acoustic waveforms generated from a set of cascaded acoustic cavities of known dimensions.
SPEECH ANALYSIS BY GENERALIZED INVERSE OF MATRICES

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ABSTRACT
The so-called "Covariance Method" for speech analysis is discussed from a point of view of generalized inverse (g-inv) of matrices[1]. Adaptive procedures for flexible time window and for order incrementing in LP analysis are proposed.

FORMULATION OF LINEAR PREDICTION IN G-INV
Assume that the sampled speech signal $y_j$ can be approximated by eq.(1),

$$y_j = \sum_{i=1}^{p} \theta_i y_{j-i} \quad (1)$$

where $\theta_i$ denotes $i$'th predictor, $p$ is the order of prediction, and $n$ is the observation period. Eq.(1) can be rewritten in matrix form as eq.(2). As eq.(2) is a set of $n$ equations with $p$ unknowns, the type of solution appropriate will be different according to the relative sizes of $p$ and $n$. In the general case, the minimum-norm least-squares solution of eq.(2) is obtained as eq.(3), where $+\text{ denotes the Moore-Penrose g-inv.}$ In the case of $\text{rank}(\mathbf{Y}) = p$, we can get eq.(4) by multiplying by $(\mathbf{Y}^T\mathbf{Y})^{-1}\mathbf{Y}^T$ from the left hand side of eq.(2). Here, $\mathbf{Y}^T\mathbf{Y}$ is identical to the covariance matrix of $\{y_j\}$, and $\mathbf{Y}^T\mathbf{y}$ is the auto-correlation function of $\{y_j\}$. Therefore, eq.(4) is in effect the solution of the Yule-Walker equation.

Having been formulated in equations (2) and (3), this method can use certain theorems[1] concerning the g-inv of augmented matrices. Consequently, special procedures for data addition, data deletion and order incrementing are available for this method.

RESULTS AND DISCUSSIONS
The performance of this method with short flexible time windows is quite satisfactory, especially for the rapid transient parts of speech. However, it takes a much longer time to carry out the computation than does ordinary LP analysis at present. The main problem left is to develop some special method of computation which exploits the structure of the matrix $\mathbf{Y}$.

REFERENCE

Adaptive Predictive Coding of Speech Signals at Low Bit Rates

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This paper describes an improved method of quantizing the prediction error in adaptive predictive coders for speech signals. In our earlier work on predictive coders, we discussed methods 1) for efficient prediction of formant and pitch redundancies in speech signals and 2) for shaping the spectrum of the quantizing noise to reduce its loudness in the decoded speech signal (B. S. Atal and M. R. Schroeder, IEEE Trans. ASSP, June 1979, pp.247-254). In a predictive coding system, a significant number of bits per second is used for transmitting the prediction error signal. Efficient quantization of the prediction error is thus essential in achieving the lowest possible bit rate for a given speech quality.

At bit rates lower than about 10 kbits/sec, it has been necessary to quantize the prediction error with only 1 bit/sample (2 levels). Such a coarse quantization is the major source of audible distortion in the decoded speech signal. Our studies indicate that accurate quantization of high-amplitude portions of the prediction error is necessary for achieving low perceptual distortion in the decoded speech signal.

This paper reviews our earlier work and describes a new method of quantization for improving the speech quality. The improvement is obtained by center-clipping the prediction error and by fine quantization (many levels) of the high-amplitude portions of the prediction error. The center-clipping threshold is adjusted automatically for each frame to provide encoding of the prediction error at a specified bit rate. This method of quantization not only improves the speech quality by accurate quantization of the prediction error when its amplitude is large but also allows encoding of the prediction error at bit rates below 1 bit/sample.

Tape recordings of speech encoded at different bit rates using this new quantizer will be played at the Congress.
ORDER ESTIMATION OF AR MODEL BASED ON EIGENVALUES OF COVARIANCE
MATRIX OF SPEECH

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INTRODUCTION

This paper presents a method for estimating the order of AR process of the speech production model. This method is based on eigenvalues of the covariance matrix of speech wave in the interval where the glottis is closed.

THEORETICAL BASIS

The speech production model is assumed to be ARMA model( the order of AR process is $p_0$, the order of MA process is $q_0$). Let $\{s(0), s(1), \ldots, s(N-1)\}$ be a set of $N$ time-sequence speech samples obtained from the speech production model. We consider the covariance matrix $\Phi$ defined as Eq.(1).

$$
\Phi = \begin{pmatrix}
\phi_{ij} \\
\phi_{ij} & \phi_{ij} & \cdots & \phi_{ij}
\end{pmatrix}
= \sum_{n=p+1_0}^{N-1} s(n-1_0-i)s(n-j), \quad i,j=1,2,\ldots,p.
$$

(1)

Considering $s(n)$ is the $n$-th speech sample obtained from the ARMA process, the following equation is obtained from Eq.(1).

$$
\phi_{ij} = \sum_{k=1}^{p_0} a(k) \phi_{i,j+k} + \sum_{m=0}^{q_0} b(m) \sum_{n=p+1_0}^{N-1} s(n-1_0-i)u(n-j-m), \quad i=1,2,\ldots,p
$$

where $a(k)$ and $b(m)$ are the coefficients of AR and MA process of the speech production model, respectively, and $u(n)$ is the $n$-th sample of excitation waveform. If the analysis frame is set at the interval where $u(n)=0$, RANK $\Phi$ is equal to $p_0$ from Eq.(2). The interval of voiced speech where the glottis is closed can be a candidate for $u(n)=0$. In the case of natural voiced speech, however, $u(n)$ may not be zero even in the interval. But $u(n)$ may be assumed to be white noise in the interval. If $i_0 > q_0$ and the analysis frame is set at the interval, we have $\sum_{n=p+1_0}^{N-1} s(n-1_0-i)u(n-j-m) \approx 0 (i \geq 1_0, j=1,2,\ldots,p_0)$ due to the causal relation between $u(n)$ and $s(n)$. Therefore RANK $\Phi \approx p_0$ is obtained from Eq.(2). Then $|\lambda(i)/\lambda(i+1)| (\lambda(i)$ is the $i$-th eigenvalue of the matrix $\Phi$ and numbered in order of absolute value) has a maximum at $i=p_0$.

RESULT AND DISCUSSION

Figure 1 illustrates a synthetic speech applied to the simulation of order estimation. The excitation source is a mixture of glottal wave[Rosenberg, A.E.J. Acoust. Soc. Amer., Vol. 49, 2, p. 583(1971)] and white noise, as is shown in Fig.1(a). Figure 2 illustrates eigenvalue ratios of the matrix $\Phi$. When $i_0=0$(triangles in Fig.2), $|\lambda(i)/\lambda(i+1)|$ has a maximum at $i=2$, however, when $i_0=10$(circles in Fig.2), it has a maximum at $i=6(p_0)$. It may be concluded that we can determine reliably the order of AR process of speech production model.

Fig.1 Excitation source and synthetic vowel waveform.

Fig.2 Eigenvalue ratio $|\lambda(i)/\lambda(i+1)|$ of the matrix $\Phi$. 
USE OF SPECTRAL COVARIANCES IN TALKER VERIFICATION

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The present work relates to a statistical method of talker verification which has been designed to be robust against spectral weightings in telephone transmission.

Preprocessing based on estimates of the log-spectrum was chosen because these estimates should only be perturbed by simple addition of the decibel values of the transmission at the corresponding frequencies. Direct pattern matching of the averaged log spectral estimates is of course rendered prone to error by the differences between transmission paths. However, the variation (variance) of each estimate, and the covariances of the estimates should not be changed by the addition of different, constant, decibel values to each. Minor influence may persist if the spectral weighting pattern has significant variation across the sub-band contributing to each log-spectral estimate.

The 14 x 14 covariance matrix of 14 log-spectral estimates taken at 10ms intervals over about 2 seconds of speech was used as the pattern to be compared for verification. The theory of comparison of covariance matrices was developed to derive a convenient Euclidean norm which could be combined with other Euclidean norms.

Experiments confirmed the robustness of the process against spectral distortions, and showed superiority in this respect over LPC-based statistical talker verification.
SIMPLY STRUCTURED PARGOR-ANALYZERS FOR SPEECH SIGNALS

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The autocorrelation procedure of LPC has become a standard method in the analysis of speech signals. Special hardware and microprocessor realisations are known. Little is known about realisations of lattice-type analyzers because of the very high number of necessary computations. Itakura has proposed an "absolute mean" correlator using table-lookup procedures instead of multiplications and divisions for the calculation of the PARGOR coefficients in the lattice-type analysis. In addition, it is possible to simplify the procedures for calculating the necessary mean values. By using these methods and a careful design, it is possible to reduce the computational accuracy for the correlators down to 8 bits where only elementary arithmetic operations are necessary. The realisation of such a lattice-type analyzer using special hardware or microprocessors turns out to be very efficient. The speech quality remains sufficiently good.
Phonetics has long been used as a process for language acquisition and fluency. For most persons the task of learning to speak a target language depends upon acquiring mastery of the system of the spoken language in which the person has already achieved fluency. Because of breakdowns in the complete language ecosystem, fluency is not often achieved. However, by acquiring comprehensive understanding of phonetics from ecosystem language analysis, symbiotic commensalism can be achieved, and competency and fluency of speaking realized. This research paper explains the ecosystem theory of language, identifies problems, analyzes specific phonetic identities, and shows the process involved.

In the U.S. it is increasingly a problem for immigrants to acquire confidence in communication when speaking English. Application of using the ecosystem analysis theory to the problem of speaking fluent English was applied to Spanish and Asian languages. These host languages were selected due to the fact that in recent years the U.S. has incorporated an influx of Asian peoples. Then, too, Los Angeles contains a large Mexican Spanish-speaking population. Although widely variant in language components, both the Mexican and Asian populations were found to have had an immediate need to speak English fluently for personal, financial, and professional objectives.

The logopedist has the task of providing ways whereby language speaking proficiency can be realized. Languages exist in an ecosystem which includes all the factors impinging on the language contact situation. The host language is the reference point to the foreign speaker. The new or guest language entering the ecosystem becomes the target and the relationships between host and guest languages need to be carefully scrutinized in order for there to emerge a symbiotic commensalism or even mutualism.
A l'aide de 84 phrases énonciatives, lues en contexte par un homme et une femme, on propose dans cette étude qualitative, résumant les principaux points d'une analyse menée par ailleurs (Thèse de 3ème cycle, G. CAELEN 1978), une interprétation des rapports susceptibles de s'établir entre les structures textuelles (principalement syntaxe) et les structures orales.

Ces phrases non complexes, représentant les principales structures et toutes les catégories syntaxiques du français, partageant toutes avec une autre au moins, une séquence homophonique, ont été l'objet, après vérification de leur naturel, d'une segmentation manuelle en phonèmes; ces phonèmes (y compris les consonnes pour la durée) ont subi des corrections en fonction de l'effet des caractéristiques intrinsèques et de celui du contexte selon les paramètres (durée, fréquence fondamentale, énergie).

De la confrontation entre les divers plans d'analyse de la langue -analyse syntaxique, analyse prosodique- à propos des unités linguistiques des énoncés, il semble se dégager les faits suivants:

1- à structure syntaxique identique, diversité des structures prosodiques.

2- pas de corrélation nécessaire entre catégorie syntaxique des constituants et direction des pentes paramétriques.

3- pas de corrélation nécessaire entre valeurs paramétriques spécifiques (maxima par exemple...) et la hiérarchie des constituants

4- en dehors d'une certaine adéquation toujours possible entre les limites du groupe syntaxique et celles des unités prosodiques, on constate également dans le corpus deux phénomènes conjoints ou non que l'on appelle, par convention, "disyntaxe" et "asyntaxe" prosodiques.

Selon notre interprétation, les structures prosodiques ne sont pas la traduction orale des structures syntaxicosémantiques, mais possèdent leurs propres lois (que l'on définit précisément) d'agencement et de mise en relation des unités linguistiques - en un mot une syntaxe spécifique - lois qui opèrent sur les deux axes, paradigmatique et syntagmatique. De même, les relations fondamentales que l'on dégage entre locuteur (et auditeur) et discours émis fondent la pragmatique des structures orales, si tant est que la lecture soit représentative de ces dernières.
Many methods of compressing and expanding speech, in time or frequency, depend on cutting and splicing the time-waveform. The need arises for processes which avoid spurious perceptual effects due to abrupt transitions or erroneous overlaps at the points where the waveforms are joined.

We have studied the matching of waveforms by choosing suitable instants based on the analytic signal, \( x(t) = s(t) + j\hat{s}(t) \), \( \hat{s}(t) \) being the Hilbert transform of the real signal \( s(t) \). The basis of the criterion is that the euclidean distance \( |x(t_1) - x(t_2)| \) should be less than a threshold, but that \( |x(t_1)| \) and \( |x(t_2)| \) should be greater than the rms value of the signal. The latter criterion ensures that erroneous acceptances of \( |x(t_1) - x(t_2)| \) do not come about simply through the signal envelope being small. The analytic signal has attractive properties in this application in that

1. The criterion ensures matching of envelope as well as phase (within the distance criterion).

2. The envelope and phase become continuous functions (within the distance criterion) if \( s(t) \) is bandlimited.

3. \( \hat{s}(t) \) is orthogonal to, and hence uncorrelated with, \( s(t) \) although both are of identical rms value.

4. Although high quality Hilbert transformation is moderately expensive there are well-known and adequate all-pass complementary networks which can produce suitable \( s_1(t) \) and \( s_2(t) \) from \( s(t) \) over several octaves.

Alternative multidimensional signal representations, such as \( s(t) \), \( s(t + \tau) \) and \( s(t) \), \( ds(t)/dt \), have drawbacks in respect of one or more of the four properties.
AUTOMATIC SYNTHESIS OF THE PITCH CONTOUR OF GERMAN SENTENCES

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For speech synthesis by rule, there is an interest in an automatic production of the prosodic parameters, especially of the pitch contour.

An automatic procedure has been developed, using rules for sentence accent, for segmentation of a sentence into phrases ("phrasing"), and for pitch contour given by Heidolph an Kiparsky and by Bierwisch. As input, the orthographic text and the syntax of the sentence are required. Then sentence accent, "phrasing", and the pitch contour are formed. At last, the pitch levels are converted into frequency contours, using additional rules. The way of phrasing with its great effect on the melodic contours depends on speaker and speaking situation with great variability. The exact positions of the boundaries, however, depend on the syntax.

In the process of testing the naturalness of the synthesized contours, we are confronted with the difficulty that it depends on other parameters too, as for example timing, amplitude, and fast variations of the fundamental frequency (for instance "intrinsic pitch"). Therefore in control experiments one should prefer vocoded natural speech rather than a completely synthesized one, only substituting the slowly changing components of its fundamental frequency by the synthesized ones. For this superposition of natural speech and a synthetic pitch contour, a mapping function is required from the "orthographic" time axis of the synthetic pitch to the natural one of the speech. An automatic procedure for this purpose has been developed, by using only the position of the boundaries of the voiced parts in the sentence.

The procedure of using natural speech with a synthetic pitch contour performs well sounding vocoder sentences.
In previous papers 1,2 we discussed a speech synthesis by rule scheme where segments obtained from natural speech were linearly concatenated. These segments included the consonants and the transitions from consonants to vowels, vowels to vowels, and vowels to consonants. Each synthesis parameter was defined by few sets of LPC area parameters, and in the concatenative process, straight line interpolation was used to obtain the complete set of area parameters. Informal listening and some formal intelligibility testing revealed that this simplified description of the synthesis segments was not sufficient to produce the speech quality that would satisfy us. Consequently, it was decided to improve the definition and selection of the concatenative units. Specific improvements include: (1) better selection of tokens from which the concatenative units are obtained; (2) the number of phonetic segments defining a synthesis unit can vary from one to three; (3) arbitrary number of area parameter vectors can be stored for a given synthesis unit; and (4) a facility for expressing transition durations and amplitude rules for the synthesis of dyadic units. This paper will discuss in detail the scheme for synthesis employing these concatenative units.


THE SYNTHESIS OF ESTONIAN CONSONANTS

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In Estonian language there are the following constants: unvoiced plosives /p,t,t',k/, semivoiced plosives /b,d,d',g/, unvoiced fricatives /f,s,s',h/, voiced fricatives /v/, nasals /m,n,n',/ laterals /l,l'/, trill /r/ and semivowel /j/.

Some consonants have palatalized forms, which are marked with apostrophe. They are synthesized by means of i-like transitions between the preceding phoneme and an unpalatalized form of a consonant. The phonemes /b,d,d',g/ are shorter and less intensive variants of /p,t,t',k/ and when they are standing between vowels, there is very often a short voicing in the initial phase of pronunciation.

The vowels and most of the consonants have three degrees of quantity. The duration of silence when /b,d,d',g/ are pronounced is at an average 20-100 mc, while the duration for /p,t,t',k/ (the second quantity degree) is at an average 100-200 mc, and for the same phonemes in third quantity degree it is 200-400 mc, while for /d,d',t,t'/ we have the highest and for /g,k/ the lowest limits of these time intervals. The duration of explosion is shorter for /d,d',t,t'/ (6-12 mc) and longer for /g,k/ (10-25 mc and more). When synthesizing the plosives, the choice of right duration of silence is more important to have needed phoneme then the intensiveness of hiss generator. The spectrum of the noise have the concentration in the limits of 300-1000 Hz for /b,p/, 1.2-2.5 kHz for /g,k/ and 2.5-4.5 kHz for /d,d',t,t'/.

Besides in each explosion there is also some less intensive components of noise in the lower and higher frequency. The bendings of formants of vowels and voiced consonants preceding or following the plosives depend on the frequency of formants of them and the central frequency of explosion of the plosives. The central frequency of the spectrum of /h/ depends on the frequency of the formants of the neighbouring vowels and voiced consonants. The frequency of the spectrum of the noise of the isolated /h/ is concentrated in the limits of 200-2000 Hz. The lowest cutoff frequency of the spectrum of fricative /s/ for male voices is at an average 3,3 kHz and for female voices 4,0 kHz. When following the vowels or voiced consonants, in the spectrum of /s/ remains the frequencies of highest formants of these phonemes.
Due to the very complex nature of the problem, automatic speech recognition systems operating on continuous speech make errors at various stages. This paper is an attempt to study how the errors made at the segmentation and parameter estimation stages affect the performance of the recognition system.

An experiment is described where the vowels in continuous speech are recognized using their steady-state formant frequencies as recognition parameters. The speech data used consists of forty English sentences spoken by a single speaker. The duration of each sentence is between three to four seconds. The steady-state vowel segments are detected manually by visually examining the digital spectrograms and analysed using the selective linear prediction method to obtain the power spectra. Formants are extracted from these power spectra by an algorithm similar to Markel's [1] except for the decisions about spurious peaks and merged formants which are made here manually. The effectiveness of five different classifiers (in classifying the steady-state vowel segments into ten vowel classes) is studied. In general, the more complex classifiers (the Bayesian classifier is the most complex used) perform better.

Recognition performance of the system using the Bayesian classifier is found to be 77.8% when the training and test sets are the same and 72.8% when they are different. The performance of the recognition system deteriorates by more than 30% when the segmentation and formant extraction errors made by the system are not corrected manually [2,3], thus emphasizing the need for improving the segmentation strategies and parameter estimation techniques.

REFERENCES


APPLICATION OF ARTICULATORY MODEL FOR SPEECH RECOGNITION

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1. INTRODUCTION

Major difficulties which exist in the machine recognition of speech are the large variability of each utterance and the difference between talkers. Most of the feature extraction methods ever studied use the acoustic parameters calculated rather directly from speech waves. However, in order to normalize or adapt for the wide variability of speech waves, it is necessary to apply more effectively the fundamental properties of the speech production process. Several efforts have been made for the calculation of the vocal tract shape from speech waves. The method of the estimation has not been established satisfactorily, but it has been shown that the estimated articulatory parameters are useful feature for speech recognition.

We have proposed a method to estimate the articulatory motion.* In that method, the articulatory model is constructed by statistical analyses of real data and the state of articulation can be described very precisely by the small number of parameters. Using that model, it becomes possible to estimate the articulatory parameters by nonlinear optimization technique.

In this paper, our recent advances in the estimation method and its application for speech recognition are presented.

2. ESTIMATION OF ARTICULATORY MOTION FOR CONSONANTS

The estimation method can be applied not only for vowels but for nasals and other consonants. In case of nasals, an articulatory model containing the nasal tract is prepared. Since the shape of the nasal tract is different for each speaker, before the estimation, the nasal tract shape should be adjusted using nasalized sounds. After the adjustment, the articulatory movement is successfully estimated for nasals and /m/ and /n/ are discriminated by the trajectories of the articulatory parameters.

3. NORMALIZATION OF ARTICULATORY PARAMETERS

Second, a normalization procedure of the articulatory parameters for each speaker is discussed. There are two kinds of strategies for the normalization. One is to modify the articulatory model itself and the other is to obtain the transformation of the set of articulatory parameters. The latter method is simple and practical in the real situation. Linear transformations for female data and for those of children which translate positions of five vowels nearest to those of male voices are calculated by the least square method. It is shown that the normalized parameters are useful for the discrimination of vowels and several kinds of consonants.

In conclusion, the articulatory parameters estimated from speech waves are effective enough especially for the talker independent speech recognition.

* K. Shirai and M. Honda, Feature extraction for speech recognition based on articulatory model, Proc. of 4th Int. J. Conf. on Pattern Recognition (1978)
COMPARISON OF WORD RECOGNITION SYSTEMS WITH DIFFERENT DECISION UNITS

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Automatic speech recognition primarily starts with a segmentation of the speech signal into basic elements which are related to a limited number of decision units. Although many isolated word recognition systems use complete words as decision units the segmentation into smaller units can be necessary if a large vocabulary has to be processed. It has been outlined in a previous paper /1/ that parts of syllables (demi-syllables) can be used for this purpose. In order to investigate the efficiency of syllabic segmentation methods, several word recognition systems have been developed which start from the same preprocessing of the speech signal but use different segmentation procedures. The preprocessing is carried out by a special loudness analyzer which yields 22 specific loudness functions covering a frequency range of 50 cps to 8.5 kcps arranged on a critical band rate scale /2/. The sampled data of these functions represent the input pattern of the spoken utterance.

WORD RECOGNITION SYSTEMS

Four word recognition systems have been implemented which can be sketched by the following characteristics:

1. No segmentation; linear time normalization of the word pattern; classification of the total pattern.
2. No segmentation; normalization by a special dynamic interpolation; classification of the total pattern.
3. Segmentation into demi-syllables; normalization of the demi-syllable patterns; classification of the total pattern.
4. Segmentation into demi-syllables; normalization of the demi-syllable patterns; classification of the demi-syllables into vowels and consonant clusters.

The demi-syllable segmentation can be achieved evaluating the total loudness and a modified loudness function which is calculated as a weighted sum of all 22 channels. In contrast to the first three systems which recognize the total word patterns, the fourth system classifies the consonant clusters of the demi-syllables independently of the adjacent vowel. In this system, even with a large vocabulary the inventory of consonant clusters and vowels is limited.

EXPERIMENTS

All systems have been tested with a vocabulary consisting of 237 names of German cities. Additionally, first results have been obtained using a list of about 1000 German words. It is possible now to compare the efficiency of the different systems and to estimate the influence of the segmentation procedures with respect to the size of the vocabulary.

REFERENCES:
ANALYSIS AND RECOGNITION OF VOICED STOP CONSONANTS

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Although formant loci are known to be important in the identification of voiced stop consonants, no effective methods have been established for their utilization in the automatic speech recognition. The present paper describes (1) a method for extracting rapid formant transition, and (2) a method for estimating formant loci of voiced stop consonants for automatic recognition.

EXTRACTION OF FORMANT TRAJECTORIES BY SHORT-TERM LPC ANALYSIS

Because of its advantage in short-term analysis, the covariance method of LPC analysis was adopted for extracting formant frequencies. Synthetic voiced stop consonants with simulated glottal source waveform and formant transition were used to determine the optimum window size for analysis, which turned out to be approximately 3 msec. It should be noted, however, that the errors in formant frequencies and bandwidths can be quite large if the placement of the analysis window relative to the excitation waveform is unfavorable, and a method is necessary to eliminate data points with gross errors and to select those poles that correspond to vocal tract resonances. This was accomplished on the basis of bandwidth and continuity in frequency of extracted poles. Figure 1 compares the waveform and formant frequencies (x) thus determined at 1 msec intervals from a natural utterance of the syllable /be/.

DETERMINATION OF FORMANT LOCI FOR AUTOMATIC RECOGNITION

Formant frequencies extracted from the initial 40 msec-segment of a CV-syllable were used to estimate the formant loci by extrapolation. The extrapolation was based on an approximation of an observed formant trajectory by the step response of a first-order linear system. The loci of the second and third formants are shown in Fig. 2 for five utterances each of the voiced stops followed by the vowel /e/. The results of analysis indicated that the three voiced stops /b/, /d/, and /g/ can be clearly separated if the vowel context is known (i.e., if we first recognize the following vowel and utilize the information for classifying the three consonants).

![Fig. 1. Estimation of formant loci from formant trajectories of a natural utterance /be/.

![Fig. 2. Distribution of formant loci for /b/, /d/, and /g/ in /-e/ context.](image-url)
Coarticulation Across Word Boundaries

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In order to achieve good recognition scores at the acoustic-phonetic level of continuous speech recognition, phoneme recognition algorithms must be capable of dealing with coarticulation effects. An experiment will be described which investigated coarticulation across word boundaries. VCV combinations were extracted from continuous, conversational speech. All possible VCV combinations in which the consonant was a plosive and the vowels were /i/ and /ɔ/ and in which the word boundary could occur either before or after the consonant were examined for ten speakers. It was found that there were strong coarticulation effects across word boundaries. In particular the degree of coarticulation depended on the place of articulation of the consonant. Thus coarticulation effects were more noticeable for the velar plosives than they were for the bilabial and alveolar plosives.
AUTOMATIC DETECTION AND RECOGNITION OF VOICELESS STOP CONSONANTS
BASED ON THE SPECTRAL CHARACTERISTICS TRANSITIONS

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At the present time, how to extract acoustic features of voiceless stop consonants is one of the most difficult problems remaining unsolved in the field of automatic speech recognition. This paper describes a study on characteristic features of the voiceless stop consonants /p/, /t/ and /k/ and on their dynamics in the feature space.

A multi-dimensional statistical analysis method is applied to extract feature parameters. Two physically significant feature parameters are obtained. One of them is equivalent to the first order spectral moment. It gives the feature of the global distribution of energy. The other is the output of a "matched filter" for the spectral distribution of the consonant /k/. These two feature parameters make it possible to separate the stop consonant from each other.

It is said that formant transitions are important acoustic cues for the perception of consonants. Such transitions must be used positively in the automatic recognition algorithm. Though, it is very difficult to extract the formant transition. This study shows characteristic transitions of the feature parameters in the feature space. The dynamics of the feature parameters are equivalent to the formant transitions. Since the feature parameters are obtained from the whole spectral pattern of the consonant, their transitions are extracted more easily and stably than the formant transitions. A matched filter to detect the voiceless stop consonants is determined using the dynamics of feature parameters, which gives the instant of the noise onset of the voiceless stops. Experiments of automatic detection of them show that the matched filtering is a very useful method.

Experiments of automatic recognition based on the feature parameter dynamics are performed. It is found that a recognition rate as high as about 97% can be attained.
INFLUENCE OF MICROPHONE POSITION IN THE RECORDING OF SPEECH
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Sound pressure levels of various speech sounds are measured simultaneously at different distances from the mouth. The observed values for low vowels and for /s/ differ especially close to the mouth from those predicted from the distance law for sound radiation. The variation of sound pressure with distance seems to depend on the speech sound in question. Some consequences hereof are pointed out. The results are compared with calculated values of the sound pressure from different sound sources on a rigid sphere. Some of the observed deviations from the distance law seem attributable to the different frequency composition of the speech sounds. However, some of the observations (e.g. a difference between the variation of low and high vowels at positions close to the mouth) cannot be accounted for by the models.
PROCEEDURES FOR EQUATING AMPLITUDE OF SPEECH IN INTELLIGIBILITY TESTING
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There are a number of methodological problems that remain unsettled in the development and recording of speech materials for use in the testing of receptive communication. The problem of variance due to uncontrolled acoustic and linguistic factors can be overcome with statistical evaluation on sufficient test runs on adequate numbers of subjects. However in many applications of speech intelligibility testing, such as audiological clinical testing, there is a need to test individual performance reliably and in a short period of time. The solution to the problem of reliability for individual results is dependent on improved control of those factors which make inter-list results variable. Although there are a large number of contributing factors, one major source of variance is the difference in perceptual salience between words in the lists.

This paper discusses the arguments for and against the use of amplitude normalisation procedures, or the equalisation of the amplitude of speech in a list, and presents a comparison of techniques used to perform amplitude normalisation. The techniques investigated include VU measures, peak amplitude and impulse level measures and the A-weighted Leq technique.

Results of perceptual tests are presented for non-normalised and normalised word lists using different techniques. The decisions about which criteria to adopt are discussed in terms of the situation requirements for the tests.
A RE-EXAMINATION OF SOME DYNAMIC PROPERTIES OF SPEECH

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Measurements were undertaken to re-examine certain widely held assumptions about the dynamic properties of speech. These are: that, for most practical purposes, speech can be regarded as having a dynamic range of 30dB, extending from 18dB below the r.m.s. level to 12dB above this level; that this dynamic range, and the relationship of r.m.s. to peak level, is relatively constant across the frequency range which is important for understanding speech. These assumptions are used for various purposes such as in calculating the Articulation Index and in relation to hearing aid selection and evaluation.

The above assumptions were rather critical for some studies in which we were evaluating the use of frequency shaping, compression and expansion in a multi band amplification system. Although there was no specific reason for questioning the assumptions, they appear to be based exclusively on data from a 1940 study and it was, therefore considered desirable to check their justification.

A 16 second segment of running speech was subjected to a one-third octave band real-time analysis, with an integration time of 16 seconds. A similar analysis, but with an integration time of 125 msec. was performed on each consecutive 125m sec of the speech. For each one-third octave band, distributions were obtained for the levels measured in each 125m sec. segment of speech. Approximately 100 segments were analysed, after excluding small sections at each end of the tape recorded speech. These measurements were performed with four 16 second segments of speech recorded by different talkers (2 male and 2 female). The results of this study agreed quite well with the above assumptions and indicated that they were acceptably accurate even for our relatively critical purposes.

In addition, the integration time was reduced from 125ms to 10ms to determine its effect on the measured dynamics. At all frequencies, no change in the relationship between peak and average one-third octave band levels was noted as the integration time was changed. This indicates that the peak speech components arise from signals having durations in excess of 125ms. Subsequent analysis showed that the peaks are caused by vowels in the low and middle frequency regions and mainly by fricatives in the higher frequency regions.
AN ANSWER TO THE "COCKTAIL PARTY PROBLEM"

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INTRODUCTION

Speech signal separation in a multi-speaker environment, the so-called "Cocktail Party Problem", is not only pragmatically important but theoretically interesting. Here presented is a theoretical solution to the problem. Its implementation in a practically useful device will depend upon whether social circumstances are such that participants in cocktail parties are willing to accept severe constraints on their displacements in the anechoic cocktail lounge. The technique suggested is the same that we have previously proposed for sound separation using the generalized inverse of matrices in the frequency domain.

FORMULATION OF THE METHOD

The basic idea of this method is sound source separation[1] or source sound restoration[2] in a multi-source environment. This can be briefly explained as follows: Let $S(\omega)$ denote the vector function composed of the Fourier transforms of the speech waves from $n$ speakers, and $P(\omega)$ that of the waves received by $m$ microphones. Then the following equation holds:

$$P(\omega) = A(\omega) S(\omega)$$

where $A(\omega)$ is an $m \times n$ matrix with elements of the form $d_{ij} e^{-j\omega \frac{d_{ij}}{c}}$, $d_{ij}$ is the distance between microphone $i$ and speaker $j$, $d_{ij}$, and $c$ denotes the sound velocity in the free field.

The least-squares solution in the time domain can be obtained as

$$\hat{s}(t) = \mathcal{F}^{-1}[A^*(\omega)P(\omega)]$$

where $\mathcal{F}^{-1}$ denotes the inverse Fourier transform. The estimated speech of the $j$'th speaker is obtained as the $j$'th element of $\hat{s}(t)$.

COMPUTER SIMULATION AND DISCUSSION

The method achieves perfect results for stationary speech if all the distances among microphones and speakers are exactly known. However, it presents some difficulties in the transient parts of speech. Investigations of perceptual effects of the method will be carried out in the future.


SUBJECTIVE PERCEPTION OF NOISY SIGNALS
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In any PCM or ADPCM system with the quantizer in the feedback loop, the quantizer error is a white noise signal. As the speech spectrum is not white, the S/N ratio can be low at different frequencies, depending on this spectrum. This disadvantage can be minimized by modifying the spectrum of the quantizer noise using one or two filters in certain feedback loops at the transmitter. So we use the masking properties of hearing, with speech as masker signal, to minimize the subjective perception of quantizer noise. We measured the masked threshold of noise bursts, masked by one or several tones, as a function of the frequency $f_m$ of the masker. These measurements have been carried out for noise bursts of various center frequencies $f_c$ and bandwidths and for various intensities $L_m$ of the masker signal. As one of several important results, the maximum masking frequency is, for low and medium masker level, higher than $f_c$ and decreases below $f_c$ at very high masker level ($L_m$ over 80 dB). At the center frequency, the masked threshold is 20-26 dB below $L_m$ and increases for increasing $L_m$. If the masker consists of two tones, the increment of the noise threshold relative to the "single" masked threshold is important. We found this increment to be a function of the difference $L_1-L_2$ between the single masked thresholds, the difference between the two masker frequencies and the difference between the masker frequencies and the noise center frequency. The greatest increments are measured if $L_1-L_2=0$; they can reach 12 dB. We also tested three-tone complexes as masker, frequencies and amplitudes of the masker signal corresponding to vocal formants. In this case, also used the results for one and two tone masking, a wide-band noise burst will be modified to be as much as possible inaudible.
Nous avons étudié l'effet de bruits impulsifs et de bruits continus sur la compréhension et la mémorisation de textes lus. Les bruits continus étaient constitués d'un bruit rose filtré entre 2000 et 6000 Hz dont le niveau était, selon les groupes de sujets, de 40 dB A (SPL) ou de 80 dB A (SPL). Les bruits impulsifs étaient obtenus en ajoutant des impulsions régulières (30 impulsions/minute pour le bruit impulsif n°1 et 60 impulsions/minute pour le bruit impulsif n°2).

Les 70 sujets qui ont servi pour les expériences ont été exposés au bruit continu faible lors d'une séance et à l'un des trois autres bruits lors d'une autre séance. L'exposition au bruit durait une heure pendant laquelle 20 minutes étaient consacrées à la lecture de 4 textes différents d'une page chacun. Après l'exposition, les sujets remplissaient un questionnaire permettant d'évaluer la compréhension et la mémorisation des textes.

Le bruit continu fort et le bruit impulsif n°2 diminuent la performance des sujets. Le bruit continu faible et le bruit impulsif n°1 n'ont aucun effet.

Nous discutons divers facteurs pouvant expliquer ces résultats.
Adaptive Predictive Coders (APC) with two predictors for formant and pitch description, respectively, achieve a good result, even if a one-bit quantizer is used. This result is attained by shaping the quantization noise for optimal subjective performance. It raises the question, if there are equally successful methods to quantize the residual signal with less than 1 bit/sample.

A simple method was used to obtain effective bit rates of less than 1 bit/sample: Depending on the bit rate, larger or smaller signal intervals with energy as high as possible were selected for transmission with 1 bit/sample; the rest of the signal was set to zero. The real problem was found in developing a suitable APC-system. The growing quantization noise caused by the lower bit rate results in an instable system, even if the predictor filters are stable.

Including the noise shaping filter or other kinds of feedback filters in the computation of predictor coefficients, one gets a new prescription for determination of the coefficients. The new equation system, in which not only terms of the input signal appear, can be solved by iteration. The APC-system becomes stable now. A low bit rate (e.g. less than 0.5 bit/sample), however, causes a low gain of the predictor and therefore a worse speech quality. In order to get better results at low bit rates, a system was found which in the limiting case of zero bit rate becomes a vocoder. For this purpose the pitch predictor was replaced by a feedback loop, generating a periodic signal P. The excitation signal for the formant filter was gained by adding a scaled version of P to the quantized residual signal. Furthermore there is a mechanism to renew the signal P out of the same quantized signal.
NOISE REDUCTION OF THE SPEECH SIGNAL BY ADAPTIVE KALMAN FILTERING

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INTRODUCTION

Optimal extraction of the signal from noise must be based on the estimation of the signal characteristics. In particular, noise reduction of speech requires adaptive techniques because of temporal variations of acoustic characteristics of speech. This paper presents an adaptive Kalman filtering technique when the additive noise is a white Gaussian noise with an unknown variance.

PRINCIPLE OF THE METHOD

The process of speech production is approximated by an autoregressive moving-average process described by a state equation and an observation equation. In the state-space representation of the system, it is assumed that the input signal is either a train of volume velocity impulses or random pressure fluctuations and the system parameters are unknown. The problem is to estimate the speech signal from an additive noise environment. The state transition matrix that characterizes the autoregressive scheme is determined from the short-term autocorrelation function of the speech signal. Using the estimated state transition matrix, the speech signal can be estimated by the Kalman filtering. The optimal Kalman gain is then directly estimated without estimating covariances of the speech signal and the measurement noise.

EXPERIMENTAL RESULTS

The speech signal was first low-pass filtered at 4.8 kHz and then sampled at a frequency of 10 kHz. Figure 1 shows an example of estimation of the speech signal uttered by a male speaker in a noisy environment where the SNR is about 3 dB for the voiced segments. The result clearly indicates the effectiveness of the present method. Subjective listening tests were also conducted to optimize some of the system parameters.

REFERENCE


Fig. 1. An example of estimation of the speech signal /korekara raigetsu/ uttered in a noisy environment by a male speaker.
Dans un cadre plus vaste de reconnaissance de la parole, nous proposons dans cet'article un module de reconnaissance acoustico-phonétique fondé sur l'extraction des indices et des traits phonétiques. On utilise comme analyseur acoustique un modèle d'oreille qui permet d'accéder avec une bonne fiabilité aux paramètres acoustiques tels que : formants (amplitude, fréquence, transition), énergie du signal filtré, fréquence du fondamental, rapport grave/aigu. Ces paramètres servent dans un deuxième temps à calculer des indices instantanés tels que : indice grave/aigu, friction, nasalité, explosion, occlusion, "burst", "buzz", fermé/ouvert, compact/écarté, avant/arrière, diésé/non diésé, vocalique/non vocalique, voisé/non voisé. On construit alors une fonction d'instabilité à l'aide de tous ces indices hiérarchisés. Cette fonction permet de découper le signal en : blocs stables, blocs frontières et blocs instables. De tels segments ne sont encore que des étapes vers le segment phonétique, ils sont faits de blocs homogènes sur un plan acoustique. Ces segments sont maintenant considérés dans leur succession temporelle afin de pouvoir les regrouper ou les scinder en phonèmes. Pour cela on calcule de nouveaux indices à long terme, extention des indices instantanés. Une table de décision à score majoritaire permet d'orienter les segments vers les grandes classes acoustico-phonétiques : silence, occlusion, explosion, friction, voyelle, vocalique, transition longue, transition courte, fluidité, nasale, consonne. La classe obtenue pour chaque segment renvoie à une procédure spécifique de reconnaissance qui permet d'identifier les blocs contenus dans le segment. Ces procédures de reconnaissance utilisent en particulier les paramètres acoustiques et les indices instantanés (nuages dans \( \mathbb{R}^n \), \( n \geq 6 \)). Les informations ainsi construites sont synthétisées par un automate lexical qui procède à la segmentation finale en phonèmes et à la reconnaissance phonétique définitive.
REAL-TIME SPEECH RECOGNITION BY COMPUTER

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Automatic speech recognition based on a minicomputer (Varian 620/L) is described. The input speech is preprocessed by the real-time 1/5 oct. band frequency analyzer (B/K 5347). An idea of spectrum stimulus is established, upon which the nonlinear normalization of speech pattern in time domain is realized by a simple linear method. The transient sound is especially emphasized in our model. To compress the information of reference sounds, a new adaptive technique is used to obtain the binary spectrum. The EOR distance is adopted to measure the difference between the two binary spectrum to be compared. For each reference spoken command there is only one template in memory.

The results of recognition for a specific speaker are as follows: 10 spoken Chinese digits - 99.7%; 20 sentences (7 syllables for each) - 99.7%; phrases (4 syllables for each): 100-99.5%; 150-99.3%; 200-98.85% (in real time). 400-97.7% (in approximate real-time).

A voice control system VCCV620 has been implemented with a vocabulary of 31 spoken commands having various syllables. It is used to control the computer calculating +, -, x, /, and ↑. It has demonstrated for visitors over 100 times successfully. It can run perfectly in English as well as in Chinese.
TWO SYSTEMS FOR AUTOMATIC WORD AND SPEAKER RECOGNITION
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Calavrytinos, Pantelis; Lewien, Thomas; Helling, Detlef;
Strube, Hans Werner; Schroeder, Manfred R.

In word recognition, experience has shown the difficulties of making the correct segmentation. This problem can be avoided, if the parameters describe a whole word. In our first investigation, the short-time spectrogram of a whole word is estimated as a 14 x 14 matrix M. The so called "singular value decomposition" is then used to reduce this matrix [1]. This method reduces M to a sum of dyadic products of singular vectors, weighted by decreasing singular values. The parameters used are the first three pairs of singular vectors.

For the recognition, the ten numerals 0 to 9 of the German language are used, spoken by 21 male speakers. Word recognition is realized using several classifiers. We obtained the best results, over 98 % recognition rate, with a word-dependent maximum-likelihood classifier.

In another investigation, the word parameters consist of 39 two-dimensional modified cosine-coefficients of the short-time spectrogram of the word. The same parameters are used for word as well as speaker recognition. In speaker recognition all 21 speakers are identified correctly. For both systems, if the word recognition is made in a speaker-dependent way, the word recognition rate exceeds 99 %. The speaker identification system will now be tested for a greater number of speakers. After computing a great number of cosine coefficients, we also test a "knock out" strategy to select the best parameters for the speaker identification.

A SPOKEN WORD RECOGNITION SYSTEM FOR UNSPECIFIED SPEAKERS

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High rate of correct recognition is known to be attainable in spoken word recognition by registration of word templates from individual speakers. No method has yet been established, however, for the recognition of spoken words from unspecified speakers with an accuracy sufficient for practical use. This paper describes a new method for speaker-independent recognition of spoken words, based on an extension of the principle\(^1\) and a hardware system\(^2\) previously proposed by the authors utilizing word templates from individual speakers.

**PRINCIPLE**

The present system utilizes four characteristic parameters, viz. vowel parameters \(M_1, M_2\) and consonant parameters \(X_1, X_2\), to conduct pattern matching between an unknown input word and registered words. The parameters \(M_1, M_2\) correspond to first and second formant frequencies, respectively. The major part of individual differences of vowels can be eliminated by linear frequency warping.\(^3\) Thus the speaker adaptation is accomplished in the present system by linear frequency scaling of the vowel parameters, as shown in Fig. 1. The validity of the principle has been demonstrated by computer simulation.

**HARDWARE SYSTEM**

The hardware system for real-time recognition is composed of three parts. The first part is a filter bank for frequency analysis of input speech. The second part (voice parameter extractor) calculates the four parameters, and the last part (speech recognition unit) is for the registration of word templates and for matching of an unknown input word against stored templates. The latter two parts are constructed using microprocessors and microprogramming techniques for high-speed processing. The performance of the total system for real-time recognition is being tested by using various vocabularies including railway station names and spoken digits.

**REFERENCES**


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![Diagram](attachment:diagram.png)

Fig. 1. A speaker-adaptive recognition system for spoken words.
Bei der Auswertung von Sprachaufzeichnungen, aber auch bei Verhaltensfor-
schungen über Lautäußerungen von Tieren ist es häufig notwendig, den Hörr-
eindruck exakt dem Signal zuzuordnen und diese Zuordnung zu dokumentieren.
Hierzu wurde ein Programmsystem für einen Minicomputer (PDP 11) mit Platten-
speicher und Graphik-Prozessor entwickelt, das eine interaktive Bearbeitung
derartiger akustischer Signale auf visueller und auditiver Basis ermöglicht.
Folgende Anwendungen wurden bisher erprobt:
Textübertragungen von stark gestörten Gesprächsaufzeichnungen, z.B. aus dem
Funksprechverkehr bei Flugzeugunfällen;
Analyse von Diskussionsgesprächen für phonetische Zwecke mit Auswertung des
Pegei- und Tonhöhenverlaufs sowie der Pausendauer;
statistische Auswertung der Lautäußerungen von Hühnern bei unterschied-
lichen Arten der Käfig- und Freilandhaltung.

Die Signale werden zunächst kontinuierlich mit einer maximalen Taktfrequenz
von 40 kHz abgetastet, mit 12 Bit Auflösung digitalisiert und automatisch
auf Magnetplatten gespeichert. Die akustische Wiedergabe der Signale und
die Darstellung auf einem Bildschirm kann kontinuierlich oder in frei wähl-
baren Ausschnitten erfolgen. Auf dem Bildschirm wird in der oberen Hälfte
ein Signalausschnitt von bis zu mehreren Sekunden Dauer dargestellt, aus
dem Teile ausgewählt werden können, die mit größerer Auflösung in der unter-
ren Bildschirmhälfte zur Darstellung kommen. Mit Hilfe von insgesamt 7 über
ein Tastenfeld steuerbaren Lichtzeigern kann die Auswahl von Signalabschnit-
ten erfolgen, deren Darstellung jeweils optimiert werden kann, die abgehort
oder auch herausgelöscht werden können, z.B. wenn es sich um Störungen
handelt.

Unterprogramme können über Tasten aufgerufen werden, die z.B. den Tonhöhen-
der oder den Pegelverlauf berechnen und ihn am unteren Bildschirmrand in exak-
ter Zuordnung zum Zeitsignal zur Darstellung bringen. Sprachsegmente können
einzelmarkiert werden, der auditiv erkannte Laut kann auf dem Bildschirm
über die Bedienungstastatur eingetragen werden.

Entsprechende Eintragungen erfolgen automatisch auch bei dem auf der Magnet-
platte gespeicherten Zeitsignal, indem der jeweils 4 Bit umfassende freie
Speicherplatz, der bei der Abspeicherung der 12-Bit-Datenworte in 16-Bit-
Speicherplätzen entsteht, ausgenutzt wird. Es läßt sich daher nach Abschluß
der interaktiven Bearbeitung eine automatische Auswertung vornehmen. Der zu-
geordnete Text kann über einen Drucker ausgeschrieben werden, er läßt sich
aber auch zusammen mit dem Zeitsignal auf einen Streifenschreiber, dem ein
Druckkopf aufgesetzt ist, ausgeben. Über einen mehrkanaligen D/A-Wandler
können abgeleitete Größen, wie z.B. der Tonhöhenverlauf, gleichzeitig
dokumentiert werden. Es lassen sich u.a. auch Markierungen der Pausen auto-
matisch statistisch auswerten, indem z.B. die Häufigkeitsverteilung berech-
net wird.
RECOGNITION OF STOP CONSONANTS

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1. INTRODUCTION

The manner of articulation of stop consonants forms a silent or buzz part and burst noises in speech waves. While, the place of articulation is reflect-
ed in the vocal tract shape and frequency spectra. We have tried to use these characteristics for segmentation and discrimination. In order to avoid effects of coarticulation we discriminate them knowing the following vowel. It seems to be very troublesome, but our methods do little depend on individuality.

2. SEGMENTATION OF STOP CONSONANTS

i) Segmentation of unvoiced stops: A frame (10 msec interval), where speech wave amplitude becomes large after silence, high frequency components become dominant, similarity to vowels and nasals is low and also it has a similar shape to some stop consonant, is decided to be an unvoiced stop.

ii) Segmentation of voiced stops: In the acoustic analyzer of our speech recognition systems, buzz parts are recognized and stationarity of speech waves is estimated for every frame. After a buzz sequence, the stationarity is much disturbed at the instant of explosion, so we can find out the frame of explosion of a voiced stop consonant.

3. DISCRIMINATION AMONG UNVOICED STOP CONSONANTS

At the onset frame segmented as an unvoiced stop, a pseudo vocal tract shape is estimated. Some characteristic parameters $f_x^i$ ($x=$/p,t,k/, $i=1,2,...n$) are decided from the average vocal tract shape of each stop consonant preceding each vowel. If an estimated shape has a characteristic $f_x^i$ we put $c(f_x^i)=1$, otherwise $c(f_x^i)=0$. Similarity $y_x$ of an input phoneme to a phoneme $x$ is

$$y_x = \frac{1}{\sum_{i=1}^{n} w_i^1} \cdot \sum_{i=1}^{n} \frac{w_i^1}{x_i} f_x^i,$$

where $w_i^1$'s are weights and do little depend on individuality.

Final decision is done chiefly using $y_x$ and occasionally taking account of some other characteristic features such as a fact that the duration of /p/ in /pe/ is shorter than those of /k/ and /t/ in /ke/ and /te/.

4. DISCRIMINATION AMONG VOICED STOP CONSONANTS

In the first step we discriminate among voiced stops using ratios of power spectra and stationarity of speech waves. In the second step we use a pseudo vocal tract shape at the segmented frame in the same way as for unvoiced stops. Finally we discriminate them combining the results obtained in both the first and the second steps.

5. RESULTS

Discrimination rates of unvoiced and voiced stops are 86.2 % and 83.3 % respectively for stop consonants (in which the ones that have not been segmented are included) contained in each 56 words uttered by 8 adult males.

REFERENCE

The modern speech encoding procedures, originally developed in order to improve the efficiency of speech and video signal transmission, can also be used for an amelioration of the existing subscriber-oriented services and for an introduction of new ones. These new techniques can be demonstrated by the example of the automation of inquiry services. This procedure involves three different problems: The automatic recognition of the request spoken by any subscriber (male or female), an appropriate response spoken by a speech synthesizer and the automatic transcription of normally written texts into a string of phonetic symbols to be used by the synthesizer. Additional difficulties for speech recognition arise from the fact that the conversation between man and computer is band limited (300 - 3400 Hz). A survey will be given of the attempts at speech recognition; and the limitations of the existing solutions of speaker-independent speech recognition procedures will be described. But as the telephone as a terminal is almost everywhere available, it is possible to use dial or push buttons for transmitting the subscriber's request in a coded form. This paper also reports on different methods of speech synthesis, depending on the size of the vocabulary, the desired speech quality and the existing multichannel hardware models. Some tape recordings will illustrate the quality of the synthetic speech which can be used in the commercial, scientific, industrial, and social fields. Finally, a report will be given on the activities aimed at an automatic transcription of the different languages in a phonetic writing suitable for the computer and on the efforts made to obtain the rules for automatically describing the intonation and grammar of the different languages.
Speaker verification experiments have been made on utterances of 8 male speakers. Spoken words were Japanese digits 4/yon/ and 0/rei/ and each digit was uttered 8 times by each speaker. Each utterance was analyzed by linear prediction and time series of predictor coefficients $a_k$, PACOR coefficients $K_k$ and log area ratio $L_k$ were obtained ($k=1$-10).

The utterance of the time length closest to the mean for each digit of each speaker was selected and the other 7 utterances were matched to the selected utterance by dynamic programming\(^1\). A reference spectrum time pattern for each digit of each speaker was obtained by averaging his matched time series of spectrum parameter coefficients at each time section.

Eight utterances were matched again to his reference pattern by D.P. and variances of coefficients were calculated for each corresponding time section. Variances of each coefficient vary in time considerably as seen in Fig. 1.

Time axis of each utterance was warped by D.P. matching. This time warping is measured in terms of the time warping strain defined by.

$$\epsilon_j(i) = \{j(i)-j(i-1)\}/2 + \{j(i+1)-j(i)\}/2,$$  \hspace{1cm} (1)

where $j(i)$ is the number of time section of the test pattern, matched to the $i$-th time section of the test pattern.

The mean and variance of $\epsilon_j(i)$ at each time section were calculated for 8 time patterns matched to the reference and Fig. 1 is one of examples.

Taking into account the variances of spectrum coefficients and $\epsilon_j(i)$ at each time section, speaker verification performance is improved. And it is shown that any choice of spectrum parameter coefficients $a_k$, $K_k$ and $L_k$ gives nearly the same verification performance.


![Fig. 1. Time patterns of variances.](image)
Spectral speaker adaptation by matrix transformation for automatic speech recognition

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There are several ways in which an automatic speech recognition system can be made suitable for use with several different speakers. Assuming that the classifier has been trained on a reference set one possibility is to transform all the test speaker's utterances so that they are more similar to the reference patterns.

A special loudness analyser is used for the preprocessing of the speech signal. This device generates 24 specific loudness functions /1/ which cover a frequency range of 70 Hz up to 14.5 kHz, representing the spectral energy in 24 critical bands. The loudness analyser approximates the characteristics of human hearing, as far as the loudness sensation as a function of time is concerned.

The initial investigations are restricted to the adaptation and classification of eight German vowels. There exist two possibilities to adapt incoming loudness spectra at the reference set:
- either the spectral amplitudes can be altered,
- or the spectral position of this amplitudes can be shifted. /2/

Under restriction to linear transformation, a general linear matrix transformation includes both adaptation methods:

\[ \hat{x}_{Ta} = T \hat{x}_T + \hat{t} \]

\( \hat{x}_T \) Test spectrum
\( T \) Transform matrix
\( \hat{t} \) Translation vector

The coefficients of the transformation have to be calculated so, that the adapted testspectra \( \hat{x}_{Ta} \) are more similar to the corresponding spectra \( x_R \) of the reference set. An estimate of this similarity is the mean square error:

\[ e_m = \frac{1}{n} \sum \frac{1}{i} (x_{R,i} - \hat{x}_{Ta,i})^2 \]

\( x_{R,i} \) Reference spectrum

As for the mean square error is a quadratic function, a minimum can be determined by: \( \text{grad} (e_m) = 0 \). This condition yields a system of linear equations. By using corresponding test and reference spectra this system of linear equations determines the parameters of the transformation.

Applying this adaptation, improvements of the recognition score at the amount of 70 to 90 percent of the theoretically reachable improvement could be achieved.

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AUTOMATIC SEGMENTATION OF SPEECH CONTROLLED BY A QUASI-PHONETIC TRANSCRIPTION

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The automatic segmentation scheme is an essential part of a word recognition system based on segmentation and labelling of speech into quasi-phonetic units. The segmentation is controlled by a quasi-phonetic transcription of all the words in the used vocabulary and is performed on the acoustic level by use of three spectral bands: total pre-emphasized energy, low pass energy (300 Hz), and high pass energy (500 Hz). It discriminates four classes: silence, unvoiced segment, voiced vowel-like, or voiced consonantal segment and is carried out in a way similar to the dynamic programming technique used in pattern recognition for time normalization, i.e. when hypothesizing a vocabulary word as the input word all possible segmentation paths for that word are tried out and the path having the best overall score is chosen for further analysis. In practice, however, paths having very low probabilities are excluded before reaching the end of the utterance to reduce the amount of computations. The strategy used makes it possible to deal with deletions and insertions of phonetic segments by testing them in parallel and choose the best match. The criterion for scoring segmentation is based on simple and relatively stable phonetic-acoustic relations.

The quasi-phonemes resulting from the segmentation of a hypothesized word are characterized by estimates of the first three formants calculated from filter bank spectra and the energies in the spectral bands also used for segmentation. Dynamic parameters, measuring the transitional properties of the speech, have been found to be a very useful complement to the ordinary static parameters. The phonetic approach makes the system more insensitive to inter- and intra-speaker variations. Recognition results will be presented together with data on the segmentation performance.
B. PHYSIOLOGICAL AND PSYCHOLOGICAL

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With moderate acoustic stimuli, measurements of basilar-membrane vibration (especially, those using a Mössbauer source attached to the membrane) demonstrate:
(i) a high degree of asymmetry, in that the response to a pure tone falls extremely sharply above the characteristic frequency, although much more gradually below it:
(ii) a substantial phase-lag in that response, and one which increases monotonically up to the characteristic frequency;
(iii) a response to a 'click' in the form of a delayed 'ringing' oscillation at the characteristic frequency, which persists for around 20 cycles.

This paper uses energy-flow considerations to identify which features in a mathematical model of cochlear mechanics are necessary if it is to reproduce these experimental findings.

The response (iii) demands a travelling-wave model which incorporates an only lightly damped resonance. Admittedly, waveguide systems including resonance are described in classical applied physics. However, a classical waveguide resonance reflects a travelling wave, thus converting it into a standing wave devoid of the substantial phase-lag (ii); and produces a low-frequency cutoff instead of the high-frequency cutoff (i).

By contrast, another general type of travelling-wave system with resonance has become known more recently; initially, in a quite different context (physics of the atmosphere). This is described as critical-layer resonance, or else (because the resonance absorbs energy) critical-layer absorption. It yields a high-frequency cutoff; but, above all, it is characterized by the properties of the energy flow velocity. This falls to zero very steeply as the point of resonance is approached; so that wave energy flow is retarded drastically, giving any light damping which is present an unlimited time in which to dissipate that energy.

Existing mathematical models of cochlear mechanics, whether using one-, two- or three-dimensional representations of cochlear geometry, are analysed from this standpoint. All are found to have been successful (if only light damping is incorporated, as (iii) requires) when and only when they incorporate critical-layer absorption. This resolves the paradox of why certain grossly unrealistic one-dimensional models can give a good prediction of cochlear response; it is because they incorporate the one essential feature of critical-layer absorption.
At any point in a physical system, the high-frequency limit of energy flow velocity is the slope of the graph of frequency against wavenumber* at that point. In the cochlea, this is a good approximation at frequencies above about 1 kHz; and, even at much lower frequencies, remains good for wavenumbers above about 0.2 mm\(^{-1}\) (which excludes only a relatively unimportant region near the base).

Frequency of vibration at any point can vary with wavenumber either because stiffness or inertia varies with wavenumber. However, we find that models incorporating a wavenumber-dependent membrane stiffness must be abandoned because they fail to give critical-layer absorption; this is why their predictions (when realistically light damping is used) have been unsuccessful. Similarly, models neglecting the inertia of the cochlear partition must be rejected.

One-dimensional modelling becomes physically unrealistic for wavenumbers above about 0.7 mm\(^{-1}\), and the error increases with wavenumber. The main trouble is that a one-dimensional theory makes the effective inertia 'flatten out' to its limiting value (inertia of the cochlear partition alone) too rapidly as wavenumber increases. Fortunately, a two-dimensional, or even a three-dimensional model can readily be used to calculate a more realistic, and significantly more gradual, 'flattening out' of this inertia. All of the models give a fair representation of the experimental data, because they all predict critical-layer absorption. However, the more realistic two- or three-dimensional models must be preferred. These retard the wave energy flow still more, thus facilitating its absorption by even a very modest level of damping. The paper indicates many other features of these models.

* In any travelling wave, wavenumber is the rate of change of phase with distance; for example, it is \(2\pi/\lambda\) in a sine wave of length \(\lambda\).
Although many observations relevant to the mechanical tuning properties of the basilar membrane have been made, the resulting picture is incomplete, being drawn from heterogeneous species with disparate techniques. In particular fluid has sometimes been drained from scala tympani, complicating the evaluation of various theoretical models of cochlear mechanics. Investigators have often used mechanical parameters which suit their model, rather than physiologically accurate values.

A new displacement-measuring transducer has been developed, based on optical fibre technology. This convenient technique gives measures of basilar membrane motion both with and without perilymph in scala tympani. It has enabled a direct evaluation of theoretical formulations of basilar membrane motion, in particular the effect of the scala tympanic fluid loading on basilar membrane tuning.
ELECTROPHYSIOLOGICAL STUDY OF HEARING IN MICE WITH GENETIC HEARING DEFECTS

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Anatomical studies using light microscopy have shown that mice affected by the deafness or jerker genes develop cochlear hair cells which at first appear to be normal. The hair cells then begin to degenerate, and by about three months of age most of the Organ of Corti has degenerated. The object of the present study was to determine whether the cochleas of affected animals are functionally normal during the period when they appear structurally normal. Mice from the two strains aged between 12 and 50 days were anaesthetized and audiograms were determined by measuring thresholds for detecting the compound action potential recorded at the round window. No action potentials or cochlear microphonics could be recorded from affected animals at any age but littermates unaffected by the respective genes had normal thresholds. These results indicate that the apparent structural integrity of the cochleas of young deafness and jerker mice is not accompanied by normal function. Mice affected by these genes appear to be completely deaf at all stages of cochlear development.
Le modèle d'oreille que nous avons mis au point sur des bases physiologiques a été étudié en vue de l'analyse des sons et plus particulièrement des sons de parole. Il comprend :

1- un modèle d'oreille externe et moyenne constitué d'un filtre variable adaptable. On tient compte du réflexe stapédienn et de fonctions d'adaptation à long terme comme les mouvements de la tête.

2- un modèle de la propagation non-linéaire de l'onde sonore dans la périlymphe. Le milieu est supposé faiblement compressible et visqueux, la propagation est donc amortie, dispersive.

3- un modèle de la vibration de la membrane basilaire constitué par un banc de 24 filtres couplés non-linéaires

4- un modèle du système afferent avec codage impulsionnel des sorties constitué par des filtres très sélectifs augmentés d'un étage de codage en modulation d'intensité et modulation de fréquence selon la bande considérée.

5- un modèle du système afferent et de l'inhibition latérale qui permet de sélectionner les informations sortant du système afferent. Plusieurs modèles sont ici comparés après simulation avec des sons quelconques.

Parmi toutes les performances de l'oreille son adaptation aux sons extérieurs a mobilisé notre attention. A l'issue d'une simulation complète et minutieuse nous avons pu constater que les autres performances découlent de cette capacité d'adaptation surtout pour des sons aussi complexes que la parole. Les analyses acoustiques ainsi offertes par l'oreille, à grande résolution temporelle vers l'extérieur et à grande résolution fréquentielle vers l'intérieur, permettent d'expliquer son adéquation pour l'analyse de signaux non stationnaires.
In order for the cochlear prosthesis to transfer sufficient information to primary auditory nerve fibres for recognition of speech-like sounds, it is necessary to electrically excite discrete fibre populations at several points along the cochlear spiral (Tong et al., 1979). In addition, because of the large number of stimulus electrodes and the small volume of the scala tympani in which the electrodes are usually implanted, current transfer must derive from "passive" electrochemical processes in order to preserve the normal biochemical environment of the nerve fibres. From measurements made in the human cochlea during implantation of the cochlear prosthesis, it has been shown that the longitudinal ground current distribution arising from a multielectrode system incorporating a common intracochlear ground electrode is accurately replicated by measurements made in a saline-filled uniform tube of diameter 3-5 mm. Further measurements in such tubes are described which investigate the possibility of improved current localisation by reduction of the ground electrode impedance and the possibility of producing greater electrochemical "passivity" by the use of voltage sourcing in preference to current sourcing.
A PHYSIOLOGICAL MECHANISM OF NOISE INDUCED HEARING LOSS
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Over 16 years, 5736 pure tone audiometric tests were conducted on industrial workers. Noise exposure levels were also obtained. The results were analysed by the Chi-Square test and were highly significant (P < 0.001). Noise induced hearing loss (NIHL) eventually affected all tested frequencies.

Hearing loss, measured in dB for 3000, 4000 and 6000 Hz form an approximation of a negatively accelerating slope, while the lower frequencies of 500, 1000 and 2000 Hz form a positive accelerating curve. The initial loss in the 6000, 4000 and 3000 Hz range is soon followed by a more rapid accelerating loss of the lower frequencies, e.g. 500, 1000 and 2000 Hz.

The location along the basilar membrane of the hair cells in the cochlea for the high to mid-frequencies is positioned closer to the oval window than for lower frequencies. For the high frequencies, the basilar membrane is relatively narrow and well anchored at the promontory in comparison with the low frequencies located at the apical end toward where the scale vestibuli and scala tympani meet at the helicotrema. Also the basilar membrane is much wider at the apex than closer to the oval window.

Consider a high energy standing wave composed of numerous fundamentals. The arrival of the first fundamental at the oval window travels along the scala vestibuli and back along the scala tympani. Subsequent fundamentals naturally follow the same course. There will be a phase difference between the fundamentals across the basilar membrane. As any two fundamentals pass each other, there will be a point of maximum stress and consequent tearing of the basilar membrane when: (1) the fundamentals are in-phase, thereby displacing the membrane's anchor, and/or (2) the fundamentals are slightly out-of-phase causing the membrane to move large distances in a short period of time, thereby producing transverse tearing. The same two fundamentals can cause one or both kinds of damage depending upon: original intensity, frequencies of the fundamentals (rise-time), and the point of intercept along the basilar membrane. When the harmonics and heterodyning of the original sound fundamentals are considered, a complex interaction of destruction can take place. It is possible that different fundamental waves will produce maximum damage, due to the phase relationships and consequent tearing effect upon the basilar membrane, and may be removed from its loci (locus) of neuronal activation.

The basilar membrane at the apex is broad, hence more elastic and experiences a much shorter phase shift (displacement), regardless of the stimulating frequencies. It is therefore more resistant to standing wave form but not totally immune. The damage to the basilar membrane progresses away from the point of maximum phase shift or tearing effect, at the higher frequencies to ultimately the lower frequencies. This would explain the dichotomic noise effect of the presented data.
ON FREQUENCY RESPONSE OF EXTERNAL EAR AND THE MODEL OF ARTIFICIAL EAR

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The frequency response of external ear measured for 616 ears is described. The results show that the frequency response, the model of artificial ear and the approximate analog electrical network are in good agreement. This paper contains comparisons of analytical, numerical and experimental results. This work presents a method that opens up the possibility of measuring the response of human ears with an "unclosed seal".

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MULTIFREQUENTIAL IMPEDANCE MEASUREMENT OF THE EAR FROM 0.1 TO 4 kHz

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The greatest limit to an extensive clinical application of multifrequentual measurements is the use of sine-waves as probe tones which makes the procedure excessively time consuming. Using an impulsive method (Arslan & Coll. Scand. Audiol., 8, 127, 1979) one may deduce from a single excitation the impedance function in the range from 0.1 to 4 kHz. Thirty normal ears were examined at different pressures in the external ear canal (-200 through +200 mm H2O in steps of 20 mm H2O). In order to obtain the plane of measurement at the eardrum, external ear canal contribution was computed and subtracted. A good agreement is found between our results, previous observations and theoretical previsions, mainly for frequencies in the range 0.1 through 2 kHz. Most significant information is:

1 - An overall description of the impedance function (magnitude and phase). This enables an easy identification of the resonances of middle-inner ear systems (Fig. 1).

2 - From 800 to 1500 Hz a zone of secondary resonances is clearly identifiable. This frequency range is characteristic of the tympano-ossicular system. At positive pressures the peaks are higher than at negative ones, for a non-symmetrical behaviour of the ear impedance (Fig. 1).

3 - Finally our results are compared with Zwislocky analog of the middle ear. Some subjects show good agreement with model impedance function (Fig. 2-3). On the contrary others significantly differ in the phase due to an inductive component not present in the model results (Fig. 4-5).
A hybrid correlator was constructed to measure the transfer functions of the pinna, in an attempt to externalize dichotically presented sounds. The correlator used pseudo random noise as the interrogative signal and was provided with 4086 delay steps of 1μsec each. The auto correlation function consisted of a main peak of 10μsec width and a number of relatively small side peaks (-26dB). The spatial resolution was therefore in the order of 3-4mm.

The transfer function of the pinna consisted of up to 20 dominant peaks. They were separated by 10 - 40 μsec, which corresponded to a spatial separation of 3.4 - 13.6mm. The peaks appeared to devide into 4 distinct groups corresponding to the reflections from the upper part of the helix, the antihelix, the concha, and the direct sound. The transfer functions were found to vary in a complex pattern with the azimuth an elevation angles.
MIDDLE-EAR FUNCTION, THERMAL NOISE AND HEARING THRESHOLD LEVELS
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At frequencies below 2 kHz, the function of the human middle-ear is well described by lumped element acoustical networks developed by Zwislocki and others. At higher frequencies the situation is less satisfactory since the eardrum no longer behaves as a simple piston but breaks up into zones vibrating with widely different amplitudes (1). As a better approximation, one can treat the eardrum as two pistons linked by a frequency-dependent coupling impedance $Z_{do}$ (2). The smaller represents the area $S_o$ overlying the malleus and is directly connected to the ossicular chain while the larger represents the surrounding area $S_d$. The corresponding network contains an ideal transformer shunted by $Z_{do}$. These network elements couple the pistons tightly at low frequencies where $Z_{do}$ is large but permit substantially independent motion at high frequencies where $Z_{do}$ is small.

Many of the network values can be estimated from anatomical information and from eardrum impedance data for normal and pathological ears (3). Unfortunately, reliable high-frequency impedance data are difficult to obtain due to the presence of an air column of uncertain size and shape between the transducers and the eardrum. The absorption coefficient as determined from observations of ear-canal standing-wave ratio is, however, comparatively insensitive to variations in canal geometry. The available data for normal ears indicate that, at 8 kHz, more than 50% of the sound energy incident at the eardrum is absorbed (SWR=13 dB). To account for so much absorption it is necessary to assign 80% of the eardrum area to the larger piston ($S_d/S_o=4$), postulate a piston mass no greater than 4 mg and assume that the coupling impedance provides near-critical damping ($R_{do}=300$ cgs-ohms) at 8 kHz. In effect, it is inferred that the eardrum is viscoelastic at high frequencies except over the malleus. When the network is adjusted to meet these various requirements it is found that, at 8 kHz, only 3% of the power absorbed at the eardrum reaches the oval window and, at 15 kHz, less than 1% (4).

In the free field the interaction between the external and middle ear can be expressed in terms of a radiation impedance which shunts the input terminals of the middle-ear network. The thermal noise in the combined system can then be calculated by applying the Nyquist noise generator theorem. The calculation shows that most of the noise appearing at the oval window is associated with the cochlea not with the external ear. Hence knowing the detectability of pure tones in noise (5), one can estimate the detection limit imposed by thermal noise. At 500 Hz this limit is approximately 20 dB below the observed median hearing threshold SPL at the eardrum. Between 8 kHz and 16 kHz, however, the thermal curve matches recently-determined threshold levels for young ears.

AUDITORY LOCALIZATION OF A SOUND SOURCE
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Localization of a tone-1kHz 650ms-and a series of 7 identical tone pulses 1 kHz (total duration 620ms) is studied in an anechoic room in presence or in absence of a plane reflector (see fig.). Fronts durations of a tone are 2.5 ms and 75 ms; single pulse duration is 20 ms, and its front is 1 ms. A sound source is omnidirectional, a listener's head is fixed. At azimuth -22° VS +22° a regular deviation G in an estimate of position of a sound source is equal to 1° and a random deviation S is equal to 1.5° independently on the front duration and on a type of a signal (a tone or a series of pulses). In presence of a reflector G increases and oscillates (curve 1) and S rises up to 3°+5°; for a pulse series is almost same as in absence of a reflector (curve 2) and S is equal to 1°+2°. Increase of G and S for a tone is, though to be due to a stable interference of signals arrived from a source and from a reflector, which creates false sound sources. In this case auditory processing is determined mainly by a "slow acting" mechanism of localization (Bekesy, 1960); the shorter a front of a tone, the bigger contribution of a "fast acting" mechanism, leading to a decrease of S and G. Increase in quality of localization for a pulse series is assumed to be due to multiple action of a mechanism of fast localization.

* Symbols OE and UE on figure mean "overestimate" of Ŷ, and "underestimate" of Ŷ.
When the tympanic muscles perform a reflex contraction in response to any stimulus, the activity in the muscles does not begin immediately after the arrival of the stimulus, but starts after a latent period. This period is termed the latency of the reflex.

In this paper, the latency studies for the ipsilateral reflex in man are made for different ipsilateral acoustic stimulus durations, intensities, and frequencies. The effect of these results on the acoustic impedance of the ear, if any, is also investigated.

The latency of the ipsilateral reflex in man appears to have a value of 220 msec for the stimulus intensity of 1000 Hz, 90 dB SPL. Also this (Latency) is found to decrease with an increase in the stimulus intensity and to increase with an increase in the frequency in the range of 1000 Hz to 2000 Hz, used in the present work. No effect of stimulus duration longer than the latency is found as that part of the sound which occurs after the start of a response is not able to influence the latency. This property is used to develop a model of the ipsilateral reflex.

The air pressure variation in the ear canal is found to have no effect on the latency and hence no effect on the acoustic impedance of the ear, in the range of measurements of the present investigation.
Relaxation of a reflex is defined as the lessening of the contraction of tympanic muscles due to cessation of acoustic stimulation. In this paper, relaxation time which is one of the dynamic properties of the ipsilateral reflex, is described for different stimulus durations.

The relaxation time for the ipsilateral acoustic reflex of 1000 Hz, 90 dB SPL, is found to be 0.25 sec for a stimulus duration of 2 seconds in a particular normal ear. Mainly, the relaxation time is found to depend directly upon the duration of the sound stimulus.

In case of a series of repetitive stimuli, the relaxation time is determined to increase every time because the muscles, in this case, do not come to their perfect relaxed position after stopping of the first stimulus whereas the second stimulus starts immediately.

The effect on the acoustic impedance change is also investigated. The relaxation time of 0.6 seconds is observed for a typical normal tympanogram generated completely under the influence of continuous ipsilateral sound stimulus with the duration of 14 sec. Thus, a correction factor for finding the absolute value of the acoustic impedance change is found with the help of the present technique.
DYNAMIC PROPERTIES OF THE IPSILATERAL ACOUSTIC REFLEX III ADAPTATION

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Adaptation is one of the dynamic properties of the acoustic reflex and is defined as a process, in the lower levels of the nervous system, which lessens the reflex response during the continued acoustic stimulation. The adaptation of the ipsilateral acoustic reflex is discussed here.

The response peak of the ipsilateral reflex 1000 Hz, 90 dB SPL used in the present work is found to decrease during a continued stimulation but is found to vanish after 200 seconds of stimulus duration, for a particular normal human ear with average characteristics of ten normal ears used in the present investigation.

The tympanograms generated under the influence of continuous ipsilateral sound stimulus for 14 seconds are studied for determining the acoustic impedance change during adaptation. An increase of 0.01 cc is found in the compliance component of the ear at or near zero pressure where there is maximum change in the acoustic impedance change.
ON THE ORIGIN OF "N₂" IN TRANSTYMpanic AND SURFACE AUDITORY EVOKEd RESPONSES

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Introduction

Despite the large number of studies dealing with the growth of the early auditory evoked electrical responses (0-10 ms post stimulus time), the identification of the neural structures contributing to the response recorded by gross electrodes is still a matter of speculation. An attempt to understand the mechanisms of generation of the second response peak, N₂, is described in this paper.

Methods & Results

The auditory evoked activity has been recorded in man at the promontory (trans tympanic approach) and on the scalp (vertex-mastoid-forehead derivations), in response to clicks delivered at different rates.

Latency and amplitude of the first two peaks, N₁ and N₂, as a function of the repetition rate, have been measured and compared. The difference between the latencies of transtympanic and surface N₁ are very small, at any rate, with a maximum value of 0.08 ms. In the transtympanic recording, the latency difference between N₂ and N₁ is remarkably constant throughout the whole range of rate values, from 3 to 100 clicks per second. In the surface responses, on the contrary, the latency difference between N₂ and N₁ tends to increase as the rate is increased. The amplitude of the transtympanic N₂ is consistently reduced at click rates above 20-50 per second (more markedly than the amplitude of N₁), while the amplitude of the surface N₂ is much more insensitive to the rate increase.

Implications of these results are discussed, with respect to the possible sources of "N₂" in the two recording conditions, in terms of the available models for the response generation (see, e.g. 1-2).

Acknowledgments

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References

Components of brainstem evoked responses to auditory stimuli especially of low intensity show a distinct time-dependent variability. This variability is in part due to a nonstationarity of the superimposed biological noise the spectrum of which overlaps partially that of the evoked response. Consequently, band-pass filtering cannot fully eliminate this time-dependent variability. As demonstrated by BEAGLEY and McA SAYERS (1974, 1978, 1979) for the slow cortical evoked response, a phase constraint exists operative on specified harmonic components of the recorded individual post-stimulus epochs. They used a statistically significant phase aggregation as an objective measure for detecting the presence of a cortical evoked response normally to within 10 dB or better of the subjective threshold. Starting from this idea we developed a technique for the processing and analysis of brainstem evoked responses. The principal features of the technique are as follows: Windows of individual epochs with a duration of 16 ms, beginning 1 ms before stimulus onset, are tapered using a cosine profile, and transformed into the frequency domain. For each harmonic component amplitude and phase values as well as their ensemble distributions are computed. Using the deviation of the phase distribution from a uniform one as a weighting measure for the respective harmonic component, the complete spectrum is recomposed from all weighted amplitude values, and all phase values. After transformation of the amplitude and phase spectrum into the complex spectrum, inverse FFT is applied to transform the composed signal into its time domain representation.

The main advantages of the described technique are:

- The technique does not employ a band-pass filtering in terms of fixed cut-off frequencies and so does not possibly reject decisive information.
- The technique also avoids disadvantages inherent in optimal filtering in terms of a fixed template.
- The technique is adaptive in the sense that the actual phase spectrum determines the actual filter characteristic.
- The technique preserves the phase information of the harmonic components of the response and so avoids latency shifts.

First results obtained with this technique turned out to be very encouraging, and it may become the foundation of an objective, automatic pattern recognition for the different components of brainstem evoked responses, even in marginal recordings.
Evoked responses have been recorded from the scalp of normal human subjects to continuous sinusoidally-modulated amplitude-modulated tones. The responses were periodic in nature, and a Fourier transform was used to quantify the amplitude and the phase of the constituent fundamental frequency and harmonic components. The fundamental frequency of the response equalled the frequency of the modulation envelope and the amplitude of the harmonic components was usually less than the fundamental. The responses were also found to be invariant with time for periods exceeding 30 minutes.

Responses could be recorded for carrier frequencies from 250Hz to 8kHz and for modulating frequencies from 25Hz to 200Hz although the largest responses were obtained when the carrier frequency was in the range 500Hz to 2kHz and with the modulating frequency between 40Hz and 100Hz.

With the modulating frequency held constant, variation of the carrier frequency led to phase changes in the response's fundamental frequency. The phase changes corresponded to time delays of approximately 1 msec for each octave decrease in the carrier frequency. Responses were also found to be largest when the carrier frequency was an integer multiple of the modulating frequency.

An investigation was also made of the relationship between the response amplitude and the stimulus sound pressure level (SPL). It was found that the logarithm of amplitude of the fundamental frequency of the response was directly proportional to the stimulus SPL, whereas the phase of this component did not significantly vary. The responses were generally recordable down to sensation levels of 20-30dB.

The physiological, psychoacoustical and audiological implications of these results will be discussed.
ERROR ESTIMATION FOR THE SECOND DECONVOLUTION WITHIN MODELS FOR THE COMPOUND ACTION POTENTIAL OF THE AUDITORY NERVE

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In recent years our research group has developed the deconvolution technique with the aim to detect frequency-specific information at the level of primary neurons from the click-evoked compound action potential (CAP) of the auditory nerve. The technique consists mainly of three steps:

I. Deconvolution of the CAP with the postulated unit response (UR) to obtain the compound PST histogram (CPST). This has been proposed and performed by ELBERLING in 1976.

II. Logarithmic transformation of the CPST into a transformed PST histogram (TPST).

III. Second deconvolution of the TPST with a norm PST histogram (NPST) which is expected to yield the excitation pattern E as function of the location on the basilar membrane and of the corresponding characteristic frequency (CF), resp.

In several papers by the authors the value of this technique was derived and demonstrated for model CAPs under the following conditions:
(1) a UR exists uniformly for all primary neurons,
(2) latency of and peak-distance in the single fibre PST histogram is essentially reciprocal to the CF of that fibre,
(3) inter-peak ratio in the single fibre PST histogram is constant for all CF.

Conditions (2) and (3) are necessary for the feasibility of the second deconvolution (step III). In a preliminary study it could be shown that slight irregularities with respect to condition (3) did not change the result E which is a measure for the contribution of individual nerve fibres to the CAP.

What looked nicely in the model seemed difficult to be judged in recorded data. Especially the second deconvolution with its postulation of an NPST asked for a thorough examination of the errors due to slight changes of the deconvolution kernel, the NPST.

Besides studying this matter strictly mathematically, also different models for the formation of the CAP have been examined, and the deconvolution technique has been applied to the respective model CAPs. Thus, a guideline was at hand to judge the results of processing recorded CAPs with the described technique.
A thorough analysis of the discharges of single auditory nerve fibres can give a lot of information on the encoding mechanisms taking place in the peripheral auditory system. Since entirely different mechanisms can effect the discharge pattern in a similar way, inferences from the discharge pattern require a synopsis of the nerve fibres responses to different input stimuli including the zero-input (spontaneous discharge). Because of the complexity of the peripheral auditory system such a synopsis can be made only in terms of a model. The considerations which will be presented are restricted in the main to mechanical-to-neural transduction taking place between the hair cells and the auditory nerve fibres. A model of this transduction must explain so different phenomena like rectification and saturation of the probability of discharge, phase-locking as well as adaptation. All these phenomena have been investigated for several models. It has been tested to what extent nonlinear phenomena like rectification and saturation can be explained by models based upon simple physiological concepts as for example refractoriness of auditory nerve fibres. It could be shown, for instance, that rectification of the probability of discharge can be explained by very simple nerve models, and that saturation is to a great extent the result of refractoriness of auditory nerve fibres and a limited rate of quantal production in the synapse between hair cells and auditory nerve fibres.
On the mechanism of perception of bone conducted ultrasonics by the human ear

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INTRODUCTION

It is known that ultrasonics with frequencies up to 200 kc/s may produce auditory sensations if it will be applied as bone conducted sound to different parts of the head or to the region of the large vessels of the neck. The produced hearing impression corresponds to that of a very high tone whose pitch is nearly independent of the ultrasonic frequency.

THEORETICAL BASES

The functional mechanism responsible for this kind of hearing is not known up to now. The bibliography gives only hypothetical interpretations hereto. As we know from different investigations it seems to be that normal hearing persons have always on principle the ability to "hear" bone conducted ultrasonics, while persons with an impaired hearing, according to the nature of the impairment, can have this ability or also not. An extension of the measurement of the threshold for bone conducted sound on the ultrasonic range too could give an additional possibility for the differential diagnosis of hearing impairments.

EXPERIMENTAL INVESTIGATIONS

In order to advance by the clearing up of this phenomenon it have been made a few of investigations which should give an answer to the question whether the ultrasonic induced auditory sensations are caused by the formation of a radiation pressure difference between the both sides of the Basilarmembrane near the oval and round window or by a direct stimulation of the auditory nerve at the site of contact with the skull bones. Furthermore it has been investigated the influence of the large vessels of the neck on the perceptive process of ultrasonics. - The results obtained from this investigations will be presented and discussed.

REFERENCES

A functional model of auditory analysis based on temporally evolving lateral inhibition

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It appears that time evolution of critical bands in response to an incoming auditory stimulus is a crucial process in perception of transient sounds. The mechanical analyzing system represented by the basilar membrane of the inner ear cannot alone account for the frequency resolution of hearing. Some additional mechanism, possibly lateral inhibition located in the auditory neural network, is needed to achieve the frequency selectivity obtained in experiments investigating the dynamic properties of hearing.

A functional model of a lateral inhibition neuron network which simulates the organization of critical bands in time is suggested. In this network, schematically represented below, short-term adaptation is simulated by differentiators $D$ in the direct neural pathways. Integrators $I$ in the lateral inhibitory branches delay the action of the inhibitory mechanism. For stimuli comparable in duration with the time constant of the integrators, the frequency selectivity is enhanced by gradual superposition of the neural excitation function in the direct pathways with its negative second spatial derivative originating in the lateral branches. Steady state is reached in about 300 msec after the onset of a stationary stimulus. Such a system simultaneously enhances contrast both in the frequency and time domains.

This model was implemented on a digital computer. Its parameters were determined by simulating psychoacoustic experiments of several other investigators.
THE MEASUREMENT OF AUDIOMETRIC ZERO LEVEL

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MENG Chao-huei **

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** Peking Institute of Ear, Nose and Throat

The audiometric zero level of air conduction for 100 healthy youths was determined over a frequency range from 20 Hz to 10 KHz in 1971-1973 under laboratory conditions. These results have been further verified recently from a selected group of 12 persons. The transfer experiments from the prime values of telephone TDH-39(MX-41/AR) with the test artificial ear to the other six earphones commonly used all over this country with coupler NBS-9A and artificial ear IEC-318 are also presented. The deviation of our data thus obtained from those recommendations by ISO are discussed at some length.

References

5. ISO Recommendation, R 389, 1712.

The persons taking part in this work are:

Chen Jian-rhang
Fen Gen-quan
Zhang Ru-weiy
Luo Zhang-zhen
Chou Shao-lin
Dai Lin-pin
Jian Wei-xin
Xu Xian
Zhou Jia-qiu
He Yu-kui
Lin Qui-zhen
ON THE OPTIMUM LEVEL OF MUSIC LISTENED IN THE PRESENCE OF NOISE


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When we listen to music in noisy environment as in a car, it is required to keep the music at some higher intensity than noise. As a result of the experiment in which ten subjects adjusted the levels of five kinds of music to the optimum ones in the presence of noise with a spectrum similar to noise inside a car, we found that the adequate increment of the level of music was about 3 dB for 10 dB(A) increase in noise level, though the adjusted level of music itself was different among subjects.

The relative optimum level is shown in Fig.1 where the level of music for 65 dB(A) of noise was taken as the reference level. We tried to calculate the masked loudness of music based on the following assumptions; (1) We perceive the loudness of music corresponding to a certain percentile exceeded sound level, though the loudness of music varies incessantly. (2) The optimum level of music increases along with an increase in the level of noise, and in the process of adjusting the level we intend to keep the music at a constant loudness according to our preference. (3) The total loudness of music is the sum of the difference between loudness of music and that of noise in each critical band.

As a result of calculation of masked loudness for every ten percentile exceeded sound level of music, we found that subjects adjusted the music to keep its $L_{10}$ (10 percentile exceeded sound level) at a constant loudness.

Fig.2 shows the perceived level (Stevens' Mark VII) for $L_{10}$ of music for various noise levels. The pendency of loudness for the higher noise means that the intensity of music attained at intolerable level to subjects.

![Fig.1](image1.png)

**Fig.1** Relative optimum level of music vs. level of noise. The optimum level at 65 dB(A) of noise was taken as the reference level.

![Fig.2](image2.png)

**Fig.2** Perceived level of music at its optimum level. It was calculated for $L_{10}$ (10 percentile exceeded sound level) of music.
Continuous judgment of level-fluctuating sounds.
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Most of the noises in our surroundings are level-fluctuating sounds. Listening to such level-fluctuating sounds we have the impressions of loudness corresponding to the fluctuation at each moment. These instantaneous impressions are thought to be affected by the sound energy of the preceding parts as well as the sound level at that moment. In order to determine how long the duration is during which the sound energy is summed up resulting in the instantaneous impressions, instantaneous loudness of road traffic noise of 20-minutes duration was obtained using the newly developed method called "the method of continuous judgment by category".1,2

PROCEDURE

Eight subjects judged the loudness at each moment using seven categories from "very loud" to "very soft" by touching one of seven micro-switches on a response box corresponding to each category.

RESULTS AND DISCUSSION

Sound level and subjective responses were measured every 100 msec and the coefficient of correlation between them were calculated changing the interval between them (TL) in three different averaging conditions. In condition A, both subjective and physical values were the instantaneous values measured every 100 msec (Fig.1). The highest correlation was obtained when TL was 1.0 sec (p < .01), which suggests that reaction time was about 1 sec. In condition B, only physical values were averaged during 0.5 to 6 sec (TI, Fig.2). The highest correlation was obtained when TI was 2.5 sec and TL was 0 sec (p < .05). This fact suggests that the instantaneous loudness is determined by the sound energy averaged 2.5 sec preceding that moment. In condition C, subjective values of 1 sec duration were averaged and physical values remained same as in B (Fig.3). The highest correlation was obtained when TI was 3.5 sec and TL was 0 sec (p < .01). This result confirms the result of B considering that the subjective values were the average of 1 sec duration. From these results it was suggested that the instantaneous loudness is not always determined by the sound level at that moment, but is much affected by the sound energy averaged 2.5 sec preceding that moment.

References
A NEW EXPERIMENT ON THE "MISSING 6 dB"

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It is now well known that, at certain audiometric frequencies, there is a significant and persistent difference between monaural thresholds in earphones and binaural free-field thresholds, such that the sound pressure in earphone listening is higher.

The mean difference was valued up to 9-10 dB [1], [2]; 3 dB of the difference can be accounted for by the gain of binaural over monaural listening (binaural summation); 6 or 7 dB (i.e. the "missing 6 dB") still remains unaccounted for.

Anderson and Whittle [3] have studied and explained this effect in the low frequency range. They demonstrated that in this range the pressure threshold is raised by the masking effect of the physiological noise produced under the earcup.

The purpose of this study was to verify the presence of the "missing 6 dB" effect in the range 500 ÷ 4000 Hz.

Experimental Techniques

Four students ranging in age from 15 to 25 years, with normal hearing acuity, were tested. The stimuli were pure tones of 500 Hz, 1000 Hz and 4000 Hz; white noise was added to the stimuli in order to have a S/N ratio of 0 dB. The stimuli were presented at 5 intensity levels which differed by 1.5 dB.

The type of subjective test used in this experiment was the two alternative-forced-choice method (2. AFC) [4]. The rate of right answers was evaluated taking into account the chance the listener had to correctly guess a certain number of test items [5].

The threshold value for each subject and each frequency was chosen as the flex point absciss of a logistic function interpolating the five experimental results. This function, shown in Fig. 1, represents the percentage of correct answers versus the intensity level in a given condition test.

Results

The stimuli SPL were measured with a probe-microphone fitted at the entrance of the external auditory meatus. The mean monaural and binaural thresholds and their difference are shown in table 1. The difference between monaural pressure and binaural free-field thresholds is only given by the binaural summation effect.

Further measurements by earphones of monaural thresholds, carried out on the right and left ears, were compared with the binaural threshold. They confirmed the binaural summation effect.

These results are quite in accordance with new estimates of auditory thresholds [6], which have been made by taking into account physical factors such as physiological noise, transducer distortion, mechanical vibration coupled to the subjects and so on.

References;


Tab. 1 - Monaural pressure thresholds; binaural free-field thresholds and their difference

<table>
<thead>
<tr>
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<th>500 Hz</th>
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<tr>
<td>Difference</td>
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<td>4.1</td>
<td>1.9</td>
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Fig. 1 - Example of logistic function interpolating the five experimental results obtained for one subject
The loudness of brief pure tones
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The effect of the duration of a pure tone on its loudness was investigated for three frequencies: 500, 1000 and 2000 Hz and three levels: 20, 40 and 80 dB. The duration of the comparison-tone was always 800 ms and its level 20, 40 or 80 dB SPL.

Two young and audiometrically normal subjects were used. They were in forced-choice situation and their task was to do a loudness comparison between the test-tone and the standard-tone.

Results show that auditory temporal integration estimated by the index $A^2/B$ (Pedersen, C.B. and Elberling, 1972) depends jointly on frequency and level. Indeed the auditory temporal integration is reduced at high frequencies and/or at high levels, at least in the range studied here. These data must be considered like an experimental confirmation of the results of the Round Robin Test on Impulsive Noise (Pedersen, O.J., Lyregaard and Poulsen, 1977) in which the trading relation between intensity and time is steeper at lower levels.
REDUCTION OF NOISE HAZARD DUE TO INTERMITTENCE

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Considerable evidence indicates that an intermittent noise exposure will produce less temporary threshold shift (TTS)--often dramatically less--than a single continuous exposure having the same total energy. It does not necessarily follow, however, that the same will be true of permanent threshold shifts (PTSs). Therefore experiments have been conducted to test the validity of the total-energy principle for PTS in the chinchilla.

It was first established that the "critical level" for a 220-min exposure to 700-2800-Hz noise is 105 dB SPL, in the sense that chinchillas given this exposure display neither PTS nor an increase in missing hair cells (MHC) over the normal value of 50-100. A 220-min exposure at 108 dB SPL produced an average MHC count of over 1000, 111 dB developed not only a 700-MHC count but also a 10-dB PTS, and 114 dB produced 1500 MHC and a 30-dB PTS. Single exposures for 2200 min to 102 dB and for 22,000 min (15 days) to 92 dB produced PTS and MHC values that would be expected to result from 220 min of 112 dB, thus confirming the validity of the total-energy principle for the evaluation of single uninterrupted exposures.

However, breaking up a 114-dB exposure in time, i.e. into 22 10-min exposures given at 3- or 4-day intervals for 11 weeks, reduced the MHC count to 300, with negligible (less than 5 dB) PTS, implying a reduction of effectiveness of about 7 dB. Less effective reduction, only on the order of 2 dB, was produced by breaking the 114-dB exposure into 40 30-sec exposures separated by 30 sec (that is, an on-fraction of 0.50); a 20-dB PTS accompanied by an MHC count of 800.

These data suggest that a relation seen in TTS studies involving more moderate exposures--i.e. that TTS is proportional to the on-fraction of the exposure--may apply to PTS and MHC counts as well. This hypothesis is now being tested by employing various on-fractions and burst durations. It is clear, however, that intermittent exposures are not as deleterious as steady exposures, even when the total energy entering the ear is constant.

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Temporal modulation threshold curves (TMTC's) for intensity modulated noise bands.

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INTRODUCTION

Psychophysical experiments were done to provide answer to the question: what value must the modulation index m of intensity modulated (IM)-noise have to make it possible to distinguish between modulated and unmodulated noise, as a function of modulation frequency \( f_m \), noise bandwidth B and bandcentre frequency \( f_c \)? In literature are a few experiments which provide answer to this question using broadband IM-noise as a stimulus e.g. (1). For small bandwidths only amplitude modulated noise was used e.g. (2). As a modulation detector an "energy detection model" was proposed e.g. (2). This model consists of a bandfilter followed by a square-law device which in turn is followed by a RC-integrator. As a detection decision rule was proposed: the integrator output ripple must exceed a fraction of the ripple existing for an unmodulated noise input. The model predicts a low-pass characteristic for the curve relating the modulation threshold to \( f_m \) (TMTC). The constant level of the curve is predicted to be proportional to the inverse square root of the filter bandwidth and the slope is predicted to be 6 dB/oct.

EXPERIMENT AND RESULTS

For a broadband filtered IM-noise and 3 B-values at 3 \( f_c \)-values TMTC's were measured. The results are displayed in the figure. They show a low-pass characteristic with slopes of 2-3 dB/oct. At constant \( f_c \) the level of the TMTC's is independent of the used B-values. At constant B this level decreases significantly with increasing \( f_c \). The broadband TMTC is within experimental error the same as the TMTC's for the \( f_c = 4.0 \) kHz bands.

DISCUSSION

Interpreting the data in terms of the "energy detection model" we conclude that only a limited portion of B is used for modulation detection. And that this "receptive field" lies in the highest frequency region of B. Rodenburg (2) suggested that this field is two times the critical band (CB)-width. Our data do not contradict the hypothesis that the TMTC's are governed by the highest CB that lies in B. But assigning this CB to the model as initial filter, from the constant level of the TMTC's a RC-constant in the 100-400 ms range must be concluded, whereas from the TMTC's low-pass characteristics a time constant in the 1-4 ms range can be calculated. The model predicts a wrong slope too. (1) I. Pollack, JASA (1951) 23, 650; (2) M. Rodenburg (1972), Thesis, Rotterdam.
Spectro-temporal modulation threshold curves (STMTC's) for intensity modulated noise bands. Dept. of Medical and Physiological Physics, Utrecht State University

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Modulation thresholds were measured as a function of temporal modulation frequency (STMTC) for two overlapping frequency bands of spectral modulated (rippled) noise. The threshold is defined as the lowest spectral modulation index value for which alternating +/- cosinus-noise is discriminable from "white"-noise (modulation index zero). The STMTC's show a low-pass characteristic. Their constant level as well as their (temporal) cut-off-frequency depends on the (spectral) ripple density. The slope is about 3 dB/oct.

The concept of auditory bands, each followed by a power integrator as a spectro-temporal pattern processor will be discussed in the light of these data.
Determination of the Frequency Selectivity of the Impaired Ear

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A good frequency selectivity is an essential prerequisite for discriminating language and music. Two methods were developed by measuring the frequency-resolving power of normal and damaged ears: The psychoacoustical Tuning-curves and the $\Delta$-f measurements. The psychoacoustical Tuning-curves as well as the $\Delta$-f measurement of patients, suffering from various hearing disorders such as conductive hearing loss, noise-induced hearing loss; Menières disease, sudden deafness, congenital non progredient degenerative hearing loss and progredient degenerative hearing loss, demonstrate frequency resolution power with varying dysfunctions. The differences are so characteristic that the results of these two tests can indicate possibilities for exakt diagnosis and subsequent therapy.
FREQUENCY DISCRIMINATION PROCESS IN HEARING
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Frequency discrimination for brief tone signals was investigated as a function of the inter-stimulus-interval (ISI) and frequency separation between the interference tone and signal ones. The results were discussed in terms of discrimination model based on the short-term memory. Frequency discrimination was determined through a two-alternative forced-choice experiment. The stimuli used are shown in Fig.1. The value of Δf required for the correct response of 75% was adopted as the measure of difference limen. Changes in frequency discrimination without the interference tone (T1) are shown in Fig.2 as a function of ISI. As ISI increases, frequency DL becomes greater. It is due to the decay of the pitch memory. We drew up a model for this process as follows: In the discrimination task, the first tone should be compared with the second one. The pitch perception of the tone could be assumed to be normally distributed on the psychological continuum. In our model, the decay of pitch information is assumed that the standard deviation of the distribution increases exponentially with time (t), as

$$\sigma(t) = \sigma_0 \exp(\alpha t). \quad (1)$$

The distribution of the difference between two random variables, therefore, is expressed by

$$\sigma(t) = \sigma_0 \sqrt{1 + \exp(2\alpha t)}. \quad (2)$$

The dotted line in Fig.2 shows the DL calculated after Eq.(2). Each value is almost consistent with the measured one. In the second experiment, the effect of T1 was investigated. Fig.3 is an example of the results which represents the effect of the frequency of T1. It shows a dip around the frequency of test tones. We obtained almost the same results in other ISI condition, too. These results show that T1 with almost same frequency as test stimuli rather gives a clue to the discrimination than interferes it. We expanded the model to include the effect of T1 on the frequency discrimination, and it can account for the obtained data well.
FREQUENCY DEPENDENCE OF POST-STIMULATORY PITCH SHIFTS

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INTRODUCTION

It has been noted /A. Rakowski and I. J. Hirsh, J. Acoust. Soc. Am. 63, 851/A/ 1979/ that pitch of a short 1000 Hz tone pulse may change if the pulse is immediately preceded by another tone. The observed pitch shift was usually directed reversely from the pitch of the preceding tone. The aim of the present experiment was to investigate this effect at various probe-tone frequencies.

METHOD

Three professional musicians, highly experienced in psychoacoustic experiments were used to tune frequency of a variable-tone pulse V to achieve a pitch match with preceding probe-tone P. Both tone pulses were 25 msec long and were separated by 700-msec time interval. The probe tone P was preceded by 1250-msec long leading tone L whose frequency was always 3% lower than that of P. Time interval P-L was 3, 10, 25, 100, or 3000 msec randomly chosen. The series of three tone pulses L-P-V was repeated continuously until the subject reported satisfactory pitch match. Frequency difference between the tones V-P, after the pitch has been equalised, was used as a measure of pitch shift of P due to the post-stimulatory effect induced by L. It was expressed as percent change f_v. Measurements were performed at probe-tone frequencies 0.5, 1, 3, and 5 kHz, monaurally at 50 dB SL.

RESULTS

In the figure median and interquartile values for 16 pitch matches are given at each time-frequency combination. It can be noted that poststimulatory pitch shifts exist in wide frequency range. At time separation 3 sec no post-stimulatory effect is observed.
THE EFFECT OF SIGNAL DURATION ON TIMBRE DISCRIMINATION

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The attack, or initial transient portion of a musical sound has been shown to contain spectral features which are not contained in the steady state portion of the waveform. It has been claimed that the details present in the initial transient can be characteristic of the instrument generating the sound, or even of the family of instruments from which that instrument is drawn. Before such spectral information can be held even partly responsible for the characterisation of a particular instrument's sound, it is necessary that the information can be clearly perceived by the human ear and brain. Such perception involves the analysis of quite brief acoustic stimuli, but it has been shown that at least some details in an initial transient are perceived. Most previous experiments have used natural, complex stimuli (or modifications of them) to measure the importance of initial transients.

The experiment reported here used synthetic stimuli so that more systematic measurements could be made. Stimuli having the spectral characteristics shown in figure 1 were presented to the subjects in triplets in an ABX paradigm. The subjects were requested to respond by pressing buttons to indicate whether they thought the third stimulus was the same as the first or second stimulus.

Each subject received a range of signals of different durations, and at each duration the level of the third harmonic was also varied over a range of values. The acuteness of the subject's timbre discrimination ability at each duration was estimated by noting the level of harmonic three which resulted in 75% of the trials being correctly answered. The average results are shown as the circles in figure 2, which show that steady state discrimination is maintained down to duration values as low as 20 ms. A model has been developed which calculates the effectiveness of the other nine harmonics in masking the third harmonic. Naturally the masking is more effective for brief signals than for long, since the spectra of the harmonics overlap more fully for brief signals. These results are shown as the unbroken curve in figure 2.

The present investigation examined the effect of \( f_2/f_1 \) (where \( f_1 \) and \( f_2 \) are the frequencies of a two-tone input, \( f_2 > f_1 \)) exerted on the slopes of the functions relating simple difference tone level [\( L(f_2-f_1) \)] and cubic difference tone level [\( L(2f_1-f_2) \)] to input level (\( L_1=L_2 \), where \( L_1 \) and \( L_2 \) represent the levels of the tones at \( f_1 \) and \( f_2 \), respectively). Two normal-hearing young adults served as subjects. An adaptive 2AFC nonsimultaneous gap-masking paradigm was utilized to obtain estimates of distortion product magnitude. Parameters of the two-tone input were: \( f_1=1550 \text{ Hz; } f_2/f_1=1.08 \text{ and } 1.41; \text{ and } L_1=L_2=35 \text{ to } 85 \text{ DB SL}. \) Results revealed that slopes for the functions relating \( L(f_2-f_1) \) and \( L(2f_1-f_2) \) to \( L_1=L_2 \) were approximately 1.0 dB/DB for \( f_2/f_1=1.08. \) For \( f_2/f_1=1.41, \) however, a slope of 2.0 dB/DB was observed for \( L(f_2-f_1) \) while a slope of 3.0 dB/DB was obtained for \( L(2f_1-f_2). \) It is suggested that some combination of nonlinearities is needed to account for these data.
THRESHOLD OF PERCEPTIBILITY OF REPETITION COLORATION FOR WHITE NOISE IN FREE FIELD

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A remarkable change in sound quality is perceived at certain values of time delay when listening monaurally or diotically to a white noise signal together with its delayed coherent repetition radiated via one or more loudspeaker(s) in an anechoic environment. This quality change is called Repetition Coloration, and is associated with the perception of an evoked Repetition Tone having a pitch sensation corresponding, generally, to the reciprocal value of time delay. The perception of repetition coloration is a result of the physical interference (comb-filter) effect between an original sound (noise, speech or music) and its delayed coherent repetition. The perception of the phenomenon depends upon the degree of the fluctuation in and the spacing of the adjacent peaks of the combined sound spectrum. The threshold of perceptibility of repetition coloration for white noise is defined as being equal to the critical value of the relative level of the delayed repetition, with a particular time delay and type of presentation, for which the repetition coloration is just noticeable. The means of the averaged threshold measured for different types of presentations are shown in the figure, which are considerably dependent upon time delay, type of presentation and the subjects.

These results are of practical importance, because they can be applied in room acoustics and electroacoustics in order to predict whether repetition coloration will be perceptible or not.
MIMIC VOICE OF THE MYNAH AND HUMAN AUDITORY SENSE—A POSSIBLE EXPLANATION OF ITS PERCEPTION

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As it is well known, a mynah's mimic voice has no apparent first formant in its utterance of some vowels, however it sounds just as a human voice. Five Japanese vowels extracted from the mimic voice of a mynah were compared with those of its tutor by means of masking technique. It was found that the human auditory sensation caused by the mimic voice of the mynah showed similar characteristics to that caused by its tutor's voice, though their frequency spectra differed each other. This phenomenon, same masking for different spectrum, seemed to be explained by the possible sensation of \( mf_1 - nf_2 \) beats.

METHOD

A part of vowel was extracted from mimic voice of a mynah with the duration of 200 msec and 10 msec rise and fall time. This was used as the masker to evaluate the sensation caused by itself. A signal of 25 msec with 10 msec rise and fall time was superposed at the center of the masker. Varying the frequency of the signal, amount of masking was measured in terms of dB and plotted against frequency. The same procedure was carried out for the tutor's utterance and its result was compared with that of the former at the same level of the maskers.

For experimental proof of frequency difference theory, two pure tones of same level were mixed and applied to a pair of earphones and masking characteristics were measured by the same method mentioned before if it would possible to obtain considerable peak masking at difference frequency.

RESULT

1. For vowels /a/, /e/, /i/ and /u/, mimic voice of the mynah had not any lower frequency component corresponded to the first formant of the tutor's voice, however it had complicated higher frequency components instead.
2. For vowel /o/, mimic voice of the mynah was very likely comparing that of the tutor.
3. For mixed sound of 2 kHz and 2.6 kHz, its masking characteristics showed distinct peak value at 600 Hz.

CONCLUSION

It was supposed to be able to explain why mimic voice of the mynah could be heard just as that of the tutor might be a perceptual effect of frequency differences to produce the same effect as of the tutor's voice. Our proof experiment of masking against mixed tones seemed to support this hypothesis.
Effects of Visual Contact on Aural Perception using Complex Sound

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Introduction
Aylor and Marks\(^1\) have demonstrated the effects of visual contact on aural sensitivity using simple signals and visual images. If similar effects occur for more complex sounds and images, a fundamental reappraisal of environmental design methods will be required. This paper reports the results of some experiments in loudness perception of traffic noise whilst subjects looked at different visual images.

Experiments
Eighty undergraduate Architecture students were used as subjects in the experiments to view five pairs of thirty seconds video segments (with four seconds fade-in fade-out) of traffic flows. All segments were adjusted to have the same L\(_{eq\} (\pm 0.5\text{dB})\) value. A Heron two-part personality measures\(^2\) was used to gauge the personality types of the subjects. Each subject viewed six different traffic segments arranged in random order to form a total of five pairs. Loudness comparisons were made between the segments of each pair. In some cases one of the segments of the pairs had no visual image. Besides the loudness comparison experiments a further series of tests was carried out, using twenty-five subjects, where the acceptability of the various traffic flow segments was assessed on a ten-point scale.

Results
Both scenery content and personality type are found to affect aural judgements or assessments (F=4.60, p<0.01 and F=14.37, p<0.01 respectively). The results are summarised as follows:
1/ Concealed sources are judged as louder than exposed sources, to all personality types; 2/ ambiverts are much more easily influenced by special events, such as motorbike, truck or bus noise, in making their judgements; 3/ extroverts showed the tendency to be influenced like the ambiverts but to a very minor extent; 4/ introverts are more stable type of personality and least influenced by any contaminating factors such as the special events mentioned; 5/ traffic flows partially concealed by dense green hedges are judged, by all personality types, as more acceptable than those where the road is in full view; 6/ traffic flows at highly associated locations are judged as more acceptable, to all personality types, than those at less associated locations.

Comments
Noise susceptibility level of an individual was found to be dependent on not only the noise level but also his personality type\(^3\). However, from the above findings, it is apparent that visual contact is important too.

Acknowledgement
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References
Mapping Linear Traces into the Auditory Space
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The illusion of moving, "point-like" sources has been successfully created in the past by experimenters using synthetic binaural signals. Such sources are localized by the listener as "inside" or "outside" his head, depending on the nature of the stimuli. Moving images can thus traverse "sound paths" in an intra- or extracranial "host space".

We hypothesize that by temporal integration of the stimuli listeners could retain in short-term memory entire sound paths and thus perceive them as gestalten. If so, mapping linear (i.e. line-like) contours from a two or three dimensional visual space into an auditory space might be accomplished by letting the trajectory of a moving sound source coincide with the contour. Motivation for exploring this possibility is provided by the needs of the the blind: while the printed word is gradually becoming accessible to them via spoken-output reading devices, no efficient method exists for the auditory (or, for that matter, tactual) presentation of even the simplest graphic images. Yet a rapid mode of access to graphs and plots would be of great value to blind scientists, engineers, technicians etc.

Results of exploratory experiments will be presented: subjects listened to auditory versions of oscilloscopic displays of triangular and sinusoidal pulses. The traces were swept slowly, about 3-5 seconds per sweep. Horizontal displacement was conveyed by variations in the relative intensities of the dichotic stimuli. Vertical displacement was translated, after exponentiation, into frequency variations of a voltage-controlled oscillator. We found that time epochs (e.g. onset and termination of the displayed pulse) were adjusted quite accurately by our blind subject, and he could also distinguish fairly well between triangular and sinusoidal pulses.

Our use of pitch as a substitute "vertical dimension" was an expedient imposed by instrumental limitations, though one well founded in the analogy between spatial and tonal "height" in Western music. Future experiments will, however, center on the exploration of pseudo-spatial effects attainable by introducing into dichotic signals "altitude" cues such as pinna shadows, reverberation from a "ceiling" defined in the synthetic sound space and Doppler-effect.
Infrapitch Echo

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When noise is mixed with its echo having a delay of $\tau$ sec, a pitch of $1/\tau$ is heard for delays corresponding to pitches from about 50-2,000 Hz. The spectrum of this mixture is "comb-filtered" with a series of spectral bands having peaks at integral multiples of $1/\tau$ Hz. When the echo is phase-shifted 180°, the peaks of the spectral bands are shifted downward by $1/(2\tau)$. This antiphase echo mixture has two simultaneous pitches, roughly 0.9/$\tau$ and 1.1/$\tau$ (Fourcin, 1965; Bilsen, 1977).

I reasoned that a conflict between temporal (neural periodicity) and frequency (neural place) analyses might be responsible for antiphase echo pitches. The time delay of $\tau$ sec would indicate a pitch of $1/\tau$ Hz, but the lowest spectral peaks are at 0.5/$\tau$ and 1.5/$\tau$. This unusual conflict between temporal and place cues could result in displacement of the temporal value toward the place values of each of the neighboring peaks, resulting in the double pitches of 0.9/$\tau$ and 1.1/$\tau$. Note that this explanation requires a neural antiphase equivalency, permitting identification of restatements with 180° phase shifts.

If temporal information plays an important role in echo pitch detection, it would be anticipated that long period infrapitch echo of noise might be heard,* even though the peaks of the rippled power spectrum could not be resolved. These predictions were verified: listeners, after practice, could match unknown echo delays with periodic pulse trains for values of $\tau$ as long as 0.5 sec (echo periodicity of 2 Hz, spectral peaks spaced every 2 Hz). It was also found that antiphase and normal infrapitch echo were indistinguishable, in keeping with the absence of conflicting spectral cues.

*The existence and characteristics of infrapitch echo are in agreement with a general theory considering that pitch-range perceptual phenomena employing neural temporal analyses have infrapitch analogs extending down to 4 or 5 octaves below the limit of pitch. Other such infrapitch analogs have been reported (Warren & Bashford, 1977; Warren, 1978).

Decay of Auditory Sensations

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The decay of auditory sensation has been in the past been investigated using pre and post masking paradigms. An alternative paradigm has been investigated and is assessed. This paradigm involves the presentation of an interrupted tone burst for comparison with a step function containing the same tone. Preliminary psycho auditory testing has shown that subjects can identify similar sound components in each of the test signals and make equilisation adjustments. This testing has also suggested an auditory decay time constant of 50-60 ms for the first 10 dB of decay.
The beating threshold of mistuned consonances

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The phase sensitivity is one of the fundamental characteristics of the ear. Without this property the ear would not be able to detect whether the frequency ratio $f_1:f_2$ of e.g. an octave complex is exactly 1:2 or if it is only approximate 1:2. In the latter case the octave complex is mistuned and beats are perceived if only both the components are presented to the same ear.

The investigations of the phase effects which concern the pitch, the loudness or the pulsation-threshold of the octave tone have lead to the development of essentially two hypotheses. The first one assumes that the phase effects observed can be ascribed to the vector summation of an internally generated aural harmonic and the external octave tone. The second refers to the phase dependent waveform variations. An additional proposal /1/ shows how a nonlinearity acting on the waveform of the applied signal may evoke a phase dependent perception.

In the present work the thresholds are investigated where the beats of mistuned consonances disappear in order to obtain additional criteria for the acceptance or the rejection of a specific hypothesis. The beating thresholds are measured as a function of the levels of the primaries in the two-tone complex. The experiments show that phase effects of a 200/396 Hz complex are still perceived at rather low primary levels. The results will be reported in detail and e.g. the relation to the 'audibility region of combination tones' /2/ will be discussed.


BASE WIDENING IN STEREO LOCALIZATION
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DESCRIPTION

According to known standards, if the stereo listening angle supersedes 90° or if the stereo base is over 5 m, a dead spot is formed and a central imaginary source cannot be localized. Quadrophonic systems although producing more realistic listening conditions, taking the listener's head as immobile, leave the problem unsolved.

Research on the psychoacoustic phenomenon of head-turning in response to source movements, gave evidence that the 5 m basis or 90° angle can be widened.

MEASUREMENTS

Under the assumption that the basic listening phenomena are taking place in an 180° wide field in front of the listener, three loudspeakers, as two stereo pairs were positioned in front of listeners, fed by a 3-channel signal.

DISCUSSION

The base was thus widened up to almost 9 m without dead spot formation.

Using a variable listening basis the angle of head movement was determined through a number of tests on listeners and a non-linear relation was established. Further on it was found that there is a max angle beyond which the head does not follow source moves continually along the line, the head stays in a central position.
TENTH INTERNATIONAL CONGRESS ON ACOUSTICS

MOBILITY AID FOR THE BLIND USING AUDITORY LOCALIZATION OF SOUND SOURCES

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A new method of mobility aid was proposed for the blind by using auditory localization of sound sources. In our method, obstacles may be perceived as localized sound images corresponding to the direction and the distance of the obstacles. In this report, we investigated how the auricles work from the viewpoint of auditory localization when the sound sources are placed vertically and horizontally. Furthermore, a mobility aid device was designed by using ultrasonic devices and tiny sound generators which were arranged around two auricles.

APPARATUS and EXPERIMENTS

Short burst tones were produced from positions corresponding to the direction of obstacles by use of speaker matrix, and its sound intensity was proportional to the distance of obstacles. Subjects with normal ears answered the position of perceived sound images. In our speaker matrix, 96 tiny speakers were arranged in 16 columns with 7 cm pitch and in 6 rows with 10 cm (Fig.1).

Obstacles were detected by two ultrasonic devices, and one transmitted 40 kHz-200 μsec ultrasonic bursts 30 times per sec, while the other received the reflected sound waves. The direction of ultrasonic beam was varied vertically and horizontally by moving the head of ultrasonic device in order to detect the direction of obstacles. The distance of obstacles could be measured from time difference between transmission and reflection ultrasonic waves if obstacle was placed within 3 m distance and its size was larger than 5 cm x 5 cm.

RESULTS and CONCLUSION

(1) When burst tone was displayed up and down on speaker matrix, the ability of vertical localization depended remarkably on bandwidth of its sound spectrum. In the case of 200 μsec-white noise, the height of perceived sound images was almost coincident with the position of sound sources (solid line of Fig.2). Sound images were, however, localized on the upper position when the auricles were covered with ear mould (broken line of Fig.2).

(2) When two burst tones, masker and signal, were displayed simultaneously on horizontal plane, the perceived position of signal deflected to the opposite side of white noise. The maximum deflection level was beyond 15 cm in the case of 600μsec-white noise masker and 200μsec-1kHz-signal. This phenomenon, however, was not affected so much by the ear mould.

From above experimental results, it seemed most appropriate method to display multisound sources corresponding to obstacles not simultaneously but successively by tiny sound generators which were arranged around two auricles.

![Diagram](image-url)
NEW ACOUSTIC TRAFFIC SIGNAL FOR THE BLIND

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INTRODUCTION

An acoustic traffic signal which makes it possible for blind people to cross the street in an easy and safe way has been constructed. The system consists of two signals: 1) a weak attention signal used to inform the blind about the existence of an acoustic traffic signal in that particular crossing and 2) a louder leading signal used to lead the blind in the correct direction in the crossing. Both the weak attention signal and the loud leading signal are pulsed, making it possible to distinguish between RED and GREEN. The RED signal consists of repeated sound pulses lasting 2 sec interrupted by pauses lasting 0.2 sec. The GREEN signal consists of repeated sound pulses and pauses each lasting 0.2 sec.

SOUND SIGNAL

The sound is a 900 Hz square wave. This signal has been chosen because: 1) it is easy to localize, 2) it is well attenuated by windows and buildings, i.e. the annoyance is minimized, 3) also elderly people will be able to hear the sound, 4) it is easy to distinguish from the traffic noise and 5) it is easy to construct a square wave generator in digital technique.

THE SYSTEM

A loudspeaker is mounted on each side of the street. These loudspeakers are for the leading signal. From the box which normally contains a push button for pedestrians the weak attention signal is generated. This sound is only audible within 2 or 3 m from the box, and thus a blind person walking at the pavement will be able to find and press the button. By pressing the button the leading signal will be generated, but from the opposite side only. When the GREEN interval begins, the blind can simply walk towards the sound source.

BLIND SUBJECTS

The system has been evaluated by 11 blind subjects who had only little training in walking around in the traffic. All subjects were satisfied with the system. If the system was widespread installed, the subjects would be able to walk around also in areas they were not familiar with.

MODIFIED SYSTEM

Investigations are planned to take place late in 1979 with a system containing an automatic volume control. The RED pulse in this system is made shorter and the RED pauses correspondingly longer.
Fisher and Freedman (1968) demonstrated the apparent efficacy of free head movements (FHM) in restoring the accuracy of horizontal plane localization when listeners were tested under conditions of functional absence of the pinnae. The present author reports two experiments that examined aspects of this finding in more detail, using standard earmuffs to effect removal of pinnae function. The efficacy of FHM in vertical plane localization was also examined. Listeners sat facing two intersecting semicircular arrays of loudspeakers - one vertical, one horizontal - their aural axes in line with the two extreme horizontal sources at the start of each trial. Signals were 1/3-octave continuous noise bursts centered on 1kHz, and under listener's offset control.

Allowing for design differences between the present study and the previous one, Fisher and Freedman's result was broadly confirmed. In the restricted head movement (RHM) condition, horizontal sources not in or close to the median plane were again found to be perceived as lying close to the interaural axis, and in the FHM condition a marked improvement in accuracy was observed.

However, the earlier authors' view that FHM appears to "wash out" functional absence of pinnae needs modifying. The present experimental task required finer discriminations than Fisher and Freedman's: 18° as against 45° distances between sources. While absolute accuracy for the equivalent range of sources, in the RHM condition, was 24% in both studies, accuracy improved to only 50% in the FHM condition in the present study, compared with 86% in the earlier work. Also, listeners required an average 6 sec to make accurate decisions in the FHM condition. In a second, separate experiment, allowing functioning pinnae, listeners required an average 2 sec only to make accurate horizontal plane decisions in a FHM condition. Absolute accuracy was 95% (cf., Fisher & Freedman, 87%).

Vertical plane results from the first ("no pinnae") experiment showed only 19% accuracy in the FHM condition, and an average accurate decision time of 11 sec. In the second ("functioning pinnae") experiment, vertical plane accuracy was 72% under the FHM condition, and average decision time was nearly 5 sec.

Removal of pinnae function thus appears not to be wholly compensated for, in the horizontal plane, by free head movements. Vertical plane accuracy is severely disrupted, even with FHM, but this result should be viewed against the generally greater difficulty of vertical plane discrimination.
INTRODUCTION
Latera'izati on due to interaural time differ-ences, in general, is possible mediating time differences between oscillations of the temporal fine structure of the signal or between slow variations of the signal envelope. Lateralization based on the signal fine structure is limited to frequencies below about 1500 Hz while lateralization on the signal envelope covers a wide frequency range.

EXPERIMENTS
Experiments with antiphase band filtered noise showed that if fine structure and envelope clues provide contradictory informa-tion, the lateralization occurs on the basis of the fine structure below 1500 Hz. Latera-lization of high frequency stimuli on the basis of the signal envelope has been report-ed in the literature for different types of stimulation (e.g. [1], [2], [3]). Here we will report experiments comparing the latera-lization of amplitude modulated tones and noise as well as 2-tone complexes. Therefore the existence regions have been deter-mined as a function of the modulation frequency, the carrier frequency and the modulation depth.

RESULTS AND CONCLUSIONS
In the figure the modulation threshold has been plotted for the lateralization of AM-noise, AM-tone and 2-tone stimuli for two observers. A different behaviour appears for the band limited stimuli compared to the AM-noise. This can be explained with the peripheral frequency analysis. Optimal latera-lization for two- and three-component high frequency stimuli is found at a modulation frequency of 400 Hz where the modulation threshold is in the order of magnitude of 20 dB.

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INTERAURAL ARRIVAL TIME DIFFERENCES: ENVELOPE VS. FINE STRUCTURE

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In binaural hearing interaural arrival time differences as well as interaural level differences of the sound signals at the two ears determine the sidedness (lateralization) of the perceived sound image.

In classical psychoacoustics it was hypothesized, that interaural time differences govern the lateralization of frequency components up to about 1.6 kHz whereas components of higher frequency are only lateralized by means of interaural level differences. The data to support this hypothesis have been obtained by pure tone experiments.

Later tests with more complex signals gave evidence that in the frequency region above 1.6 kHz interaural arrival time differences still play an important role in lateralization. However it was assumed, that within this region not the fine structure but the envelopes of the signals at the two ears have to be presented with an interaural time difference to obtain sidedness of the image.

Some more recent experiments showed however, that also above 1.6 kHz images can be lateralized on the basis of interaural time differences even if they have a flat envelope, provided that their fine structure shows the attributes of phase angle modulation, i.e. non-equidistant zero crossings.

It is the purpose of this paper to show, that the different findings of early and recent experiments are not contradictory but can be understood on the basis of the facts we know about the function of the peripheral ear.

A more important effect which has to be considered in this connection is the conversion of phase modulation into amplitude modulation by means of band pass filters with sufficiently steep slope. Obviously the conditions for such a conversion are given in the cochlea.

Deutsch's octave illusion and handedness

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An auditory illusion has been reported by Deutsch (1974). This illusion appears during dichotic presentation of a 400-800 Hz stimulus alternating from one ear to the other and presented continuously during 20 seconds. According to Deutsch, the right-and-left hander's percepts are not the same: right handers have a tendency to perceive the high tone in the right ear and left handers, the opposite.

We have reproduced the stimulus used by Deutsch in changing the total duration of the stimulus which is in our experimental condition, 1.5 second. 37 Ss (20 right-handers, 17 left-handers) listen ten times the stimulus and described their percept. In these conditions, we found no significant difference between left-and-right handers. Deutsch and Gregory have pointed out that correlation between type of percept and handedness is clear for long stimulus but does not appear with a single repetition of the stimulus (0.5 sec.).

So it would seem that the correlation between handedness and type of percept depends on the total duration of the stimulus. At the moment, it seems difficult to explain this fact.
A DETERMINATION OF THE NORMAL THRESHOLD OF HEARING BY BONE CONDUCTION
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INTRODUCTION

Whereas the reference threshold of hearing by air conduction has been internationally standardized since 1964 (1), the specification of the corresponding threshold for hearing by bone conduction is still on the program of the responsible ISO committee. The pre-conditions for solving this problem seemed to be given when an international agreement upon the requirements for a mechanical coupler for objective calibration of bone vibrators was reached in 1971 (2) and both an "artificial mastoid" and several bone vibrators with characteristics as specified became commercially available. However, thorough investigations proved that the artificial mastoid, though suitable in principle, initially showed a severe lack in uniformity of the mechanical impedance in current production (3,4). Meanwhile, its characteristics have been improved (5). Now, there is a strong need for subjective determinations of the normal threshold of hearing by bone conduction to obtain data for international comparison and averaging.

EXPERIMENTAL WORK

Threshold measurements on a group of normal hearing subjects are being carried out both with earphones and with bone vibrators. A Prücitronic KH 70 type bone vibrator has been chosen as a transducer with the required characteristics (5) but later on investigations using other models will be added. First, the vibrators are applied to the human mastoid, which is internationally favoured in routine audiometry. Measurements on the human forehead which are more reliable due to fewer variations in the mechanical impedance are intended in a second stage. In both cases the non-tested ear is masked according to the rules given in (6). The threshold determinations are made with the aid of computer-controlled laboratory audiometric equipment (7).

RESULTS

The final results will be given at the verbal presentation in terms of equivalent threshold force levels and will be compared with the data of other investigations (4, 8).

REFERENCES

MORE DETAILED HEARING LOSS CURVES

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The hearing losses of young girls working beside power-looms were measured in a greater detail than usually, using AP6 audiometer. Fig.1. shows the detailed audiogram of a 16 year-old girl working at the same machine for 2 years. A rather sharp dip around 5 kHz occurs in the audiogram indicating the early stage of acoustic trauma. Fig.2. shows the conventional audiogram of the same subject. This octave audiogram does not reflect the sharp dip, neither does the additional measurement at 3 and 6 kHz /dotted X in Fig.2./. This example well indicates that the octave audiometry is not sufficient for thorough screening or for experimental research and only continuous audiometry guarantees the required accurate testing of hearing.

Continuous audiometry has already been developed by von Békésy /1947/. Unfortunately, nowadays most of Békésy-audiometers operate only at fixed octave frequencies, in the case of polar audiograms of van Dishoeck /1956/ the frequency has changed continuously, too. With continuous audiometry Gravendeel & Plomp /1959, 1961/ also examined soldiers exposed to impulsive noise and workers exposed to continuous noise. In school children Gjaevenes et al. /1974/ measured the hearing losses at 1/12-octave frequencies.

Several authors have examined noise induced or ototoxic hearing loss in animals, guinea-pig/6/, cat/7/, monkey/8, 9, 10/. Using the up-to-date technics highly precise cochleograms can be recorded. However, in these experiments the frequency spectrum of the audiograms could be improved.

In further research - in the animal tests, as well as in the screening of people - there is going to be a greater demand for a new audiometer with more accurate frequency resolution /author's plan/.

When examining people exposed to noise using continuous audiometry one could record hearing loss curves as detailed as required. However, the dB/A/ or any other one-number noise valuating process are not suitable for describing impulsive noise. To examine impulses the noise has to be recorded and digitalized /with proper sampling frequency/ and analysed with a computer. E.g. the sharp cracking noise of the power-loom's shuttle affecting the worker's ears for years is the most dangerous.

REFERENCES
TOWARDS A NEW INTEGRATED HEARING AID

School of Electrical Engineering, University of NSW, and National Acoustics Laboratories, Sydney.

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Although hearing aids have been widely used for many years, existing designs with linear amplification and linear frequency equalization bring disappointingly little benefit in many situations (e.g. when the user wishes to understand speech in noisy environments). They do not adequately correct such problems of hearing loss as frequency dependent dynamic range distortion, threshold increase and lack of separation of speech from background noise.

A new type of hearing aid is being developed by the IC Laboratory, School of Electrical Engineering, University of New South Wales, in collaboration with the National Acoustics Laboratories. In the final form it is intended that the circuit will be integrated. It will have field-programmable parameters so that it can be fitted and re-fitted to match each patient's individual impairment.

The new circuit will use elaborate nonlinear signal processing operations to overcome the problems of existing aids. Several schemes based on multiband nonlinear filters are under consideration. The signal is split into multiple bandpass channels with appropriate nonlinearities in each channel. The nonlinearities will be controlled by the signal characteristics (spectrum, level, transients, etc.) to discriminate against noise and favour speech signals. Other schemes using adaptive noise cancellation are also being investigated.

The proposed schemes are being tested by computer simulation. Psycho-acoustic tests are performed on a group of hearing-impaired subjects, using processed and unprocessed speech in various background noises. The results of some of these tests will be presented.
A COMPARISON OF VARIOUS PARAMETERS OF AMPLITUDE COMPRESSION

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This study deals with the so-called short-term or syllabic amplitude compression ("compression") in which the signal's short-term dynamic range is reduced. We assume compression might increase intelligibility only if a reduction of level differences between strong and weak sounds is perceivable. Difference limen for attack and release times (T_A, T_R) of compression for various compression ratios (CRs) based on a quality change of a 1.7-s long utterance was determined by the adjustment or ABX methods. The compression threshold was set 25 dB below the average maximal RMS level of the utterance (speech level). Noise added after compression equalized the background for various compression conditions. The smaller CR or T_R, the shorter the T_A had to be for discrimination. Experienced normally hearing subjects (Ss) needed T_A = 420 and 250 ms to discriminate a quality change between linear amplification and compression with CR = 5, T_A = 10 and 3 ms, respectively. Naive normally hearing Ss needed a much shorter T_A (about 30 and 40 ms for the same conditions). Testing of hearing impaired Ss is presently in progress.

A change in quality does not necessarily alter intelligibility. Results of a series of speech discrimination tests with hearing impaired Ss seem to support this notion. Modified rhyme test words, CVC nonsense syllables were processed through a wide-band compressor with CR = 2.5 and various T_A's (1 to 42 ms) and T_R's (10 to 370 ms). The compressed signal, amplified with individually fitted frequency responses, was delivered to 9 Ss via a circumaural earphone. The speech was always presented at the same individually set comfortable level. The signal was compressed either without noise and presented in quiet and in noise at -10, -5, and 0 dB re speech level, or with noise at L = 50 dB. There were no significant differences between data for compressed and non-compressed stimuli. Performance slightly increased only when noise was added after processing at L = 5 dB and the T_A and T_R were shorter than 3 and 90 ms, respectively.

The lack of positive effects of compression might be caused by the loudness function of sensori-neural Ss. Their loudness function often has a slope steeper at low levels and approaches the slope of the loudness function of the normally hearing persons at higher levels. Compression of the whole range of speech or of the high levels only (HLC) makes an abnormal loudness function effectively more abnormal, however a compression of low levels only (LLC) should result in a more normally appearing loudness function. Three such conditions with an average CR = 1.7 and T_A/T_R = 1.5/30 ms were compared. The performance of 9 impaired Ss for all three conditions was similar.

These data seem to indicate that broad band, short-term compression does not improve intelligibility. Discrimination testing revealed that compression with T_R's upto about 100 ms compensates for the natural level drop toward the end of the sentence and that T_A = 10 ms produces large overshoots. As very short T_R can cause distortion, compression with CR about 5, T_A = 3 ms, and T_R around 100 ms should provide adequate protection, level fluctuation reduction and absence of distortion. (Work supported by NIH NS 12946.)
Computer-Based Aids for the Hearing Impaired
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The past several years have seen a number of significant advances in methods of prescribing hearing aids, in the development of systematic procedures for teaching speech to the hearing impaired, and in the development of special-purpose sensory aids. These developments have resulted from a combination of improved technology and a rethinking of fundamental concepts. Two examples are given. In the first, an experimental computer-based system has been developed using the results of a recent study on prescriptive fitting of a wearable master hearing aid (WMHA). In this case, the WMHA is used as a reference hearing aid to obtain basic audiometric data on the subject. An estimate of the optimum frequency-gain characteristic for that subject is then made. The frequency-gain characteristic and other electroacoustic characteristics of the prescribed aid are specified relative to those of the reference hearing aid. A computer-search procedure is then used to find a commercially available hearing aid that approximates the prescribed aid as closely as possible. The second example relates to the use of computer-based systems in studying the speech of the deaf. Digital processing techniques are used for both analyzing and synthesizing the speech of deaf children. Common speech errors are systematically eliminated from the synthetic speech in order to investigate their effect on intelligibility. The effect of correcting durational errors will be reported as well as the results of an ongoing study on the correction of deviant pitch patterns. [Research supported by National Institute of Neurological and Communicative Disorders and Stroke].
This paper describes the results obtained in a series of psychophysical studies conducted with our first multiple-channel cochlear implant patient. The variations of apparent loudness and pitch for single-electrode stimulation were determined by the method of magnitude estimation. The results showed that the loudness growth due to increases in current level was much steeper than the growth for acoustic stimulation in normal hearing subjects. The pitch produced by electrical stimulation was found to increase with pulse rates below 200 pps, while the increase in pitch with pulse rate was less pronounced above 200 pps. For a constant rate of stimulation, the pitch varied with the electrode position. Furthermore, the same pitch estimate could be obtained by driving single electrodes with different pulse rates.

The hearing sensations at different electrodes were consistently described by the patient as "different sounds". For a fixed pulse rate, the hearing sensation at electrodes with low pitch estimates were described as "dull sounds", while "sharp sounds" were reported at electrodes with high pitch estimates. The nature of the "different sounds" produced at the electrodes was further studied by an identification experiment. The results of this experiment showed that the patient was able to associate the hearing sensations produced by single electrodes with different vowel colours. For 200 ms pulse trains at 75 pps, the hearing sensation produced at the electrode with the dullest sensation was associated with /b/ (as in hot), an /ɛ/ (as in get) was associated with the electrodes characterized by a medium sharpness, while an /I/ (as in hid) was associated with the electrodes with the sharpest sensations.

The results of these studies indicated that the hearing sensations produced by single-electrode periodic pulse trains might be, as a first approximation, equivalent to the sensations produced by acoustic single-formant synthetic vowels. The pitch estimates obtained in the pitch scaling experiment might be interpreted as being influenced by two pitch components. The first component is related to the pulse rate and corresponds to the "voice pitch" of an acoustic complex signal, while the second is related to the electrode position and corresponds to the "spectral pitch" of an acoustic signal.

In the light of these results, a possible speech coding scheme operating on similar principles as a formant vocoder may be proposed. In this scheme, the formant frequencies could be transformed to a pattern of activated electrodes, fundamental frequency to pulse rate, and amplitude envelope to current level.
Basic Guidance for Use of Prostheses

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Reports on prosthetic device use for both the United States and Australia indicate that a significant percentage of those fitted with hearing aids make limited use of such devices. Furthermore, severely or profoundly post-lingually deafened persons, if fitted with prosthetic devices, receive little or no specific instruction in their use. A 26 hour basic guidance program is discussed which offers systematic and sequential training in the use of auditory cues and in behavioral modification techniques for hard of hearing or severely impaired adults. This program enables more in depth case evaluation and orientation for prosthetic device use. This program is in use at the Ear Research Institute in Los Angeles and in 10 other centers in the United States fitting hearing aids and cochlear implants on deafened adults. An outline of this program is presented as well as data on pre/post training performances by severely and profoundly deafened adults.
AUSTRALIAN AIDS FOR THE HARD-OF-HEARING TELEPHONE USER.

TELECOM AUSTRALIA RESEARCH LABORATORIES

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A recent survey has confirmed that a significant portion (7.4%) of the
Australian population has a hearing problem. It is estimated that of those
who may need to use the telephone, approximately 200,000 persons would
experience some difficulty at sometime in using the telephone because of
their hearing impairment. About 150,000 of these would use a hearing aid.

Telecom Australia provides a number of standard facilities for assisting
the hard-of-hearing telephone user, including telephones with receive
amplifiers, a magnetic field coil for the handset, several loud extension
bells, a gliding tone caller, and a visual-indicating handset.

The considerable advantage provided to a hearing aid user of magnetic
coupling to the telephone, has in recent years prompted special efforts to
improve the efficiency and availability of magnetic coupling facilities.
All new public telephones are now being fitted with a handset coil and an
associated driving amplifier. A box-type coil for providing magnetic
coupling to body-worn aids has performed well in field experiments.

A further aid, which is to be marketed commercially, is a small battery-
operated coupler which clips over the telephone receiver and converts the
acoustic signal to a magnetic field for use by hearing aid users. This
was developed initially in the Telecom Research Laboratories with the
assistance of the National Acoustic Laboratories, and then further by a
private manufacturer.

To match existing hearing aid pickup coil characteristics, and to ensure
a satisfactory signal-to-noise ratio in the presence of spurious hum
fields, it has been necessary to provide a coupling field of at least
100 mA per metre from all of these magnetic coupling aids.
INVESTIGATIONS CONCERNING THE INFLUENCE OF HEARING AIDS ON THE INTELLIGIBILITY OF SPEECH WITHIN NOISE

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INTRODUCTION

It is known that hard of hearing people though they are using hearing aids have problems in speech understanding within noise. Their ability to concentrate on one source and to exclude others (the so-called "cocktail party effect") is disturbed. This study deals with the following problems: First to find an experimental set-up which allows the description of this handicap by results of reproducible hearing tests. Second to investigate different attributes of hearing aids due to their contribution to this effect.

EXPERIMENTAL TECHNIQUES

The Binaural Intelligibility Level Difference (BILD) is determined in real freefield or simulated freefield conditions. Changes in BILD's due to the hearing aid can be assumed to quantify the handicap of people wearing those aids. Only normal hearing people are chosen as testpersons in order to exclude all other parameters except the influence of the hearing aid. We are using head related stereophonic transmission technique with headphone sound reproduction. This technique, i.e. accurate reproduction of earsignals, allows in a very simple way to simulate the basic properties of hearing aids. In the anechoic room 12 loudspeakers are arranged on a horizontal ring (diameter: 3,5 m) around the testperson with an angle distance of 30° one from the other. Each loudspeaker can be fed with the speech signal, a noise signal, or the sum of both. The power density spectrum of the noise is shaped so that it approximates the longterm power density spectrum of the used speech signal. The following noise conditions were employed: one noise source at varied positions in the horizontal plane, 6 uncorrelated noise sources in the horizontal plane. Speech signal is a modified monosyllable wordtest, a standart test in audiometry.

RESULTS AND CONCLUSIONS

First conditions without hearing aid were investigated. Maximum BILD's were found in case of one speech and one noise source only. Values reach about 10 dB. This maximum value of BILD is reduced to 6-7 dB when a diffuse noise sound field is approximated by 6 independent noise sources. For one-sided deafness BILD shows negative values on side of the deaf ear and positiv values on side of the normal ear if identical reference condition as above is used. Results of experiments simulating various hearing aid conditions will be found within these limits. They will be reported in detail in our paper.
Investigators in East Germany (W. Kraak and colleagues) have established recently a direct relationship between the time integral of noise-induced temporary threshold shift (known as integrated TTS, or ITTS) and eventual permanent threshold shift (PTS). We made use of this relationship between ITTS and PTS to evaluate the potential hazards to hearing sensitivity associated with the use of hearing aids. ITTS resulting from hearing aid usage was measured in nine normal-hearing young adults. A variety of hearing aid output settings and exposure durations were explored under well-controlled exposure conditions. The data provide strong support for the notion that hearing aids can cause deterioration in the hearing sensitivity of typical hearing aid wearers. Guidelines for safe output limits for hearing aids based on these and other data will be discussed.
Past attempts at using the skin for recognition of tactile patterns derived from acoustic speech signals have largely been unsuccessful for perception of running speech. Problems facing researchers in this field include: frequency discrimination, especially for electrical stimulation, temporal and spatial resolution, real time speech processing and tactile pattern configuration strategies. It is considered that recent developments in speech processing which allow real time estimation of formant frequencies and vocal tract area functions will enable a successful speech aid to be developed. Based on results of the Tadoma (or Hofgaard) Method, in which speech is perceived by the deaf-blind using tactile and kinesthetic senses to determine movements of a speaker's articulators, a model is evaluated which enables a tactile display of articulatory information derived from parameters extracted from the speech signal by real time speech processing. Psychophysical measurements of percepts of computer derived patterns were carried out concentrating in particular on patterns more likely to be important for phonemic and speech discrimination. In this way it is hoped to validate the model as a useful speech aid for the profoundly and partially deaf.
TACTILE SPEECH CONVEYING AIDS FOR THE DEAF, A COMPARISON.

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Several systems for conveying speech information via the skin to deaf and hard of hearing people exist today in different laboratories spread around the world.

This investigation is a comparison between seven different systems performed without lipreading. A combined training and test procedure measured the capacity of the aids to convey information about the ten Swedish numerals 0-9. The table shows the results obtained after training.

corr.resp.
1. Finger-stimulating Optacon system (Spens 1977) 75%
2. Hand held, single vibrator system (Traunmüller 1977) 71%
2b. No. 2 applied on the abdomen 42%
3. Electrotactile matrix on the abdomen (Sparks et al 1978) 67%
4. Electrotactile spectral belt-array (Saunders et al 1976) 66%
5. Vibrotactile spectral array on the thighs (Engelmann & Rosov 1975) 64%
6. Vibrotactile 3-vibrator system on the abdomen (Scott) 51%
7. Hand held single vibrator, no signal processing. (Schulte 72)46%
7b. No. 7 with distortion which makes fricatives perceivable 58%

Among other things the results indicate that:
(a) Not unexpected, fingers and/or hands are the most effective tactile input areas.
(b) There is no significant difference between vibrotactile and electrotactile stimulation.
(c) Processing of the signal in order to extract relevant features of the speech signal and to match the skin characteristics increases the effectiveness of the system significantly.

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K-E. Spens: "Is there an optimal time window for tactually conveyed spectral patterns derived from the speech signal?", paper at Res.Conf. on Speech-Processing Aids for the Deaf, Gallaudet Coll., May 1977.
A MODEL FOR THE GROWTH OF THE RECEPTOR POTENTIAL IN THE PACINIAN CORPUSCLE

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This paper is a critical review of the most relevant electrophysiological data obtained from the Pacinian corpuscle, which is considered a kind of prototype mechanoreceptor. The analysis is based on a mathematical model of the receptor; it consists of two components: the capsule and the peripheral segment of the afferent axon. According to this structure, the following aspects have been studied in detail:

i) responses of the intact corpuscle

ii) responses of the decapsulated nerve terminal.

Analysis and simulation of the model are shown to be a powerful tool to describe the growth of the receptor potential generated by typical stimuli. In addition, some peculiar phenomena which were previously misunderstood are successfully explained by the present approach, in particular:

i) the presence and the behavior of an off-response rising after the decompressing phase of a triangular or trapezoidal stimulus

ii) the properties of the responses generated by sustained sinusoidal stimuli

iii) the directional dependent phenomena, i.e. the relations between the direction of the stimulus and the properties of the responses.
MECHANICAL IMPEDANCE OF HUMAN SKIN - RESULTS FROM SERIES OF CIRCUMAURAL AND INTRAURAL MEASUREMENTS

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Dummy heads which are used to measure the sound attenuation of ear protectors need a simulation of human skin and underlying soft tissues at those places where the ear protectors touch the head. The mechanical impedance is that property of both natural and artificial tissues which is required to be equal in original and replica. That is why the impedance of human skin at relevant places must be known. Therefore measurements, using a minicomputer for the recording and handling of the impedance data, were made at each ear of 100 subjects.

A piezoelectric impedance head served as the measuring device. It was applied to the soft tissues at three places around the human ear by a flat stylus (contact area = 1.75 cm²) with static application forces of 0.2 N, 0.5 N and 5.4 N. The first two values correspond to static pressures of earmuffs; the last one refers to the IEC recommendation 373 (concerning mechanical couplers).

For the measurement of the mechanical impedance of the ear canal walls a stylus with a conical shape was vertically introduced into the ear canal. The static forces with which it was pressed down were 0.2 N and 0.5 N.

All circumaural measurements covered the frequency range 100 Hz to 10 kHz. The upper frequency limit for the ear canal impedances only ran up to 4 kHz.

The results of the circumaural measurements are similar to those of impedance data of human mastoids. The soft tissues around the human ear principally act as a compliance combined with a mass and a resistive component. Both resistance and reactance vary with application point and static force. Those places of the skull which are covered by relatively thick tissues show smaller values of resistance and larger values of compliance than those ones covered by thin skin. A decrease in application force leads to a decrease in resistance but to an increase in compliance.

The ear canal walls represent an impedance that is a combination of a compliance with a resistance in the considered frequency range. The dependance upon the static force is the same as for the circumaural tissues.

The work was sponsored by the Bundesanstalt fuer Arbeitsschutz und Unfallforschung, Dortmund.
NEW ARTIFICIAL HEAD FOR MEASUREMENT OF HEARING PROTECTORS

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For testing ear-muffs, some artificial head methods have been proposed in the last years. Until now there has been a lack of an artificial head for the test of ear-plugs. Additionally existing heads show poor conformity in attenuation compared with subjective data. To overcome both above stated aspects, some basic acousto-mechanical properties of the human head have to be taken into account:
- the mechanical impedance of circumaural soft tissue
- the mechanical impedance of the skin in the earcanal
- the acoustic impedance of the eardrum and the influence of the earcanal
- the directional characteristic of the human ear in the non-occluded case
- bone-conduction

To receive the desired data, measurements were performed (see Els, Schroeter and Hudde and Blauert et.al. presentations). Bone-conduction was considered with an extensive attenuation and compensation-technique using ten subjects. Mechanical impedances were imitated by layers of two component elastomers. The influence of the eardrum and earcanal on attenuation data is considered mathematically (1). So the originally measured data can be corrected. The directional transfer functions are met by extensive modelling of pinna and the outer head. The measurements of actual ear protectors are done with the modified impulse method, controlled by a microcomputer, which also calculates corrections (eardrum and bone-conduction).

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DEPENDENCE OF ACOUSTIC ATTENUATION OF HEARING PROTECTORS ON INCIDENT SOUND LEVEL

MARTIN ALAN M

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With increasing dependence being placed by industry on personal hearing protection as a means of noise reduction, there is a considerable need for the accurate determination of their attenuation characteristics which may be applied with confidence to the real practical industrial situation. The majority of national standard measurement procedures (eg. ASA Z24.22-1957, BS5108:1974, ASA STD-1:1975) rely upon subjective threshold-difference techniques which necessarily measure attenuation for steady-state sounds at low incident sound levels. These data are then routinely applied to practical occupational situations where hazardous high levels of noise are present, often with impulsive components. The validity of applying such standard attenuation data to the practical situation is open to doubt. It is therefore important to examine the relationships between attenuation and incident sound level for both steady-state and impulsive sounds over a range including the hazardous noise levels found in industry.

These relationships have been investigated for four types of earplugs and four types of earmuffs using freshly prepared and instrumented cadaver ears. Pure tones and 1/3-octave bands of random noise in the frequency range 125-8000 Hz were employed as steady-state stimuli with sound pressure levels between 75 and 125 dB. Impulses with peak sound levels in the range 135-175 dB(P) were also presented.

For the steady-state signals employed, the eight hearing protectors have been shown to have constant attenuation characteristics over the range of incident sound levels investigated. This was also the case for the six conventional protectors (with intentionally linear characteristics) for the impulse stimuli. The two intentionally amplitude-sensitive protectors provided attenuation which increased with incident sound level for impulse noises. Comparison of the protector attenuation-frequency characteristics determined for steady-state sounds shows good agreement with those obtained from subjective (threshold difference) national standard measurement procedures.

It may be concluded, therefore, that the six conventional hearing protectors studied here have attenuation characteristics that are equal for incident sound levels at about 40 and 75 dB, and that they are constant for levels between 75 and 175 dB. Consequently, the results of national standard laboratory procedures, although measured at low sound levels, may be applied with confidence to occupations where hazardous high-level noises are present.
THE FACT UPON THE ACOUSTIC ATTENUATION PROVIDED BY EAR MUFFS OF THE SIMULTANEOUS USE OF OTHER ITEMS OF SAFETY EQUIPMENT.

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This paper presents results from experiments designed to quantify the loss of acoustic attenuation experienced when modern ear muffs are worn with either safety spectacles, or perspiration covers for the ear muff seals.

METHOD: The acoustic attenuation provided by five combinations of commonly available new ear muffs and safety spectacles was measured in accordance with the method described in British Standard 5108 (British Standards Institution, 1974), at eight frequencies from 63 Hz - 8 kHz, using 15 otologically normal subjects and compared with the attenuation offered by the ear muffs when worn alone. A similar comparison was also made using a two-year old ear muff, which had been worn in a hot environment, and exhibited a measured relaxation of the headband tension, and stiffening of the seals. Hardening of the earmuff seals can be retarded by the use of perspiration covers. The loss of earmuff acoustic attenuation when worn with one of two types of perspiration cover, the first a sleeve enclosing the seal, and the second an oval pad of material, was measured.

RESULTS: In the case of the new ear muff and safety spectacle combinations, the mean of the attenuation losses reaching statistical significance (5% level) or better, was 3.1 dB with a range of 2.5 dB - 4.2 dB. Losses occurred in general at frequencies between 63 Hz and 1 kHz, and at 6.3 kHz and 8 kHz. The aged earmuff showed a more pronounced deterioration of acoustic attenuation when worn simultaneously with a pair of safety spectacles, at 250 Hz and 6.3 kHz, reaching 5.2 dB at these frequencies. No statistical difference in the s.d. of the attenuation data was observed between the two experimental conditions. The perspiration covers were tested whilst dry, and when damp, on one model of earmuff. The results showed that only the damp sleeve caused no loss of acoustic attenuation. Use of the dry pad or dry sleeve, resulted in a mean loss of acoustic attenuation over the major portion of the frequency band tested of 6.8 dB with a range of 4.6 dB - 12.8 dB, and 5.5 dB with a range of 3.4 dB - 12.5 dB, respectively. Use of the damp pad, gave a mean loss of acoustic attenuation of 4.8 dB between 2 kHz and 4 kHz, with a range of 2.9 dB - 6.2 dB.

CONCLUSIONS: Hearing conservationists issuing earmuffs to individuals using either safety or prescription glasses should make an allowance for the loss of earmuff attenuation described above. In view of the relative stability of the loss of attenuation effect, it might be possible to direct further work towards the generation of a "spectacle factor" in decibels, which could be applied during any assessment of ear muff for suitability of use in a given noise environment.

It is not recommended that perspiration covers be worn with earmuffs, unless the fulfillment of comfort requirements makes this a necessity. It is suggested, as an alternative, that the earmuff seals be renewed at frequent intervals.
THE EFFECT OF HEARING PROTECTORS ON THE ATTENTION DEMAND OF WARNING SOUNDS

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An attitude persists amongst many noise-exposed workers that when wearing hearing protectors their ability to hear important sounds such as acoustic warning signals is impaired. As this is one of the reasons given by individuals and employee groups for not wearing their protectors it is important to resolve what basis if any there is for the attitude. A review of the relevant literature (1) indicated that in general hearing protection does not impair the detection of signals in the levels of noise for which they are normally required. Problems in detection may however arise if they are worn in intermittent noise or in conjunction with a temporary or permanent threshold shift. There is however no evidence as to whether hearing protection may affect the perception of largely unexpected sounds. This might be predicted from the reduction in loudness of the signal and the change in spectral character of the signal and noise associated with their use. This experiment therefore investigated the effect of wearing ear muffs on the attention demand of a typical industrial warning sound.

PROCEDURE
The warning sound, a tape recorded electromechanical siren was presented to 12 normally hearing subjects four times at each of five different levels in a broad band random noise of 75 dB, with and without the subjects wearing ear muffs. In a vigilance condition the sounds were presented in random order and at randomised intervals (Mean ISI=93s, Range = 20-180s). This vigilance condition was compared with a similar condition during which the subjects also performed a loading task, a modified version of a television tennis game. The inattention produced by these two conditions provides an assessment of the effectiveness of the sound as a warning.

RESULTS AND DISCUSSION
The results indicated that there was no large effect of wearing the ear muffs on the attention demand or detection of the warning sound in either the vigil or task loading conditions. The results also indicated that inattention, due either to temporal uncertainty alone, or in conjunction with the loading task, did not affect perception of the warning sound. This difference with the findings of two previous studies may be due to methodological differences. In particular these latter studies incorporated an element of recognition since several warning sounds were tested in randomised order in the same session.

CONCLUSION
In terms of attention demand this experiment found no basis for the attitude that the wearing of hearing protectors impairs the perception of warning sounds. This conclusion may however apply to users of hearing protection with an existing noise-induced or other hearing loss. The results also only apply to relatively simple sound environments, and further research is to be aimed at examining the effect of hearing protection on the perception of warning sounds in more realistic acoustic environments.

LONGITUDINAL STUDY OF INDUSTRIAL WORKERS WEARING EAR PROTECTION
PUBLIC HEALTH DEPARTMENT OF W.A., OCCUPATIONAL HEALTH, AUSTRALIA.

HICKS RONALD G.


Nearly 6000 pure tone audiometric tests were conducted on industrial workers over a 16 year period. Noise exposure levels were also obtained. The results were analysed by the Chi-Square test and were highly significant (P < 0.001). Noise induced hearing loss (NIHL) eventually affected all tested frequencies. Personal hearing protection, as reported by the worker, does not alter the rate nor degree of NIHL. The data as presented is far from ideal, as there are numerous variables that enter into the analysis that have not been controlled: firstly, the degree of deafness that occurred before hearing protection was worn; secondly, was the hearing protection worn continuously or intermittently; thirdly, the level and degree of managerial support; and fourthly, the quality of the personal hearing protection.

There are numerous possible reasons why personal hearing protection has not been effective in the prevention of NIHL in this population. In an article by Regan (1977), he concluded about the effectiveness of earplugs as worn in the workplace, "Results of this study indicate that attenuation provided to the worker is significantly less than manufacturers' specifications. These results indicate the manufacturers' specifications do not reflect the amount of protection actually provided to the worker while on the job line, daily, for the ear protective devices investigated in this study. In fact, these protectors provide an inefficient means of protecting the employee from intense noise exposure." Podilla (1976) concluded "...that laboratory tested standard earplug data do not represent the actual field conditions studied...the effectiveness of the device under field usage may be grossly over-estimated." More recently, Edwards et al., (1978) stated that "...industrial workers were receiving, on the average, only 33-54% of the protection available from the three types of earplug designs evaluated...at low frequencies was non-existent in virtually half the cases tested."

Workers wearing personal hearing protection are probably exposed to very high dB(A) levels of noise. Our data starts at 105 dB(A) for noise immision. Possibly, in such an environment, sound waves can be transmitted by bone conduction, hence bypassing the protective devices of the middle ear and stimulating the cochlea directly. Consequently, protective hearing devices are less effective and may even aid bone conduction by providing a resonance frequency. This explanation is more probable with high intensity impulse noise (Nixon, 1976).

Personal hearing protection must be proven to be an effective noise attenuator in the workplace and never assumed to be effective. Consequently all hearing conservation programs that use personal hearing protection must be well-planned and constantly monitored in order to determine the effectiveness of the devices in preventing NIHL. The basic goal is to eliminate noise at its source.

Verwendet man die vorübergehende Hörschwellenverschiebung (TTS) als Maß für die Beanspruchung durch Geräuschbelastung, so lassen sich die in der Literatur bekannten Formeln für die TTS auf das obengenannte Konzept zurückführen. Einige Versuche zeigen ebenfalls, daß sich eine bestimmte TTS nach diesem Konzept aus Belastungshöhe und Belastungszeit relativ gut berechnen läßt. Aufgrund dieser Versuche wird festgestellt, daß sich auch der Erholungsverlauf durch eine Potenzfunktion beschreiben läßt, wobei die Erholungszeit allerdings von der Art der Belastung abhängt: Eine TTS, die über ein langdauernd einwirkendes Geräusch niedriger Intensität erreicht wird, klingt langsamer ab als eine TTS, die durch kurzzeitig einwirkende Geräusche hoher Intensität entsteht.
C. NOISE: ITS EFFECTS AND CONTROL
NOISE GENERATION IN DUCTS BY FLOW-AcouSTIC COUPLING

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DAVIES PETER OWEN ALFRED LAWE
AS ABOVE

Significant sources of duct flow noise result from interactions between flow generated disturbances and acoustic fields. Whenever a flow separates from a sharp edge a thin shear layer is formed. Such layers are always very unstable and develop waves which grow until the layer rolls up to form a train of vortices. Rayleigh (1) analysed the stability of vortex sheets. He showed that a plane uniformly sheared sheet of thickness b was unstable to all disturbances with wavelength greater than 5b, the most strongly amplified having a wavelength of about 8b. Recent observations (2) with unexcited free jet shear layers confirm Rayleigh's predictions.

The basic mechanism of flow acoustic interaction as a sound source appears to be the development of an ordered train of vortices in a thin shear layer. Though not themselves strong radiators of sound, periodically generated vortices can excite resonators very strongly. The resultant resonant acoustic field then controls the behaviour of the vortex sheet, producing a self sustained oscillation.

Two types of such excitation of acoustic resonance have been studied; the first by a jet formed at a flow duct expansion and the second by flow across a slot. With the former, the resultant spectrum is essentially broad band, though the radiated noise spectrum is modulated by acoustic resonances of the duct system. The source strength varies as flow velocity $V^6$, with observed acoustic power output of $10^{-4}$ of the total flow power, at low flow Mach numbers. In the latter, with flow across a slot backed by a cavity, the spectrum consists of one or two discrete tones. These represent coincidence between a Strouhal frequency $f$ and a resonant mode of the cavity. The corresponding Strouhal number is defined by

$$\frac{fL}{U} = (M - 0.5); \ m = 1, 2, 3 \ etc.,$$

where $U$ is the convection (phase) velocity of the vortices and $L$ the streamwise slot length. Acoustic outputs up to $10^{-3}$ of the total flow power have been observed.

(1) Lord Rayleigh The theory of Sound Macmillan 1896.

WELDING-TORCH NOISE RADIATION

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The frequency spectrum for combustion roar, as distinct from combustion-driven oscillations, has no specific frequency peak. Although it is flatter, the broad-band spectrum is similar to that of jet noise. Since the preferred direction of sound emission is in the range of 30° - 80° from the flame axis, high-angle directivity makes acoustic shielding an attractive possibility in industrial applications. The noise directivity was specifically studied in this research.

A small LPG/oxygen welding torch was used to generate single flames at four different test conditions by varying line pressures and mixture ratios. One flame was essentially laminar while the other three were turbulent. The type of flame varied from diffusion to premixed.

Each flame was rotated through 0° - 180° relative to the sound-level meter in a horizontal plane. Sound-pressure levels were measured at various distances for each 15° increment of rotation. Frequency spectra were obtained with 6% band width.

In all four test cases, the maximum sound-directivity angle occurred between 75° and 82° from the flame axis. Schlieren optical photographs were taken at 0.5 millisecond intervals with the torch positioned at its maximum directivity angle and also at 45° to the flame axis for each test case. Direct colour photographs were taken at 90° to the flame axis.

Le bruit des turbines basse pression des turbo-alternateurs est dû en grande partie à l'émission sonore des corps de turbine basse pression. Le rayonnement des sources de bruit interne (roue du dernier étage basse pression et écoulement de la vapeur dans les fonds de corps) est renforcé par le fait que le volume interne à ces matériels est fortement réverbérant.

Une étude expérimentale a été entreprise sur un corps de turbine basse pression d'un turbo-alternateur de 700 MW afin de connaître l'influence sur le bruit de ces machines de certains paramètres de fonctionnement (puissance du groupe, vitesse de rotation, pression au condenseur).

Des mesures de bruit ont été faites à l'intérieur des fonds de corps ainsi qu'à l'extérieur de la machine. Les résultats montrent (figure 1) que le bruit externe n'est pas seulement dû à la source interne qui transparaît au travers de la paroi du corps, mais aussi au rayonnement acoustique de la paroi qui vibre sous des excitations indépendantes du bruit interne. De plus, on constate que le bruit dépend des conditions de fonctionnement et que, par exemple le bruit augmente lorsque la pression au condenseur augmente par rapport aux conditions normales de fonctionnement.

Ces mesures effectuées dans plusieurs conditions de fonctionnement permettent d'établir une loi expérimentale du niveau de bruit à l'intérieur de ces machines en fonction des paramètres de fonctionnement. Cette loi, comparée à une loi expérimentale du bruit des réacteurs d'avions utilisée par la SNELCMA (Société Nationale d'Études et de Constructions de Moteurs d'Aviation), mais adaptée au cas des turbines à vapeur, permet de prévoir le bruit des futurs matériels de ce type, et d'agir au niveau de leur conception pour en réduire l'émission sonore.

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TENTH INTERNATIONAL CONGRESS ON ACOUSTICS

METHODOLOGIE DE MESURAGE DU BRUIT DES TURBO-ALTERNATEURS DE
GRANDE PUissance

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Le bruit émis par un turbo-alternateur est dû à de multiples sources d'origines diverses dont les spectres d'émission couvrent des zones étendues de la gamme audible.

Actuellement, les exploitants de ces machines demandent qu'elles soient plus silencieuses. Ils incitent les constructeurs à étudier en détail chaque sous-ensemble et à en prédéterminer le rayonnement. Pour cette raison, ils exigent le respect de niveaux sonores maximaux qui sont exprimés soit en valeur de niveaux de pression à une certaine distance de la machine sur un contour de mesure défini préalablement, soit sous forme de niveaux de puissance acoustique rayonnée dans des conditions de fonctionnement spécifiées.

Lors des contrôles de réception de ces machines, plusieurs problèmes se posent généralement : comment obtenir ces conditions de fonctionnement spécifiées compte tenu des autres contraintes sévères qui peuvent entraver le bon déroulement des tests acoustiques ? Comment s'assurer que le rayonnement à mesurer est bien dû à la machine en cours d'essai car d'autres sources directement liées au turbo-alternateur, indispensables à l'obtention des conditions de fonctionnement exigées, sont susceptibles de perturber les mesures ? Comment s'assurer que le rayonnement de la machine en essai est bien stable ? Comment éliminer l'influence du local ? Enfin, en cas de dépassement des valeurs spécifiées, il est utile d'en pouvoir déterminer l'origine précise.

A partir d'exemples précis tirés des expériences récentes et fructueuses d'Electricité de France et du Constructeur Alsthom Atlantique, de même que de confrontations avec d'autres expériences analogues, les auteurs montrent que la détermination de la puissance acoustique de chaque sous-ensemble, grâce à la mesure directe de l'intensité acoustique et la prise en compte de cette intensité acoustique elle-même, permet de s'affranchir de plusieurs de ces contraintes et de parvenir plus rapidement à une connaissance plus précise du rayonnement de ces grandes machines.

Ces méthodes permettent en outre de mieux situer les responsabilités des excès de niveaux sonores qui peuvent être observés et de mettre en œuvre les techniques de réduction de bruit appropriées.

Il peut en être conclu qu'un travail important de réexamen des méthodes de mesures actuelles sera peut-être nécessaire mais qu'une période de transition sera également indispensable, pour tenir compte des intérêts économiques des parties en cause, qui ont basé leurs cahiers de charges sur une certaine technique d'évaluation du bruit.
ADAPTATION LEVEL - A MODEL TO PREDICT ANNOYANCE

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Numerous indices have been derived to improve the correlations of physical measurements with annoyance from environmental noise. The most widely used index ($L_{eq}$) does not perform too well, especially at moderate levels. Robinson (1) has corrected $L_{eq}$ for time variability by $L_{NP}$, and Matschat (2) has introduced the slope and spectra of variations. More recently Gjestland (3) suggested that noise below a certain threshold cannot contribute to annoyance. Zwicker and Fastl (4) have drawn attention to the temporal masking patterns of impulses and of reverberation; and Stein (5) has shown that impulsive signals produce forward masking of some 500 msec, similar to $T_{50}$ values common indoors.

A model that integrates energy emerging above masked levels (adaptation level, $L_{ad}$) should yield an "annoyance level", $L_{ann}$ correlatable to subjective judgments of annoyance. Further, an "annoyance threshold" $L_D = 35$ dB(A) blocks integration in the model whenever $L_{ad}$ descends below that threshold.

We have designed all these conditions in a microprocessor, in order to compare it with $L_{eq}$ values obtained from conventional instrumentation for identical signals.

Laboratory tests using recordings of environmental noises as heard indoors are presented to subjects at our anechoic room and $L_{eq}$ and $L_{ann}$ are computed for each sequence. We hope to have evidence of the advantages of this model before long.

STATISTICAL CRITERIA FOR COMMUNITY NOISE, AND MEASUREMENT WITH A SOUND LEVEL METER

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As can be seen by the graph below of typical cumulative distribution histograms, the use of two statistical descriptors such as L90 and Leq has obvious advantages. The use of only one descriptor could cause under-kill or over-kill of a noise to the extent of 10 dB(A). Under-kill to that extent would result in a continued environmental problem, while over-kill to that extent may quadruple the cost of noise control.

The main disadvantage of the use of such statistical criteria is the high cost of automatic statistical analysis instrumentation. A pre-printed graph sheet has been formulated by the Commission enabling sampling to be made manually with a simple sound level meter on a read-tick-read-tick basis. Using this system, an overall accuracy of ± 2 dB(A) may be consistently achieved.
ANNOYANCE OR ACCEPTANCE - THE INFLUENCE OF THE QUESTION ON CHOICES AMONG TRAFFIC NOISES

DPG

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The increasing complaints of people being annoyed by noise have initiated investigations on the subjective annoyance and correlated physical sound field parameters.

A number of different methods have been applied to the problem and the results are not consistent yet. Even under comparable test conditions as subjective choices among different sound fields represented in the laboratory considerable deviations from author to author are observed. Our results /1/ e.g. show a rather high correlation between the $L_{eq}$ and the judged annoyance, whereas Cermak /2/ reported a smaller crosscorrelation coefficient under a comparable experimental set-up.

It can be argued that even larger differences in the results are obtained by asking different questions to the subjects. E.g. the question to choose the more annoying noise presupposes a priori that there is really an annoyance caused by the noises. Our aim is to investigate the influence of the special task that is given to the subject upon the results. The subjects have to decide for the more acceptable noise (under the assumption of a possible acceptability). The results of the annoyance and the acceptability test will be reported.


/2/ Cermak, G.W., Exploratory laboratory studies of the relative aversiveness of traffic sounds, J. Acoust. Soc. Amer., 65, 1979, 112
Evaluation of the Reliability of Nominal Aircraft Noise Indices.
National Acoustic Laboratories

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A noise measurement program has been undertaken around several Australian airports in conjunction with a social survey. In the process the accuracy of nominal values of some noise exposure indices was evaluated. Variations in noise level between different overflights due to differences in thrust settings, take-off profile, and other factors, were evaluated.

Although exposure indices must ultimately be judged by their reliability in predicting subjective response to aircraft noise, some criteria of validity may be proposed _a priori_. First, the value taken by an index must be determinable, in principle, entirely from acoustic measurements. Indices specifically invoking aircraft type fail this test.

Second, it must be possible, from published data, to find the value of the index over a wide area to within reasonable accuracy, and to experimentally determine this accuracy using currently available technology. Indices requiring knowledge of details of the probability distribution of noise levels at a given point appear to be questionable on these grounds.

The application of these principles to some existing aircraft noise indices, and to some proposed indices, is discussed.
EVALUATION OF INHABITANT NOISE IN MULTIFAMILY DWELLINGS USING TNEL

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In designing multifamily dwellings which are adjacent each other through a party wall or floor, it is natural that the greatest importance should be attached to the prevention of internal noises.

According to the answers to the questionnaire put to people living in multifamily dwellings, most of the complaints centered around noise coming from the above floor, e.g. jumping or running of children, as well as plumbing noise in the lavatory and bathroom above.

On the other hand, A-weighted sound level which are commonly caused in the apartment house by the daily life of the neighbouring families having children were measured for 24-h period in the 13 vacant dwelling units.

During the 24-h the noise exposure from adjacent dwellings associated with general daily living activities was recorded. Typical examples of actual data are shown in Fig.1.

On the basis of the measurement, the basic data on inhabitant noise were arranged, and averaged peak sound levels, durations and numbers of impacts each 30 minutes were obtained for respective noise sources. In order to evaluate the inhabitant noise, Total Noise Exposure Level (TNEL) for individual noise sources were calculated.

According to the sound insulation grades of building, obtained TNEL data were classified into 3 groups and compared with the frequency of noise complaints (Fig. 2). As the result, TNEL30', calculated by formula (1) showed a quite well agreement with the frequency of internal noise complaints pointed out by residents.

\[ TNL_{30}' = \frac{TNL_{30} + TNL_{30\text{max}}}{2} \] (1)

From this viewpoint, it is considered the evaluation of inhabitant noise using TNEL should have been one of the useful index in designing multifamily dwellings, and the 24-h average TNEL would have been reduced to 55 or so.

REFERENCE Y. MITSUDA and S. KIMURA: Transactions of the Architectural Institute of Japan No.272, p.75, 1978, 10
THE SPECTRAL FACTOR IN THE VALUATION OF THE ANNOYANCE OF TRANSPORTATION NOISE

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It is generally accepted that the $L_{eq}$ is some measure for the subjective annoyance of transportation noise. Our own study /1/ shows that the loudness after Zwicker is an equal appropriate parameter for the annoying effects of this class of noises. However, it is insufficient to describe the annoyance by a sound level dependent quantity only, though the loudness after Zwicker already takes spectral properties into account.

Furthermore other properties like tonality, impulsiveness and especially the time history of the noise have to be considered. A lot of proposals have been made how to account for additional properties of the noise by adding some dBs to the measured level.

The factor analysis of subjective judgements yields a second factor for the subjective annoyance. This one can be interpreted as a decision criterion which valuates the spectral characteristics of the sound field presented /1/. For the investigation of this second factor and especially its physical correlates a subjective comparison of a set of transportation noises with equal $L_{eq}$ is performed. The results of this choice experiment are reported and suggestions for possible physical parameters are made.

SUBJECTIVE VALUATION OF INDUSTRIAL NOISE: CORRELATION WITH PHYSICAL PARAMETERS

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The annoyance of noise from different industrial plants is investigated. A very convenient method under laboratory conditions, the paired comparison, is used for judgement of noises. The subjective results are analysed with the aid of the principal component method, which renders a certain number of independent factors.

The aim of this paper is to interpret the factors by correlating them with physical parameters of the original sound field. Level dependent characteristics of the sound field as $L_{eq}$, $L_{10}$, $L_{99}$ etc., are determined from the actual noise level which is simultaneously recorded on the third track (FM-Mode) of tape used. In addition, the recording of the original level serves to extract time dependent properties of the level, e.g. pulses per second, rise and fall times and periodicities.

Spectral features of the sounds are measured on the basis of the recorded condenser microphone signals of the artificial head. This is possible because a pilot experiment gives no important difference between the artificial head signals and recordings in which a stand-alone condenser microphone was used /1/.

Besides a significant correlation of the first factor with level dependent values of the sound field, it is expected that other factors will correlate with spectral or time dependent parameters /2/.


/2/ Weber/Mellert, Vergleichende Beurteilung von Verkehrsgerauschen - Korrelation mit Lautstaerkeparametern in: Fortschritte der Akustik, DAGA 78, S. 271, VDE-Verlag Berlin 78
COMPARATIVE STUDY OF INDUSTRIAL NOISE FOR LOUDNESS AND ANNOYANCE

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A set of different types of industrial noises (e.g. printing-works, machine-tools etc.), especially with distinct audible spectral components, is recorded with the aid of a stereophonic mannequin head system. The reproduction allows a subjective natural impression of the original sound field /1/. The sound level is recorded in FM-mode on a third track of the tape.

A selection of the noises is presented in random order to groups of subjects. The original sound levels are reproduced in the original dynamic range by a special volume control system. Pairs of the industrial noise situations are presented to the subjects alternatively for comparison. Every subject has to decide which one of the noises is the more annoying (or the louder one resp.) or whether no difference is perceived. No additional explanation of 'annoyance' and 'loudness' is given. The results are factorially analysed.

In a previous research of transportation noise /1/,/2/ this method of investigation proved to produce results with small tolerance in the answers of the subjects.

The results are discussed with respect to the influence of the spectrum of the noise on the subjective valuation.


A CRITERION FOR LOW FREQUENCY NOISE ANNOYANCE

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INTRODUCTION

Over the last few years it has become apparent that low frequency noise annoyance is more widespread than originally believed. Sources of high level low frequency noise (0-100 Hz) have now been widely documented and many exhibit a spectrum which characteristically shows a general decrease in sound pressure level with increase in frequency.1 Such items as boilers, compressors, oil and gas burners and ventilation and airconditioning equipment are typical examples.

ANNOYANCE

Annoyance due to low frequency noise has been found to be high even though the dB(A) level measured has been relatively low. Typically, annoyance is experienced in the otherwise quiet environs of residences, offices and factories adjacent to or near low frequency noise sources.2 It is concluded that the dB(A) measure is not a valid basis for determining the justification of a complaint for those cases where the intruding noise is unbalanced in that it contains most energy in the lower frequencies. The common assumption that loudness and annoyance are equivalent is also seen to break down in such cases.

EXPERIMENTAL DATA

As part of an investigation into low frequency noise annoyance at Chelsea College, London, unacceptability ratings for stimuli of 10 Hz bandwidth within the 20-90 Hz range were obtained. Results show that for an overall SPL of 55 dB, an average 20% of a sensitive group of respondents expressed unacceptability while for 65 and 75 dB OASPL, the percentages rose to 50% and 80% respectively.

THE CRITERION

Based on the above experimental evidence and on field annoyance data obtained by Vipac it is proposed that where an unbalanced noise immersion occurs, the overall SPL inside residences should not exceed 50 dB. For offices and other industrial type environments, the SPL of any given one-third octave below 100 Hz should not exceed the surrounding one-third octaves by more than 10 dB.

REFERENCES


Some considerations on impulsive noises

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One considers some kinds of impulsive noises trying some analytical relations that permit to examine some aspects of their behaviour.

By a very simplified processing one evaluates the equivalent level produced by a pulse in a reverberating room and its variation as a function of the reverberation time.

One considers also the behaviour of a given ambient with respect to some kind of impulsive noises when the equivalent level is determined according to an exchange factor of both values 2 and 5.

By the same processing one investigates an evaluation of the effect of an insulating wall on the sound energy transmitted through it by an impulsive sound.

One proposes a damage risk criterion for noises due to the presence of many pulses of different peak value.
INTRODUCTION

S'il est généralement admis que les bruits industriels présentant un caractère impulsionnel sont plus nocifs que des bruits stables de même niveau de pression sonore pondérée "A", le terme correctif à attribuer aux mesures demeure controversé. Or les critères dont on dispose reposent essentiellement sur des essais de fatigue auditive pratiqués chez l'Homme, dont on n'a pas la certitude qu'ils soient représentatifs du risque réel de surdité, d'où l'intérêt d'utiliser d'autres méthodes d'évaluation complémentaires et parmi celles-ci, l'expérimentation sur animaux.

MODE OPERATOIRE

Des groupes de 16 cobayes sont exposés 8 heures par jour et durant 4 semaines à des bruits stationnaires de différents niveaux sonores d'une part, à des bruits impulsionnels industriels d'autre part. On admet que 2 bruits, l'un stationnaire, l'autre impulsionnel, qui induisent le même déplacement du réflexe de Freyer, sont de nuisance équivalente, pour l'Homme comme pour l'animal.


RESULTATS

Les résultats sont présentés sur le graphique ci-contre. Pour les trois premiers bruits industriels testés et pour un niveau sonore mesuré au sonomètre intégrateur de 95 dB(A), on a obtenu les facteurs correctifs suivants :
- Bruit n° 1: bruit de presse à cadence rapide (260 cps/mn) – $\Delta L = 1,2$ dB(A)
- Bruit n° 2: bruit de presse à cadence lente (60 cps/mn) – $\Delta L = 3,2$ dB(A)
- Bruit n° 3: bruit de pilon (cadence : 3 cps/5s) – $\Delta L = 5,5$ dB(A)
In the course of a comprehensive study in comparative evaluations of worldwide used noise rating procedures a simple scheme resulted for splitting up each of these ratings. This scheme is a powerful tool for classifications and shall be presented. On the other hand these structural features yield a good basis to put up a more comprehensive noise evaluation procedure. As part of this a general view of widely differing kinds of average level values is given. This includes currently used ratings e.g. $L_{eq}$, $Q$, CNR, NEF etc. and also new approaches in this field. In the discussion clear preferences of recently presented loudness averages to current used average power values will be outlined.
DISCRIMINATION DU RAYONNEMENT ACOUSTIQUE D’UNE VANNE SUR UN CIRCUIT DE VAPEUR
Ingénieurs à E.D.F.

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Les méthodes classiques d'évaluation de la puissance acoustique basées sur la mesure de la pression surestiment parfois largement les niveaux en ambiance industrielle. Dans l'exemple présenté ci-après il s'agit de déterminer la puissance acoustique rayonnée par une vanne, installée en centrale sur un circuit de contournement de turbine, et dont le niveau propre est voisin de celui du bruit ambiant. Pour résoudre ce problème on détermine la composante normale de l'intensité sur la surface de mesure entourant la source par la méthode du "gradient de pression" qui utilise la partie imaginaire de la densité interspectrale des signaux de pression délivrés par deux microphones [1].

Résultats expérimentaux : les conditions expérimentales complètes sont développées en [2] et les résultats seront présentés lors de la communication. L'indépendance des résultats par rapport aux perturbations ambiantes a été montrée par des essais en laboratoire (voir figure [3]). En l'absence de perturbation les mesures d'intensité concordent avec celles de pression à ± 1 dB dans la bande 250 - 4000 Hz.

Conclusion

Sous réserve de satisfaire des exigences sévères sur le déphasage introduit par l'appareillage de mesure, la méthode du gradient de pression permet de s'affranchir pour une grande part des conditions extérieures autorisant la détermination de la puissance acoustique dans des conditions qui la rendaient jusqu'alors impossible.


In 1977 the Department of Housing and Construction embarked on the development of a computer program and associated design manual for the design of air conditioning ductwork. To make this program and manual of value to designers it was considered that a complete acoustical analysis should be provided for. To achieve this it was first necessary to establish a design procedure suitable for computerization and a comprehensive set of data on self generated noise and noise attenuation for the range of fittings and ductwork components normally encountered in practice. This necessitated a thorough working knowledge of current practice, a survey of available data and a considerable amount of brain storming and innovative thought.

Available data was confirmed as being very limited particularly in the area of noise generation, and as money, facilities and resources were not available to undertake the necessary experimental research, an investigation was undertaken to determine means of interpolating, extrapolating and applying engineering judgement to expand this data and consolidate and extend existing manual design methods to fully utilise the power of computer techniques.

This paper outlines in broad terms the work undertaken in this investigation and the resultant methodology and data incorporated in the program as a consequence. The detailed results of the investigation are described in the Department's Air Conditioning Duct Design Manual which is the technical support document for the program.
INTERACTION OF LOCAL TURBULENCE AND SOUND WAVES IN AIR DUCTS
Acoustical Institute of the Acad. Sci. USSR

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Vodopjanov Vjacheslav G.
Shvernik str. 4, Moscow B 36, USSR

INTRODUCTION
A distinct reduction of resonances in an air duct with a local narrowing, when a steady flow and acoustics excitation are superimposed, were observed. This phenomenon can be explained if one calculates the pressure loss in an air flow caused by a local change of duct cross section.

CALCULATIONS
Following reference 1 one may compare the pressure drop in a flow with a local change of cross section and in a corresponding laminar flow with smooth cross change. The result is

$$\Delta P = \Delta P_0 + \Delta P_{ac} \quad \Delta P_0 = \frac{P_2}{2} \left( \frac{s_2}{s_1} - 1 \right)^2 \quad \text{(static pressure loss)}$$

$$\Delta P_{ac} = \frac{c_0}{s_0} \left( \frac{s_2}{s_1} - 1 \right)^2 M \cdot U_\infty + \frac{1}{2} \rho' \frac{\partial U_\infty}{\partial x} \quad \text{(acoustic pressure loss)}$$

$P_2$ - static pressure downstream cross section change, $M = \frac{V}{U_\infty}$ - flow Mach number, $S_2/S_1$ - duct cross section up/downstream accordingly, $U_\infty$ - acoustic particle velocity, $\rho'$ - local inertia. Thus a local impedance $Z_0 = \frac{\Delta P}{P_2}$ with an active component proportional to $M$ occurs. When a diafragm is inserted in a duct $S$ will be 1.5 - 2 times greater than the theoretical one.

EXPERIMENT
The quality factor of a duct with open ends and a diafragm in the middle, with and without air flow was measured. The difference between these two values allows to calculate $Re[3]$ and compare with the theoretical value (see Fig. 1)

CONCLUSION
The theory and the experiments show that the interaction of turbulence due to an obstacle in a flow with the acoustics wave results in a considerable loss of energy of the wave.

REFERENCE
L. Prandtl "Führer durch die Strömungslehre" 13, 111 Aufl., 1949
NOISE INDUCED HEARING LOSS IN INDUSTRIAL WORKERS.

PUBLIC HEALTH DEPARTMENT OF W.A., OCCUPATIONAL HEALTH, AUSTRALIA.

HICKS RONALD G.

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In this study 5736 audiograms were analysed. Most of these audiograms represent retests. For some clients as many as six or seven retests were conducted over a sixteen year period. Some of the items coded from each audiogram were: worker's name, date of birth, if hearing protection was worn, if so for how long, noise level of exposure, date of audiometric testing and right and left ear pure tone "threshold" for 500, 1000, 2000, 3000, 4000 and 6000 Hz. From this data, the following data was computed: age of the client at the time of testing, his length of employment, the worker's daily dose of noise, and the worker's noise immission, total lifetime exposure to noise (Burns and Robinson, 1970). The data was computer analysed by Statistical Package for the Social Sciences using the Chi-Square test. All results were highly statistically significant ($P < 0.001$).

When each industrial worker's noise immission (cumulative energy received by the ears of the exposed person, Burns and Robinson, 1970), is plotted against his hearing test, 4000 Hz shows the predominant loss. After a noise immission exposure of over 100 dB(A) the hearing loss at 6000 Hz is greater than at 3000 Hz. All the results show the negative acceleration effects for the mid-frequencies (3, 4 and 6 kHz) while the lower frequencies (0.5, 1 and 2 kHz) form a positive accelerating slope. The lower frequencies show a considerable hearing loss after 105 dB(A) noise immission exposure.

This study does not assess non-auditory effects of noise, which must be considerable as noise induced hearing loss (NIHL) progresses: see Jensen (1961a, 1961b, 1979), Grandjean (1960, 1969), Lipscombe (1972a, 1972b, 1973). More attention should be directed to the secondary effects of NIHL. Cantrill (1974) concluded, that at levels of 80 dB SPL, the blood, cholesterol and plasma cortisol levels begin to rise significantly ($P < 0.01$) and continued to rise after exposure to 90 dB noise. Barr and Miller (1979) also Barr and Mullin (1979) included other non-auditory effects of NIHL which include: firstly, decline in key neurotransmitters of the Central Nervous System (norepinephrine and serotonin) and a corresponding rise in monoamine oxidase levels (see Mullin, 1979, as well); secondly, reduction in brain alpha rhythm; thirdly, cardiovascular changes, increased heart rate and vascular constriction, and fourthly, ulcers due to stress. Indeed deafness due to NIHL appears to be only the tip of the proverbial iceberg.
LOW FREQUENCY SOUND IN LOCOMOTIVE DRIVING CABINS
AUSTRALIAN FEDERATED UNION OF LOCOMOTIVE ENGINEERS
WILLINGALE                BERNARD JACK

A.F.U.L.E., 126 Chalmers Street, Surry Hills 2010

Research relating to sound pressure levels of infrasound and low frequency noise in the human environment, have been somewhat low and restricted to a small number of fields.

This paper discusses the levels of high infrasound and low frequency noise found in locomotive driving cabins. Both diesel-electric and electric locomotives, suburban and inter-urban type trains are dealt with.

Cab noise (and in some type trains and locomotives vibration) has been a serious source of discomfort to enginemen, since the introduction of this type traction following the demise of the steam locomotive. Whilst the steam locomotive may have given rise to high noise levels generally, quite apart from the effects of the sedentary type tasks in the modern locomotive cab, noise and the obviously different type mental work load compared to the steam locomotive, appears to be attributed by locomotive crews to the main cause of fatigue and other disabilities experienced in the operation of modern traction.

The concern by most authorities to date in respect to noise levels in the locomotive cab, has concentrated on hearing impairment problems. This paper deals with a field in which it is felt the need for much more work to determine the presence of low frequency noise and its effects is much overdue.

The paper does not attempt to establish of itself those affects other than to make reference to the findings of other workers who have turned attention to that area of study.
Man's HABITAT (from Latin "habitare", living space) should include, besides the buildings in which man works or rests, enclosures within vehicles and partial enclosures in which he spends increasing amounts of his lifetime.

The ICA dealt with the effects of the acoustic environment on man and the 9ICA put the emphasis on the results of research applicable to building acoustics. We should be ready in the 80's to design man's Habitat in an integrated approach that will not merely avoid negative effects but will rather create the necessary acoustic qualities which will allow man to enjoy work, communication, cultural life, creative thinking and relaxation.

We have analyzed the functions and activities of Habitat in the broad sense above defined as well as the acoustic sensitivity for each type of task and environment in order to select the simplest and at the same time the most accurate criteria in which to base design.

On the physical side we have included not only the control of immission from the environment into the enclosure but also the emission from its sources as well as the acoustic quality appropriate to each situation. The available theory from recent research has been selected and the techniques and materials rank ordered.

This operational procedure leads to sets of criteria, calculation procedures and construction techniques to optimize any type of environment significant in today's urban Habitat.

The failure to transfer criteria and design theories into practice has been traced to a lack of adequate control mechanisms. Materials, sources and finished environments are not controlled properly, mostly due to complexity of measuring standards and lack of trained technicians and incentives to attract public acceptance. This serious gap may be bridged with our proposed criteria.
INTERNATIONAL STANDARDIZATION
CURRENT AND FUTURE ACTIVITIES OF TC43 "ACOUSTICS"

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Ingerslev Fritz H.B.

The Technical University, 2800 Lyngby, Denmark

The scope of the Technical Committee TC43 "Acoustics" of "The International Organization for Standardization (ISO)" reads:

"Standardization in the field of acoustics, including methods of measuring acoustical phenomena, their generation, transmission and reception, and all aspects for their effects on man and his environment." TC43 has established two Sub-Committees: SCI "Noise" and SC2 "Building Acoustics."

It is clearly emphasized that standardization related to the influence of noise on man and his environment shall play an important role in the work of TC43. The long list of International Standards dealing with acoustical subjects proves that this is the case.

Although it is difficult to rank order the standards, it may not be incorrect to stress the importance of the two documents: 1) ISO 1999 "Assessment of Occupational Noise Exposure for Hearing Conservation Purposes (1975)" and 2) R 1996 "Assessment of Noise with respect to Community Response (1971)". These two documents are widely used by national authorities as basis of the establishment of noise criteria or limits. They are currently under revision.

The series of standards ISO 3740 to ISO 3746 includes standards which prescribe methods to be used to determine the sound power level of noise sources.

ISO/DIS 6081 "Guidelines for the Preparation of Test Codes of Engineering Grade Requiring Noise Measurements at the Operators' Position" is another document which will be adopted and published in the near future.

An important object of TC43/SCI is to develop test codes for measurement of the noise emitted by various noise sources. The task must be accomplished through collaboration with other Technical Committees within ISO or other organizations. A test code includes two main parts, one part describing the technique for carrying out the acoustical measurement, another part describing the conditions of operation of the machine or the equipment. It is to be expected that the above-mentioned standards in the near future will be applied in connection with a voluntary or compulsory noise labelling of noise machinery. ISO 4871 "Noise Classification of Machinery and Equipment" shall be utilized when labelling.

ISO published in 1978 the revised document ISO 140 "Measurement of Sound Insulation in Buildings and of Building Elements". This document is an extensive document divided into seven parts. The full written version of this paper will include information about a number of other International Standards, Draft International Standards and Draft Proposals.
DATA BANKS FOR PRODUCT NOISE EMISSIONS

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LANG

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During the last few years, interest has focused on the noise emissions of specific products in order that these products do not create levels of environmental noise which exceed national objectives and requirements. Legislation to protect the public health and welfare and the labeling of products for their noise emissions are expected in many countries. In addition, customers are demanding more information about product noise emissions. How can this information best be stored so that it is readily available for dissemination to interested parties?

Two different approaches have been taken to the accumulation and dissemination of noise data. The hard copy approach requires the assembly of all information for a particular product or group of products and the listing of the pertinent acoustical data in a catalog-like format. The required information is then made available to the customer, but it is difficult to keep the catalog up to date. An example of the catalog approach has been developed under the guidance of the Scandinavian institution Nordforsk.¹

An alternative approach is to implement a real-time, computer-based, interactive data bank system with users being able to access the system via dial-up communication terminals over leased lines. If the concept of data banks for product noise emissions is to become viable, agreement is necessary on the structure and content of these data banks.

Considerable progress has been made during the last decade, both nationally and internationally, to establish standardized methods for determining product noise emissions.²,³,⁴ It is the data obtained according to these standardized methods that should provide the input to noise data banks. As a minimum, such data banks should include the following information:

- Description of product and its mode of operation
- A-weighted sound pressure level at the operator’s position (if any) for operating and idle modes
- A-weighted sound power level for operating and idle modes
- Presence of prominent discrete tones and/or impulsive noise.

Additional information (e.g., noise spectra) should be provided by the product manufacturer directly to the user.

2. International Standards ISO 3740 through 3746: Determination of Sound Power Levels of Noise Sources.
GRADIENT METHOD OF SOUND POWER MEASUREMENTS IN SITU

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1. INTRODUCTION

Two classical methods of sound power measurements are in use: in reverberation chamber with the sound level independent of the distance from the source, and in anechoic chamber with the sound level decreasing according to the rule \(1/r^2\) i.e. 6 dB/dd.

2. THEORY

The method is based on the experimental determination of the SPL gradients \(\Delta L/\Delta d\) at given directions for the particular spectrum bands. Hence, the equivalent exponent \(k\) for the rule of sound spreading at given distance \(r\) can be found as

\[
\log 2 = \frac{0.1 \cdot \Delta L/\Delta d}{\log 2}
\]

Then, the overall sound power level of machinery can be calculated using the formula

\[
W = 10 \log \left[ \frac{1}{\pi} \sum_{i=1}^{n} 10^{-0.1 L_{ir}} \right] = L_{\text{ref}} + 10 \log \left[ \sum_{i=1}^{n} L_{ir} + 10 k \log r_i + 8 \right] [\text{dB}]
\]

where \(L_{ir}\) is the SPL at the distance \(r_i\) for \(i\)-th direction, \(n\) is the number of directions under measurements, \(r_0 = 1\ m\).

3. EXPERIMENTS

Three kinds of sound sources were tested:

a/ omnidirectional /without enclosure/,
b/ directional in the horizontal axis/half-cylinder enclosure/,
c/ directional in the vertical axis /cylindrical enclosure/.

The experiments were performed in two halls: I /24m x 8m x 6.6m/ \(T_{2000} = 3s\) and II /70m x 13.5m x 13m/ \(T_{2000} = 4.1s\).

![Fig.1. The example of the sound source spectrum at the distances 1m, 2m, 4m.](image)

Table. The results of sound power measurements for octave band component 2000 Hz by use the gradient and classical methods

<table>
<thead>
<tr>
<th></th>
<th>HALL I</th>
<th>HALL II</th>
<th>reverber.</th>
<th>anechoic</th>
</tr>
</thead>
<tbody>
<tr>
<td>a/ without enclosure</td>
<td>88.8</td>
<td>85.7</td>
<td>89.0</td>
<td>87.0</td>
</tr>
<tr>
<td>b/ half-cylinder enclosure</td>
<td>85.7</td>
<td>85.3</td>
<td>84.2</td>
<td>87.1</td>
</tr>
<tr>
<td>c/ cylinder enclosure</td>
<td>82.6</td>
<td>82.0</td>
<td>79.5</td>
<td>85.0</td>
</tr>
</tbody>
</table>

1. The results obtained with the gradient method are not dependent on the kind of hall and are in good agreement with these obtained by classical methods /reverberation and anechoic/.

2. The gradient method can be used for the omnidirectional sources as well as for the directional ones. In the last case the number of the measuring axes should be increased e.g. \(n=9\).
The Use of Modulated Reverberation for Acoustical Measurements
Center for Building Technology
National Bureau of Standards, Washington, DC, U.S.A.

Cook
Richard K.

8517 Milford Avenue, Silver Spring, MD 20910, U.S.A.

Reverberation chambers are widely used in applied acoustics principally for the measurement of absorbed and radiated sound power. The traditional (standard) method is based on measurements made in two steps. In the first step, the time rate of exponential decay of sound energy in the chamber is measured at various frequencies, from which absorption cross sections are obtained. In the second step, the steady sound pressures caused by a radiating source are measured, and then combined with the known cross sections to obtain the absorbed sound powers at the same frequencies.

The basic idea of the relatively new method of modulated reverberation is to amplitude modulate, at a low infrasonic frequency, the sound power radiated by an audiofrequency source. There will be a corresponding amplitude modulation of the total acoustical energy in the sound field, but reduced in amplitude and lagging in phase due to the absorption in the chamber. Measurement of the phase angle yields the "instantaneous" time rate of exponential decay.

The new method can be applied with an absolute sound power source. Its sound power output is absolutely determined, as a running function of time while it is radiating acoustically, through electroacoustical measurements conducted directly on the source. The modulated reverberation method will then yield (a) an absolutely measured total energy, potential and kinetic, of the reverberant sound field, and (b) the "instantaneous" time rate of energy absorption.

The traditional method suffers from the limitation that only the potential energy is measured; the kinetic energy of the sound field is assumed to be equal to the potential energy. This false assumption might be the origin of the systematic errors in measured sound power sometimes observed with the traditional method.

We will present the analysis and some experimental results for measurements made with the modulated reverberation of an absolute sound power source. The basic analysis is applicable to the use of such sources in three-dimensional reverberation chambers. A more detailed analysis will be given for a simple "thought" experiment; the modulated reverberation of the sound field in a standing-wave tube. This will show quantitatively the precise relationships between the kinetic and potential energies, and the "instantaneous" time rate of energy absorption in a one-dimensional reverberation chamber.
Pouvoir déterminer le niveau de puissance acoustique des machines en milieu industriel, de façon aussi simple que possible et en toutes circonstances, en particulier en présence de champs acoustiques perturbés par des réflexions, est devenu un objectif prioritaire pour la métrologie. A cet effet, des méthodes dites de "comparaison", de "substitution", ou de "juxtaposition", ont été proposées [1], faisant appel à l'utilisation d'un source sonore de référence (SSR) aux caractéristiques acoustiques déterminées à l'avance, selon une procédure d'étalonnage appropriée. Des travaux de normalisation ont été entrepris au niveau international par ISO/TC43/SC1/WG6, tant dans le domaine de l'étalonnage des SSR [2] que dans celui de leur utilisation[3]. L'attention est attirée ici sur ce dernier point ainsi que sur l'élargissement de l'expérimentation en cours, en vue d'une simplification des procédures projetées.

Dans cette première étape de normalisation [3], on s'attache aux déterminations en champ acoustique fortement perturbé par les réflexions (facteur de correction d'environnement $k > 7 \text{ dB}$) ; c'est une méthode de faible précision, de type "contrôle"[5], pour détermination "in situ". La méthode intéresse les machines inamovibles sur lesquelles, ou le long desquelles lorsque le dessus n'est pas accessible, on doit placer la SSR. Elle fait appel à l'utilisation d'une surface de mesure parallélépipédique enveloppant la machine soumise à essai (MSE), avec une distance de mesure $0,25 \text{ m} < d \leq 1 \text{ m}$. On cherche tout d'abord à déterminer, en fonction de la présence de parois ou d'obstacles réfléchissants à proximité immédiate de la MSE, combien des faces du parallélépipède de mesure théorique sont utilisables pour y implanter les microphones. Le nombre d'emplacements de microphone est déterminé en fonction du nombre de faces disponibles. Il en faut d'autant moins qu'il y a moins de faces (plus grand nombre de réflexions). À la limite, lorsqu'aucune face du parallélépipède n'est disponible, on opère dans une ouverture du local. On détermine ensuite le nombre et l'emplacement des SSR. Il dépend de la taille de la MSE, dont la plus grande dimension est de toutes façons $< 15 \text{ m}$, et de la possibilité de placer la SSR sur la partie supérieure de la MSE. Ce nombre de positions peut atteindre 6 dans le cas le plus compliqué. Une large expérimentation du projet permettra sans doute de réduire ce nombre.

Pour la détermination de la puissance acoustique de la MSE, on applique l'équation suivante : 

$\text{L}_w(\text{MSE}) = \text{L}_{p}(\text{MSE}) + \text{L}_w(\text{SSR}) - \text{L}_p(\text{SSR})$  

avec : 

- $\text{L}_w(\text{MSE})$ : niveau de puissance acoustique recherché pour la MSE 
- $\text{P}_p(\text{MSE})$ : valeur d'étalonnage de la SSR 
- $\text{L}_p(\text{SSR})$ : niveau de pression acoustique quadratique moyenne sur le contour de mesure 

Les corrections tenant compte des différences de caractéristiques de la SSR entre étalonnage et utilisation pratique (voir[3]), ainsi que des différences spectrales entre le bruit de la SSR et celui de la MSE, n'ont pas été prises en compte dans cette méthode. Il faudra sans doute le faire si on veut améliorer la procédure et la rendre plus précise. L'expérimentation déjà suggérée pourrait en fournir l'occasion.

CARACTERISTIQUES DES SOURCES SONORES DE REFERENCE
DIFFERENCES ENTRE VALEURS THEORIQUES (ETALONNAGE) ET VALEURS
PRATIQUES (UTILISATION "IN SITU")

FRANCOIS

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On a traité ailleurs [1] des problèmes liés au mode d'utilisation des sources sonores
de référence (SSR). Nous voudrions évoquer ici quelques questions restées pendantes
relativement à l'étalonnage de ce type de matériau. Les caractéristiques acoustiques
ainsi obtenues et que l'on doit prendre en compte lors de la détermination de la
puissance acoustique des machines soumises à essai, diffèrent en effet des caractéristi-
tiques déterminées dans les conditions d'utilisation pratique des SSR. Il convient alors
de savoir si ces différences entrent dans le domaine de dispersion de résultats acquis
avec une méthode peu précise du type "contrôle" [2] et que l'on se limite à ce type de
détermination, ou s'il est intéressant d'introduire les corrections correspondantes afin de
pouvoir prétendre à une précision supérieure dans le cadre d'une
méthode plus précise, du type "Expertise" [2].
Les différences constatées proviennent du fait que l'étalonnage des SSR est fait en
champ libre au-dessus d'un plan réfléchissant [3], alors que dans la pratique industrielle
on est amené à utiliser ce matériel en champ plus ou moins diffus, à une certaine
distance du plan réfléchissant horizontal (positionnement sur la machine soumise à
l'essai lorsqu'elle-ci est inamovible) ou à proximité d'obstacles réfléchissants
verticaux.
Des essais systématiques ont été entrepris à ce sujet dans le cadre d'un test
interlaboratoire national et donneront lieu prochainement à la diffusion d'un rapport
détailé. L'expérimentation a été faite par 6 laboratoires qui ont testé le même
ensemble de SSR, toutes du type à axe de symétrie acoustique verticale : EDF/AIRAP
(type A), CIMPO/INRS, B & K (type 4204). L'essentiel des résultats est donné dans le
tableau joint, sous forme d'écart de niveau de puissance (global A et octave) entre les
valeurs d'étalonnage et les valeurs obtenues sur site, pour les différentes variables
indiquées plus haut. Il s'agit ici de valeurs moyennes lues sur des courbes liées :

\[
\Delta i (dB) = L_w(\text{étalonnage}) - L_w(\text{variable i, "in-situ")}
\]

<table>
<thead>
<tr>
<th>Variable i (positions SSR)</th>
<th>Global A</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>0,12</td>
</tr>
<tr>
<td>Champ diffus, selon test</td>
<td>+ 1,3</td>
</tr>
<tr>
<td>Champ diffus, selon [4]</td>
<td>+ 1</td>
</tr>
<tr>
<td>à 1 m du sol, champ libre</td>
<td>- 0,2</td>
</tr>
<tr>
<td>à 1 m du sol, champ diffus</td>
<td>0</td>
</tr>
<tr>
<td>au sol, 1 écran latéral, d = 0,125 m</td>
<td>+ 1</td>
</tr>
<tr>
<td>au sol, 1 écran latéral, d &gt; 0,125 m</td>
<td>+ 0,2</td>
</tr>
<tr>
<td>au sol, 2 écrans latéraux //, d &gt; 0,125 m</td>
<td>+ 0,1</td>
</tr>
</tbody>
</table>

Dans le cadre de méthodes de "contrôle" [2], où l'on ne détermine que la valeur globale
du niveau de puissance, des corrections ne semblent pas nécessaires pour tenir compte
des différences existant entre les conditions d'étalonnage de la SSR et celles de son
emploi, alors que dans le cadre des méthodes "d'expertise" [2], avec détermination par
bande d'octave, ces corrections apparaissent nécessaires.

[1] P. FRANCOIS : Mode d'utilisation des sources sonores de référence pour la
détermination de la puissance acoustique des machines - 10e ICA SYDNEY 1980.
[2] Termé défini par la norme internationale ISO 2204
Noise Control Engineering (USA), Vol. 9, n° 1 (July-August 1977), pp.6-15.
Noise From Kitchen Blenders

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One of the most versatile appliances in a typical American kitchen is the food blender. It is used for a variety of mixing and whipping functions. Even though a blender is not used for long periods, the noise produced by this appliance can substantially add to other kitchen appliance noise and produce an annoying environment.

For this project, six popular brands of kitchen blender were purchased from retail stores. These units were then measured for dBA and 1/3 octave spectra in several typical kitchens and in a 292 m$^3$ reverberation room. The blenders were operated over the complete speed ranges and with different viscosity fluids to simulate various mixing tasks.

Most of the blenders exhibited approximately 2 dBA variation in operator noise level between the four test kitchens. When operating half full of water, the change in sound level from lowest to highest speed setting reached 10 dBA. The quietest model produced 86-88 dBA levels in the four kitchens while the noisiest unit developed 87-93 dBA.

One unit was selected for the application of noise reduction techniques. The major noise sources were motor, fan, cooling air flow, and structural vibration. Several techniques are presented which reduced the overall noise level 10-12 dBA.
NOISELESS HOUSEHOLD REFRIGERATION IN THE EIGHTIES?

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In published studies of the noise levels of domestic appliances, refrigerators normally take first place with the lowest noise levels in dB(A), if mentioned at all (1). It is generally known that annoyance is not merely a question of noise level but rather a question of "how quiet is quiet enough in a given situation" (2,3). In the case of automatic refrigeration equipment, where we cannot influence operation, it is natural that user and authority alike should enforce the most rigorous requirements. In Denmark, the authorities permit max. 40 dB(A) in kitchens and max. 35 dB(A) in living rooms and open kitchens. In Sweden, the corresponding levels are 5 dB(A) lower.

Despite many improvements through the years, many of the refrigerators and freezers on market today would undoubtedly have difficulty in observing even the Danish requirements (4).

Noiseless household refrigeration is within reach, for example, by using absorption refrigerators. However, the energy crisis renders this solution un economical, and we must therefore assume that household refrigeration in the eighties (and even longer) will still be based on compressor-driven refrigerators. The question is, then: is it technically and economically possible to achieve a marked reduction in noise level on an already highly-developed mass product?

Some years ago, we made some experiments to find out how factors such as compressor vibrations, gas pulsations and airborne noise affect the noise level of a refrigerator (5). The results were combined with existing knowledge to form the basis for a new generation of hermetic compressors. This compressor is now on the market and is characterized by a noise and vibration level (6) which is lower than ever before.

In this paper we shall document the noise levels which can be obtained with refrigerators and freezers using the above mentioned compressor. The reduction in noise is so pronounced that even the rigorous Swedish requirements can be more than met. The theme "How close are we to a noiseless appliance" will be discussed.

REFERENCES

DISTURBANCE CAUSED TO SCHOOL TEACHERS BY NOISE

BUILDING RESEARCH ESTABLISHMENT

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INTRODUCTION

A survey has been undertaken to determine the disturbance caused to school teachers by noise, particularly road traffic noise. A short self-administered questionnaire was completed during the summer term by 999 teachers in 73 schools. Some teachers taught in classrooms on the noisy side of the school facing the road while others taught on the quiet side. Teachers were asked to indicate how bothered they were by noise from a number of sources using the four point scale, 'not at all', 'a little', 'quite a lot', 'very much'. During the following summer holidays noise recordings were made at each school in order to establish a level of traffic noise exposure outside each classroom facing the road and of general noise outside each 'quiet' classroom. Most schools were also exposed to aircraft noise and the recordings were analysed to yield measures of aircraft noise exposure.

RESULTS

The data was analysed in two ways to derive relationships between teachers' response and noise level. First, the degree of bother was scored by assigning a number from 1 (not at all) to 4 (very much). Regression analyses of individual bother scores and of mean bother scores in 1 dB(A) bands against external traffic noise level L10 were obtained. In both cases the correlation was highly significant with r = 0.66 (individual) and r = 0.97 (grouped). In the second method of analysis sigmoid curves were fitted to the data to provide a relationship between the proportion of teachers bothered and the noise level. These curves showed, for example, that 50% of teachers were bothered 'quite a lot' or 'very much' by an external traffic noise level of 64 dB(A)L10. The corresponding noise levels for responses of 10% and 90% were 55 and 72 dB(A)L10. Above an external traffic noise level of about 60 dB(A)L10 a higher proportion of teachers were bothered by traffic noise than by any internal source (eg, the class next door). Also at this level of traffic noise there was an increase in response to questions about noise in general and a higher proportion of teachers considered the classroom to be unsatisfactory as a working environment. It appeared that high levels of traffic noise disturbed a higher proportion of teachers than occupants of dwellings. The difference between the noise levels in L_{eq} dB(A) from road traffic and aircraft which bothered a given proportion of teachers was small.
LE BRUIT À L'ÉCOLE PRIMAIRE
UNIVERSITE PARIS VIII

MOCH-SIBONY
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PRINCIPAUX OBJECTIFS :
Nous avons étudié les effets, à long terme, du bruit d'avions sur de jeunes enfants (6 ans) fréquentant le cours préparatoire.

Nous avons comparé deux écoles situées dans une même zone d'exposition au bruit [zone A], proches d'un aéroport, dont l'une avait bénéficié de travaux d'insonorisation, l'autre pas.

Nous nous sommes intéressé à l'aspect subjectif de gêne (exprimée par les individus) et à son aspect objectif (perturbation des performances et des comportements).

EPREUVES PROPOSEES :
Une épreuve de mémorisation de chiffres, de résolution de problèmes ; un test d'attention, de tolérance à la frustration, d'apprentissage de la lecture et de discrimination auditive.

RESULTATS :
Nous retiendrons les résultats contradictoires entre la gêne exprimée par les enfants et la perturbation des performances. En effet, alors qu'aucun sentiment de gêne ne se dégage des déclarations verbales des élèves, nous avons observé, dans l'école insonorisée, la dégradation de certaines épreuves (résolution de problèmes, apprentissage de la lecture), des comportements (agitation psychomotrice et démobilisation), des attitudes (tolérance à la frustration).

Nous pensons qu'une partie de nos résultats peut être interprétée par un phénomène de surcharge, "overload", qui serait à l'origine d'une fatigue physique et physiologique. Nous estimons ainsi, d'après notre étude, que le développement normal des facultés d'attention peut être perturbé chez de jeunes enfants à la suite d'une exposition, d'une année, au bruit d'avions.
Noise and other causes of interruption to work in offices

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Although the development of acoustics has made it easy to design office interiors so as to minimise problems associated with noise, office premises nevertheless continue to be designed, built and furnished in ways that fail to eliminate annoying and disruptive levels of noise.

The blame for this can be laid on the architect, the developer and the lessee, and to a lesser extent, on furniture manufacturers. Each will pass the responsibility for a good acoustic environment to another. The architect will not forcefully advise a developer client wishing to produce "competitive" rentable space, and the lessee is frequently unaware of noise problems caused by the building fabric and/or furniture until he has taken possession and is receiving complaints from his staff.

A request was made by a large Australian Government department for assistance in isolating noise problems in its newly leased centre-city office space. Anecdotal evidence from office staff itemised a number of alleged sources of noise which were irritating and/or which interrupted on-going work. A diary method of surveying the office staff was evolved which would yield a rank ordering of the frequency of interruption due to various different kinds of noise, and the degree of annoyance attributed to each, as well as pin-pointing the noise source, and the locations within the office which were most affected.

The method and results in this particular office are detailed in the paper. Its importance is that it is a method which can be carried out by the management of almost any office without requiring sophisticated acoustic equipment and it then allows a consultant to provide remedial specifications for each separate noise source. This provides the office management with a series of options to improve the environment according to finance available.
SCALE MODEL STUDIES OF SOUND PROPOGATION IN INDUSTRIAL SPACES

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It is now generally considered that the techniques of acoustic scale modelling are well established for enclosures such as auditoria and music studios (1,2). The present investigation uses similar techniques to study the behaviour of sound fields in industrial enclosures (3).

The situation in industrial spaces is rather different from auditoria in that the sound field is usually non-diffuse and so cannot be analysed by Sabine's reverberation theory. This is principally because industrial halls are usually very low compared to their length and breadth. In such enclosures the sound level from a single sound source drops continuously with increasing distance and does not approach a constant reverberant level. Furthermore, there are usually many sound sources together with scatterers distributed over the whole floor space. The complexity of analysing these sound fields theoretically is the fundamental reason for resorting to acoustic models.

A model has been built at 1:16 scale of an existing industrial building which houses 12 similar production lines manufacturing domestic light bulbs. The shape of the building is typical of many industrial structures and has the following dimensions: Length = 120m, Height = 10m, Width = 45m.

Model measurements of sound propagation with distance from one of the production lines have been made with the help of a motorised microphone carriage. Some measurements have been made from a single point source whilst others have been made using a line source. These sources have been constructed using various types of air-jet. Initially, the model results were compared with similar ones made in the prototype factory to ensure that the accuracy of modelling was adequate. Subsequently, the effect of various noise-reducing treatments on the sound propagation is being studied so that the most efficient measures can be singled out.

REFERENCES
1. F. Spandock, 5th ICA (1965) 11, 313;
2. H.D. Harwood and A.N. Burd, BBC Engineering (1972) 22 25;
REDUCTION OF NOISE FOR SAWING AND ABRASIVE CUT-OFF OF ROLLING STOCK

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The use of circular saws for cutting red hot rolling strands for manufacture of merchant bars causes a sound level up to 115 dB(A), measured at a distance of 1 m. The causes were studied. These are, in addition to the marked increase of the air-borne sound pressure level with larger feed and a minor influence of the cutting speed, above all instabilities at the metal-cutting blade in form of natural saw blade vibrations, lateral jamming of the workpiece in the cutting gap and hence flexural vibrations of the saw blade. The following constructional possibilities to reduce the sound level were found:

- use of saw blades with maximum possible number of teeth,
- insulating intermediate layers at the workpiece clamping device to reduce the excitation of the substructure,
- vibration damping of the saw blade by means of inserts in the clamping flange,
- vibration damping of the saw blade by means of back-up rollers,
- laminated saw blades of composite structure.

However, these measures enable only sound level reductions of 1 to 5 dB(A). If abrasive cut-off disks are inserted in the hot iron sliding frame saw instead of saw blades, a reduction of the sound power level of maximum 15 dB(A) is obtained at unchanged operating conditions. Further advantages of hot abrasive cut-off are short cutting times and hence high specific cutting capacity and consequently better economics, especially for cutting special steels. Hot abrasive cut-off needs no subsequent deburring.

The work on which this report is based was supported by funds made available by the Minister of Research and Technology of the Federal Republic of Germany (Support No. 01 VA 035-B 13 TAP 03). The author is however responsible for the content.
A method is proposed for the diagnostics of sound and vibration signals from reciprocating machines such as internal combustion engines. Such signals consist of a series of impulsive events, e.g. explosion, piston slap, etc., (ref. 1) and very short impulses, although noticeable to the ear, do not always have a measurable effect on a conventional frequency analysis, where the energy is "smeared out" over the whole cycle time. Moreover, the time of occurrence of an event may be just as important diagnostically as its frequency spectrum, since many excite the same structural resonances (ref. 1).

The solution proposed is to make a series of overlapping short-term analyses, distributed throughout the cycle, by moving a window function (e.g. Hanning) with respect to a once-per-cycle trigger pulse. In each window position an averaged power spectrum over several cycles gives a stable repeatable result. Fig. 1(a) and (b) shows a 3-dimensional representation of typical results from a 4-cylinder, 4-stroke diesel engine operating at 1500 rpm (80 ms cycle time), for delayed and advanced injection, respectively. The spectra were obtained from an FFT analyzer and have a linear frequency scale, but were plotted digitally using a desktop calculator. The latter can also be used to convert the spectra to constant percentage bandwidth on a log scale for data reduction and noise evaluation purposes. As will be seen, the method facilitates identification of the various events both with respect to their frequency content and time of occurrence, and is applicable to both noise source identification and condition monitoring.

REFERENCE:

Fig.1 Frequency-time representation of diesel engine cycles.
Le traitement acoustique de locaux industriels à l'aide de baffles suspendus permet d'obtenir des coefficients d'absorption élevés. L'efficacité acoustique de ces baffles est conditionnée par leur disposition, mais également par la forme du toit sous lequel ils sont suspendus. Une faible quantité de matériaux absorbants peut conduire à un traitement acoustique efficace et économique ; l'étude entreprise par le laboratoire d'acoustique de l'I.N.R.S. a pour but de préciser les propriétés acoustiques de telles structures absorbantes périodiques afin d'optimiser leur utilisation.

APPROCHE THEORIQUE

Les phénomènes physiques existant lors de l'interaction des ondes sonores avec une structure périodique, s'apparentent à ceux décrits dans la théorie des réseaux optiques. A la réflexion spéculaire viennent s'ajouter des réflexions secondaires plus ou moins nombreuses, dont les directions de propagation et les intensités dépendent de la direction de l'onde incidente, de la géométrie de la structure et de la fréquence considérée.

Pour des ondes planes, l'apparition d'ordres spectraux est observée, comme dans le cas des réseaux optiques. Une approche rigoureuse du problème a été proposée par de DE BRULJN (1), puis par ANDO et KATO (2). Ces derniers donnent une solution analytique au calcul de l'intensité des divers ordres spectraux, dans le cas d'une onde plane incidente sur une structure périodique de profil quelconque. Ce calcul est précis mais sa programmation sur ordinateur est délicate à mettre en oeuvre. Une méthode plus simple et plus souple d'emploi consiste à considérer la diffraction due à chaque élément du réseau, comme satisfaisant aux critères de la théorie géométrique de la diffraction développée par KELLER (3).

CONSTATATIONS EXPERIMENTALES

Un certain nombre de mesures ont été effectuées sur des maquettes et dans des locaux industriels dont les plafonds avaient une forme régulière et périodique. Les ordres spectraux sont effectivement observés pour des structures simples éclairées par une onde quasi-plane en champ libre. Pour un local fermé, les nombreuses réflexions sur les diverses parois et le caractère sphérique de l'émission sonore rendent négligeables les phénomènes d'ordres spectraux. On constate cependant une absorption acoustique élevée, même pour une faible quantité d'absorbants, comme le montre la figure jointe (4).

PROPOSITION POUR UN CALCUL SIMPLIFIÉ DE L'EFFICACITÉ ACOUSTIQUE DES BAFFLES SUSPENDUS

Si le champ sonore dans le local est considéré comme diffus, le traitement acoustique par baffles suspendus est supposé constitué un couplage entre deux locaux réverbérants, l'un situé sous les baffles, l'autre au-dessus. Cette hypothèse conduit à une expression simple d'un coefficient d'absorption apparent de la structure. Des vérifications expérimentales de cette expression ont été faites en chambre réverbérante. Elles confirment le bien fondé de cette approche. Pour des cas plus complexes, baffles suspendus sous des toits de forme spéciale (sheds par exemple), on propose de considérer le plafond comme équivalent à une structure diffusante dont le diagramme de directivité est calculé au préalable, dans le cas d'une émission en ondes planes. L'introduction de ce concept de paroi équivalente plane dans un programme plus général de calcul de la propagation de bruit dans un local industriel par la technique des rayons sonores (5) devrait être relativement aisé. Ceci sera l'objet de travaux futurs.

(1) DE BRULJN, Acustica, vol. 18, n° 3, p 123-131 (1967)
(4) CORMIER, Thèse de doctorat 3ème cycle, Université du Maine (1980)
(5) LEBLOND - LECOCQ, INTER-NOISE 78, et Notes Scientifiques et Techniques INRS.
Introduction  The addition of sound-absorbing material to rooms has always been regarded as a fundamental method of controlling noise in industrial and commercial buildings. Yet the alleviation of a noise problem is not always apparent, if the calculated or measured noise level reductions are considered.

Subjective Survey  In a subjective survey undertaken in a workshop, after sound-absorbing treatment was added, employees estimated that the noise level had dropped to between one half and one quarter of the level before the treatment was added. Questionnaire returns also indicated that conversation was much easier, after the addition of the absorbing material which reduced the overall reverberation time from 2.3 seconds to 1.1 seconds.

There was no significant difference in the $L_{10}$ sound level before and after the addition of absorbing material but there was a small reduction in the $L_{90}$ level.

Theory  The theoretical investigation of impulse noise levels in a room showed that the percentile sound levels should change when absorption is added. The percentile levels will depend on the time between pulses and the reverberation time of the room, for a given pulse duration (See Fig. 1).

Conclusion  It would seem from these results that the subjective assessment of impulse noise probably depends on the time between pulses as well as the characteristics of the pulses and the room in which they are heard, i.e. steady state sound level reduction predictions and sound level measurements made using 'slow', 'fast' or 'impulse' response do not give a good indication of the subjective assessment when sound absorbing material is added to rooms where impulse sound sources operate.
The introduction of noise legislation in industries has meant that machinery designers and purchasers must consider more carefully the noise emission from their machines. In many industrial situations the age old solution of enclosing the noise source is not possible and an attempt has to be made at noise control at source.

While studies of the vibrations characteristics of a machine or a fluid flow are not possible in detail, the dependence of noise output upon simple parameters such as the rate of change of applied force, the number of wavelengths in a typical machine component dimension, the damping factor, the bulkiness or stiffness, permit simple laws to be established. These can be used as diagnostic tools by design engineers to include noise control alongside the more conventional design rules of strength, accuracy and reliability.

The paper presents these elementary laws and gives two examples of how useful they can be.
A QUIET ROTARY LAWNMOWER THAT CATCHES GRASS

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INTRODUCTION: The noise produced by rotary mowers is caused mainly by aerodynamic disturbances associated with the cutting blades [1], [2]. This paper describes an investigation of aerodynamic blade noise and methods by which it can be reduced. Part of the work was supported by Victa Ltd.

AERODYNAMIC BLADE NOISE AND GRASS CATCHING: A popular domestic brand mower of 0.457m cutting width was investigated. The dominant aerodynamic noise source was interaction between the airflow around the rotating blades and features on the housing (blade-housing interaction). It can be eliminated by avoiding salient features close to the path of the blade tips. The other important source (wake shedding noise) is due to disturbed airflow around the cutting blades. Features which produce wake-shedding noise, that is, paddle-like blades and high tip speeds (84 ms⁻¹), are essential for grass catching in current rotary mowers. The blade tip speed necessary to cut grass is about 60 ms⁻¹. A substantial noise reduction would result if grass could be caught at the lower cutting speed. Measurements indicated that for successful grass catching, an air swirl velocity of about 30 ms⁻¹ near the blade tips was necessary. A newly designed vortex mower utilizes a fan mounted above a curved blade disc to generate the necessary air movement at a blade tip speed of 60 ms⁻¹.

RESULTS AND DISCUSSION:

Fig.1 shows aerodynamic blade noise and grass catching capacity versus operating speed for three mowers. They are the standard (commercial) mower, the commercial mower modified to reduce blade-housing interaction (modified commercial) and the Vortex mower. The improved grass catching performance of the modified commercial mower results from its deeper housing and catcher. The housings and catchers of the modified commercial and Vortex mowers are identical in size and shape. The Vortex mower can operate at a speed 30% lower than that of the modified commercial mower while maintaining the same grass catching capacity. Overall it is 11 dBA quieter than the original commercial mower, and it catches grass better.

REFERENCES:
2. PHELPS, R. Victa Ltd., personal communication, 1974.

Fig.1. Noise and grass catching capacity vs operating speed (O vortex mower; + modified commercial mower; A commercial mower).
PREDICTION OF ACOUSTIC PERFORMANCE OF MUFFLER SYSTEMS

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Prasad, M.G.

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INTRODUCTION

There are several ways to describe the acoustic performance of muffler systems. The most useful ones are: prediction of the sound pressure level, SPL, radiated at a given distance from the tail pipe and the insertion loss, IL, of the muffler system. These parameters are easy to measure but difficult to predict [1,2,3]. Schemes for SPL prediction and insertion loss of any general exhaust system and sample results for a model system are given [2,3].

PREDICTION SCHEMES

RADIATED SOUND PRESSURE [3]: The sound pressure radiated from a straight pipe system (without muffler) is measured, and using the theoretical four-pole matrix of a straight pipe and measured source impedance, \( Z_r \), the source strength \( V_e \) is evaluated. For a given muffler system, the transfer impedance \( Z_e \), with respect to the position of SPL prediction is evaluated. Then the radiated sound pressure level is:

\[
\text{SPL} = 20 \log_{10} \left( \frac{Z_r V_e}{\rho\omega} \right)
\]

INSERTION LOSS [2]: The insertion loss is predicted using the measured source impedance and theoretical four pole matrices of the muffler and a straight pipe (primed quantities) system. The insertion loss is:

\[
\text{IL} = 20 \log_{10} \left| \frac{(A'e_z + B' + C'z_e + D'e_z)}{(A'e'_z + B' + C'z_e + D'e'_z)} \right|
\]

RESULTS AND CONCLUSIONS

The above schemes can be used in general for any exhaust system. Results are given for a model system comprised of an electro-acoustic driver, expansion chamber and tail pipe. The source impedance was measured and radiation impedance, \( Z_r \), evaluated. Results are in good agreement provided measured rather than assumed source impedance is used.

REFERENCES


Fig. 1 Radiated SPL at 0.406 m
— theory, --- expt.

Fig. 2 Insertion Loss
— theory, --- expt.
Acoustic optimization of a vent silencer.

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Recent research work (e.g. 1) indicates that noise reduction by means of overcritical expansion in a vent silencer can be optimized by modifying the type of expansion (axial or radial), the free area of flow (diameter, spacing and the number of holes) and the number of stages of expansion. It is known that radial expansion offers a larger area of flow for a given diameter than the axial type. The free area of flow is optimized by distributing it over a large number of small holes.

Preliminary experiments were made with a single stage and a two stage radial expansion silencer with natural gas. The total pressure ratio was varied to 9 : 1. The silencer was fitted with three perforated cylinders with 6, 5 and 3 mm holes. These cylinders could be fitted in any combination. The velocity of flow across the cylinders varied from about 30 m/sec to about 100 m/sec. The silencer casing was lined with a 50 mm rockwool packing.

The maximum insertion loss of about 40 dB obtained from a series of tests is shown in Fig. 1. The concentric perforated cylinders covering the expansion unit varied in performance; these differences were attributed to the flow noise generated by the flow through the outermost cylinder.

REFERENCE:
(1) R.M. Ellis, M.V.H. TOOKEY, The noise generation and silencing of high intensity steam discharges, Paper E7, ICA-Madrid 1977
A TUNED MUFFLER/SILENCER FOR LOW FREQUENCY FAN NOISE CONTROL
VIPAC & PARTNERS PTY. LTD.

DR. BRONER
NORMAN

30-32 CLAREMONT STREET, 5TH YARRA, 3141, VICTORIA, AUSTRALIA.

INTRODUCTION
Following a community noise complaint regarding the noise emission of a paper-coating plant, it was determined that the major source of annoyance was a large paddle fan used to transport scrap paper. The blade passing frequency tone of 168 Hz and up to five harmonics were quite clearly seen in the community noise spectrum. Due to stringent space constraints, a standard type of solution could not be utilised to achieve the required reduction at 168 Hz so it was decided to install a tuned muffler as described by Ver and Biker\(^1\) with its high selective attenuation per unit length. Two absorptive faces were also added to achieve overall attenuation at higher frequencies.

DESIGN
The tuned cavity, with porous lining on the walls, has the advantage of providing high selective attenuation without directly exposing the porous dissipative material to contaminated flow. The dimensions were scaled from Ver and Biker\(^1\) resulting in five cavities of 265 mm depth, overall length of 1325 mm and breadth of 200 mm (so as to maintain the required flow velocity). The absorptive faces were 1325 mm long and presented 50% open flow area.

LABORATORY TEST
An Insertion Loss test was performed on the tuned muffler alone, on the two absorptive faces alone and on the combination. Table 1 shows the results.

<table>
<thead>
<tr>
<th>Frequency of Attenuation (Hz)</th>
<th>168</th>
<th>335</th>
<th>505</th>
<th>672.5</th>
<th>840</th>
</tr>
</thead>
<tbody>
<tr>
<td>Tuned Muffler</td>
<td>-13</td>
<td>-1.5</td>
<td>-3.5</td>
<td>-0.5</td>
<td>-5</td>
</tr>
<tr>
<td>Two Absorptive Faces</td>
<td>-6.5</td>
<td>-9</td>
<td>-15</td>
<td>-16</td>
<td>-6</td>
</tr>
<tr>
<td>Muffler/Silencer</td>
<td>-13</td>
<td>-8.5</td>
<td>-20</td>
<td>-10</td>
<td>-11</td>
</tr>
</tbody>
</table>

CONCLUSION
Following installation of tuned muffler/silencers on both inlet and outlet sides of the paddle fan and appropriate lagging of the fan casing, no further community complaints were registered. This solution thus provides a successful means of noise control of fans given a tight space constraint; in this case a 50% space saving.

REFERENCE
Development of Noise Reduction Technology for Armored Track Laying Vehicles.
US Army Human Engineering Laboratory, Aberdeen Proving Ground, MD 21005, USA
Carinther Georges R.

US Army Human Engineering Laboratory, Aberdeen Proving Ground, MD 21005, USA

INTRODUCTION

Excessive noise in military tracked vehicles is known to cause hearing loss, degraded speech intelligibility and early aural detectability. Because of these effects the army is pursuing a program to develop noise reduction concepts and to provide the technology necessary to produce a light, track laying vehicle of the M113 family which has an interior sound level of 100 dB(A).

METHOD

In a paper presented at the 9th ICA we rank ordered the major noise sources of the M113 Armored Personnel Carrier and developed a preliminary mathematical model of the track and suspension system. Vibration-to-noise transfer functions were experimentally derived for inclusion in the model.

The primary noise sources of the M113 are, in order of noise level, the idler, the sprocket and the roadwheels. As a result of this information a high compliance idler has been designed and fabricated. This experimental idler consists of a series of eleven "paddles" each mounted radially in rubber-in-shear springs made of two 1/2" thick rubber pads.

A statistical power flow analysis was conducted to determine the interior radiation pattern of the hull and its various panels. The internally radiated sound power level of each panel was determined to aid in determining which panels should be acoustically treated. Reverberation time of both the hull structure and the interior volume were determined along with the acoustical absorption of the crew area, both occupied and unoccupied. Based upon these data, both normal damping and constrained layer damping were tried on the most promising areas.

A computer model of the track and suspension system was designed. The objective of this model was to predict interior noise levels induced by the track and suspension system.

RESULTS AND CONCLUSIONS

Testing of the high compliance experimental idler indicates that it provides noise reduction of 10-12 dB(A). Cycling the rubber for 20 minutes at a simulated speed of 30 mph produced no evidence of damage due to fatigue or overheating. Based upon these promising results, a practical prototype idler is being fabricated which consists of a solid outer hoop mounted on rubber-in-shear springs. Also, an experimental sprocket is being constructed which is anticipated to provide greater than 12 dB(A) noise reduction. Based upon the results of the statistical power flow analysis, damping treatment of various parts of the vehicle provided modest noise reduction. Interior sound absorptive treatments of major portions of the vehicle provided little noise reduction below 500 Hz and 3-5 dB at higher frequencies.

It is anticipated that a quieted track vehicle can be built in which suspension noise has been reduced by 10 dB(A), hull noise by 5 dB(A) and in which track changes produce a noise reduction of 4 dB(A).
DESIGN AND DEVELOPMENT OF A QUIET PNEUMATIC ROCK DRILL

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The percussive-type pneumatic rock drill is a major noise source in mining. A new drill is described in which substantially quieter operation has been achieved.

Design of this drill was based on two broad guidelines. The first was that it be a truly new design, not necessarily influenced by existing drill design practice. The second guideline was that the design throughout be such as to minimise noise.

This approach resulted in the selection of an expansive operating cycle, which is unusual in modern drills. Other unconventional features were the elimination of some noise sources altogether and a noise-insulation type of construction.

The operating cycle was developed by means of analog simulation and model testing. Prototype drills were built and tested extensively. From test results a comprehensive computer simulation was developed and used to optimise drilling rate.

It is considered that the design is now at a stage suitable for commercial development.
SCHALLREDUKTION MITTELS RESONATOREN

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A REVIEW OF MILITARY NOISE CRITERIA AND STANDARDS

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INTRODUCTION

Three types of noise criteria have evolved over the years. (1) Hearing Damage-Risk Criteria (DRC) are statements of the relation between various parameters of noise exposure and the risk of hearing loss. (2) Hearing Conservation Criteria (HCC) are noise limits which, when reached or exceeded, are indication for the employment of hearing conservation measures. (3) Materiel Design Standards (MDS) provide specific noise limits to equipment designers and manufacturers.

Damage-Risk Criteria and Hearing Conservation Criteria are aimed primarily at prevention of hearing loss. Materiel Design Standards include design limits intended to conserve hearing but, in addition, they cover such other topics as: insuring adequate communications; minimizing community annoyance; long-term habitability; and denying aural intelligence.

EVOLUTION OF U.S. MILITARY NOISE CRITERIA

In general, DRC have not been independently developed by the US military services. Rather, the services have tended to adopt or adapt industrial or other civil DRC as the basis for their HCC and MDS.

Hearing Conservation Criteria and MDS have tended to evolve independently within the three military services. Undoubtedly, this has been due, in part, to differing operational requirements of the services.

Recent promulgation by the US Department of Defense of DOD Instruction 6055.3 has resulted in activity, particularly on the part of the Army and Navy, to bring all the services' HCC into compliance with the DODI. Simultaneously, the services are upgrading their respective MDS to reflect the new (lower) permissible exposure levels.

Discussions are also being conducted among the services regarding the desirability of joint publication of HCC and/or MDS.

SCOPE OF THIS PRESENTATION

This review will present the current status of the three services' noise criteria, summarize the revision activities that are in progress, and contrast the limit provisions for various applications. Representative comparisons will be made with other nations' military standards, as well as national military versus civil exposure criteria.

REFERENCES

Rating the Hazard from Intense Acoustic Impulses

U.S. Army Human Engineering Laboratory, Aberdeen Proving Ground, MD 21005, USA

Price, G. Richard

INTRODUCTION

Accurately rating the hazard from intense acoustic impulses, such as those produced by weapons, remains a problem because of the lack of a theoretical understanding of the mechanisms of hearing loss at high intensities and how they relate to the exposure conditions. The experiments reported here were undertaken in order to begin filling these gaps in knowledge and move toward a theoretically based damage-risk criterion for intense impulse noise exposure.

EARLIER WORK

Work reported at the 9th ICA (1) was extended to establish the susceptibility of the structures on the basilar membrane to spectrally narrow impulses. Expressed in stapes displacements, susceptibility increases at 5.4 dB/oct with increasing frequency (2). Furthermore, the data were interpreted as supporting the contention that the mechanism of loss at high sound pressures is mechanical stress at the level of the hair cell. Following this lead, arguments were made that for intense sounds there is a spectrum-dependent critical level at which the primary loss mechanism changes to one of mechanical stress (3). Given the foregoing, a new method for rating the hazard from weapons impulses was developed, based in part on the spectral content of the impulse (4). This method predicts that the ear will be most susceptible to impulses with peak energies in the mid-range (1.0-3.0 kHz) and that higher peak pressures can be tolerated as the frequency departs from this region.

EXPERIMENTS

In order to test this prediction, cat ears were exposed to impulses from the recoilless rifle, 105 mm cannon, and 5.56 mm rifle. Changes in sensitivity were measured by combinations of brainstem and electrococchleographic audiometry. Measures were made before, immediately after, 1 day after and at intervals up to 2 months following exposure.

RESULTS

Preliminary results are consistent with the contention that impulses with peaks in the low-frequency region are relatively less hazardous than previously thought. For cannon fire (one exposure of 60 rds) the threshold of permanent loss is in the vicinity of 155 dB peak SPL for the cat ear. Data will be presented comparing the hazard from cannon fire with that from rifle fire. This comparison, from a theoretical standpoint, is a crucial one for the validation of the concept developed above of a spectrally based rating of hazard for weapons impulses.

REFERENCES

INFRASOUND IN INDUSTRY

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INTRODUCTION
The noise and vibration investigation in industry has indicated that the effect of LF and infrasonic components on human should be taken into account.

INFRASOUND IN INDUSTRY
Numerous industrial plants and installations in various industries were measured and analysed. In many cases the components in the range from 2 up to 50 Hz were strongly expressed (Fig.1). Some investigations have shown that the workers exposed to the noise where the low frequency and infrasonic components were more expressed, suffered more frequently from discomfort and diseases in comparison to those exposed to the audible noise, even if this was for 10 dB higher. The measurement results accomplished by means of standard sound level meter applying weighting filter A can not give data on infrasonic component value. Therefore, the measurements should be carried out by special measuring equipment.

CONCLUSION
Wherever in industrial plants great movable masses are involved, it is possible that the LF noise and infrasonic components will be strongly expressed; in this case a detailed overall measurements in broader frequency range should be undertaken. In addition, further investigations in order to determine the limiting values of sound pressure level for infrasonic components or possible LF extension of NR-curves, are required.

Fig. 1. Third-octave spectra in the vicinity of 1) generator in hydro-electric power station, 2) fans in oil refinery and 3) burner in factory power station
ON OVERLOADING SOCIETY'S ACOUSTIC COMMUNICATION CHANNELS

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In the course of evolution the human species has adapted itself to communicating meaningful lexical meaning via the acoustic information channel; while hearing capability evolved prior to the evolution of the class Mammalia, speech capability is a more recent development. Thus, it is not surprising that ordinary human communication is limited by speech capability. It is the practical input capacity to man's acoustic information channel - - - limited to the primary speech frequency band as an uppermost limit. For the majority of (if not for all) humans those centers of the cortex evolved for speech/language processing are utilized not only for speaking and hearing language but also in writing and reading codified speech symbols communicated via optical or tactual communication channels. That is, such indirect communication of speech can be considered indirect acoustic communication.

The abuse of the acoustic communication channels of a society should be of concern to the acoustician. This paper presents some results of investigating the restrictions imposed on society and its structures by the inherent limitations of - - - direct and indirect - - - acoustic communication. But it should be noted that the results would remain essentially unchanged if optical, tactual, olfactorial, algesial, thermal, etc. channels had also been taken into consideration.

The constraints imposed by the acoustic communication channel require that a collection of individuals whose communications exceed the capacity of this channel must necessarily organize itself geographically and structurally into an unequal society. This is so even if all individuals were otherwise initially identical (which they are obviously not). But unless such a structuralization is allowed to proceed slowly and selectively it is certain to result in a very inefficient (even ineffective), uncompetitive structure which frustrates its individual members instead of maximizing their inherent capabilities.

Any attempt to overload the acoustic communication channel with excessive protocol (that is: detailed instructions and regulation; rules to govern and control individual behavior) results in social instabilities. In ordinary language this means society undergoes crises. It attempts to cope with each crisis by creating more crises. It also must develop substructures (such as bureaucracies) which benefit from crises they have caused. Such substructures must necessarily strive to prolong crises and to create new ones - - - quite independently of any desires or conscious efforts to the contrary. (Good, competent individuals necessarily tend to become effectively eliminated from such organizations.) The creation of too many instabilities (and the process is self-accelerating) must necessarily lead to the involvement of the military or change-control function; that is, to some sort of war. Another consequence of such loads are some very detrimental effects on the health of individuals exposed to such overregulation. Another effect is that the protocol becomes 'contradictory'; this has an extremely pernicious influence on society's legal/judicial system.

The only way to avoid overloading modern society's acoustic communication channel is to have maximum effective and self-regulatory freedom at the individual and local levels. This requires that regulations sent through the acoustic channel must be quite limited in extent and remain relatively constant over time. This implies that most regulation - - - in so far as any such regulation is really an unavoidable physical necessity - - should be accomplished indirectly via non-acoustic variables or parameters.

Acousticians have learned to use some of the results of mechanics, thermodynamics, mathematics, etc. in solving acoustic problems. It is to be hoped that in the 1980's they will begin to use cybernetic results in a similar manner. It is indeed regrettable that some acousticians have been in the forefront in advocating the abuse of society's acoustic communication channels rather than the converse.

(See the "full written version" for details and references.)
THE RELATION OF NOISE TO CARDIOVASCULAR FUNCTION

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In addition to hearing loss, many categories of illness and dysfunction have been attributed to long-term and intense noise exposure.

The most extensive and cohesive documentation for non-auditory noise effects relates to various forms of cardiovascular disorder. Alterations in cardiac electrical activity as well as anomalies of blood pressure regulation are frequently reported in workers exposed for many years to high levels of industrial noise and in laboratory animals exposed to similar levels of noise for shorter times. Despite an impressive uniformity of positive findings, certain inconsistencies and contradictions in the present literature and inherent limitations associated with both human and animal-model research render accrued information concerning this subject highly provocative but not definitive.

Nevertheless, the economic implications for industry on the one hand and the importance of workers' health on the other are of sufficient moment to warrant specification of as firm a causal link between noise exposure and cardiovascular function (or structure) as possible. Direct observation of human responses within the industrial environment may not alone provide the necessary link.

It has long been established that a useful supplement to human experimentation is carefully designed animal experimentation. In order to maximize the utility of animal data in predicting human responses, a number of design criteria should ideally be met. These criteria will be discussed in relation to a program of animal research recently initiated at the University of Miami.
VASOCONSTRICITING EFFECT OF INTERMITTENT NOISE

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It has been established that sound stimuli produce vasoconstriction of peripheral blood vessels via sympathetic nervous system but the relation of the response to perceived noisiness of sound has not been well studied. In the present studies, pulse amplitude of peripheral blood vessels was recorded by the photoelectric plethysmography from index fingers of human subjects who were exposed to intermittent noises. Amplitude of pulse waves decreased by the onset of noise and recovered slowly afterwards. Relative decrease of the amplitude and time for its recovery were measured and the relationships among the intensity of response, the nature of sound, and perceived noisiness judged by the subjects were studied.

In the first experiment, sixteen subjects of both sexes were exposed to intermittent pink noises having peak level (PL) of 62, 70, 80, or 89 dBA, duration of peak level (PD) of 2, 4, or 8 sec, and rise time (RT) of 0, 1, 2, 4, or 10 sec/10 dB. Noises were applied in a random order with an equal interval of 90 sec for 90 min. Intensity of the response was dependent on PL and RT but not on PD. The response showed a high correlation with the energy summation of noise expressed in dB but prompt noises of RT=0 sec/10 dB and slowly growing noises of RT=10 sec/10 dB produced more intense responses because of their startle effect and irritation effect, respectively. The response also highly correlated with the perceived noisiness judged by the subjects.

In the second experiment, a same number of subjects of both sexes was used. They were exposed to 30 sec noises with an equal interval of 90 sec for 90 min. Noises were tones, 1/3 octave band-noises, and 1 octave band-noises having PL of 60, 70, or 80 dBA and central frequency (CF) of 250, 500, 1k, 2k, or 4k Hz. The order of exposure of noises was randomized. Vasoconstricting response became more intense according to PL, band-width, and CF of noise. The response, here again, highly related to the perceived noisiness judgment made by subjects.

In the third experiment, effect of whole body vibration on the vasoconstricting response to noise was studied. Six male subjects, sitting on chairs fixed on shaking table, were exposed to noise and vibration of 30 sec every 2 min for 90 min. Sound level was 75 dBA (N1) or 85 dBA (N2). Vertical, sinusoidal 10 Hz vibration of 85 dB, 30 sec was applied alone (V) or in combination with noise (V+N1 or V+N2). Control experiment (C) without noise and vibration was also performed. Above six experiments; C, N1, N2, V, V+N1, and V+N2, were made on separate days. Vasoconstriction was produced by V as well as by N1 and N2. The response to N2 was larger than to N1 and that to V was between them. When noise and vibration were applied simultaneously (V+N1 and V+N2), the intensity of the response was inhibited. Further experiments on the combined effect of noise and vibration are being carried out.
INTRODUCTION

This paper reports on an investigation into noise generation due to orifices in pipelines with flow.

EXPERIMENTAL ARRANGEMENT

The investigation was conducted with 72-54mm I.D. cold drawn seamless tubing with three test orifices, with area ratios of 0.3, 0.57, and 0.8. Air was the working fluid and nominal centreline flow Mach numbers of 0.2, 0.35, 0.4, 0.45, and 0.5, were obtained from a vacuum induced flow rig. These Mach numbers correspond to a range of centreline velocities of 65m/s to 140m/s. The orifices could be placed at various positions upstream and downstream of the measurement position.

Measurements were taken of the internal wall pressure spectra, the vibrational response of the pipe wall and the acoustic radiation from the pipe for the various orifice sizes and flow speeds.

RESULTS AND CONCLUSIONS

Results obtained indicated that maximum pressure fluctuations occur immediately downstream of the orifice, in the region of separated flow just before the reattachment position. These pressure fluctuations excite higher order acoustic modes and plane waves which propagate down the pipe. The higher order acoustic modes were found to readily excite a vibrational response in the pipe wall and to dominate at distances greater than 10 pipe diameters from the orifice. Excitation by the individual modes could be identified. The response in the pipe wall, due to excitation by higher order modes, was increased by the phenomenon of coincidence. Close to the orifice, non-propagating pressure fluctuations were also significant.

An increase in the flow speed resulted in an increase in acoustic power radiated from the pipe by a power of $U^5$ to $U^4$ at low frequencies, $U^5$ at the cut off frequency of the first higher order mode and $U^8$ at higher frequencies.
PRESSURE DEPENDANCE OF JET NOISE

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1. Total noise of the jet. At low or high stagnation pressures, the jet noise consists only of the turbulent noise, and in the middle range of pressures, the noise is dominated by the shock-associated noise. A plot of the total noise power vs pressure reveals that the condition at the nozzle lip plays an important role to the sound emission. Nevertheless, the sound pressure level in 90° (jet axis) remains approximately constant, as

\[ L = 77 + 20 \log d + 10(R - 1.9)(R - 1.5), \quad 2 < R < 3.1 \]

\[ 97 + 20 \log d, \quad 3.1 < R < 8.6 \]

at 1 metre, \( L \) being in dB re 20\( \mu \)Pa, \( d \) nozzle diameter in mm, and \( R = P_t / P_o \), ratio of stagnation and ambient pressures.

2. Turbulent noise. The Lighthill's 8th power law at low velocities may be expressed in terms of pressure ratios, as

\[ L = 80 + 20 \log d + 20 \log^2(R - 1)/(R - 0.5)/ \]

Experiments show that the formula holds for \( R - 1 = 0.01 \) to 100.

3. Annoyance of turbulent jet noise. It is suggested that this is to be assessed by the A-Weighted sound level. It is found that

\[ L_A = L - 10 \log(12/\pi) (\tan^{-1}X - X/(1 - X^2)) \]

and \( X = 0.165d \) for choked jet. When \( d \) is small, say a few mm, the last term reduces to 30 log \( X - 4 \), viz. halving the diameter reduces the A-level by 9dB besides the reduction due to area change. This is the basis of multipole and porous diffuser.

4. Shock-associated noise. By taking the narrow-band spectrum of the jet noise, both the discrete-frequency and wide-band shock-associated noise are observed. The screech has harmonic as well as inharmonic components, and the basic Strouhal number

\[ Sh = \frac{fd}{c} = 0.36(R - 1)/(R - 0.5)^\frac{1}{2}(R - 1.893)^\frac{1}{2} \]

For the wide-band shock-associated noise, the peak agrees well with above formula in 180° direction (upstream). The 3dB band width, however, is only about half octave as against 1.8 octave for the turbulent noise. One notable phenomenon is that in the upstream direction interference exists between the turbulent and shock-associated noises, at frequencies some what below the peak of the latter. But one observes no interference above the peak, nor for any frequency in the 90° direction.

5. Reduction of the shock-associated noise. This can be done easily by slight modification at the nozzle, e.g., by adding a few vanes, knife edges, gauze cylinder, by cutting a V-notch, etc. The whole shock cell system is thus destroyed, as shown by the noise field.
ANNULAR JET OF DIFFERENT DIAMETER RATIOS

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Ko
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INTRODUCTION

In basic annular jet wake vortices in the centre and jet vortices in the other mixing region have been observed [1, 2]. It was also observed that the disturbances of the wake vortices in the centre propagated into and excited the outer mixing region. This resulted in the formation of another train of vortices. However, the above study was based on the diameter ratio which Strouhal number of the wake vortices was near that of the most preferred mode of the jet [3]. The present study will investigate the vortices of basic annular jet of different diameter ratios.

EXPERIMENTAL TECHNIQUES

The nozzle has an outer diameter $D_0$ of 20 mm. Two diameter ratios $D_0/D_1$ of 2.9 and 1.5 were used. The exit Mach number of the jet $U_f/U_0$ was 0.4. Schlieren technique was used to flow visualize the jet. The photographic averaging technique of 30 superimposed schlieren images on a single photographic negative was adopted.

RESULTS

For the basic annular jet of a diameter ratio of 1.5 the Strouhal number $fD_0/U_0$ of the jet vortices and of the wake vortices is 0.50 and 0.29 respectively. Because the Strouhal number of the wake vortices is near the most preferred mode of excitation of the jet, another train of wake excited vortices inside the outer mixing region is observed from the schlieren photographs.

For the jet of diameter ratio of 2.9 the Strouhal number of the jet vortices is 0.78. It is higher than that of the lower diameter ratio. For the wake vortices inside the jet the Strouhal number is only 1.19. This Strouhal number is far above that of the most preferred mode of the jet. Because of this, no wake excited vortex is observed in the schlieren photographs obtained.

The phase velocities, as obtained from the schlieren photographs, of different types of vortices of the two basic annular jets are shown in figure 1. They seem to suggest increasing phase velocity with the axial position of the vortex.

REFERENCES

NOISE AND YOUNG PEOPLE'S HEARING

NATIONAL ACOUSTIC LABORATORIES SYDNEY AUSTRALIA

CARTER, Norman L; KEEN, Keith; WAUGH, Richard L; BULTEAU, Volney G; MURRAY, Narelle M; KHAN, Albert E.

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On the basis of audiometric data on sixth and ninth grade children, high school seniors and university students Lipscomb [1, 2] has said that ".... is tangible evidence of the toll being exacted by high intensity recreational and environmental sounds...." and that ".... establishing or validating audiometric standards seems virtually impossible at present in the major industrial areas of the civilized world because of already damaged hearing." These are serious claims, and data by Hanson and Fearn [3, 4] and Skurr and Bulteau [5] appear to lend some support to them. However it will be argued that these data are equivocal and that survey data of the type supplied by Roberts and Ahuja [6], while essential for the purpose it was intended, necessarily confounds noise and aural pathology with demographic variables.

Some years ago a study was begun at MAL of several groups of young people under age 21 which included audiometric, acoustic impedance and ear, nose and throat examinations, history taking and administration of a questionnaire concerning recreational and occupational noise exposure. The groups were 10-12 year olds (now being followed up at age 16-18 years), and four groups selected to represent various degrees of recreational and occupational noise exposure. The four groups were university students under age 21 (mainly exposed to recreational noise); office workers in the Commonwealth Public Service (recreational noise in employed young people) and first and third year apprentices (recreational and occupational noise exposure).

Selected data on hearing levels and estimated noise immission from various sources will be given for each of the above groups, and trends in occupational and recreational noise immission compared.

1. D.M. LIPSCOMB. Audiology 11, 231-237 (1972)
2. D.M. LIPSCOMB. Clinical Pediatrics 11, 374-375 (1972)
HAZARDS OF NOISE EXPOSURE IN MUSICIANS USING HIGH POWER ELECTRONIC EQUIPMENT
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INTRODUCTION

The danger of hearing loss due to exposure to high intensity rock music has been discussed in numerous reports over the past three decades. However, the results and implications by various investigators differ both with respect to intensity levels and to the degree to which such exposures are dangerous. This report deals with the danger of hearing loss in musicians and with usefulness of DRC in the determination of permissible exposures.

METHOD

Live records of a routine rock performance of the recognized Polish group /12 typical music pieces/ were analysed using conventional methods. TTS₁ was measured in the performers /four subjects/ immediately after the performance had finished and used to find TTS₂. Resting hearing levels were measured 48 hrs after the performance. The measurements of both were repeated after 12 months.

RESULTS

The results of intensity measurements are given in Fig. 1. Example of resting hearing level and TTS₂ in one of the subjects are shown in Fig. 2. Moderate permanent hearing loss reaching 20 dB re. audiometric zero was found in all musicians. TTS₂ reach 40 dB or more in both ears /70 dB in one case - subject A/. The measurements repeated after 12 months show no increase of hearing loss in all musicians but one /subject A/. In that case the additional loss was 15 dB at 2 kHz.
"ROCK" MUSIC - IS IT DAMAGING YOUR HEARING?

JAMES A. MADDEN ASSOCIATES PTY. LTD.

COOPER STEVEN EDWIN

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Is exposure to high levels of "rock" music a major health hazard to patrons and staff?

To assess the situation, noise level measurements were taken at 25 entertainment venues in Sydney and surrounding areas. To simplify the assessment, the music was divided into "Live Music" or "Recorded Music (disco)".

During the course of the investigation over the last 18 months, the increasing demand by younger age patrons for extremely loud live music rather than recorded music has been observed in the majority of entertainment venues.

Noise levels measured have involved $L_{90}$, $L_{10}$, $L_1$ and $L_{eq}$ A Scale levels slow response, and have been presented in terms of an Occupational Noise Dose limit of 1.0 for patrons and staff.

Typical octave band spectrums and exposure periods encountered by patrons and staff during the investigation are presented, together with a 'desired' noise level by the different age groups of patrons frequenting the various establishments.

Patron attendance and reactions have been noted where there have been significant reductions to the level of music, to satisfy legislative requirements, in some cases with violent crowd reactions resulting.

What is the answer for a reduction of high noise level "Rock" music - Legislation or Education?
A table has been prepared which illustrates the way in which the incidence and degree of hearing loss in populations exposed to noise can be expected to increase as the level and duration of the noise exposure increase.

In calculating the values for the table, Robinson's (1971) equation relating threshold levels to noise exposure and the National Acoustic Laboratories (1976) procedure for evaluating percentage loss of hearing from threshold levels were used and it was assumed that, in distributions of the threshold levels of noise-exposed populations, individuals with threshold levels corresponding to a particular centile at one frequency tend to have threshold levels corresponding to the same centile at other frequencies.

Apart from the effects of noise exposure and ageing, no other factors affecting the threshold levels of noise-exposed populations were taken into account, so the table values are likely to be less than those encountered in any actual population. In subsequent calculations, an allowance was made for further threshold shifts due to additional causes, in accordance with the findings of Burns et al. (1977), and another table was prepared which thus provides estimates of the incidence and degree of hearing loss in noise-exposed populations affected by additional causes of threshold shift.

Comparison of the two tables highlights the importance of thorough otological and audiological assessments of compensation claimants to ensure that they do not receive compensation for components in their hearing loss not caused by industrial noise exposure.


Tenth International Congress on Acoustics

Impulse Noise-Induced Hearing Loss Due to Driving-in of Steel Bolts

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Special techniques are required for the measurement of acoustic impulses and for assessment of their hearing damage risk. Gun shots, impacting bodies and explosions all fall within this category. The difficulties encountered with these transient pressure changes are caused by its high spectrum levels at low frequencies, non-repetitive nature and very short duration. In addition, the present knowledge on effects of impulses on hearing is still rather scanty.

To collect more information on the development of permanent threshold shift due to impulse events one may decide to examine the population of building workers employed at driving-in of steel bolts into concrete walls.

Field measurements of impulses generated by gun shots during driving-in of bolts were performed by means of impulse precision sound level meter and tape recorder Nagra IV L Kudelski, and then analysed by the computerized system of data processing to obtain time history and spectral values.

The evaluated impulses were characterized by a peak pressure level 137,1 dB, decay time 60 ms and rise time 1,0 ms.

The hearing thresholds of 60 subjects /after excluding of those with otological abnormalities and previous noise exposure/ were analysed by years of employment at driving-in of bolts, number of gun shots per month and by age of subjects.

The audiogram configurations were typical of noise-induced hearing loss /vs. normal audiograms of 53 non-noise exposed controls/ and showed increasing deterioration with length of employment.

x/ This investigation was supported by Program Project Grant 05-335-C PL 480.
DISTRIBUTION OF SHORT-DURATION NOISE LEVELS IN FACTORIES

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In certain factories, short-term impulses are measured and related to the total noise levels in the room. In metal working industries, astonishingly high levels of very short duration (in the order of 30 μs) are found to occur very often. This short-duration noise, often containing peaks up to 150 - 165 dB, is initiated from hammer blows and punch presses. The average sound level in the workshops is often in the order of 90 - 95 dB(A).

Other industries – especially those working with wood where the average noise levels can be higher than in the metal working industries – normally experience no high peak levels of short duration, as in these industries there is very little metal-to-metal contact.

In earlier publications it has been shown that these high, short-duration noise levels can contribute severely to noise induced hearing loss, because the time constants of the outer ear and middle ear up to the nerve ends at the Basilar membrane are very short. This results in the fact that the nerve ends are exposed to the full level of the short-duration sound, even though the subjective impression of the sound is low, as the time constant of that part of the brain which senses the sound is long. The figure shows the time constants of the human ear before and after the Basilar membrane.

An instrument which can be used to determine these very short impulses consists of an amplifier which holds the peak levels of impulses longer than 10 μs and which records these maximum levels twice a second. Another suitable instrument is a level analyzer which divides the level into 40 different levels working with a sample frequency of 10³ Hz, simply recording a sample every 10 μs.

REFERENCE: Brüel, P. V., "Do We Measure Damaging Noise Correctly?"
THE ECONOMICS OF INDUSTRIAL NOISE IN AUSTRALIA

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Important sources of industrial noise, and the corresponding worker exposure levels in the Australian manufacturing industry are surveyed. The magnitude of the noise problem is assessed in terms of both social and economic consequences. Particular reference is made to Australia as there is no published data available on the cost-benefit balance of industrial noise in this country.

The offensive industrial noise sources stem from four fundamental categories of machine types or processes: (1) continuous workshop machinery noise; (2) impact noise of a working tool (or machine) on the workpiece; (3) high speed repetitive actions that create intense 'pure tone' sounds; (4) unsteady flow induced noise. Figure 1 shows the estimated incidence of hearing impairment of exposed workers in Australia for several Standard Industrial Classification (S.I.C.) codes, including the 90 dB(A) legally permissible occupational noise level.

It is evident from our survey that: (1) Average continuous sound pressure levels are greatest in industries associated with primary metal works; (2) The average compensation payable for hearing impairment in Australia (1979 $'s) varies linearly up to $17,500 for 100 percent loss of hearing (P.L.H.); (3) If the 90 dB(A) noise exposure for an 8 hour day were achieved, 39% of exposed workers would still have a compensable hearing claim; (4) Since 1972 there have been consistent increases in workers compensation claims associated with hearing loss; (5) The estimated industry wide workers compensation liability is $200 million, and the recurrent liability with unexposed workers entering the workforce each year is $6.0 million/year; (6) A hearing protection programme employing earmuffs, audiometry and noise monitoring would cost about $5.0 million/year; and (7) The cost of enclosing all potentially harmful plants using existing barrier technology would be $380 million.

The present levels of compensation and the present costs of financing production activity offer no monetary incentive to industry to reduce its noise. Despite the increasing number of compensation claims, the total payments for industrial deafness are a small fraction of Australia's workman's compensation payment. If noise reduction costs per worker could be reduced below compensation costs per worker, the situation would change.

Fig.1 Fraction of exposed population F(PLH) with a percentage loss of hearing less than PLH.
RESPONSE TO N.S.W. HEARING CONSERVATION REGULATION
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The introduction in June 1979 of the NSW "Factories (Health & Safety Hearing Conservation) Regulation, 1979" gave specific criteria to eliminate employees' hearing hazards, where the hazards are defined by legislation.

The Regulation requires the occupier of a factory to determine the noise levels and noise doses which persons receive while employed in the factory. Where noise criteria is not met the occupier must use one or more of the following methods to satisfy the Regulation:

Engineering Noise Control
Administrative Controls
Issue of Hearing Protection.

The following points have been developed from independent surveys recently carried out by our firm, to determine an appreciation for the protection of employees' hearing as a direct result of the Regulation.

- Total aggregate employed, who regularly work in noisy areas, < 85; 85-90; 90-95 dB(A).
- Total aggregate employed, who receive peak noise levels above 115 dB(A).
- Of those who regularly work in noisy areas, the percentage who consistently wear hearing protection.
- Of those who regularly work in noisy areas, the percentage protected by:
  Engineering Controls
  Administrative Changes
  Muffs.
- Noise Industries classifications, numbers employed, noise exposure, exposure patterns.

The information presented represents an up-to-date evaluation highlighting current practice in NSW industry.
In 1970 audiometric examination of a random sample of over 1,100 Coal Miners revealed a 30% compensable level of hearing impairment in one or both ears. In the same survey, 10% of 496 recruits had a similar impairment. The percentage impairment was calculated according to tables provided by the Commonwealth Acoustic Laboratory. Of the recruits, some of whom were quite young, 50% had lost their hearing through exposure to industrial noise. In the remainder the loss was due to a number of medical conditions, head injuries or through injury to the tympani. In both groups, I found that the greatest loss of hearing acuity occurred in those working with, heavy earth moving equipment; washing, crushing and screening plants; winch and winding engines; in saw mills; while boiler-making and blacksmithing and while using compressed air driven equipment.

In two subsequent surveys of recruits to the Industry again about 10% had a compensable impairment in one or both ears. In the first group numbering 411, 60% had lost their hearing through exposure to excessive noise, and in the second, numbering 632, noise was the cause in 80%. In these groups loud music was responsible in three and the use of a shotgun in one who was only 18 years of age.

100 recruits to the Industry were invited to respond to a questionnaire about exposure to loud music and gunfire, the presence of tinnitus and whether protection had been worn, for however short a period, during exposure to noise.

<table>
<thead>
<tr>
<th>Exposure to Loud Music</th>
<th>Exposure to Gunfire</th>
<th>Presence of Tinnitus</th>
<th>Hearing Protection at Work</th>
</tr>
</thead>
<tbody>
<tr>
<td>36%</td>
<td>64%</td>
<td>11%</td>
<td>38%</td>
</tr>
</tbody>
</table>

In a larger group of 220 working coal miners, the same questions were put.

10%        61%                 15%                  51%

Strangely those who protected their hearing at work generally failed to do so when exposed to gunfire or loud music. Very few appreciated the importance of a temporary threshold shift, tinnitus or ear pain in association with an exposure to noise.

In a survey being undertaken at present, using National Acoustic Laboratory tables a far greater percentage with an impairment has been found in both recruits and working miners. A serious problem confronts us, education is essential and this should commence at high school.
VARIOUS APPROACHES FOR DETERMINING AND IMPLEMENTING AGE CORRECTION FOR
PURPOSES OF MONETARY COMPENSATION FOR NOISE INDUCED HEARING LOSS
Air Force Aerospace Medical Research Laboratory, Aerospace Medical Division,
Air Force Systems Command, Wright-Patterson AFB, Ohio 45433, USA

Daniel L. Johnson, Colonel, USAF
AFAMRL/TS, Wright-Patterson AFB, Ohio USA

A theoretical study was conducted into the potential approaches to compensate for hearing loss. A measure, named a Unit of Potential Compensation, is introduced to determine the average number of decibels of compensatable hearing loss expected in a population. Using this measure, several approaches for calculating age corrections are proposed and evaluated. Compensating individuals at the time they quit a noisy occupation is shown to be an unreasonable practice unless some procedures for adjusting for age is used. However, age corrections based on large populations are not necessarily equitable for the individual. A better approach is to compensate all individuals at one standard age. The best approach is shown to be one which compensates an individual dependent on the actual hearing level at various times during the person's lifetime.

(Work supported by the US Environmental Protection Agency and the USAF)
ESTABLISHMENT OF INDUSTRIAL NOISE CRITERIA BASED ON THE CONCEPT OF HUMAN FACTORS

Beijing Municipal Institute of Labour Protection, China

FANG, Dang-chung, FENG, Ceng-chuen, SUN, Jar-chi, CHEN, Chian and DONG, Jin-ying

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Beijing, China

This paper refers to the main results of a series of studies on the establishment of industrial noise criteria based on the concept of human factors.

With measurements of noise in 109 factories, we obtained the distribution of noise level in different industries, and found that, as a rule, the spectrum of noise in these industries varied as relating to the difference of (LC-LA). Based upon this fact, we divided the industrial static noise spectrum into 10 modes.

We recorded ECG and invested the appearance of neurathenia syndrome of 10021 workers in some factories, tested their hearing ability and obtained 10021 audiograms of these workers. Using these data, we established an equation to indicate the fact that the appearance of hearing impairment in workers increased as the noise level increased. We also obtained some rules on the ECG and neurathenia syndrome relating to noise level.

We recorded the EEG, pulse rate, blood pressure and hearing acuity, and also the measurement of CAMP of urine of workers. Some indication of relation between these indexes to noise level was found.

We also obtained the estimation of the dominant frequency of Chinese common, through analysing the relation between workers permanent hearing loss and their speech hearing loss.

Based upon all these studies, we proposed an industrial noise criteria for our country. With the viewpoint of human factors, this criteria considered many aspects of the effect of noise on the human body. The criteria was accepted and promulgated by the Ministry of Health and Bureau of Labour of the Government in 1979.
Noises containing impacts and distinctive tones have been known for a number of years to be more annoying than continuous broadband noises with the same physical level. Generalised corrections to account for such effects have been recommended in a number of standard rating methods. The aim of this research was to investigate these allowances by determining those physical parameters of recurrent impact noise which affect their loudness to human observers. This should lead designers to make more appropriate allowances for impact noise when using present criteria.

Twenty subjects with normal hearing were asked to adjust impact noise from a pulse generator (specifically developed for the purpose) until it was equal in loudness to a reference signal. In fact each subject was asked to adjust the impact noise using the "double staircase" method due to Cornsweet until it was "equal in loudness: to 1 kHz reference signal (loudness level). The impact noises used were recurrent being simulated to fall within the range of impacts normally met: 1) Repetition rates 5 to 100 i.p.s.; 2) Decay times 2.8 to 28 ms; 3) Rise times less than 5 ms; 4) loudness level equivalents from 40 to 80 phons.

The loudness effect of changing the frequency content of the impact shaped carrier signal could also be measured.

Results - A summary of the loudness enhancement caused by the impact and/or tonal characteristics of recurrent impact noise is shown in table 1 below.

<table>
<thead>
<tr>
<th>Noise Type</th>
<th>Repetition Rate (i.p.s.)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Recurrent Impact Noise having a pure tone carrier (present study)</td>
<td>5, 10, 20, 40, 100</td>
</tr>
<tr>
<td>Recurrent Impact Noise having a random noise carrier (present study)</td>
<td>5, 10, 20, 40, 100</td>
</tr>
</tbody>
</table>

Conclusions - It is clear that from this research there are three basic determinants of the loudness level of recurrent impact noise. An Energy Determinant, $\Delta L_e$, a Spectral Determinant $\Delta L_{SP}$ and a Temporal Determinant $\Delta L_T$. They appear to be additive so that:

$$\text{Loudness Level of Recurrent Impact Noise} = \Delta L_e + \Delta L_{SP} + \Delta L_T$$

To use this equation in practice $\Delta L_e$ and $\Delta L_{SP}$ can be equated to $L_{Aeq}$ and $L_T$ simply read off table 1 (or in short $\Delta L_e$ to enhanced loudness provoked by a recurrent impact noise having a pure tone carrier is 6 dB (A) or for an impact noise having a white noise carrier the allowance is 5 dB (A).

Thus for making the crude sort of allowances needed in BS4142 and ISO8996 on the basis of loudness considerations alone, the influence for either pure tone or impact characteristics alone should still be 5 dB; if they both occur together an allowance of 6 dB (A) might be more appropriate.
On the basis of extensive research the Kosten-Committee has established a relation between aircraft noise level and the disturbances and the annoyance caused by it. This relation was established for a large civil airport (Schiphol, Amsterdam airport), in 1963. In view of the new Dutch noise abatement act, which includes noise zoning around all Dutch airfields, the study was replicated in the late seventies both for Amsterdam airport and for four military airbases.

A major finding was, that the dose-response relations, established by the Kosten-Committee, still hold true for a large civil airport. For military airbases the situation is different. Not so much in the nature of the disturbances and the annoyance felt, but to the extent in which the disturbances occur at certain noise levels. On the whole, military aircraft causes more annoyance than civil aircraft at similar noise levels.

Also big differences in the shapes of the dose-response relations are found between the military airbases. Only part of these differences can be attributed to the numerous individual and situational variables covered in this study. Sensitivity to noise seems to explain more variance in the annoyance than the noise level itself does.

Near one military airbase nearly the same enquiry was held twice with almost all respondents. One enquiry was held at the end of a long, hot summer and the other at the end of the cold autumn. The results differed widely: at the end of the summer, after the respondents had had a learning period of several months in living outside or with windows open, more people felt annoyed, and to a greater degree, than at the end of the autumn. This finding questions the hypothesis that annoyance, like the attitude-concept, is fairly stable over time.
The annoyance due to road traffic noise (levels varying from 56-75 dB(A) $L_{eq}$) has been investigated in 36 houses for the aged in The Netherlands, 220 m east and west of the traffic, and women of old age have been questioned.

The relation found between the noise level outside the homes and the (non-specific) annoyance is as follows:

- 56 - 60 dB(A) $L_{eq}$: 3% highly annoyed ($n = 32$)
- 61 - 65 dB(A) $L_{eq}$: 32% " (n = 28)
- 66 - 70 dB(A) $L_{eq}$: 17% " (n = 84)
- 71 - 75 dB(A) $L_{eq}$: 29% " (n = 76)

The average noise annoyance in homes for the aged correlates weakly with the equivalent noise level ($r = .30$).

Comparing these results with those of "average" people in "normal homes" leads to the conclusion that on the basis of the answers given, the aged are not more susceptible to road traffic noise.

Considering the afternoon sleep as the daily activity which is most sensitive to noise, it appears that at a level less than 65 dB(A) $L_{eq}$ hardly anybody gets disturbed. The same applies to the nighttime sleep at a level lower than 50 dB(A) $L_{eq}$ over that period. (night)

Interesting is that old people who used to live in noisy surroundings (a city) prefer a lively neighborhood including some noise which belongs to it, and visa versa.

A strong desire to keep in contact with people they know living outside the houses for the aged has been clearly found.

More sensitive to noise (following a self-rating procedure) appeared to be the aged who:
- do not use a hearing-aid(!);
- report to get no visitors;
- want to return to the place they lived before;
- are not satisfied with their way of living;
- and, as a non-psycho-social category, people, who are not satisfied with the sound insulation at home.

No relation has been found between the subjective sensitivity to noise and the equivalent noise level; nor between sensitivity and living alone or sex.
The annoyance due to road traffic noise (levels varying from 60-75 dB(A) $L_{eq}$) has been investigated in 25 schools in The Netherlands. 77 Teachers and 753 pupils have been questioned.

The so-called non-specific annoyance experienced by the teachers is found to reach a maximum already at 61-65 dB(A) $L_{eq}$. 75% of them are very annoyed here, especially by the noise of trucks. At this noise level 10% of the pupils are annoyed. However, differentiating the pupils to educational level, large variations in susceptibility to road traffic noise have been found, especially at the relatively higher noise levels. For example, pupils of higher vocational training-schools are much more annoyed (70%) than those who are at the lower vocational training-schools (0%) at the same noise level of 70 - 75 dB(A) $L_{eq}$.

The intrusion by traffic noise upon the performing of specified school-activities (the specific disturbance) appears to be higher to the pupils than to the teachers in some cases, especially regarding the children of the primary school.

At a level of 61-65 dB(A) $L_{eq}$ the disturbance to reading already reaches its maximum level: for 50% of the pupils (against 10% of the teachers) this often happens. At this noise level problems of concentration are also maximal, 55% of the teachers are suffering themselves, 70% of them perceive these with their pupils. At 61-70 dB(A) $L_{eq}$ 30% say, at $> 70$ dB(A) $L_{eq}$ 40% of the pupils say often loose the thread. Speech interference problems are already large at 61-65 dB(A): 30% of the pupils and 50% of the teachers in this noise level class mention it.

A conflict between wishes to ventilate and noise can be expected.
At 61-65 dB(A) $L_{eq}$ 40% and at 71-75 dB(A) 70% of the teachers say that it is always or often impossible to ventilate.

It appeared that at a noise level of 40-44 dB(A) $L_{eq}$ inside the schools the noise annoyance problems are comparable with those at the level of 61-65 dB(A) outside. Worthmentioning is the negative correlation found between looking out of the windows by the pupils (self-rating) and the noise level inside the classroom.
INFRASOUND IN DAILY LIVES

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Recently, infrasound has been treated as one of the environmental pollutions, in parallel with noise and vibration pollutions. The effects of infrasound are divided into two categories. One is the physiological effects which may cause some damages to the human body. The other is the annoyance against the audible noise emitted by the vibration of windows or fittings which are excited by infrasound. This paper describes the results of the general survey on infrasound existing in our living environments caused by different sources.

METHOD OF MEASUREMENT

Usually, infrasound is defined as sound having frequencies below 20 Hz. However, our experiences show that low frequency sound having frequencies above 20 Hz would cause similar effects which are caused by the above-defined infrasound. For this reason, measurement frequency range was set up from 2 Hz to 125 Hz. Effects of infrasound, especially infrasound-induced vibration characteristics would depend closely on the frequency of sound. So, frequency spectra were obtained in the form of power spectrum or 1/3 octave band sound pressure level. Also, the measurements of weighted sound pressure level were carried out.

MEASURING OBJECTS

According to the purpose of this survey, measuring objects were chosen from the important environments in our daily lives. They are divided into four groups: (1) traffic environments, (2) indoor environments, (3) factories and (4) outdoor public environments.

RESULTS OF MEASUREMENTS

Sound pressure levels measured in various environments are summarized in TABLE I. Here, sound pressure level indicates overall level corresponding to the frequency range from 2 Hz to 125 Hz. As shown in this table, there exists fairly high intensity infrasound in our living environments. Frequency spectra depend on the type of important source which contributes to each environment.

<table>
<thead>
<tr>
<th>Environments</th>
<th>Sound pressure level (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Elevated roads and bridges</td>
<td>75 -- 98</td>
</tr>
<tr>
<td>Passenger cars (inside)</td>
<td>90 -- 118</td>
</tr>
<tr>
<td>Bus (inside)</td>
<td>91 -- 108</td>
</tr>
<tr>
<td>Railway stations</td>
<td>75 -- 107</td>
</tr>
<tr>
<td>Trains (inside)</td>
<td>79 -- 105</td>
</tr>
<tr>
<td>Harbors</td>
<td>81 -- 102</td>
</tr>
<tr>
<td>Ships (inside)</td>
<td>103</td>
</tr>
<tr>
<td>Around airports</td>
<td>89 -- 100</td>
</tr>
<tr>
<td>Dwellings (inside)</td>
<td>58 -- 83</td>
</tr>
<tr>
<td>Public spaces</td>
<td>65 -- 91</td>
</tr>
<tr>
<td>Factories (inside)</td>
<td>89 -- 118</td>
</tr>
<tr>
<td>Construction sites</td>
<td>90 -- 93</td>
</tr>
<tr>
<td>Waterfalls</td>
<td>74 -- 78</td>
</tr>
<tr>
<td>Parks and streets</td>
<td>70 -- 84</td>
</tr>
</tbody>
</table>
Some countries (Norway and Sweden) have established limitations for the max. permissible level of infrasound and ultrasound in factories. In Sweden, the infrasonic frequency range is from 2 Hz to 20 Hz, and the max. permissible limit in factories for an eight-hour working day is 110 dB regardless of the frequency. The ultrasonic frequency range is from 20 kHz to 200 kHz, and the max. permissible third-octave levels are: 105 dB for 20 kHz, 110 dB for 25 kHz and 115 dB for 31 kHz and all higher frequencies, all re 20 μPa.

In Norway, infrasound is defined in the octaves of centre frequencies from 4 Hz to 31.5 Hz with a max. permissible level of 120 dB re 20 μPa for an eight-hour working day. Ultrasound is defined in the octaves of centre frequencies from 31.5 kHz and up to 125 kHz with a max. level of 120 dB. This means that in Norway the frequency range of infrasound is from 2.8 Hz to 45 Hz and for ultrasound from 22 kHz to 187 kHz. Comparing the Swedish and Norwegian regulations, it is thus obvious that international standardization is urgently needed.

It clearly appears that it is wrong to consider the 2 Hz levels as being just as important as the 20 Hz levels, which has been done in the Swedish and Norwegian regulations. It is therefore suggested here that the 2 Hz should be allowed to have a 20 dB higher level than the 20 Hz. In other words, infrasound should be measured with a slope of 6 dB per octave, when permissible infrasound limits are introduced in factories, and limits of 110 dB to 120 dB for an eight-hour working day seem to be sensible figures.

When the considered problem is speech interference, then it is logical to have a weighting curve sloping 12 dB per octave, because this follows completely the established thresholds for infrasound. The figure shows a possible weighting curve for determining infrasound. For ultrasound measurements third-octave analysis is recommended.

REFERENCE:

The introduction of a noisy open air sporting activity into a residential or rural area is cause for nearby residents to be concerned. The predicted noise impact and assessment of annoyance is usually found to be most difficult. Regulatory authorities find it hard to frame statutory requirements to balance the interests of all parties concerned in such situations. A sporting facility may serve a large number of people as a community facility in a manner not too unlike a freeway. Irrespective of whether the facility serves a large or small section of the community a penalty is imposed on a few local residents. Careful scrutiny is necessary if the interests of the local community are to be protected. Very strong resentment may be felt when the added activity extends into the evening. In some instances an area already influenced by noise from one public facility may look attractive for the introduction of another noisy activity. People living nearby can react strongly about successive noise impacts likely to happen.

Two sporting activities which involve evening operation are horse trotting and clay pigeon shooting. The possibility of imposing the evening activity of trotting on an existing gallops racecourse in a residential area already influenced by a freeway caused strong reaction. Prior to the introduction of trotting it was estimated that ambient noise levels at 10 pm in the area of 39 to 45 dB(A) would rise to cause levels of 44 to 50 dB(A). It was further predicted that crowd roar and other sounds characteristic of trotting would be most noticeable. Measurements taken after the introduction of trotting showed levels of 47 to 57 dB(A) and crowd roar 54 to 70 dB(A) at 400 metres from the racecourse and near its boundary respectively.

A clay pigeon shooting range operating on afternoons and evenings in a semi-rural residential area has obvious repercussions. Sound levels at the dwellings were 48 to 62 dB(A) “fast” for typical gunshots against background levels of 36 to 40 dB(A). The most affected property experienced gunshot levels of up to 75 dB(A). Community reaction has been vigorous. The nuisance effect has been felt most strongly during the evening periods.

In both cases the noise character was an important factor in the overall assessment of community response.
THE DAILY NOISE EXPOSURE OF POPULATIONS
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Air Force Systems Command, Wright-Patterson AFB, Ohio 45433, USA

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AFAMRL/BB, Wright-Patterson Air Force Base, Ohio 45433 USA

While past research has primarily concentrated on industrial noise exposure, it has become increasingly important for hearing risk assessment and other health-related concerns to consider the total daily noise exposure of a population composed of the occupational and non-occupational (environmental, leisure time) exposure. This paper reports on the results of a multifaceted program using and evaluating various types of personally-worn dosimeters by subjects of: (1) a general urban population, (2) a military population, (3) a children population (age < 18 years) and (4) Subjects of special interest. Results of the general urban population study conducted on 50 Americans over a 7-day test period showed average L_{eq}(24)'s (or L_{eq} (week)) among these individuals ranging from a low of 66 dBA to a high of 85 dBA, with a median of 74.7 dBA. Sixteen (16) persons from a military population were monitored and 73 total separate 24-hour dosimeter measurements were obtained on them. These L_{eq}(24)'s ranged from 60 dBA to 97 dBA, with the median daily A-weighted average sound level being 77 dBA. An ongoing study currently being conducted involves monitoring the daily 24-hour Leq's of children. To date 187 useful measurements have been obtained ranging from a low of 67 dBA to a high of 104 dBA, with a median level of 82.2 dBA. These results are compared with each other and with previous estimates for such populations and recommendations for the future use and development of personally worn noise dosimeters are presented. (Work supported by the US Environmental Protection Agency and the USAF).

REFERENCES
Psychological scaling in the assessment of subjective reaction to aircraft noise.

National Acoustic Laboratories

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The traditional approach to the assessment of subjective reaction to aircraft noise involves the use of scales composed of annoyance ratings for a number of disturbances. While such scales are typically very reliable they are of dubious validity. Analysis of data from a pilot survey of 160 Sydney residents provides empirical evidence of the methodological inadequacy of disturbance-item scales of annoyance. If it is to be valid an annoyance scale must consist of items which provide ratings of overall subjective annoyance.

In a model of subjective reaction to aircraft noise it is proposed that annoyance be viewed as one of several components of a more general reaction. This is supported by the results of a separate survey of 100 Sydney residents which showed: 1) a general rating of 'affectedness' correlated more highly with nominal noise exposure (NEI) than did a rating of annoyance; 2) there was a significant partial correlation between exposure and affectedness at constant annoyance. It is argued that an adequate social study of the effects of aircraft noise must include ratings designed to assess general reaction as well as specific reactions such as annoyance and fear.
COMPARATIVE STUDY OF THE STRUCTURE OF ATTITUDE TO NOISE PROBLEM
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INTRODUCTION
Usually we relate objective measured value of noise (stimulus) to subjective unpleasant feeling (response). But noise is actually only a part of personal circumstances and consciousness to noise is a part of individual attitudes. So we must take care to consider the relation between stimulus and response.

This study was undertaken to find out the structure of attitude to noise problem by comparing five field surveys. These surveys were carried out with the method of personal interviews at various areas from 1976 to 1978. Table 1. shows contents of five surveys.

<table>
<thead>
<tr>
<th>Survey</th>
<th>Contents of five surveys</th>
</tr>
</thead>
<tbody>
<tr>
<td>I</td>
<td>People of Kanagawa ward of Yokohama</td>
</tr>
<tr>
<td>II</td>
<td>People of the sites exposed to road or railway noise</td>
</tr>
<tr>
<td>III</td>
<td>People of the surroundings of Atagi Air Base</td>
</tr>
<tr>
<td>IV</td>
<td>People of the surroundings of Camp Fuji</td>
</tr>
<tr>
<td>V</td>
<td>People of Kanagawa ward of Yokohama</td>
</tr>
</tbody>
</table>

Concerning subjective responses, 36 items were selected and classified into patterns in accordance with the third formula of quantification (Multi-Dimensional Analysis in case of No Outside Criterion) of Dr. Hayashi.

CONCLUSION
In conclusion, we can say as follows.

1. Basic elements of the structure of attitude to noise problem are "discontent" to outdoor noise, "necessity" of main source of outdoor noise and "interest" in noise problems.
2. The characteristic of each area can be expressed by projection of the space of consciousness.
3. In general, "discontent" to outdoor noise is main element, but in areas of specific sources of noise, "necessity" of the source is main.
4. "Discontent" is strongly connected with the matters of noise surroundings and "interest" is related to demographic matters.
We measured the amount of noise exposure of about 700 persons in Sendai, Tokyo and Nagoya and examined the problem from various angles. The purpose of the study was to evaluate comprehensively the significance of noise exposure, for in our daily lives we are being exposed to a variety of sounds including the so-called "noise" and sounds which originate from our own activities.

We had the subjects carry the compact and light noise exposure meter for 24 hours and obtained the total number of 144 of Leq every ten minutes and Leq(24) of sounds recorded through the microphone clipped on the lapel of the subjects. At the same time we asked the subjects to record their activities for 24 hours.

Average Leq(24) for 462 persons with occupation is about 73 dB(A), and the Leq(24) is mainly determined by Leq during their working hours, especially the amount of exposure for skilled workers is larger than other occupations. For some kinds of occupation noise exposure during commutation also contributes greatly to Leq(24). As for the Leq during commutation by various means, the largest value is found in motorcycle.

Average Leq(24) for 140 housewives is about 70 dB(A), and Leq during cooking shows the largest contribution to it among different activities.

The largest amount of Leq(24) on an average is found in elementary school children among all samples.

Fig.1 shows the population of Sendai distributed by Leq(24) calculated from the results of our survey. The calculation is based on the average Leq(24) and division of 26 groups of the whole population classified by job and age. The cumulative curve shows a stepwise tendency around 80 dB(A), which is caused by the large amount of exposure in kindergarten and elementary school children. We are afraid these higher levels of noise exposure found among school children should have serious effects on the hearing ability, even though we know the levels will not last for 40 years, they also include their own voices.

Fig.1 Population of Sendai Distributed by Leq(24) based on the survey.


COMMUNITY RESPONSE TO RAILWAY NOISE IN GREAT BRITAIN

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Residents' reactions and railway noise levels have been measured in 403 neighbourhoods along 75 sections of railway routes in Great Britain. The reactions of 1453 residents were measured in 45 minute interviews. The descriptions of railway noise levels were based on complex computer analyses of tape recordings of over 1,700 pass-bys from the 403 measurement sites. The use of a probability sample design has enabled statistics to be computed which are statistically representative of the British population near railway lines.

MAJOR FINDINGS FROM THE STUDY

1. The 24 hour Leq dB(A) noise index appears to be the most practical choice of indices for representing railway noise. The noise and number trade off implicit in Leq fits the data better than any of the other established indices tested. Linear, D and B weightings are slightly more highly correlated with annoyance than the 'A' weighting. The 'A' weighting appears to do less well than a linear weighting in weighting some acoustical aspects of overhead electrified routes.

2. Different measures of railway noise impact are related differently to noise level. General railway noise annoyance increases as noise level increases. As a result there is no particular 'acceptable' or 'target' noise level. In general the lower the noise level, the less the annoyance.

3. It is estimated that about 40,000 to 50,000 dwelling units in Great Britain are at noise levels above 65 Leq dB(A).

4. The comparison of these railway data with three aircraft surveys (around Heathrow) and two English road traffic surveys, suggests that, at least above 60 Leq dB(A), railway noise is less annoying than noise from these other sources. The estimated size of the difference in reactions depends upon the survey with which the comparison is made as well as the noise level. As noise level increases the gap between reaction to railway and other noise sources increases.

5. At high noise levels people alongside overhead electrified routes report less annoyance than people near third rail or diesel routes. In the 55-75 Leq dB(A) range the difference in general annoyance is equivalent to at least a 10 dB(A) difference in noise level. The difference in reactions is greatly reduced if the linear frequency weighting network is used.

6. Noise from railways is rated as the most important impact of a railway in a neighbourhood. Vibration is the most important non-noise impact. Of the various noises associated with a railway's operation, maintenance is rated as the worst, even more of a problem than the noise from through trains.
ON LOCATING RAILWAY NOISE SOURCES WITH AN ACOUSTIC TELESCOPE

DFVLR - Berlin

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Radiated noise generated by a high-speed electric train traveling at controlled speeds ranging from 100 to 250 km/h was measured with a linear array of 15 1/4" microphones mounted both along the wayside and above the train. To establish known ground conditions, wayside measurements were made with the array mounted on a ground plate. For each complete series of pass-bys, the array was arranged with the line of microphones positioned (i) perpendicular to a plane tangent to the upper surfaces of the rails and (ii) parallel to the rails. In order to determine the relative role played by aerodynamically generated noise with as little influence as possible from wheel/rail noise sources, the array was suspended above the train from a boom such that the line of microphones was positioned (i) longitudinal and (ii) transverse to the track centerline. All data were recorded for subsequent computer processing. By positioning the array orthogonally, albeit at different times, it was possible to correlate the results and essentially obtain two-dimensional resolution. With the acoustic signals measured with the longitudinal array above the train and with the parallel array on the ground plate, the array beam pattern was slewed to track any section of the train by using appropriate time delays in the data processing. Measurements were also made with infrasonic microphones designed to sort out pressure wave effects from those due to acoustic sources. Results show the location of the apparent sources of wheel/rail noise as well as the relative importance of aerodynamically generated noise sources.
Dynamics of Track Fastened Resiliently to Floated Slablets

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Ungar, Eric E.

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Rail that is fastened resiliently to elastically supported slablets (or "floated" ties) has found increasing use for protecting buildings near tracks from noise and vibration due to passing trains. The chain-system model of Fig. 1 was devised in order to overcome limitations inherent in modeling the rail/slablét system as two lumped masses or as a beam that is elastically fastened to an elastically supported limp layer*. Here the slablets are represented by point-masses $m_S$, each supported on a stiffness $k_S$, and the rail is represented by point-masses $m_R$ interconnected by stiffnesses $k_T$ and fastened to the slablets via rail fasteners of stiffness $k_R$.

One may determine the salient features of the dynamic behavior of this system by dealing with sinusoidal motion at a constant propagation constant $G$ that relates the complex amplitude $Y_{n+1}$ of any mass point to the amplitude $Y_n$ of the next mass point nearer to the excitation. One may find that $Y_{n+1}/Y_n \equiv G = b \pm \sqrt{b^2 - 1}$ (where the sign is chosen to make $|G| \leq 1$), and $2b = 2 + \gamma_R T \nu_R T (1 - \omega^2/\omega_S^2) - \gamma_R T T \gamma_S (\nu_R^2 + 1 - \omega^2/\omega_S^2)^{-1}$; $\gamma_R T = k_R/k_T$, $\gamma_S = k_S/k_R$, $\omega_S = k_S/m_R$, $\omega_R = k_R/m_R$. Analysis of this result reveals the occurrence of five frequency regions in which occur different types of dynamic behaviour—standing waves and travelling waves in the rail, with slablet motions either in phase or out of phase with the rail motions.

This paper is based on work that was financed in part through a grant from the U.S. Department of Transportation, Urban Mass Transportation Administration.

Prediction of the Propagation of Train-Induced Ground Vibration

VERHAS

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In the study of the propagation of ground vibration, the ground is considered a homogeneous half space.

Propagation in the ground is attended by a decay of the vibration amplitude in two ways: a decay occurs due to geometrical attenuation of the wave front, and a decay occurs due to damping of the wave along its path. The first decay, i.e. the geometrical attenuation, varies according to the type of wave, and is proportional to:

\[ r^{-x}, \text{ where } r = \text{distance to the source} \]
\[ x = 0, \frac{1}{2}, 1, 2 \]

As situations may require a source can act as a point source, as a line source, or as a combination of both. In this last case we talk about the superposition of a point source and a line source. There are several reasons for the earth not to be the homogeneous half space as supposed in the theoretical approach of the propagation of ground vibration. One reason is the stratification of the ground. Reflection of the body wave may occur at the boundary of a layer. Considering only one layer, the pathlength of the reflected wave may vary as the depth of the top layer varies, and this will result in a different vibration level.

Summing up, vibration levels in the ground can be altered by the following phenomena:

a. the repartition of energy in the different wave types;

b. the damping in the ground;

c. the type of the source;

d. the layering of the ground.

By considering two receiver positions at a given distance to the source, it is possible to express the level in one position in function of the level in the next position by evaluating the above phenomena. This leads to a concept of level difference (ref.).

By evaluating the expressions for level difference, for a distance of 25 m between measuring positions, it was possible to determine a model that was in good agreement with the collected data from two sites. A superposed model, with an energy repartition over both body waves and Rayleigh waves, and with damping proportional to the number of wavelengths, gave the best fit.

AIRCRAFT - FUTURE NOISE ENVIRONMENT

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The demand for aircraft travel continues to increase. In the last year and a half almost 1000 new commercial aircraft have been ordered. From today until 1990 current fleet numbers will double as passenger demands continue. An examination of the development of civil air transport from 1958 shows a steady increase of both freight and passenger revenue traffic. Economic influences have caused small perturbations to the growth but the trend remains upwards. Whether in the face of the current fuel situation this trend can be maintained is speculative. But the continued rise in demand is a strong influence on the future noise environment. If the trend in the United States is indicative of world wide air transportation activities in 1978 with a total civil transport fleet of approximately 2500 aircraft, two thirds were uncertificated. In 1990 aircraft numbers will have risen to nearly 3500 with the uncertificated component representing 25% of the total. 50% of all civil transport aircraft will have complied with the original United States or ICAO noise certification regulations. Of the remainder of the fleet, 25% will achieve levels near to the latest FAA and ICAO recommendations. But further reduction in emission levels appear to be relatively modest. Coupled with the continued expansion of passenger demand and aircraft sales, the projected rate of reduction in aircraft noise exposures must diminish as the fleets become equipped with the new technology airplanes. The time at which all stage 1 and a substantial number of stage 2 aircraft will disappear from operation is impossible to forecast but it must project beyond the end of this century. At that time aircraft noise should cease to be a major source of noise nuisance, although it may never achieve complete environment compatibility. Even so most of the high levels of noise exposure will be confined to a few square kilometres beyond the airport boundary. At this point compatible land usage around the airport may assume a level of economic feasibility.

These conditions exclude the operations of a significant number of supersonic transport aircraft. Even the most optimistic forecasts of the application of noise control technology cannot lower future SST noise levels below current noise certification requirements and the financial investment would be enormous. There are no reliable estimates of the total investment by industry or government in noise control technology although the Boeing Company alone Estimates its financial investment to date at $153,000,000. Expenditure on future noise control technology is regarded as expensive for modest noise reductions. But the investment may yield considerable gains in fuel economy perhaps a more important factor in the future. For example, a one percent decrease in fuel consumption represents a saving of between 200 and 300 million US gallons of kerosene per annum.
LIGHT PROPELLER-AIRCRAFT NOISE CERTIFICATION - PITFALLS AND SUGGESTIONS

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On the basis of close to 200 noise-certification measurements of light propeller aircraft conducted by the DFVLR Department Technical Acoustics it became quite obvious that both methodology and procedure of data acquisition and reduction to eventually produce a "noise certification level" to be weighted against a maximum noise level not to be exceeded, leaves yet a lot to be desired.

In view of the fact that fractional amounts of deziBels may decide upon passage or failure of an aircraft to meet a noise specification, one needs to establish the realistically achievable accuracy and reliability of aircraft noise data through currently applied procedures. This paper will discuss several potential problem areas.

Particular emphasis will be given to the rather questionable measurement-accuracy and reliability of data obtained through the standard 1.2 m-above-ground microphone-position. In fact comparatively small deviations from that standard position may lead to significant variations - in terms of required certification accuracies - of A-weighted levels depending on the characteristics of the particular aircraft. First results of a fairly substantial study, where in a systematic manner microphone positions were altered above concrete and/or grassy surfaces and the ensuing effects on aircraft noise signatures will be reported and implications towards an improved data acquisition procedure be discussed.

Secondly the question of a physically plausible temperature-correction will be addressed. Here the currently applied procedure relies heavily on an empirical propeller-noise prediction scheme that was in the first place not intended for light propeller-driven aircraft. Correspondingly doubtful are the temperature-correction methods applied to obtain a (certification-)level for reference temperatures. Thus the potential benefits of an alternate approach are discussed, where for the aircraft to be certified its own individual helical Mach-number dependence of the (A-weighted) level is first established and then employed to correct for temperature effects.

Thirdly the question will be answered of the realistically achievable reproducibility of certification-data for identical aircraft when determined through either different (though equally experienced) measurement-crews, or by the same crew, however at different times and locations.

In conclusion then, the currently employed methods to certify (light) propeller driven G.A.-type aircraft are scrutinized and suggestions for improvements are offered.
INVESTIGATION OF NOISE CONTROL TECHNIQUES FOR HIGH SPEED TRAINS

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EXPERIMENTS

Test runs were conducted with a fast train of Deutsche Bundesbahn to investigate the abatement of wheel/rail interaction noise by

a) shrouding of the bogies with several configurations of sound absorbing baffles,

b) reduction of sound generation by wheel mounted vibration absorbers which were tuned to the resonant frequencies of the wheel.

The effect of these measures was investigated in detail by means of a highly directional microphone system, the acoustic mirror telescope [1], see Fig.1. This technique allowed to determine the spatial and spectral distribution of sound source intensity along the train. In addition, omnidirectional microphones were applied to measure the wayside noise levels. The train velocities varied between 80 km/h and 160 km/h.

RESULTS

Shrouding of the bogies diminished the noise radiation by 1.5 dBA to 5.5 dBA, depending on the baffle configuration. The different effectiveness of the baffles can be explained by the results of the directional microphone measurements.

The wheel mounted vibration absorbers reduced the noise by 3 dBA to 6 dBA, the noise reduction growing almost linearly with train speed in the velocity range investigated. This is due to the fact that, with increasing speed, the peak of the noise spectrum is shifted to higher frequencies, where the vibration absorbers are more effective.

Fig. 1
Acoustic mirror telescope for investigation of sound sources of a train passing by.

[1] Grosche, F.-R.; Stiewitt, H.,
Acoustic mirror technique for source location of train noise.
ON AERODYNAMIC NOISE GENERATION BY HIGH SPEED TRAINS

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While train noise at low velocities is caused mainly by wheel/rail interactions, aerodynamically generated noise should contribute significantly at higher speeds because it increases with a higher power of velocity than wheel/rail noise. Therefore the major noise sources of a high speed train were investigated at velocities between 160 km/h and 250 km/h by means of a directional microphone system consisting of an elliptical mirror with a vertical array of 7 microphones which allowed to measure the distributions of sound source intensity in seven horizontal planes about 0.75 m apart from each other, see Fig.1. The train consisted of an electric locomotive and four passenger cars, it was provided by the Deutsche Bundesbahn. The first car was instrumented for measuring speed and power consumption, it was also equipped with pantographs.

The following sources of aerodynamics noise were identified:
   a) The forward (collapsed) pantograph of the locomotive.
   b) The forward (raised) pantograph of the instrument car.
      The flow noise of the pantographs was also observed in wind-tunnel model tests.
   c) The bumper region of the locomotive.

The velocity $U$ dependence of the identified noise sources was estimated from the measurements assuming that the sound intensity $I$ varies with speed according to $I \propto U^n$.

The velocity exponent was found to be:
   a) for the wheel/rail interaction noise: $n \approx 3.5$
   b) for the flow noise of the pantograph and of the bumper region: $n \approx 6.8$.

Fig.1
Mirror microphone system with 7 microphones for simultaneous measurement of acoustic source strength at different vertical positions.
APPLICATIONS DE L'ANALYSE DE FOURIER A LA CARACTÉRISATION DES BRUITS ROUTIERS FLUCTUANTS

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INTRODUCTION-OBJECTIFS DE L'ÉTUDE

Parmi les nombreuses méthodes pratiquées pour caractériser les fluctuations des niveaux de bruit routier (énergétique, statistique...), on a étudié les possibilités d'utilisation de l'analyse harmonique de Fourier, dans une double optique : - mieux quantifier la gêne due au bruit ; - fournir de nouveaux paramètres pour une typologie des situations acoustiques.

ANALYSE DES AMBIANCES ACOUSTIQUES DE COURTE DUREE

Différents types d'ambiances acoustiques routières, comportant du bruit de trafic fluide... pulsé par des feux, en sens unique ou double sens, en voirie routière ou urbaine, ont été analysés à l'aide des indices classiques (Leq, L10, etc...) et d'une analyse de Fourier (densité spectrale de puissance sur la bande de fréquences 10^{-3}Hz-1 Hz). On a pu mettre en évidence que la D.S.P. permet de différencier des ambiances acoustiques pour lesquelles les indices classiques sont équivalents, mais qui peuvent avoir sur la gêne une incidence différente. Certaines ambiances présentent notamment des pics caractéristiques de l'intervalle entre véhicules, entre pelotons, ou bien du cycle des feux de circulation. Cette approche a également permis de comparer aux mesures in situ les résultats d'un programme de simulation (1,2).

ANALYSE DES AMBIANCES ACOUSTIQUE DE LONGUE DUREE

On a également analysé un ensemble d'enregistrements sur 24 heures de différentes ambiances (voirie chargée, zone résidentielle...) (fig.1). L'analyse de Fourier appliquée à ces profils a permis de fournir des paramètres caractéristiques permettant de les différencier dans le contexte d'une analyse typologique pour classifier ces ambiances. Une telle approche, qui apporte des éléments à la définition du contraste jour/nuit et du "silence nocturne", peut avoir un intérêt dans l'élaboration d'indices susceptibles de mieux caractériser l'effet du bruit sur le sommeil par exemple. (bruit sur 24 h. consécutives

Fig.1-Exemple d'un profil de bruit aux abords des carrefours routiers - Résultats d'un programme de simulation - 9ème I.C.A. - Juillet 1977 Madrid.


(2) B.FAVRE Noise at the approach to traffic lights; results of a simulation programme - J.S.V.(1978)58(4),563-578.
An Investigation of Diesel Engine Noise Using the Transfer and Coherence Function Technique

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In general, in any diesel-engined road vehicle, engine noise is the dominant noise source. In heavy truck applications this can cause an 'environmental' problem outside the vehicle and produce unacceptably high noise levels inside the cab, the driver's workplace. Suitable acoustic treatments can reduce the severity of the problem but reduction at source is a preferable approach. In order to do this effectively, a reliable, analytical method for identifying the contribution of the individual noise sources of the engine to the total engine noise, is used and presented in this paper - that is using the "transfer and coherence function" technique.

The ideas of structural transfer function testing and the use of coherence and correlation techniques have been known for some time but their application has been limited due to practical computing difficulties, and only in recent years have instrumentation and mini-computer system development made application of these methods practical and economically feasible. Even so, comparatively little use had been made of the methods in the field of engine noise research.

Crocker, Hamilton, Sybert and Chung at Purdue, U.S.A., were amongst the first to apply this analytical approach to study the mechanism of noise generation in the diesel engine where they considered the noise generating process as a complete system from the combustion excitation of the engine structure, producing engine block surface vibration, generating acoustic pressure fluctuations detected as engine noise. That is the system under consideration (diesel engine noise) is modelled as a single output system with multiple mutually-correlated inputs, although uncorrelated noise (i.e. noise not coherent with the inputs) present at the output can be accounted for. This multiple-input model assumes that the system inputs are related to the coherent output by a set of linear systems with associated frequency responses.

This paper presents results of applying this method to a small 1.8 litre, naturally-aspirated, indirect combustion, four-cylinder in-line automotive diesel engine showing good correlation with previous experimental studies conducted on this engine.

It is concluded that, with further development, this analytical technique could become a reliable approach towards fundamental understanding of diesel engine noise control.
THEORY OF BY-WAY-TRANSMISSION OF NOISE IN VEHICLES

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To declare the experimental results of noise reduction in passenger cabs of vehicles could be deduced the equation

\[ L_2 = L_1 - R - 10 \log \frac{A}{S} + 10 \log \left[ 1 + \frac{W^2 \left( \sum \frac{E_{el} S_{el}}{n} + \sum \frac{v_{n}^2 H_{n} S_{n} \chi_{n}}{E_n S_n} \right)}{E_1 S_1} \right] \]

To solve this equation the following acoustical properties must be known: the radiating areas \( S_n \), the radiation-coefficient \( \chi \), the transfer-function \( H \), which describes the acoustic properties of all connexions between any sound source and the car-body, the transmission loss \( R \) of the partition between engine-room and passenger-room and the equivalent absorbing surface \( A \) of the passenger-room. The values of this properties are individuel for each type of vehicle. Therefore it is necessary in the future to find by attemps relations between these values and the construction.
NOISE ANNOYANCE REDUCTION in RESIDENTIAL AREAS by TRAFFIC CONTROL TECHNIQUES

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In residential areas of 6 cities of Western Germany a number of traffic safety measures were introduced with the intention to reduce the through-traffic and to make residual-traffic more safe for residents.

Partial aim within the frame of the complex study was the analysis of noisepollution and annoyance reaction of inhabitants before and 1 year after the introduction of the measures. The measures consisted of systems of one-way-streets and cul-de-sacs, loopsystems and wall-to-wall-surface, alternating parking and speed limitation. For all residential areas acoustic noise measurements were made before and 1 year after introduction.

Noise maps of the areas were computed and plotted. By means of a social survey 1700 residents were interviewed before and after. The traffic-noise-annoyance-reaction of the residents was assessed by quantitative parameters.

The analysis of data showed, that the traffic volume and velocity of vehicles had been reduced in some extent, depending on systems of measures in the area. The mean noise level $L_d$ for the 1700 residents was diminished only from 59,8 to 58,9 d(A). The reduction of noise-annoyance was computed to be equivalent to a theoretical reduction of noise level of 6-14 dB(A). The further analysis showed, that the change of noise levels within streets could not explain the general decrease of annoyance. It is assumed, that residents view of traffic and its impact on daily life has changed as a consequence of the measures. This reduction of environmental threat of traffic may result in a reduction of negative evaluation and annoyance of traffic noise.
MEASUREMENT OF TRAFFIC NOISE IN THE CITY OF MELBOURNE

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MURPHY, Brian


INTRODUCTION

Melbourne is the capital city of the State of Victoria. Within the central business district are approximately 60 thousand people (business, shop employees and shoppers). One of the major city streets which also serves as an access road between the city and suburbs is Swanston Street. It is also a route, on two centre-of-the-road tracks, for electric trams.

MEASUREMENT SITE

Bordering one section of Swanston Street is the R.M.I.T. complex where it is proposed to erect a multi-storey building. This work reports the measurement and study of traffic noise adjacent to the proposed building line at a location approximately half-way between two controlled intersections 200 metres apart. The location is also adjacent to an uncontrolled T-intersection.

MEASUREMENTS

The survey was conducted at various times during working days between 7 am and 6 pm. In general, traffic counts were made during the measuring periods of 10 minutes.

Traffic noise was detected by a single microphone located on the proposed building line and recorded on a portable Nagra IV-SJ tape recorder. Statistical analyses of different types were carried out using the HP 21MX analysis system and analog RMS conversion.

In addition to studying the various possible indices for describing traffic noise this work was to serve as a basis for advising the architectural consultants concerned with the design of the proposed building.
COMPUTER MODEL FOR PREDICTION OF ROAD TRAFFIC NOISE LEVELS

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INTRODUCTION

In connection with planning of new roads or residential areas it is necessary to be able to predict the road traffic noise in surrounding areas, to avoid noise problems. In connection with noise reduction installations it is necessary to have a model for calculation of the attenuation.

In the Nordic countries a joint model for prediction of road traffic noise has been developed (1), enabling us to predict $L_{Aeq}$ and $L_{Amax}$. The model is developed as a manual model. Therefore the first step is to calculate the necessary variables (e.g. the height of barriers relative to a reflection plane). Finally, the results are calculated by means of nomograms or formulas. Larger projects take long time and require specialists. Therefore development of a computer model has been started. The computer model makes use of data concerning the terrain, the roads, barriers and receiver points.

THE MANUAL MODEL

The model consists of five independent stages, e.g. we get $L_{Aeq} = L_1 + \Delta L_2 + \Delta L_3 + \Delta L_4 + \Delta L_5$. $L_1$ is a basic value giving the unattenuated value of $L_{Aeq}$ at a distance of 10 m from an infinitely long straight road. $L_1$ is dependent on the number of vehicles, indicated speed limit, and percentage of heavy vehicles. $\Delta L_2$ is the distance attenuation. $\Delta L_3$ is the attenuation caused by surface and barriers. $\Delta L_4$ gives the facade insulation. $\Delta L_5$ is different corrections e.g. corrections for gradients, reflections, calculation in streets, thick barrier etc.

THE COMPUTER MODEL

The requirements for the computer model is terrain data and data for positioning of roads and barriers. The model should give the same results as the manual model. The terrain, road, and barrier data are collected from maps, air photographs, or field surveys. By means of these data we get a digital terrain model. In the figure you see an outline of the model.

Problems in connections with the calculations, e.g. interpolation in the digital terrain model, insertion of reflection plane, and segmentation of the road, will be mentioned during the presentation. The computer programme should be ready during the first part of 1980.

REFERENCE

SOUND POWER LEVELS OF FREELY RUNNING VEHICLES MEASURED IN REVERBERANT TUNNEL

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INTRODUCTION

It is important to know the sound power of the vehicle noise precisely to predict the road traffic noise and to make counterplan for it. We have invented a new measuring method using reverberant tunnel and reference sound source. Based on this method we have carried out some measurements of the sound power levels of vehicles running in constant speed.

MEASURING METHOD

As shown in Fig.1, sound pressure of a vehicle is measured at a point inside a tunnel. By comparing the squared and integrated value of the sound pressure of the tested vehicle and that of the reference sound source, sound power (W_x) of the tested vehicle can be determined. In this study, as the reference sound source, a cubical loudspeaker system mounted on the roof of a car was used, and a submerged tunnel constructed under the Tokyo Bay was chosen for the testing field.

RESULTS

The results of this measurement are shown in Fig.2, and the conclusions are as follows.

(1) As for passenger cars, fairly good correlation exists between sound power levels and velocities.
(2) As for heavy vehicles, sound power levels have large variation and small dependence on velocity.
(3) Heavy vehicles wearing lug tires are noisier as compared to those wearing rib tires.

Fig.1 SCHEMATIC MEASURING PRINCIPLE

Fig.2 PLOTS OF PWL(A) AGAINST VELOCITY (V) FOR 179 PASSENGER CARS AND 490 HEAVY VEHICLES
IN-SERVICE VEHICLE NOISE TESTING IN VICTORIA (AUSTRALIA)
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INTRODUCTION

The Victorian Environment Protection Authority's motor car noise Regulations were the first Australian Regulations to set maximum permissible noise levels for in-service vehicles and were first implemented in September 1977. Additional Regulations controlling noise from in-service trucks, buses, and motor cycles came into effect in August 1978. The paper reviews the background to these Regulations, summarizes the experience gained since testing began and briefly outlines some improvements to the Regulations that seem necessary in the light of this experience.

SOME BACKGROUND TO THE REGULATIONS

The administration of the Authority's Regulations is entirely an E.P.A. matter being completely independent of the Police. E.P.A. inspectors patrol the Melbourne road system and note the registration number of any vehicle they subjectively assess as being unnecessarily noisy. The owner is then traced through the registration records in the Motor Registration Branch and is served a notice requiring the vehicle to be presented for a noise test at a particular time at a specified test station. The date of the test is determined so that the owner has at least 14 days to carry out work on the exhaust system of the vehicle. If, when tested, the vehicle exceeds the relevant maximum permissible noise level by a small margin a further test is offered. If the vehicle is blatantly noisy when tested, or is not presented for test, the owner is liable to prosecution in a Magistrates' Court. The maximum penalties for excessive noise and failure to present a vehicle are $400 and $200 respectively. All prosecution work is handled by the E.P.A.

Two testing stations serve the metropolitan area of Melbourne. One, a permanent installation, is located in a western suburb. The other station is mobile and serves the eastern and south eastern suburbs. This station is also occasionally used in country areas. Under the Environment Protection Act a vehicle cannot be directed to a testing station located more than 50 km from where it is ordinarily kept. As a matter of policy this distance has been reduced to 30 km. Once at the testing station the vehicle can only be kept for one hour. In all cases the vehicle is tested when stationary, the actual test procedures being adaptations of E.E.C., United Kingdom and United States procedures. The maximum permissible noise levels were determined from the results of large scale surveys conducted in Victoria from 1974 to 1976 involving 2900 vehicles.
A VIDEO TAPE AND MICROCOMPUTER ANALYSIS SYSTEM FOR NON-UNIFORM VEHICLE MOTIONS AND NOISE LEVELS.

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The noise levels emitted by individual vehicles over a range of driving conditions are a vital input to noise simulation models. Individual vehicles’ noise levels cover speed and acceleration effects, but no data were found to cover the deceleration effect. Measurements were made on the TRRL test track in which two cars and a lorry were used.

THE EXPERIMENT AND ANALYSIS PROCEDURE
Runs were made under the following conditions: (1) a range of constant speeds, (2) acceleration from rest, (3) accelerations from various speeds, (4) decelerations from various speeds.
Noise and video recordings of the runs were made and a modified traffic counter was used to synchronise the two.

A technique was devised to evaluate the position of the vehicle on the ground by knowing its coordinates on the photo by using the video tape equipment. The method is based on the transformation of picture coordinates to the ground coordinates. Two equations have been used in which eight unknown parameters were evaluated using the coordinates of four points on the ground and the coordinates of the corresponding four points on the picture. High resolution movable cross bars were generated by an Apple II microcomputer and then superimposed on the video picture with the time. A T.V. studio facility was used for the purpose. The video tape recorder permits easy analysis of distance, velocity and acceleration for comparison with the level recorder noise trace.

RESULTS AND DISCUSSION
Results of the constant speed runs were compared with those of Ref. (1). The results of the lorry runs were scattered, whilst the results of the two cars have shown similar trends to those of Ref. (1). Results of the acceleration runs have shown similar trends to those of Ref. (2). Linear regression lines were fitted for the deceleration runs for which the correlation coefficients were 0.82, 0.85 and 0.75 for the cars, lorry when applying brakes and/or lower gear and lorry in high gear respectively.

REFERENCES
A commonly used technique for the measurement of individual vehicle noise is known as the passby method. It involves monitoring the roadside noise radiated by a vehicle as it moves, under specified operating conditions, past a measurement station. Typical vehicle operating conditions are 'driveby' (driving past at constant speed), 'coastby' (coasting or rolling past with the engine off and the transmission in neutral) and various modes of acceleration or deceleration. Other relevant parameters such as the test track or road surface macrotexture and the separation of the measurement station from the vehicle trajectory line influence the data collected by this method.

Noise data, ranging from the peak sound pressure level read directly off a sound level meter to the complete frequency spectrum may be monitored. In so doing, the effects of two phenomena - the attenuation of sound with distance from its source and the Doppler shift in frequency of sound associated with movement of its source - must be considered and minimised where possible. In this paper these effects are considered and quantified in somewhat more detail than has previously appeared in the open literature.

One procedure for minimising these effects involves the use of very small data sample periods (200 ms for example). Frequency analysis of such small samples necessarily requires relatively expensive digital instrumentation. Data presented in this paper have shown that an equally satisfactory result may be obtained for constant speed tests when a much higher (approximately 10 times) sample period is used. Selection of such a sample time allows data recording and analysis via older, conventional magnetic tape loop techniques. This has the consequent financial advantage of being able to employ analogue frequency analysers in the instrumentation system.

Having selected a sample period and a vehicle test speed, the following two parameters may be calculated - the distance travelled by the vehicle during the sample period and the resulting range of distances through which the monitored sound travels from the vehicle to the measurement station. From there the variations in sound pressure level due to distance attenuation may be calculated for a range of measurement station - vehicle trajectory separations. Equally simple calculations (documented in any reputable acoustics text) reveal the Doppler change in centre frequency of any spectral peak of interest. Examples of both these calculations are given in the paper.

By combining the results of these calculations the joint effect of distance attenuation and Doppler shift are demonstrated. Data collected during both driveby and coastby tests are presented to verify those effects. Specifically, it is shown that for a measurement station-vehicle trajectory separation of 15 m and a sample period of 1850 ms the above effects have a minimal influence on data collected using the tape-loop, analogue frequency analyser technique.
HIGHWAY NOISE CONTROL IN BRAZIL
Brasilian Acoustical Society - ABRAC

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This work intends to show the efforts of DER-RJ. (Rio de Janeiro Highway Department) to control the noise in our city. That is a pioneer project in our country, and a representative project of the noise control in its categories.

This work analyses standards, practices and research to contain noise at tolerable levels without expensive costs.

The analysed element is the vehicle traffic between two important areas in Rio de Janeiro, crossing the Catholic University Campus, near important laboratories.
GENERATION OF NOISE BY FREELY FLOWING TRAFFIC

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A number of summaries of the formulae developed to predict the noise generated by freely flowing traffic have been given in the literature (e.g. Alexandre et. al., 1975). These formulae are not in good agreement and the scatter of the data used enables a prediction accuracy rarely better than 3 db (see for example Delaney, 1972).

Some possible sources of these variations have been identified by carrying out measurements and developing a computer model simulating random traffic flow. These results are compared with theoretical prediction formulae developed to calculate the noise levels generated by a long traffic stream composed of uniformly spaced identical vehicles:

\[ L_{10} = PWL - 10 \log \left( D^2 + \left( \frac{50}{Q} \right)^2 \right) - 8 \text{ db} \quad \text{(1)} \]

\[ \text{Leq} = PWL + 10 \log \frac{Q}{VD} - 33 \text{ db} \quad \text{(2)} \]

where PWL is the sound power level of each vehicle, db, \( v \) is the vehicle velocity, km hr\(^{-1}\), \( D \) is the distance from the carriageway, m, and \( Q \) is the vehicle flow rate, vehicles hr\(^{-1}\). Use of the computer model has shown that these equations can be used for random traffic flow, with the restriction that eqn (1) is only valid for \( DQ < 5000 \).

It has been demonstrated that currently used prediction formulae underestimate the effect of heavy vehicles. Further it is shown that the variation of noise level with traffic flow rate and distance from the carriageway can not be adequately represented by logarithmic functions over the full ranges of these parameters encountered in urban areas.

REFERENCES


INTRODUCTION
A ship, especially a large one, represents an unique environment essentially different from all other ambients man lives and works in. It involves very prominent levels of audible noise, infrasound and LF vibration, affecting a human continuously and simultaneously the whole day during his working hours, rest and sleep periods, over many months. These features require careful application of assessment criteria in regard to the influence of these phenomena on human body and as well, the introduction of new ones, if necessary.

NOISE AND VIBRATION ON BOARD SHIPS
The ship can be divided in two basic environments - the engine room and superstructure, respectively. On board large ships overall condition can be represented as given in Fig.1, where the ranges of vibration and noise measured levels of four ships are plotted down.

![Graph of vibration and noise levels](image)

**Fig.1.** The spread of a) vertical vibration levels and b) infrasonic and audible noise levels on board four various ships (100 to 200 m, 8000 to 50000 t, 7000 to 17000 BHP)

CONCLUSION
The criteria applicable, for industrial noise and vibration are insufficient for the evaluation of these phenomena effects on board ships. Accordingly, there is a rising need for introducing such methods which will take into consideration the annoying and harmful effect of low frequency noise and vibration in parallel with the audible noise, especially over long exposure periods.
COMPARISON BETWEEN SOME NOISE REDUCING MEASURES ON SHIPS

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Nilsson, A.C.
Ulvund, K.E.
Det norske Veritas, P. O. Box 300, N-1322 Hoevik, Norway

Recommendations or requirements concerning maximum permissible noise levels in cabins on ships have been introduced in many countries today. This has increased the demand for efficient noise reducing measures on board. However, not only the acoustical properties of a noise reducing construction, but also the additional weight and space as well as labour and material costs and fire regulations must be considered.

There are some obvious difficulties in making full-scale comparisons on board between various noise reducing measures. However, simulated field measurements can be carried out in a test rig. The rig shown in Fig. 1 is a full-scale section of a superstructure of a ship. The rig can be excited by structure-borne sound, and the resulting sound pressure level in a cabin can be measured. In this rig the effects of four deck constructions have been investigated. Bulkheads and ceiling elements were the same in all four cases. The following deck constructions were considered:

A. 6 mm steel deck.
B. 6 mm steel deck plus 20 mm levelling compound (30 kg/m²).
C. Same as B with an additional floating floor consisting of 50 mm mineral wool and 30 mm of a concrete compound. Total weight excluding steel deck is 88 kg/m².
D. Constrained visco-elastic layer consisting of a layer of 1.5 mm visco-elastic material and 20 mm levelling compound (31 kg/m²).

The cabin was completely furnished. For the configurations C and D, an inner window was mounted in the cabin. The result is shown in Fig. 2. Curve 1 represents the sound pressure level difference measured in the cabin between constructions B and A for the same excitation, curve 2 the difference between B and C, and curve 3 between B and D. The results indicate that if - as is generally the case - the dB(A) level in the cabin is determined by low-frequency noise, the construction D can be the most advantageous. In addition floor D is less expensive and heavy than the more conventional floor C.
MEASUREMENT OF NOISE EMITTED BY VESSELS ON INLAND WATER WAYS
Physikalisch-Technische Bundesanstalt

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Bundesallee 100; 3300 Braunschweig, Germany

The procedure, described in the ISO Standard 2922 is sometimes not so easy to perform and it was the goal of the investigation to develop a measurement procedure that facilitates the measurement and provides measurement values that are in sufficient agreement with the measurement values at 25 m distance.

The new procedure is a kind of sound power measurement under free field conditions over a reflecting plane. A measurement surface was used that consisted of a hemi cylinder with a radius of 8 or 10 meters enclosing the main noise source of the vessel. Six microphone positions were chosen at one side of the hemi cylinder in two different heights above the water surface.

Measurements of the A-weighted sound power level were performed at 30 cargo vessels on the river Rhine. The microphone was attached to a rod to bring it to the specified microphone positions.

It proved, that the performance of the measurements was not very difficult. The main problem was to board and to leave the vessels during their drive. For this purpose a small motor boat was available.

When the results of the sound power measurement were used to calculate the A-weighted sound pressure at 30 m distance from the center line of the vessels good agreement was found with the measured values. The difference was not greater than 3 dB except for one vessel, where a difference of about 5 dB was observed. This increase of the difference could partly be explained by taking into account the directivity of the sound radiation.

The general conclusion is, that sound power level measurements on board the vessels are a suitable mean to estimate the noise radiation of the vessel.

Acknowledgement: The investigation was supported by the Umweltbundesamt, Berlin. The authors are grateful for the technical assistance provided by Dr. Schäle from the Versuchsanstalt für Binnenschiffbau, Duisburg, Germany.
Any man whose hearing is not extremely defective will sometimes hear noise in his immediate environment, i.e. in and around the house.

Of the types of sounds which are heard more or less "regularly" in the immediate environment, the most common noise source in the Netherlands has been found to be road traffic, closely followed by domestic noise (both from within respondents' own dwellings and from neighbouring dwellings).

Aircraft noise is also heard in or around the homes by more than half of all residents of the Netherlands.

Road traffic noise, domestic noise and aircraft noise are the "big three" as far as noise sources are concerned. Other noise sources - such as railroad traffic, industry, fairs and the like, sports fields etc. - lag far behind those three.

The most widespread noise sources also cause the most (= to most people) annoyance or even serious annoyance. Apart from the incidence of noise annoyance from various sources in the Netherlands, also the probability of noise annoyance has been studied, given the presence of a noise source in the vicinity of a person's home.

The noise source most likely to be regarded as causing annoyance if any noise is heard from it, is formed by discotheques and the like (50% of all respondents).

About 4% of the Dutch population made a complaint about noise to the authorities during the 12 months before the enquiry.

Some subgroups of the Dutch population seem to be more sensitive to noise than others: women in general and pregnant women in particular, couples without children, the higher educated, people who live in big and medium-sized cities, people who feel dissatisfied about their health. Less sensitive to noise than average seem to be: males, people in the agegroups of 16-24 years and 55 years or older, the less educated, singles, people who live in the countryside and people who feel totally satisfied about their health.
Environmental low frequency noise is normally of low level and unlikely to have direct physiological effects. However, the psychological effects of the noise as an irritant or a stress factor may be very pronounced. Noise criteria have deficiencies when dealing with low level noise and particularly so when the noise is of low frequency as well as low level. Significant changes at the low frequency end of the spectrum might be clearly perceptible but have a negligible effect on the dB level.

Generally, low level/low frequency noises become annoying when the masking effect of higher frequencies is absent. This may occur in transmission through walls and in propagation over long distances. The assessment of subjective response to a low frequency noise is complicated by the individual differences which exist, particularly in the region of threshold. For example, only one person in a household may be affected by a noise. There is a more sharply defined on-set of sensation at the low frequencies where, for example, the whole range from threshold to feeling is contained within about a 70 dB range at 20 Hz compared with a 170 dB range at 1,000 Hz. At low frequencies and low levels a doubling of loudness sensation occurs for about a 5 dB increase compared with the 10 dB which is required at mid frequencies.

A substantial number of people are disturbed by low frequency noise, especially in the early hours of the morning and late at night. Examples of annoyance from low frequency noise indicate that, in some instances, the effect is due to low frequency tinnitus, but in other cases, there is a measureable noise. Experience has shown a need for expansion of existing criteria to give answers to the following problems. Are the criteria to cover the more sensitive people? How is a throbbing characteristic of a noise to be assessed? How are we to account for sleep disturbance, individual threshold difference and low frequency tinnitus?
TEMPORAL DISTRIBUTION OF ENVIRONMENTAL NOISE
IN RESIDENTIAL AREAS

New South Wales State Pollution Control Commission

Kateifides Michael
Kotulski Peter Michael

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The environmental noise at a number of residential sites in the Sydney Metropolitan Area was monitored for three day periods excluding weekends. Australian Standard AS-1055 Type R2 residential areas were used. The ambient dB(A) noise levels were sampled at 0.1 second intervals and processed in blocks of 11000 samples (20 minutes) to give percentile exceedence noise levels, LN, and equivalent continuous noise energy, L eq.

Three major results from the surveys are listed:

- the average background noise level (taken as L eq) for the day period 0700 to 1800 hours was 36 dB(A)
- the average background noise level for the day period 0700 to 1800 hours was lower than that of the early evening period 1800 to 2200 hours.
- in some instances the L eq noise level was so erratic as to misrepresent the acoustic climate of the site.

The results are presented, discussed, and where applicable, compared to AS-1055.
Sleep Latency, Sleep Quality and Usual Traffic Noise

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Introduction: It was the aim of this study to point out whether people usually exposed to noise show sleep disturbances which can be related to noise even after years.

Method: The sleep of 20 subjects (10 female, 10 male, age 25-63) normal hearing, healthy, living in streets with high traffic density was recorded in their own sleeping rooms during 12 consecutive nights each. In nights 6 - 10 the subjects slept under experimental conditions, 10 with windows open (normally closed), 10 with earplugs.

The following parameters had been recorded: noise level (dB(A)), temperature, real time, 2 EOG, 1 EEG.

Results: The average values for noisy and quiet conditions had been calculated for each subject separately. On the basis of these data the Wilcoxon test had been carried out.

Regarding sleep latency a significant increase was found during noisy nights. Though this was true for all subjects combined, only the female subjects reported that their sleep quality was significantly less than in quiet nights.
Sleep Quality Related to Road Traffic Noise

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Introduction: Nightly occurring environmental noise is generally considered as a factor leading to a reduction of sleep, a decrease of psychological and physical performance and finally to functional as well as organic diseases.

Method: 36 subjects (18 female, 18 male students, 20-30 years), healthy, normal hearing slept in the laboratory for 12 consecutive nights each. Road traffic noise was applied by loudspeakers with equivalent noise levels ranging from 36 to 64 dB(A) indoors. Noise level, temperature, real time, 2 EOGs, 1 EEG, and body motility were recorded throughout the night for each subject. A sleep questionnaire was answered every morning during the experimental series as well as during the week before and the week following the experiments.

Results: Compared with the sleep at home the sleep quality as judged by the subjects themselves was less during the nights in the laboratory. The subjects reported that they were significantly more tired in the morning. Relating the answers of the questionnaire to the equivalent noise level it was found that the feeling to be tired increased significantly with the noise intensity during the night.
RANKING OF ENVIRONMENTAL NOISE SOURCES

New South Wales State Pollution Control Commission

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Ranking environmental noise sources is a frequently used basis for both defining the environmental noise problem and establishing control priorities. The following three methods for ranking noise sources are described:

- Exposure; ranking noise sources according to an 'objective' investigation of the community's exposure to each noise source

- Reaction; ranking noise sources according to the community's reaction as determined by a social survey

- Complaint; ranking noise sources according to an analysis of noise complaints registered by people in the community to the control authority.

The order of importance of noise sources may differ with each method used since each has a different basis for ranking. Results obtained from each of these methods for the Sydney Metropolitan Area are presented and discussed. A comparison of these results indicates a need to establish control priorities which consider all three methods.
DETERMINATION OF AMBIENT NOISE LEVELS IN THE PRESENCE OF A DISTURBING NOISE SOURCE USING A DIRECTIONAL MICROPHONE

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A method and theory is presented which enables community background noise levels to be measured with a directional microphone in the presence of a disturbing source of noise (a large plant, for example).

The microphone is used as a spatial filter with axis of least sensitivity pointed in the direction of the plant.

The calculated ambient noise level is related very simply to the measured spatial intensity and the filtering function.

The technique would find use in situations in which a plant cannot be shut down and the ambient noise level in the absence of the intruding plant noise is desired.

Limitations of the technique are discussed.

A method is devised to determine the space integral of the filtering function to calibrate the measuring system.
THE PERCENTAGE AREA METHOD OF ZONING NOISE FROM COMMERCIAL INDUSTRIAL AND TRADE PREMISES

LAMBERT

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INTRODUCTION

In developing legislation to control noise from commercial, industrial and trade premises, the Victorian EPA decided that an objective method of noise zoning was required. A method of zoning called the Percentage area method of zoning was developed and this is currently part of the Draft Policy for the Control of Noise from Commercial, Industrial or Trade Premises within the Melbourne Metropolitan Area.

DISCUSSION

Until recently the Victorian EPA used a subjective method of zoning of noise from commercial, industrial and trade premises. However, it was found that industry etc., frequently arrived at different zonings from that arrived at by EPA officers. It was also found that in some instances EPA officers could not agree on the most appropriate zoning. It was therefore decided to objectify the subjective zoning techniques for use in legislation. In this way all parties concerned would know exactly what is the maximum acceptable noise level and could plan accordingly.

ZONING METHOD

The percentage area method of zoning requires that two circles, one of 140m diameter and one of 400m diameter are drawn around a point of assessment. The percentage of land zoned commercial within each of these circles is then determined. The percentage of industrial land is then added to one half of the percentage of commercial land for each circle. The average of the resultant values of the two circles is then obtained. This value is called the influencing factor. The acceptable noise level is then determined by reading from a graph of permissible noise level versus influencing factor. This method takes into account both the amount of commercial industrial and trade land around an assessment point as well as the proximity of that land.
EVALUATION FOR THE EXCESS ATTENUATION OF NOISE PROPAGATING OVER
THE GROUND
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Matsui Masayuki
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The problems regarding the excess attenuation of the ground absorption has come to occupy an important position in various aspects.
This report is concerned with the results of the outdoor experiments on the excess attenuation by ground surface and of the prediction of the noise propagation due to the electric train.
In order to solve these problems, the experimental equation which can be used for practical design is obtained, and the method for prediction is explained.

Outdoor experiments on excess attenuation by ground surface and also scale model experiments were carried out, varying the combination of height of an artificial noise source and receiving point.

Outline of evaluation for the excess attenuation is as follows;

1) Theory of the noise propagating over the ground has been treated already with the reflection of spherical waves from a boundary, but it is not practical in use because of difficulty in the measurement. So simply and practical equation is deduced.

2) This equation is verified in outdoor experiments and the excess attenuation vs. Log (D/(Hs+Hr)) is shown in various grounds. Thereby we rearranging the expression, the experimental equation to be used in evaluation is obtained.

3) The new prediction of the propagation is established by putting this experimental equation into the conventional method. Using our method we assess the noise propagation of the electric train.

As a result, the accuracy of prediction is more improved, particularly at the point far from a noise source.

\[ L_{ex} = K_j \log_{10} \frac{D}{(H_r + H_s)} + B_j \]

where Lex: excess attenuation by ground surface.

---

**Fig-1**

Diagram showing location of source and receiver above a flat ground of the normal surface impedance.

**Fig-2**

Excess attenuation of soft soil on log10D/(Hr+Hs) = 1 and 2 in summer at Kashima seaside industries area.
Propagation factors affecting microphone location for measuring vehicle noise
National Research Council, Ottawa, Ontario, Canada K1A OR6.

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Division of Physics, National Research Council, Ottawa, Ontario, Canada K1A OR6.

INTRODUCTION

Three wave-propagation mechanisms affect sound levels measured outdoors, even at short ranges such as 7.5 m or 15 m which are commonly used for measuring vehicle noise. These mechanisms are 1) interference between direct sound and that reflected at the ground, 2) refraction due to gradients of wind or temperature, and 3) fluctuations of sound level due to atmospheric turbulence, or due to the wake produced by the movement of the vehicle.

INTERFERENCE

The significant part of most vehicle spectra is below 2500 Hz. For source and receiver heights above the ground of \( h_s \) and \( h_r \), interference reduces the measured spectrum in specific frequency bands unless the separation between vehicle and receiver is greater than a minimum value of \( d \) (\( d \approx 30 \ h_s \ h_r \) where distances are in metres). For the common measuring height \( h_s \approx 1.2 \) m, interference reduces the sound level of a typical automobile more at 7.5 m than at 15 m. Thus sound levels at 7.5 m cannot be easily extrapolated to the larger distances of interest in community noise studies. For large commercial vehicles the sound source due to the exhaust stack can be \( \approx 3 \) m above the ground, and the so-called "near-field" of the source can extend to 200 m.

REFRACTION

Avoidance of interference effects suggests low measuring points far from the source. Unfortunately these are the locations that make measurements most susceptible to refractive effects caused by vertical gradients of wind or temperature. Measurements are often made during the daytime when temperature-lapse conditions are most common. For winds of \( \lesssim 5 \) m/s or less, sound levels are not significantly affected at 15 m distance for receiver heights \( \lesssim 1.0 \) m, nor at 7.5 m distance for \( h_r \geq 0.5 \) m.

TURBULENCE

Even on relatively calm days there is sufficient atmospheric turbulence to cause significant fluctuations in the sound field (amplitude and phase) at distances of the order of 10 m. These fluctuations increase with propagation distance up to a limiting value (standard deviation of amplitude \( \approx 6 \) dB) which is reached at a distance of the order of 100 m for sounds having a frequency of about 1 kHz. Avoidance of fluctuations argues for measurements close to the source — the opposite of the requirement for avoiding interference effects.

The optimum microphone location depends on the height of the source, e.g. car or truck exhaust, or tire noise. It also depends whether measurements are required for community noise calculations where reliable absolute values are needed, or for engineering design purposes where more repeatable but relative values are required.
A method to estimate sound attenuation in built-up areas

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Apart from meteorological influences and dissipation in the air, sound propagation in built-up areas mainly depends on the inverse-square law and the effect of reflection and diffraction of sound on various obstacles like houses, walls, trees etc. Regarding any refraction of sound from its original direction as scattering of sound, sound propagation can be shown as a scattering process. A statistical model is derived, which is used to simulate the sound propagation by the MONTE-CARLO-METHOD. The obstacles within a built-up area thus are regarded as scattering obstacles, and a simple method was found to determine the parameters of the model from the data of the real situation. The results of the MONTE-CARLO-Simulation which were carried out by aid of a digital computer, are compared with measurements in a scale-model and in several outdoor situations. It can be shown, that the MONTE-CARLO-Simulation by means of the chosen model can - within certain limits - describe sound propagation in urban areas. Thus a simple method is given to calculate sound attenuation of built-up areas: there are only a few parameters and these can be determined easily, and the whole procedure is simple, so that any planning engineer can apply it without special knowledge.

This investigation was sponsored by the ministry of science and research of the Land Northrhein-Westfalia, Germany, and practised at the Institut für Technische Akustik, Technische Hochschule Aachen, Aachen, Germany.
Experiences with the application of VDI 2714 "Sound propagation outdoors" and state of the revision

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Experiences with the application of VDI-2714 "Sound propagation outdoors" and state of the revision

In December 1976 the VDI Draft Code 2714 "Outdoor Sound Propagation" was issued by the VDI-Committee on Noise Reduction. It was intended that this Code would provide a uniform method of calculating sound immission applicable to all sound sources even at greater distances between source and receiver. In the Code, a practicable set of boundary conditions is stated for calculating sound levels for planning purposes. For other meteorological conditions the range of results is given. The sound level $L_s$ at the receiver (immission point) at distance $s$ from the centre of the source is calculated according to the following equation:

$$L_s = L_w + K_\Omega - \Delta L_s - \Delta L_L - \Delta L_B - \Delta L_D - \Delta L_G - \Delta L_Z - \Delta L_M /1/$$

where

- $L_w$ = sound power level
- $K_\Omega$ = directional index
- $\Delta L_s$ = distance index
- $\Delta L_L$ = atmospheric absorption index
- $\Delta L_B$ = ground attenuation index
- $\Delta L_D$ = Attenuation index due to woodland areas
- $\Delta L_G$ = Attenuation index due to build up areas
- $\Delta L_Z$ = screening index
- $\Delta L_M$ = Attenuation index due to meteorological conditions.

After 3 years of practical testing, the draft of VDI 2714 is now being reviewed in accordance with the results before issued as a guideline. The terms in equation /1/ will be adjusted to take into account the latest available information. Preliminary results of an error-analysis of the proposed procedure show that uncertainty about meteorological data in the lower atmosphere is the principal source of differences between predicted and measured sound levels.
A STUDY OF HEDGE ATTENUATION

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INTRODUCTION

Front fences or perimeter walls are not generally approved in Canberra under existing design and siting controls. Residents often resort to front hedges for a measure of visual and physical privacy and when interviewed, often cite the reduction in dust, fumes and noise as an advantage of a front hedge.

INVESTIGATION

The outdoor attenuation for a number of front hedges about 2m high has been measured in suburban Canberra. Moderate attenuations of 3-4dB were generally recorded for frequencies below 2K Hz with greater attenuation for higher frequencies. For a typical motor car noise spectrum, the attenuation was approximately 4dB(A) and for a typical heavy truck noise spectrum, an attenuation of approximately 3dB(A) was obtained.

INSTRUMENTATION

A small loudspeaker driven by a pink noise generator was used as a point source. Sound level measurements were made using an IVIE Precision Sound Level Meter incorporating a built-in third-octave band spectrum analyser. Source stability and sound pressure level measurements were simultaneously checked with B & K type 2203 Sound Level Meters.

With the noise source 3m from the front face of the hedge, measurements were taken at 1.5m in front and behind the hedge and 1.5m from the house which typically was 7 to 10m behind the hedge. Measurements were also taken at equivalent distances over open grassland and the differences between the two sets of results were taken to be the hedge attenuation.

CONCLUSIONS

Within a few metres of front hedges in Canberra there are small but significant reductions of traffic noise. However the benefit diminishes at positions further away from the hedge.
SYNTHÈSE DES DISPOSITIONS POUR LA REDUCTION DU BRUIT DES TURBO-
ALTERNATEURS NUCLEAIRES DE 900 MW
ELECTRICITÉ DE FRANCE, Dépt. Acoustique et ALSTOM ATLANTIQUE, Le Bourget
R. BIGRET (ALSTOM ATLANTIQUE) et J. DELCAMBRE (ELECTRICITÉ DE FRANCE)

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Le premier turbo-alternateur de 900 MW à 1500 tr/mn construit par la Société
ALSTOM ATLANTIQUE a été implanté dans la Centrale Nucléaire d'Electricité de
France, à FESSENHEIM.

Entre Octobre 1977 - après la mise en service industrielle - et août 1979, les
aménagements suivants ont permis de réduire les niveaux sonores observés initialement:

- modification de la séquence d'ouverture des soupapes d'admission de la vapeur à haute
pression (à 900 MW leur ouverture est voisine de 80 %);

- modification des revêtements de calorifugeage et d'insonorisation qui couvrent :
  . les tuyaux entre les soupapes et le corps haute pression de la turbine (admission),
  . les tuyaux entre le corps haute pression de la turbine et les sécheurs-surchauffeurs
    (échappement) (Gain 17 dB à 2000 Hz),
  . le corps haute pression de la turbine (Gain 7 dB à 2000 Hz);

- modification des barrières d'étanchéité des corps basse pression (le débit d'air aspiré
  augmente, les bruits autour de 2000 et 8000 Hz diminuent);

- obturation par des tôles garnies de laine de roche du caillebotis autour du corps haute
  pression ;

- modification du capot en tête de l'alternateur, suspensions souples et garnissage
  intérieur avec de la laine de roche et de verre (épaisseur 100 mm);

- obturation par une cloison horizontale de la trémie sous l'alternateur.

Ces aménagements ont été réalisés sans perturber l'exploitation du turbo-alternateur.

Le tableau qui suit indique les niveaux sonores les plus significatifs avant (valeurs
maximales) et après les aménagements.

<table>
<thead>
<tr>
<th>Mesures à 1 m</th>
<th>Avant aménagement</th>
<th>Après aménagement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Palier avant</td>
<td>96,5 dB(A)</td>
<td>93 dB(A)</td>
</tr>
<tr>
<td>Corps haute pression</td>
<td>96,5/98,6 dB(A)</td>
<td>87,7 dB(A)</td>
</tr>
<tr>
<td>Corps basse pression</td>
<td>97,1/92 dB(A)</td>
<td>86,2 dB(A)</td>
</tr>
<tr>
<td>Alternateurs</td>
<td>94 dB(A)</td>
<td>86,6 dB(A)</td>
</tr>
<tr>
<td>Excitatrice</td>
<td>93 dB(A)</td>
<td>88,1 dB(A)</td>
</tr>
</tbody>
</table>

Les erreurs de mesurage sont voisines de ± 1,2 dB. Au voisinage du corps haute pression,
les niveaux peuvent varier lentement de quelques dB sans que la cause en soit définie.

Les gains réels obtenus sur le contour de mesurage sont vraisemblablement supérieurs
aux valeurs annoncées mais les perturbations dues aux autres sources introduisent en
certains points une majoration du bruit d0 au turbo-alternateur, majoration difficile à
evaluer avec les méthodes habituelles de mesure. L'ensemble des améliorations et
modifications apportées sur cette première machine est maintenant mis en application
par le Constructeur sur toutes les machines de ce type.
OPTIMISATION ECONOMIQUE DE LA REDUCTION DU BRUIT D'UNE INSTALLATION

ELECTRICITE DE FRANCE - Départ. Acoustique et Serv. Études Economiques Générales

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Face aux nuisances acoustiques créées par ses installations, l'industriel doit déterminer :
- quelles dispositions apportent la meilleure amélioration pour un budget donné de lutte contre le bruit ;
- quelles ressources il faut consacrer à cette lutte, aussi bien globalement que pour chacune des installations à l'origine du bruit.

La première question, purement interne, est résolue grâce à la connaissance du coût et de l'efficacité des divers moyens d'insonorisation disponibles, efficacité qui peut être mesurée à l'aide de modèles relevant du "Génie Acoustique".

Pour répondre à la deuxième question, on cherche à déterminer le niveau d'insonorisation qui minimise le coût social moyennant certaines hypothèses sur la forme de la courbe des dommages ou coût externe de la figure 1, avec \( P(N) \) dépense de prévention pour un certain niveau \( N \) de nuisance, \( D(N) \) coût externe pour le niveau \( N \).

L'optimum économique est donné par : 
\[
C(k) = P(N) + d(N) \text{ minimum,}
\]
soit \( dP(N)/dN = dD/dN \), \( dD/dN \) s'interprétant comme la somme de ce que chacun est disposé à payer pour obtenir une amélioration supplémentaire de la qualité de l'environnement.

On peut raisonnablement admettre que le coût externe dépend du niveau de bruit \( L_{eq} \) et de la population affectée \( H \) (il n'est autre que l'expression intégrale de la somme des dispositions marginales à payer de chacun) et que, lorsque la population est homogène et soumise à un même niveau de bruit \( L_{eq} \) d'effet \( \varphi \left( L_{eq} \right) \), \( D \) est proportionnel à \( H \). On peut donc écrire :
\[
D = K \sum_{i=1}^{n} H \left( M_i \right) \varphi \left( L_{eq} - M_i \right) \text{ (avec n îlots } M_i \text{ de population)}
\]

pour comparer divers types de projets en utilisant la même fonction \( \varphi \left( L_{eq} \right) \).

Appelons \( I_0 \) la valeur prise par l'indicateur d'impact en l'absence d'insonorisation de la source, l'écart \( \Delta I = I_0 - I \) traduit le gain qui résulte de la mise en œuvre d'un équipement de lutte contre le bruit et le coût externe, dans le plan \( (D, I) \), est représenté par une droite d'équation :
\[
D = k(I_0 - \Delta I) \quad (D = 0 \text{ si } \Delta I = I_0)
\]

Les dépenses optimales d'insonorisation d'une source sonore donnée sont fournies par la recherche du minimum de la somme : \( P + D = k(I_0 - \Delta I) + P \)

Cette recherche passe par l'étape consistant à déterminer les meilleures dispositions de lutte contre le bruit, c'est-à-dire d'un coût \( P \) minimal pour une valeur donnée de l'indicateur d'impact \( I = I_0 - \Delta I \).

On obtient un ensemble de courbes \((E1)\), \((E2)\) ..., enveloppées par une courbe \((\Gamma)\), lorsque l'on considère différentes techniques d'insonorisation possibles (figure 2).

Le point optimal cherché appartient aux courbes \((E)\) ou \((\Gamma)\), et permet d'associer à chaque réduction \( \Delta I \) de l'indicateur d'impact un coût d'insonorisation minimal. Inversement, la connaissance et l'application d'un coefficient \( k \) normatif facilite le choix des dépenses d'insonorisation et les rend plus cohérentes.
METHODES D'ETUDE ET DE PREVISION DU BRUIT DES GRANDS REFRIGERANTS ATMOSPHERIQUES

Ingénieur à E.D.F.
de MONTUSSAINT

Danielle

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Pour appréhender le problème du rayonnement à distance des grandes tours de refroidissement des Centrales Nucléaires le Département Acoustique d'Electricité de France utilise un programme de calcul de Génie Acoustique dont les valeurs d'entrées sont obtenues à partir d'essais effectués sur de petites installations. Certains paramètres mal connus ont fait l'objet de mesures particulières, d'essais sur maquette, ou de tentatives de calcul, le but final étant une prévision à + 2 dB(A) des niveaux émis par ces réfrigérants, avec ou sans insonorisation, dans leur voisinage situé entre 400 et 800 m.

- Mesures sur site :
  Lors de la recherche des lois d'émission, on rencontre les difficultés habituelles : - importance du bruit de fond au voisinage des installations, absence d'une zone de champ libre en terrain plat et dégagé, dimensions des appareils qui ne favorisent pas une exploration de l'espace entourant les sources. Les incertitudes ne sont levées que petit à petit et l'on a pu récemment, en explorant une verticale depuis le sol jusqu'à la hauteur du plan de refoulement, déceler une directivité de l'aspiration.

- Essais sur maquette
  Une étude sur maquette réalisée à l'échelle 1/100 (D=1,3 m - h : 1,7 m) a été entreprise au centre des maquettes du Centre Scientifique et Technique du Bâtiment à GRENOBLE. Les conditions climatiques de la salle d'essais reproduisent les conditions de propagation normales pour la gamme de fréquences 250 à 2000 Hz, dans un rayon de 1000 m. Le bruit d'impact des gouttes d'eau à la surface du bassin est simulé par une centaine de jets d'air comprimé, calibrés, régulièrement répartis dans le bassin. Il est ainsi possible d'explorer automatiquement aspiration, refoulement, et de simuler des accidents de terrain naturels et artificiels.

- Programme de calcul :
  Le réfrigérant est représenté comme la somme d'un certain nombre de sources élémentaires identiques régulièrement réparties soit à la périphérie du bassin dans l'ouverture d'aspiration, soit à la surface de l'eau. On peut multiplier le nombre des sources, jouer sur leur positionnement, introduire une directivité ou un effet d'atténuation du rideau d'eau, suivant les cas envisagés, et tenter de prendre en compte l'effet du sol ou du gradient de température lorsque les données expérimentales sont suffisantes.

- Perspectives : Les premières comparaisons maquette-grandeur et expérimentation calcul sont encourageantes. Certains points ont été mis en évidence sur maquette et contrôlés sur site, certains paramètres observés expérimentalement s'avèrent importants et doivent être introduits dans le calcul prévisionnel.
EINLEITUNG

Kernkraftwerke in der Nähe von Wohngebieten werfen unter anderem bezüglich der Lärmerzeugung Probleme auf, die bei der Planung und beim Bau vorsorglich zu berücksichtigen sind. Durch die vorliegende Untersuchung wurde die Vorausberechnung der Schallimmission eines Kernkraftwerkes (850 MW) inklusive sämtlicher Anlagen an der Bebauungsgrenze (mittlerer Abstand 1389 m) mit unserem Computerprogramm "SCHALL" unter Berücksichtigung der meteorologischen und topographischen Daten des Standorts durchgeführt. Nach der Inbetriebsetzung des Kernkraftwerkes wurden Schallmessungen bei nicht stark unterschiedlichen Meteobedingungen durchgeführt.

BERECHNUNGERGEBNISSE UND MESSERGEBNISSE

Die Berechnungsergebnisse mit dem Programm "SCHALL" und die Messergebnisse der Schallimmission des vorerwähnten Kernkraftwerkes sind in der folgenden Tabelle dargestellt:

Tabelle: Oktavband-Frequenzspektrum der Schallimmission des Kernkraftwerkes (850 MW) in einem mittleren Abstand von 1389 m (dB)

<table>
<thead>
<tr>
<th>Frequenz, Hz</th>
<th>31,5</th>
<th>63</th>
<th>125</th>
<th>250</th>
<th>500</th>
<th>1000</th>
<th>2000</th>
<th>4000</th>
<th>8000</th>
<th>A-Schalldruckpegel dB(A)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Berechnungs-</td>
<td>50,25</td>
<td>55,42</td>
<td>47,00</td>
<td>39,50</td>
<td>35,10</td>
<td>29,81</td>
<td>18,00</td>
<td>16,90</td>
<td>16,90</td>
<td>37,50</td>
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<td>ergebnisse</td>
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</tr>
<tr>
<td>Messergeb-</td>
<td>55</td>
<td>49</td>
<td>44</td>
<td>37</td>
<td>37</td>
<td>34</td>
<td>32</td>
<td>22</td>
<td>15</td>
<td>39,5</td>
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<tr>
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<td></td>
<td></td>
</tr>
</tbody>
</table>

SCHLUSSFOLGERUNGEN

Der Vergleich der Berechnungsergebnisse (Vorhersage) mit den Messergebnissen bei jeder Oktavmitte zeigt, dass die Differenz zwischen den betroffenden Pegelwerten meistens im Bereich der Messunsicherheit (+2 dB) bleibt und dass die Differenz zwischen den beiden A-Schalldruckpegeln (berechnet und gemessen) auch für solche komplexe, lärmquellenreiche und ausgedehnte Anlagen wie ein Kernkraftwerk bis etwa 2 dB(A) beträgt.
The necessary objects in new housing estates are the energetic sources—the gas boiler houses. The space source of noise that by its noisiness $L_A = 68-78$ dBA negatively influences the creation of the life environment, is in the nearness of the dwelling houses. The acoustic comfort is influenced especially by low frequencies of the sound pressure level of the spreading noise. On the base of various experimental measurements there was found that in most cases the measured values are—in interior of the flats $L_A = 3-7$ dBA and in exterior $L_A = 8-12$ dBA higher than the legislative according to ČSN/).

When suggesting precautions we must know the character of the noise sources and that of noise spreading. Then it's possible to direct the suggestion of realization to e.g. sound isolation of external constructions, whole absorption $A/m^2$, architectural precautions etc. It's not suitable to use light—weight constructions because of their low sound isolation properties and high noisiness of the boiler houses. On the base of wide experimental acoustical measurements and theoretical knowledge there were suggested precautions for bettering the acoustic comfort in dwelling houses.
Noise Source Identification using method of Generalized Least Squares Technique

Vipac & Partners Pty Ltd

TONIN RENZO

C/o: VIPAC & PARTNERS PTY LTD, 2 Holden Street, ASHFIELD, N.S.W., 2131

The noise level contributions of a multitude of sources at a chosen environmental point may be determined as if each source were operating alone by application of the method of Generalized Least Squares.

Narrow band spectra of the noise at source and in the environment are reduced to a complement of octave band base levels and a number of characteristic narrow band lines.

The resultant linear characteristic equation is solved by the method of Generalized Least Squares on a small stand-alone computer terminal.

Sensitivity to the quantity of data input is discussed.
In measuring noise propagation over a long distance (e.g. 200m and more) two main difficulties arise. First, additive noise is also recorded from surrounding sources and secondly, the channel of propagation is not time invariant. To learn about these properties of the propagation channel the following experiment is carried out.

Outdoors (Fig 1), the output of a pseudo random noise generator is fed into a high power horn loudspeaker (120dBA at 1m). The signal is recorded at some distances (e.g. 100, 200, 300m) by a directional microphone on a self synchronizing FM magtape. The identical pseudo random noise signal is recorded in an anechoic chamber (Fig 2) in the farfield of the loudspeaker, and a cross-correlation is made with the signal recorded outdoors, reproduced from the FM tape. By this way one channel of the cross-correlator is free of interference noise. The signal-to-noise enhancement is performed by a large number of short time correlations (Fig 3) which are not disturbed by phase fluctuations caused by wind, temperature and humidity fluctuations. The samples are added after phase correction by the computer. The properties of the mentioned correlation measurement will be reported.
AN INSTRUMENT TO CALIBRATE COMMUNITY NOISE ANALYSERS.
National Acoustic Laboratories, Sydney, Australia

WEISS Franz

5 Hickson Road, Millers Point, Sydney, Australia. 2000

Community noise monitors gather noise data over extended periods of time. They do this by sampling the noise at regular, short intervals and calculating and storing the number of times each noise level occurs. The noise levels are resolved to within 0.25dB for some analysers and up to 2dB for others. The dynamic range for different analysers may vary from 60dB to 100dB.

The device described in this paper automatically checks the accuracy of noise level discrimination of an analyser. Fault conditions are detected by simple examination of the output data of the analyser.

The main elements of the calibrating device are an amplitude stable signal source, a digitally programmable attenuator and some digital control circuitry. The output level of the device is changed in synchronism with the sample interval of the analyser, so only one signal level is presented to the analyser per sample interval. In this way the "no fault condition" output of the analyser is easily predictable.

A prototype was originally designed for use with noise analysers developed for the National Acoustic Laboratories, but the system should be applicable to other noise analysers.
The noise level for various types of rooms in which noise level conditions are likely to be a significant problem has been identified by many researchers. The factors that determine the tolerability of the noise are:

(a) the level and type of the noise
(b) the customary use of the building and
(c) the time fluctuations of the noise.

The instrument that is described in this paper measures the range of the so-called acceptable noise levels in buildings and their durations. Architects and sound engineers would find it more useful than the ordinary sound level meters when it comes to evaluating the suitability of a site for a particular building. The principles and performance of the system are discussed.
A high-level impulsive noise source for test purposes

National Acoustic Laboratories, Sydney; Defence Research Centre, Salisbury

Kenna, L.C. and Smith, R.N.

5 Hickson Road, Millers Point, N.S.W., 2000. Australia.

Because an increasing proportion of the noise investigation work of the National Acoustic Laboratories involves impulsive noise, when the Laboratories were requested by the Department of Defence to help develop hearing protection systems for artillery crews, it was considered that a high-level impulsive noise source with safe operating characteristics would be an invaluable aid for research, testing and calibration purposes.

The Defence Research Centre, Salisbury, was asked to attempt development of a source with the following characteristics:

(i) capable of producing a sound pressure level of 180 dB peak at a distance of one metre, with essentially uniform level over an area of 0.5 metres square;
(ii) able to be operated remotely, at intervals of approximately one minute;
(iii) to produce no noxious fumes or high temperatures which could constitute a fire hazard.

A generator has been developed in accordance with this specification. The principle of operation is basically the sudden release, by means of an electrically-operated control valve, of compressed air to the atmosphere from a pressure chamber. Details of the design and operation of the impulse generator are presented in the paper.

Acoustic measurements show that the generator provides an impulse exceeding the specified level when operated at the rated pressure. The paper includes details of the levels achieved, the variation of sound pressure level with chamber pressure and distance from the source, and the polar pattern of distribution of sound pressure level.
Low-Noise Broadband Modulated Preamplifier for a Variety of Transducers

Corliss, Edith L.R. and Penzes, William B.

National Bureau of Standards

Rm. A149, Sound Bldg., Nat'l Bureau of Standards, Washington, DC 20234, USA

A diode can be used as an adjustable capacitor. Over a range of voltages, its capacitance changes in direct proportion to the voltage impressed upon it, so long as the region near its conductivity transition voltage is not transgressed. This property makes the diode suitable for use in an R-F bridge circuit. In particular, by making the diode part of a series-resonant arm in the bridge, the arm forms a very low-impedance source, offering a favorable condition for a high signal-to-noise ratio.

Accordingly, with the circuit as shown in Figure 1, the output of any sensor convertible into a voltage within the frequency range passed by the bridge detector circuit can be amplified in effect to a suitable level.

The change of capacitance of the diode changes the impedance of the series-resonant arm of the bridge. The bridge is excited by R-F, for example, 10 MHz from a crystal clock. The unbalanced bridge generates a suppressed-carrier modulation, that can be demodulated by conventional FM detectors, and amplified conventionally. The estimated range of frequencies for a 10-MHz bridge is essentially D-C to 5 MHz. In the circuit shown in Figure 1, the elements in the dashed-line circle are needed to preserve the proper bias on the diode.

![Figure 1](image_url)

The original form of this bridge circuit was a modification made, by Walter Koldan at the National Bureau of Standards, from a microphone detector circuit devised by Zaalberg van Zelst of Philips Laboratories. Because of the present availability of solid-state components and especially stable clock oscillators, the entire circuit can be mounted as a readily portable device. If the elements marked with asterisks are precision components, the bridge can have a stable calibration limited primarily by the thermal coefficients of the diode and coil.
DANISH ACTIVITIES AND REGULATIONS CONCERNING OCCUPATIONAL NOISE AND NOISE IN THE ENVIRONMENT

Acoustical Laboratory, The Technical University, Denmark

Ingerslev Fritz H.B.

The Technical University, 2800 Lyngby, Denmark

I OCCUPATIONAL NOISE

The Danish Labour Inspection Service has recommended to the authorities that the maximum permissible noise exposure level, determined as $L_{A,eq}^{1}$ (8 hours), shall be reduced from 90 dB to 85 dB. At places of work, where the noise exposure can be reduced below 85 dB using known and practical means, $L_{A,eq}^{2}$ (8 hours) shall be reduced to lower values, even as low as 55 dB.

It is furthermore proposed that a mandatory noise labelling of noisy machinery shall be introduced. The noise declaration shall include 1) the $A$-weighted sound power level in dB and 2) the $A$-weighted sound pressure level at operators' position, determined according to appropriate noise measurement standards.

II EXTERNAL INDUSTRIAL NOISE

The present Danish guidelines are fairly strict. In areas with one-storied and two-storied residential houses the maximum permissible noise exposure level $L_{A,eq}^{3}$ produced by a new industrial enterprise is 45 dB during daytime, 40 dB in the evening and 35 during nighttime. These values and values for other areas are based on the principles laid down in ISO 1996. The basic criterion used was 40 dB.

III ROAD TRAFFIC NOISE

The Danish regulations established by the National Agency of Environmental Protection state - as guidelines - that the environment in residential areas can be described as satisfactory if the $L_{A,eq}^{4}$ (24 hours) measured outside the building - in free field - is $< 45$ dB. The environment is described as unsatisfactory if $L_{A,eq}^{5}$ (24 hours) are $> 55$ dB. Proper values are laid down for other areas.

IV AIRCRAFT NOISE

The proposed Danish guidelines for the description of the environment in areas exposed to aircraft noise are based on the same figures as stated for road traffic noise. It shall be noted that operations between 19-22 hours and between 22-07 hours are given a penalty of 5 dB respectively 10 dB before $L_{A,eq}^{6}$ (24 hours) is determined.
PROGRESS WITH NOISE ABATEMENT ZONES IN THE UNITED KINGDOM

BUILDING RESEARCH ESTABLISHMENT

UTLEY WILLIAM ALBERT

BUILDING RESEARCH STATION, GARSTON, WATFORD WD2 7JR, HERTFORDSHIRE, ENGLAND

INTRODUCTION

When Part III of the Control of Pollution Act came into force in England and Wales in 1976 it gave to local authorities new powers to control noise from fixed premises by allowing them to set up areas of special control, called Noise Abatement Zones (NAZ) (1). The basis of control within a NAZ is a register in which existing noise levels around classified premises within the Zone are kept. These registered noise levels can be exceeded only with the consent of the local authority. The local authority may seek a reduction in noise levels when they consider that these are unacceptable and that a reduction is practicable at reasonable cost and would afford a public benefit. Regulations (2) describe where, when and how to measure the noise. The noise level to be measured is the equivalent continuous noise level (Leq). Minimum equipment requirements are specified in relation to the temporal variation of the noise.

CURRENT PROGRESS

BRE has undertaken a survey among the first twelve local authorities to set up NAZs in order to examine how they were operating their zones and to ascertain what technical problems, if any, had arisen. This survey showed that progress has been satisfactory with no major technical problems encountered. However, the survey indicated that further consideration should be given to sampling procedures and to the measurement or calculation of noise levels in situations where there is significant extraneous noise. This would enable improved guidance to be given to those operating or about to set up NAZs. While all the noise registers inspected contained the required basic information it appeared that the inclusion of more detail about site conditions would be useful for future control.

REFERENCES


   The Control of Noise (Measurement and Registers) Regulations.
   HMSO 1976.
The contents of ISO R 1996, Assessment of noise with respect to community response have been incorporated in the legislation of many countries. Several shortcomings have, however, been pointed out and for this reason ISO has appointed a Working Group (WG 18) to prepare an updated and revised version of the document.

The group has decided to divide the new document into three parts:

1. Description and measurement of environmental noise; basic quantities and procedures.
2. Description and measurement of environmental noise; land use planning.
3. Description and measurement of environmental noise; application to specific situations.

(Further parts are envisaged).

Part 1 is issued as a Draft International Standard giving detailed information on measurement procedures based on the determination of the equivalent continuous A-weighted sound pressure level, $L_{Aeq}$, with modern instrumentation.

Part 2 and 3 are still treated by the Working Group.

Part 2 will give guidance for land use planning taking into account not only the existing noise situation, but also future changes, e.g. establishment of industrial plants.

Part 3 will describe applications of standardized methods to special situations, i.e. where small groups of people are annoyed by a well-defined noise source, e.g. a fan.

The paper will give a discussion of the basic factors and problems and outline the contents of the new standards.
NOISE ZONING IN YUGOSLAVIA

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Yugoslavia

INTRODUCTION

The need to pass acts in order to regulate the permitted level of the community noise in Yugoslav towns has been very prominent last few years. The sudden development of traffic and industry, in big cities such as Belgrade, Zagreb, Sarajevo and others has led to the phenomenon that the level of community noise seriously jeopardizes the normal life and work of inhabitants.

NOISE LEGISLATION

The first regulation having the force of a law was passed in the Republic of Slovenia in 1976 and it regulated, only in principle, the problems of noise, bringing the general measures in the dwelling, recreation and other laws concerning cities. A year later, in 1977, in the same Republic, a law on the protection of human environment with respect to the noise was adopted; the law regulated the permitted levels of community noise. The urban municipalities were subdivided into six zones, and the maximum permitted levels expressed in dB(A) range from 45 to 70 for the day-time and from 40 to 70 for the night-time. The values were expressed through the equivalent noise levels.

In the course of 1978 and 1979 other Yugoslav Republics proceeded to the elaboration of the respective regulations with respect to community noise. In the Republic of Bosnia and Herzegovina in 1978, a Law was passed on the permitted level of the community noise, and in Serbia the adoption of a similar law is expected in near future.

Since the main towns of Yugoslav Republics develop and extend rapidly (Beograd had, for example, in 1948 about 350,000 inhabitants and only few thousands automobiles, while in 1979 it has 1,400,000 inhabitants and about 350,000 vehicles), it is necessary at the design and construction of new settlements to have the regulations concerning the community noise. Since few years ago the urbanists have been paying due attention to the problems of noise; we may say that serious errors occurred earlier which are the cause of difficult conditions of housing due to permanent exposure to the exceeding noise.

CONCLUSION

It may be inferred that in Yugoslavia an intensive legislative activity is in course in the field of community noise. In all big cities the existing parts are subdivided into zones; especially in designing new settlements the community noise is paid due attention.
Before 1976, the French Legislation on noise was of a general character and expressed mostly in qualitative requirements or on specific items like building insulation or car noise level.

In 1976, the publication of two basic laws and one ministerial instruction bring a great modification.

The 10 July law relative to the "protection of nature" and the 19 July law on "classified establishment for the protection of the environment" introduced the obligation of impact studies, where noise is an item to be taken into consideration, and the procedure for the publicity of its result so that every one concerne could make remarks.

Therefore, noise control studies which were mostly made on voluntary bases became compulsory. The contractors are faced with the obligation of careful impact studies of the noise of their installation.

The new texts require information on: the initial acoustical situation, the noise source level, the foreseen impact, the characteristic of the insulation devices. The result to be obtained and the cost.

Bearing in mind that all the acoustical characteristics given in the impact studies are public, the responsibility of the contractor is engaged. He can eventually be proceeded if he do not comply with his own provisions. Which actually cannot be made with a confidence interval smaller than 5 dB. This impose to acousticians to examine carefully the present methods for noise control in order to have a better estimation of their precision and to develop new methods of better precision. Special efforts are to be made in the following item.

Characterization of an acoustical situation of an area where new installations are foreseen, clarification is needed on:

- the choice of the noise evaluation level $L_{Aeq}$ or $L_x$,
- the period of measurement and its relation with the period of reference,
- the tonal and impulsive correction,
- the determination of the noise impact from the reference level and the population density,
- the corrections needed to take into account the meteorological factors, to permit comparison of the same installation measured in different meteorological situations.

- Knowledge of various noise sources power level
  A critical examination of the present text code method is to be made and new method based on the direct measurement of the intensity flux are to be developed.

- Calculation procedures to forecast noise level for future installation
  Apart from a better knowledge of the noise sources characteristics, improvement have to be made in the method taking ground effect and meteorological effect into the calculation procedures.

This need of better precision lead to the necessity of new researches on human reaction, metrology of sound, propagation of sound, in enclosure or out-door for various types of area.

One can say that the new trend regulation can have a positive impact on the development of acoustical engineering.
COMMUNITY NOISE REGULATION & ENFORCEMENT

ENVIRONMENT ONTARIO, NOISE POLLUTION CONTROL SECTION

MANUEL

JHN

ENVIRONMENT ONTARIO  135 ST. CLAIR AVE. W.,
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Community noise enforcement has traditionally consisted of a mixed response by the local authority to complaints of unwanted noise. Because most noise legislation is expressed in subjective terms, enforcement depends on the subjective assessment of the offending noise. Furthermore the courts, having to be convinced that the noise is offensive, make the outcome of many prosecutions doubtful. Prosecution is thus time consuming and costly and entirely dependent on the enthusiasm and dedication of the enforcement authority and the corroborating witnesses.

In order to avoid many of the problems described above, an attempt has been recently made to assess noise nuisance quantitatively. This paper will compare the results of procedures followed by the Ontario Ministry of the Environment, the City of Montreal and the State of Victoria in determining the acceptable or baseline ambient noise level in any community situation and thus the excess sound above the baseline level due to noise nuisance.

Some preliminary results were reported at the Windsor, Ontario meeting of the Canadian Acoustical Association in October 1979. These results have been further refined and other common community noise sources have been investigated and analysed. The noise of air conditioners, heat pumps, fans and noise of industrial equipment are typical sources of public complaint.
This paper deals with the problems confronting Technical witnesses and counsel in preparing and presenting evidence before tribunals and courts and, to a lesser extent, before local government committees.

The principal difficulty is that of mounting a case within the framework of an adversary system. This often devolves to building evidence around an uncontroversible set of measurements - known to both sides - and relying upon interpretation and often a highly subjective analysis of future attitudes to a potential nuisance. Nevertheless, it is seen to be imperative to hide such subjectivity and personal opinion either in jargon or the protection of the witnesses (official) position and standing.

We have, in fact, the juggling of measurements and "standard methods" to suit the case in hand. The paper goes on to suggest that both the judiciary and the legal profession as a whole see too much law and too little of public expectation.

It is suggested that the public good could well be better served by appropriate learned bodies providing, or nominating, panels of expert witnesses that could be servants of the courts rather than being, sometimes, unwilling servants to a cause.

Also, the problem of judicial "expertness" is dealt with. The author elaborates his remarks with some examples from his experience in New Zealand.
MELBOURNE NOISE SURVEY

Environment Protection Authority of Victoria

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INTRODUCTION

This noise survey is a part of Victoria's programme to control noise from industry as it affects nearby residents. The results of the survey are to be used as an input in the development of an area zone rating system for the setting of maximum allowable noise levels.

SURVEY METHOD

A noise survey of 40 residential sites in various land use situations was carried out throughout the metropolitan area of Melbourne. Measurements were taken at each site over a period of 24 hours. Seven of the sites were further surveyed over the weekends and one of the sites was surveyed over a period of eleven days. Several statistical indices and $L_{eq}$ were determined from the level frequency distribution which were automatically printed out each hour. 24 hour tape recordings were used to identify any significant noise events which were indicated on the graphic trace from a level recorder. A traffic counter was used to determine traffic flow patterns at sites near to main roads.

RESULTS

The average background sound levels (arithmetic average of the hourly $L_{90}$ levels in dB(A)) for the day, evening and night periods were compared with various systems of zoning areas for maximum allowable levels. The systems of zoning included subjective land use area zoning, objective land use area zoning and a system based upon the land use planning scheme issued by Melbourne's planning authority.

Allowing for the effects of traffic, it was found that the average background sound levels for day, evening and night periods were closely related to the zonings. The average background sound levels increasing for areas with a greater degree of commerce or industry.
Developing a Data Base for Community Noise Standards
School of Public Health, Columbia University, U.S.A.
Paul N. Borsky
367 Franklin Avenue, Franklin Square, N.Y. 11010

The goal of regulatory agencies is to ascertain what noise exposures would be considered acceptable to most communities. To answer this apparently simple question it is necessary to establish the dose-response relationships between noise exposures and community annoyance levels. The usual research strategy is to measure noise exposures and relate them to responses obtained from personal interviews of residents living under different noise environments. Since community noise incidents are constantly varying over time - it is essential to establish how residents exposed to these thousands of time varying noise stimuli weight each event and integrate them into a perception of noise intensity. At present there are a number of engineering solutions to this problem, none of which are based on broad community response data. There are basically three aspects of time varying noise exposures which need more precise definition: 1) How are different numbers and different levels of noise exposure integrated by residents engaged in different activities? In most existing schemes it is assumed that an energy rule operates. Recent laboratory studies suggest that this is an oversimplification of how people react. Results of these limited studies suggest that different rules apply when the number of exposures are few or many. 2) Are exposures during different time periods of equal importance or are evening and night events more annoying and, therefore, different weights should be given to each period. Those who believe in the equal importance hypothesis use an Leq noise index; others use a 10:1, 15:1 or 20:1 day-night penalty. There is, however, no firm data base for supporting any of these views. 3) How does the fluctuating ambient noise resulting from road traffic and commercial and industrial and other noise sources influence perception and annoyance with an intrusive noise, such as an aircraft noise flyover?

To provide firm answers to these questions, a research strategy was developed to find communities where day and/or evening noise exposures were the same but where night time exposures varied in number of exposures. By recording samples of 24-hour noise exposures and the use of other data, all existing and best fit formulas for integrating noise can be calculated and tested with the response data. Then responses from residents actually living under these measured different environments could be compared and the day-evening and night-time penalties could be calculated. A survey of operating conditions at about 50 of the most important airport areas in the U.S. has been completed and a sufficient variety of noise exposure conditions has been found to do a national survey.
C54 SIMPLE AND EFFECTIVE LEGISLATIVE CONTROL OF NOISE FROM DOMESTIC PREMISES

Member Australian Acoustical Society

LAMBERT, John Alexander

Environment Protection Authority, 240 Victoria Parade, East Melbourne. 3002

The primary aim of any noise control legislation should be to prevent excessive noise. However, to be effective, legislation must also be shown to be successful when used in a court of law.

The domestic noise control provisions of the Environment Protection Act 1970 have been shown to be effective both in and out of court. This has been achieved by making the provisions extremely simple for residents (the involved parties) to understand and by allowing a Magistrate complete freedom to consider all circumstances relevant to any dispute he may be hearing.

The relevant provisions, contained in Section 48A of the Act, make it an offence for a person on residential premises to cause "unreasonable noise". The term "unreasonable noise" is defined simply as any noise that is "... unreasonable having regard to the time, place and other circumstances ..." of the use of the item causing the noise.

This form of legislation is recognised as being unsuitable for the control of noise from premises other than domestic premises and for some fixed items of equipment used on domestic premises, e.g. air-conditioners. It is however recommended for the control of most noise from domestic premises.
COMMUNITY ATTITUDES TO NOISE FROM A PETROLEUM REFINERY

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INTRODUCTION

A sociological survey of a community living adjacent to a large petrochemical refinery was carried out to assess annoyance caused by refinery noise.

METHODOLOGY

A sample size of 100 was used and the target population was divided evenly between residential areas to the north and south of the refinery complex. The northern and southern residential areas were 130 metres and 450 metres from the refinery boundary respectively. A random sample was generated. The questionnaire included questions related to noise annoyance, socio-economic status and length of residence in the area. The data were analysed using the ORISIS package of computer programmes from the Institute of Social Research, University of Michigan.

RESULTS

30% of respondents were found to be annoyed by noise from the refinery with likelihood of annoyance generally decreasing with distance from the refinery. A boundary condition was found to exist with those adjacent to the refinery both to the north and south, regardless of actual distance, being more likely to be annoyed.

Almost twice as many respondents were annoyed by "characteristic" noise such as tonal or impulsive noise as were annoyed by continuous broad-band noise.

Multi-variate analysis showed that the most important predictors for noise annoyance were; the distance from the refinery centroid (the closer the more likelihood of annoyance), the number of years lived in the present house (the greater the less likelihood of annoyance) and whether to the north or south of the refinery (this being related to distance from the refinery and to the respondents' expectations of their neighbourhood).

CONCLUSIONS

(i) Noise from the refinery was found to have an adverse effect on the immediate community.

(ii) "Characteristic" noise was found to be more likely to cause annoyance than continuous broad-band noise.

(iii) For this particular refinery, a buffer zone of 1.5 kilometers was found to be adequate to reduce noise annoyance to 5% or less. This also coincided with a ceiling value of 80% of respondents very satisfied with the area.
Environmental Noise from an Asphalt Plant
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In many United States municipalities, environmental noise criteria are established for industrial plants. These criteria are often set as permissible octave band, sound pressure levels. For example, in the City of Chicago these levels range from 79 dB at 31.5 Hz. to 39 dB at 8000 Hz. in restricted manufacturing districts. In order to locate an industrial plant in such a restricted zone, it is necessary to estimate the expected noise output from the plant.

For this program, an asphalt preparation plant was to be located in a restricted noise district. Environmental noise measurements were obtained at an existing plant with similar process equipment. Octave band measurements are presented at several locations surrounding the plant. The major sound source was identified (asphalt mixer), and noise decay versus distance was measured away from this source. The required distance to the new property line was then determined. Since raw materials are piled near the mixer unit, some shielding can be derived from these piles. If property boundaries are too close for natural sound attenuation, the judicious placement of raw material piles can help in reducing boundary line noise.
HORNSBY SHIRE'S "GENERAL SOUND INSULATION CODE FOR RESIDENTIAL FLAT BUILDINGS"

JAMES A. MADDEN ASSOCIATES PTY. LTD.

COOPER STEVEN EDWIN

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The results of a survey conducted by the Hornsby Shire Council in 1973 indicated that:

"The greatest shortcoming to unit living is intrusion of noise and the resultant loss of privacy and the need for greater soundproofing measures including detailed consideration of unit and room orientation to overcome this problem."

The Council in 1975 introduced an Acoustical Compliance Code for New Home Unit Developments which is simple, practical and works. This Code specifies the required sound insulating performance of Home Unit Buildings against the intrusion of both external and internal noise, having regard to its annoying effects such as interference with rest and social activities.

Particular consideration is given to the need for effective sound insulation against road and rail traffic noise, and the need for reasonable privacy between adjoining units and like areas.

The Code describes a method for measuring and rating noise levels, time and duration of measurement, measurement location, instrumentation, shielded and exposed situation influences, and the evaluation of sound insulation performance. Of particular interest in the Code is the criteria of maximum levels of L_{10} (20 minute) of 50 dB(A) with windows 'open' and 40 dB(A) with windows 'closed' during specified periods of peak traffic movements.

This paper discusses the Code's operation over the last five years and the very successful results achieved to date.
The government of the Province of Ontario has recently issued several policy statements on noise control in land use planning. There is no longer any doubt that noise control is of prime importance in planning the development of urban and rural areas, transportation corridors and facilities. The public will no longer tolerate excessively high noise levels in the community and the government has been forced to take expensive remedial measures where problems already exist. In new situations, however, strict noise control policies are intended to minimize transportation noise impacts over the long term.

After 5 years of experience in the analysis of noise control planning proposals, the government of Ontario has developed substantial technical expertise. A comprehensive acoustical training program and training publications are also available. This paper will discuss specific aspects of the ongoing program such as noise prediction models, ongoing research and an analysis of the noise abatement achieved in certain new residential developments since the noise control program was initiated.
Flying foxes, operating silently at night while we sleep, can cause considerable damage to fruit and fruit trees. The world-wide accepted method of frightening these flying foxes away from orchards is the use of repetitively exploding gas guns. These guns emit an explosive, percussive sound every few minutes, 24 hours per day. Each blast exceeds 130 dB(A) at 60 metres: more than enough to waken neighbours up to four kilometres away. The noise problem is aggravated by several different orchardists using gas guns in an area, and also by the need to use shotguns at intervals to reinforce the foxes' association of the noise with harm to themselves. The scope of the problem is illustrated by the fact that by 1978 over 3,000 of these guns had been sold in New South Wales.

As part of a three-year project to find a suitable alternative deterrent, six silent deterrent systems and one quieter system were tested with success during the 1979-80 fruit season. These systems involved suspended wire grids, mesh netting, hanging strips, electrified wires, flashes of high-intensity lighting (triggered randomly or periodically) and high-frequency, low-intensity sound. This paper reports on these trials and the results so far, including the costs and effectiveness of the alternatives.
D. SHOCK AND VIBRATION
NARROW BAND MECHANICAL FILTERS WITH TORSIONAL RESONATORS

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This paper deals with the torsional-mode mechanical filters with differential connection of two kinds of coupling wires: that is, one is a coupling wire of in-phase mode and the other is such a coupling wire of out-of-phase mode that reduces the coupling quantity obtained by the former. Fig.1 shows the coupled resonators and its equivalent circuit for an explanation of this principle.

This configuration of the filter is suitable not only to obtain a narrow band, but also to suppress the spurious response due to a higher resonant mode of transducer resonator with piezo-ceramics. The structure, moreover, is strong in comparison with a usual narrow band torsional-mode mechanical filter with one small diameter wire for the coupling.

A principle of methods for suppression of the spurious response is that resonators should be coupled with each other at the nodal points of the higher mode intended to be suppressed, or that the force factor of unwanted mode should be made zero.

In this paper, the mechanical filters of 2 and 4 element types are discussed, which are designed under consideration of the methods mentioned above. In these filters, the spurious response due to the 3rd resonant mode is suppressed by the former method, and the spurious responses of even resonant modes are suppressed by the latter method. The experimental spurious response of 4 element type filter is shown in Fig.2. On the other hand, the 3rd resonant mode is not able to suppress by the former method in the usual mechanical filter because the weld points of coupling wire are far from the nodal points of the mode in case of a narrow band.

From these results, this paper concludes that differential connection of coupling wires is sufficiently effective to suppress the spurious response and to strengthen the structure of a narrow band mechanical filter.

Fig.1 (a) Coupled resonators and (b) its equivalent circuit by impedance analogy.

Fig.2 Experimental spurious response of 4 element type mechanical filter.
LOW FREQUENCY TUNNING FORK WITH HIGH STABILITY FOR SUPPORT

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A new-type tuning fork as shown in Fig.1 is proposed, which is used at the first flexural resonant mode and supported at a no-moving point. This no-moving point is obtained by inclining both arms of a usual-type tuning fork to the inside by a certain angle $\phi_0$.

Such a modified tuning fork keeps a high effective resonant quality factor $Q_e$ because of very small vibrational leakage energy vanishing through the support, and its resonant frequency $f_1$ is not influenced by the rigid body vibration.

In this paper, design conditions for the modified tuning fork are made clear through an analysis by use of an equivalent mechanical network theory, and all the calculated results are proved experimentally. The relation between the optimum angle $\phi_0$ and the dimensional ratio of $\xi = \lambda_2 / \lambda_1$ is shown in Fig.2. The experimental quality factor $Q_e$ to the angle $\phi$ at $\xi = 0.3$ is shown in Fig.3, in which $Q_e$ takes the maximum value at $\phi = \phi_0$. From these results, it will be seen that this modification of form is very useful to realize a tuning fork with high stability for supports. Moreover, an example applying such modified tuning forks to a mechanical filter is shown in last.

![Fig.1 New-type tuning fork](image)

![Fig.2 Calculated optimum angle $\phi_0$ to the dimensional ratio $\xi$.](image)

![Fig.3 Experimental $Q_e$ of the supported tuning fork to the angle $\phi$.](image)
ELECTROMECHANICAL SILICON BEAM FILTERS

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Recently developed integrated electromechanical silicon beam filters are based on the mechanical resonance of silicon beams and piezoresistive properties of silicon. They incorporate the good points of integrated circuits technology and mechanical resonators. Since the theory and design of single silicon beam filters are discussed elsewhere, this paper introduces and develops the theory of double beam structures.

The double beam silicon filter consists of two mechanically coupled cantilevered beams, placed over a metal plate. On the top of each beam at its free end an aluminum film is deposited. The film and the metal plate form an input transducer. Between the film and the metal plate input signals are applied. The electrostatic force produced by the input signal will cause vibrations of the structure. The vibrations are sensed by the piezoresistive bridges placed either at the foot of each beam, or on the coupling part. The bridge is formed by four equal resistors. Two of them are in the direction parallel to the length of the beams, and the other two are perpendicular to it. The bridges are in balance when the structure does not vibrate, but when it does the resistances, due to the piezoresistive properties of silicon, will change differently and an output voltage will occur. The output voltage will be maximal when the frequency of the signal is equal to one of the mechanical resonant frequencies of the structure. At all other frequencies the output voltage will be much smaller. To prove this the transfer functions of the device will be found. Since it is difficult to find the transfer functions directly, they will be expressed as a product of these terms:

\[ T_{kj}(s) = \frac{V_{ok} \sigma_{Bk} F_j}{V_{ij} \sigma_{Bk} F_j V_{ij}}, \quad k=1,2,3 \]

where \( F_j \) is the force caused by the input voltage \( V_{ij} \); and \( \sigma_{Bk} \) is the stress at the center of the corresponding bridge. The first term in (1) represents the transfer function of the bridge, the second term is the transfer function of the vibrating system, and the third term is the transfer function of the input transducer. The first and the third terms have the same forms as in the single beam case, but the second term is different. The second term has six different forms corresponding to six combinations of input-output voltage pairs. All these transfer functions have the same poles, but different zeros. The poles are determined by the mechanical resonant frequencies of the structure and can be found using the lumped equivalent model. The zeros are derived from partial differential equations for the forced vibrations of the structure.
MEASUREMENT OF SMALL MECHANICAL IMPEDANCES

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The measurement of small mechanical impedances in a wide frequency range, e.g. the mechanical impedance of phonograph pick-ups, is a rather difficult problem to deal with.

A measuring method based on the use of a stereo cutting head (intended for use in disc recording systems at frequencies up to 25 kHz) has been investigated.

The moving elements of the cutting head, each corresponding to a stereo channel, contained a drive coil in a strong magnetic field in addition to a pick-up coil in a separate magnetic circuit. Series connection of the drive coils and pick-up coils respectively made it possible to apply identical forces to the two moving elements and measure their velocity difference. By loading one of the moving elements with a phonograph pick-up stylus tip, the velocity difference curve, assuming the two channels being identical, will show the mechanical mobility curve of the pick-up.

Due to unavoidable differences in the two separate channels of the cutting head a direct recording of the impedance curve was not possible. The method, however, permitted measurement of impedance values in the magnitude of at least down to $10^{-2} \, \text{Nms}^{-1}$.

Further investigation and refinement of the measuring method are still going on.
An Automated System for Interferometric Calibration of Vibration Pickups

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INTRODUCTION

Precise measurements of the amplitude of vibratory motion are most frequently made with optical interferometers. The established techniques utilize the alternating component of the photodetector output: either by counting the number of fringes per vibration cycle or by processing the individual harmonics. In the present communication a procedure that uses only the d-c component is proposed as a simple absolute method for calibrating vibration pickups. The simplicity of the procedure and of the signal processing permit the method to be fully automated.

THEORETICAL BASIS

If the optical-path-length difference of a Michelson interferometer with a monochromatic source is modulated by sinusoidally vibrating one of its mirrors, the time average of the intensity of the light impinging on the photodetector is of the form

\[ I = A_0 + A_1 \cos\left(\frac{4\pi}{\lambda} \delta \right) \cdot J_0\left(\frac{4\pi}{\lambda} \xi \right), \]

where \( A_0 \) and \( A_1 \) are constants, \( \delta \) is one-half the optical path length difference, \( \xi \) is the peak value of the sinusoidal vibration, \( \lambda \) is the wavelength of the light, and \( J_0(x) \) is the zero-order Bessel function of the first kind. For a given value of \( \xi \), the difference between maximum and minimum intensities, as \( \xi \) is varied, becomes

\[ I_{\text{max}} - I_{\text{min}} = 2 A_1 J_0\left(\frac{4\pi}{\lambda} \xi \right). \]

When \( \xi \) assumes a value which makes \( J_0\left(\frac{4\pi}{\lambda} \xi \right) = 0 \), this difference reduces to zero. (In actuality only a minimum can be observed, which is discussed in the oral presentation.)

IMPLEMENTATION

The above concept has been applied to a He-Ne laser interferometer, for which the first zero of the Bessel function occurs at \( \xi = 121.1 \) nm. To vary the average intensity, and therefore the d-c component of the photodetector output, the "fixed" mirror is moved back and forth by a slow-sweep drive. A minicomputer has been programmed to increment \( \xi \) and monitor \( I_{\text{max}} - I_{\text{min}} \). When this difference reaches zero, the incrementation of \( \xi \) is computed. The procedure is then repeated at the next frequency of vibration.

Measurements of acceleration sensitivity have been performed by the proposed method and compared with the results obtained by other techniques. The agreement is satisfactory. Factors affecting the accuracy and precision of the method have been investigated.
A PORTABLE LASER-DOPPLER VIBRATION METER FOR ENGINEERING USE

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INTRODUCTION

The use of Laser Doppler Velocimetry (LDV) for non-contact measurement on vibratory surfaces has led to the development of a series of instruments which are now commercially available. Designs for on-axis (laser beam) velocity measurement require the optical mixing of a frequency pre-shifted (reference beam) with Doppler shifted light scattered from the surface of measurement and it is largely the method of frequency shifting chosen which distinguishes designs. Commercial instruments employ Bragg cells and diffraction gratings to achieve this frequency shift and as such virtually restrict their use to the laboratory. These optical components are expensive and cause the completed design to be space consuming and sensitive (in terms of alignment) to external vibrations. It is the expense and sensitivity of optical instrumentation which is hindering the widespread use of the Laser-Doppler technique. There is a need for an engineering instrument which is robust and inexpensive and can be used on site where swift, accurate non-contact measurements are necessary. This paper describes the final development and testing of the instrument which was first reported in (1) and is designed with these properties in mind. Details of the optical geometry and initial tests can be found in Ref (2).

RESULTS AND COMPARISONS

Figure 1 shows the noise floor of the instrument compared with that of a commercially produced design which utilises a diffraction grating in order to frequency shift. These show that the instrument can be applied in situations where the rms velocity of the surface is as low as $10^{-5}$ m s$^{-1}$. The results of final tests in proving the instrument through measurements on a diesel engine casing and loudspeaker cone are presented and discussed. The dynamic range and limitations of use is identified.

REFERENCES


THE PRESSURE FIELD INSIDE A POINT-EXCITED FINITE CYLINDRICAL SHELL
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INTRODUCTION
Statistical Energy Analysis (SEA) is used to make initial investigation into
the mechanical transmission and radiation of sound into a point-excited cylin-
der. The aim is to develop a method for the prediction of the sound pressure
level inside such structures as aircraft fuselages and spacecraft excited by
force fields which are random in time.

THEORETICAL CONSIDERATIONS
The basic relationships are derived from a consideration of the energy flows
from the point-excited cylindrical shell (C) to the surroundings (internal I,
external E), (Fig.1). Assume a steady-state source
of power supplied by a vibration source: this power (W) must equal the sum of the power lost thro-
grough internal damping in the walls of the cylinder
(W_{ic}) and the power radiated as sound into the
external and internal surrounding air(W_{ec}, W_{dc})
the power balance of the system C gives: W = W_{ic} + W_{ec} + W_{dc}.
The power W_{E} may be considered as the difference
between the power which the system C transmits to
E (W_{EC}) and the power which the system E transmits
to C (W_{CE}), that is: W_{E} = W_{EC} - W_{CE}. A similar re-
sult holds for the power W_{I}.
The power balance of system I gives: W_{I} = W_{di}, where W_{di} is the power lost insi-
de the cylinder. From these relations and according to the SEA method it is
possible to obtain the average energy density <D> inside the cylinder:

\[ <D> = \frac{<F^2> Re(Y)}{(\rho_c c_o S |\sigma_{rad}|_E + \rho_s S_{wn})\left\{ \frac{V}{\rho_c n_c(\omega)} \frac{1}{S} + \frac{\alpha S_T}{4 \rho_o S \sigma_{rad}|_I} \right\} + \frac{c_o S_{i} \alpha}{4}} \]

EXPERIMENTAL MEASUREMENTS AND CONCLUSIONS
The test cylinder was an acrylic resin cylinder and a miniature electromag-
etic vibrator was used to excite the shell. The exciter was driven by a third-
 octave band white noise. From the comparison between the theoretical and ex-
perimental results of the internal sound pressure level, it is possible to see
that the SEA method seems to give acceptable predictions (+2 dB) only above the
ring frequency of the cylinder.

LIST OF SYMBOLS

\(<F>\) RMS of the point force, Re(Y) real part of the point input admittance, \(\rho\)
medium density, \(c\) sound velocity, \(S\) area of the shell, \(|\sigma_{rad}|_E\) external radiation ratio, \(|\sigma_{rad}|_I\) internal radiation ratio, \(\rho\) mass per unit area, \(\omega\) frequency, \(\eta\)
loss factor, \(V\) volume, \(n_c\) modal density of system C, \(n_i\) modal density of system
I, \(\alpha\) average absorption coefficient inside the cylinder, \(S_T\) total surface area.
Symmetric Response of a Shallow Ring with Pinned Ends
Tufts University, USA USM Corp., USA
Nelson, F. C. Wright, H. L.

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The fundamental symmetric mode of a shallow ring segment involves extension as well as bending. Extensional behavior is important for "thick" ring segments, i.e. rings whose developed length (l) is comparable to the radius of gyration of their cross-section (k); this can be measured by the parameter \( g = (l/k) \). Inextensional behavior occurs for "thin" ring segments, i.e. for \( g \to \infty \).

Also, as the opening angle of the ring segment (\( \alpha \)) increases, the shapes of the symmetric modes change: for example, for small \( \alpha \), the fundamental mode shape has no radial nodes; for larger \( \alpha \), the fundamental mode shape has two radial nodes. Moreover, if \( g \) is very large, this mode shape transition can occur for very small values of \( \alpha \).

This phenomenon of mode transition is studied for a ring segment with pinned ends and \( g = 10^4 \). The fundamental symmetric mode undergoes transition from no radial nodes to two radial nodes between \( \alpha = 10^\circ \) and \( \alpha = 15^\circ \). The second symmetric modes undergoes transition from two radial nodes to no radial nodes between \( \alpha = 15^\circ \) and \( \alpha = 20^\circ \) and transition to four radial nodes between \( \alpha = 40^\circ \) and \( \alpha = 50^\circ \). These transition opening angles will become smaller as \( g \) increases above the value of \( 10^4 \). The above analytical predictions were tested experimentally and mode transition was confirmed.

This study should be helpful to those interested in using flat beam or flat plate mode shapes to investigate the dynamic behavior of shallow, non-thin, curved structures.
DYNAMIC RESPONSE OF A PARALLEL FLOW HEAT EXCHANGER TUBE TO NEAR-FIELD FLOW NOISE

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Dynamic response of a parallel flow heat exchanger tube to turbulence induced pressure fluctuations on its surface is presented, using random vibration theory. Displacement statistics are expressed in terms of the power-spectrum of fluctuating pressure, the tube transfer function and the acceptances of the surface pressure by the mode shape of the tube. The power spectrum of the pressure field is based on pressure measurements on the surface of a cylindrical element in annular fluid flow and a pipe containing turbulent fluid flow. Evaluation of the transfer function requires the knowledge of natural frequency and damping. Consequently, the natural frequency and mechanism of damping are investigated. The acceptance are expressed in terms of special cross-correlation of the pressure field and tube eigen functions. For a homogenous turbulence, the cross-power spectral density of the pressure field over the cylindrical surface is expressed as a modified Corcos model. The natural frequency, damping ratio and r.m.s. tube displacement as a function of mean axial flow velocities have been presented in dimensionless form. The effect of tube to grid hole clearances, relative span lengths, pressures and initial tension on the natural frequency, damping ratio and r.m.s. response is also studied. Results for single flow are compared with the results available in literatures.
A mathematical model has been developed to describe small transverse vibrations of a straight and uniform parallel flow heat exchanger tube, considering the effect of tube as well as shell side flowing fluids and tube to grid hole clearances. The various physical parameters like gravity, pressurization, internal dissipation, tensile effects, etc. have also been taken into account. The model has been used to study the dynamic stability of a three span tube, fixed at the ends. Various spans of the tube are considered to have fixed-pinned, pinned-pinned, fixed-partially pinned, partially pinned-pinned, fixed-partially free and partially free-free end conditions, depending upon the relative grid hole clearances. The critical conditions of stability have been presented in dimensionless form to illustrate the effect of various system parameters on the regions of instability. The effect of relative grid hole clearances and relative span lengths are also briefly discussed.
RECEPTANCES OF FINITE AND SEMI-INFINITE SYSTEMS: NATURAL MODES OF SYSTEMS WITH FREQUENCY DEPENDENT DISSIPATIVE AND ELASTIC BOUNDARIES

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In this paper receptance method has been used to set up general equations that govern the free vibrational response of a finite array of N mono-coupled elements terminating at general boundaries. The general equations have been used (i) to obtain end receptances of the finite array in terms of receptances of its individual elements, and (ii) to introduce frequency dependent boundary conditions offered by semi-infinite periodic arrays attached at its end.

Theory has been applied to study the end receptances and natural frequencies of some simply supported 5-span beams with their ends attached to identical and different semi-infinite beams on periodic simple supports. Both, finite periodic beams with equal support spacings and finite disordered beams with unequal support spacings (with supports arranged symmetrically and unsymmetrically), have been considered.

Natural frequencies of such combinations of beams have been computed and compared to those when the finite beams have ideal end conditions. Variation in natural frequencies of the finite beams, caused by the presence of different disorders and different combinations of ideal and frequency dependent boundary conditions, has been discussed. The conditions under which an undamped combined system can behave like a spring-mass system and a spring-mass-damper system have also been pointed out.

REFERENCES


A NEW THREE-MODE MODEL FOR VIBRATING BARS AND PLATES

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Models which represent the oscillatory behaviour of continua in terms of a finite number of vibration modes have proved to be very useful in engineering analysis and design. Such a model was proposed by Timoshenko (1) for the bending vibrations of prismatic bars. Uflyand (2) and Mindlin (3) developed a corresponding model for the flexural vibrations of plates. Each of these models involves an averaging of the shear stress across the thickness of the bar or plate, but the weighting of this average has been the subject of continual controversy with regard to the criterion to be applied in the calculation of its numerical value.

All the models presume to represent the three lowest order asymmetric modes of vibration, but it was shown by Nelson (4) that a single averaging factor does not provide enough flexibility to ensure an accurate representation of all three. A modification of the Timoshenko model, aimed at overcoming this deficiency was proposed by Bobrovitskii (5), but even this model is incapable of accurately representing all three modes.

In deference to Witham’s (6) view that in problems of wave propagation the differential equation plays a subsidiary role to that of the governing dispersion relation an empirical dispersion relation is constructed so as to fit the various continuum solutions that now exist. For many computational procedures it is convenient to have the relation defined by an algebraic expression rather than as a table of specific values.

Finally it is shown how the preceding models can be modified so as to yield a dispersion relation which satisfactorily defines all three low frequency asymmetric modes.

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(2) Y.S. Uflyand, Akad.nauk SSR, Prikl.Mekh. (1948) 12,287;
(3) R.D. Mindlin, J.Appl.Mech. (1951) 18,31;
The impact of spheres of different materials has been studied with impact velocity as a parameter.

When two spheres collide the time during which the spheres are in contact is much less than the length of the sound pulse. (1) In such a situation the sound is independent of the mass of the spheres or the material. The most important parameter is the diameter, impact velocity excepted. This point is illustrated in Figure 1 where the peak sound pressure level is plotted against the logarithm of the impact velocity, for balls of wood and steel of the same diameter of 0.4 cm. Although the ratio of the mass of the steel ball to that of the wood is greater than 20:1 the effect of the sound pressure level is small. Impact of wood with steel is also included with the group. The sound pressure depends approximately on velocity to the power 1.2, as has been shown previously. (2)

The frequency depends only upon the diameter of the spheres and is independent of the mass.

![Figure 1](image)

References


The Thevenin and Norton Theorems for Random Vibration

U. S. Office of Naval Research

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In solving electrical and mechanical network problems, it is often desirable to make use of the Thevenin and Norton network theorems. Starting with the multiterminal statements of these theorems, this paper derives versions of the theorems suitable for application to problems in random vibration.
IMPULSE RESPONSE IDENTIFICATION OF A STOCHASTICALLY EXCITED STRUCTURE

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The work presented is concerned with the identification of the dynamic characteristics of a plane structure loaded by stationary random dynamic forces perpendicular to the plane of the structure. The mathematical model for the vibration analysis of such type of structure is presented in a discrete linear form. According to the type of loading, out of plane bending and torsion of beam elements of the structure are considered. Their masses are lumped at nodes. The normal mode approach is used to isolate specific modes. For experimental verification in laboratory, few modes are isolated by appropriating the excitation.

The unit impulse response function and its Fourier transform, the frequency response function, are the main characteristics employed when describing linear dynamic systems using their input output relations. The conditional mean assessment enables a direct and simple approach for estimating these characteristics when the system is excited by a stationary random loading process. In general, the conditional mean of the output given the values of the inputs at different time lags is a function of these inputs and the system characteristics.

In the presented paper, the identification problem is stated and its solution using the conditional mean approach is proposed. The conditional mean surface is assessed using input and output records of the tested structure. The estimation of the unit impulse response function using the conditional mean surface is performed. An approximation, when using the linear least square surface instead of the conditional mean surface, is also presented.
RANDOM VIBRATION OF AN IMPACTING OSCILLATOR
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STATISTICS OF THE RESPONSE

The random vibration of the two-degree-of-freedom oscillator shown in Figure 1 is discussed. f represents a Gaussian white-noise force with spectrum level $S$ acting on the mass $m_1$. From an analytic point of view the system is of interest because we can by using the Fokker-Planck equation solve exactly for the joint probability density function $p(x, y, \dot{x}, \dot{y})$, and because, in contrast to a single-degree-of-freedom oscillator the mean-square acceleration of the mass $m_2$, $E[\dot{x}^2]$ remains finite. We can deduce an exact expression for the joint probability density function $p(x, \dot{x}, \dot{x})$. The various constants involved in the expressions and the statistics $E[x^2]$, $E[\dot{x}^2]$, and $E[\ddot{x}^2]$ can be evaluated explicitly in terms of error functions. We note that although $\dot{x}$ is distributed in Gaussian form its derivative $\ddot{x}$ is not, except in limiting linear cases such as when the gap $b$ is zero.

PROBABILITY DENSITY FUNCTION OF PEAKS

The pdf $p(\zeta)$ of positive peaks ($\dot{x}<0$) can be obtained from $p(x, \dot{x}, \ddot{x})$, at least to within a constant. The constant can be evaluated explicitly in the linear case. Typical examples of $p(\zeta)$ are shown in Figure 2. Also shown by dashed lines are the narrow band approximations to $p(\zeta)$ that are based only on $p(x, \dot{x})$.

It is convenient to normalise displacements with respect to the gap width $b$. We have plotted $p(\zeta/b)$ against $\zeta/b$. The non-dimensional parameters that affect the response are $k_2/k_1=1$, in Figure 2), $k_3/k_1=0,1,$ and 10), and $\pi S/c k_1 b^2(=4)$. Similar curves for other parameter values show that agreement between the exact and the narrow band approximation is good only when the response is indeed narrow band.

The most distinctive features of the peak pdf are the presence of positive peaks for negative values of $\zeta$, and the quite abrupt changes at $\zeta=b$ for large values of $k_3$. 

Figure 1.

Figure 2.
MEASUREMENT OF HAND/ARM VIBRATION

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The principles for measurement and evaluation of human exposure to vibration transmitted to the hand have been proposed in a Draft International Standard ISO/DIS 5349. The standard gives valuable guidelines for the measurements in general.

The practical application of the proposed exposure guidelines requires solutions to some measurement problems, which have only been dealt with in general terms in the standard. The reference point for hand/arm vibration measurement is deemed to lie in the head of the third metacarpal. As it is utterly inconvenient to have a transducer drilled into the joints of the hand, alternative practical measurement points have been tested and a preferred method is recommended.

The Draft also deals with crest factors, averaging time and impulses. In order to obtain consistency in measurements, the selection of well-determined parameters is suggested.

An exponential averaging time of 2 seconds is recommended. A definition of "crest factor" as the max. peak measured within a 60 second linear averaged time duration is proposed for this purpose.

The introduction of a standardized artificial hand/arm system is proposed in order to promote practical measurements and objective comparison between potentially dangerous handheld devices.
Method for predicting the development of vibration-induced white finger
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INTRODUCTION

Habitually exposing the hands to intense vibration may lead to recurring episodes of finger blanching and numbness (vibration-induced white finger, VWF), with reduction in manual dexterity, tactile sensitivity and damage to soft tissue subsequently occurring in severe cases. In this paper the parameters controlling the initial development of the syndrome are derived from retrospective studies of workers occupationally exposed to vibration. The desire to interrelate all published reports, the few studies linking vibration exposure to symptoms and the lack of an objective measure of the severity of VWF suggested the following analysis.

METHOD

To establish that symptoms subjectively reported by individuals can be collectively related to vibration exposure, all published studies of groups using a particular tool were compared. It was found that the average duration of employment primarily determines the average time for the first white finger tip to appear (Latent Interval) in groups using similar tools, rather than the detailed nature of the operation, including number and duration of interruptions, and epidemiological factors.

To select symptoms and data representative of the progression of VWF, all studies using Taylor's classification of symptoms (W. Taylor et al. in The Vibration Syndrome, Academic, 1975, pp 121-139) were analysed. By systematically imposing constraints on minimum group size and prevalence of symptoms, it was found that the ratio of the average time to develop stage 3 VWF, which probably contains the transition from mostly reversible to irreversible disorders, to the Latent Interval is close to 3 for each occupation.

To derive a quantitative relation between the stimulus and the average rate of development of symptoms, only studies with data satisfying the selection process above and with vibration measurements were considered. A single numerical measure of vibration amplitude was first formed, by frequency-weighting the available acceleration spectra with the equinoxious contour implied in the ISO proposal for the measurement and evaluation of human exposure to vibration transmitted to the hand. A simple power law then results for occupations involving near-daily exposure to vibration, indicating that both the appearance and progression of VWF occur at a rate determined by the vibration experienced by the hands.

If it is assumed that an individual's susceptibility to VWF remains unchanged during the course of the disorders, it is possible to predict tolerable vibration exposures for the whole population. The vibration limits so derived compare favourably with those proposed for day-long exposure by the ISO.
Effects of Whole-body Transient Vibrations on Human Emotion.

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The effects of whole-body transient vibrations in the horizontal direction on ratings of annoyance was investigated in the present study, in which vibration amplitude, crest factor (peak/rms) (C.F.) and wave form were adopted as the factors affecting the annoyance. Transmitted information in the information theory was applied to the rating to measure the amount of relation between vibration stimuli and human emotional responses. Vibration stimuli used in this experiment were artificially synthetic pulsed vibration (SV), pulsed vibration generated by a pile hammer on the ground (HV) and tractor vibration on the operator’s seat (TV), which generated on the vibration table of an electro-dynamic type. Vibration acceleration levels (VAL) employed were changed from 80 to 105 dB (20 log_{10}a/a_{ref} = a_{rms} vibration acceleration value, a_{ref} = 10^{-5}m/sec^2) at five levels and the crest factors ranged from 4 to 10. Ten male subjects sitting on the vibration table were asked to rate their responses to the stimuli according to the five-point scales (1) not at all annoying, 2) slightly annoying, 3) annoying, 4) moderately annoying, 5) very annoying). Fifty vibration stimuli (5 levels x 10 repititions) on one vibration signal were presented to the subject in random order controlled by a micro-computer. Three kinds of vibration signals were examined separately. Each stimulus was 7 sec long and separated by about 20 sec rest from the next stimulus.

The method of successive categories was applied to obtain the ratings for three kinds of the transient vibrations. The figure shows the mean ratings and the standard deviations, in which the linear relation can be obtained between the ratings and VAL values in the case of the SV. It is significant that the rating values of annoyance increase with increase of VAL on the two values of C.F.(4 and 8). The same tendency shows on the other vibrations of HV and TV. With regard to the effect of the C.F. on the rating, it is found that large values of the C.F. show severe annoyance on both SV and HV. This result comes from the fact that the peak values of the vibrations were varied without changing VAL at different categories of the C.F.. On the other hand, the ratings in TV indicate no clear change regardless of the C.F.. The transmitted information on the three kinds of vibrations increases depending upon the C.F. as shown in the table. The transmitted information which intrinsically hints the correlation function is found to be a helpful measure understanding the relation between the ratings and VAL values. Analytical procedures for the multidimensional scale method were also applied to determine the co-ordinate values of each factor( the levels, the C.F., the vibration direction and the wave form) by deriving eigen values and eigen vectors from correlation matrix. In this case, Rajski’s distance is defined after the information theory.
THE RESPONSE OF HIGH RISET BUILDINGS TO GROUND VIBRATION FROM BLASTING - AN EXPERIMENTAL INVESTIGATION

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The techniques used in the measurement of building response to blast-induced ground vibration are discussed and significant sources of error are explained.

In a recent case study, simultaneous measurements of ground and building vibration during blasting have confirmed the existence of resonance magnification of the building vibration.

It is suggested that current codes which are concerned with the use of explosives may need modification to take account of this effect in connection with human comfort and damage criteria.

The techniques used in the measurement of building response to blast-induced ground vibration are discussed and significant sources of error are explained.

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It is suggested that current codes which are concerned with the use of explosives may need modification to take account of this effect in connection with human comfort and damage criteria.

Les méthodes employées dans la mesure de la réponse des bâtiments aux vibrations du sol, provoquées par des sautages, sont discutées et des sources d'erreurs significatives sont expliquées.

Dans une récente étude des cas particuliers des mesures simultanées des vibrations du sol et du bâtiment ont confirmé l'existence d'une amplification par résonance du bâtiment.

Il est indiqué, qu'une modification des codes en vigueur pour l'emploi des explosifs pourrait être nécessaire en vue de tenir compte de cet effet relatif aux critères du confort humain et des dégâts.

Die Methoden, die in der Messung von Gebäudereaktion auf durch Sprengung verursachte Bodenschwingungen gebraucht werden, sind diskutiert und bedeutende Fehlerquellen werden erklärt.

In einer neulichen Fallstudie haben gleichzeitige Messungen von Boden- und Gebäudeschwingungen bei Sprengungen das Auftreten von Gebäuderesonanzverstärkung gezeigt.

Die Autoren meinen, dass die gültigen Vorschriften für den Gebrauch von Sprengstoffen möglicherweise zu ändern sind, um diesen Effekt im Zusammenhang mit Behaglichkeits- und Beschädigungskriterien zu berücksichtigen.
EXPERIMENTAL AND THEORETICAL STUDY OF VIBRATION SYSTEMS IN THE MINING PROBLEMS

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The report deals with the main systems of the interaction "Man–Machine–Object Being Worked" applied in the mining. An analysis performed by means of the numerical investigations in the processes of free and constrained oscillations in a system is provided and the simulation technique of the vibration systems by analog devices is shown.
Die Verfahren der komplexen Untersuchung und der Wertung der Vibrationseinwirkung

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$$f = \sum_{i=1}^{N} \left(1 - e^{-\frac{T_{2i}}{T_{1i}}} \right) \int_{0}^{T_{1i}} W(t)dt$$

wo
- $f$ - die Bedeutung der "akkumulierten" Vibration;
- $T_{1i}$ - die kennzeichnende Zeit des Rückgangs;
- $T_{2i}$ - die kennzeichnende Zeitperiode, während deren die Vibration wirkt;
- $W(t)$ - die kennzeichnende Zeitperiode, während deren die Vibration nicht vorhanden ist;
- $W(t)$ - die Kennzahl, die die wirkende Vibration (Vibrationsleistung) charakterisiert;
- $N$ - die gesamte Zahl der Zwischenzeiten mit Vorhanden der Vibration sind.

Da die Betriebsvibration in meisten Fällen nicht harmonisch ist und oft schlagartig sein kann, schlagen wir vor, ein Zwischenwert der Vibrationsgeschwindigkeit zwischen dem quadratischen Mittel und Spitzenwert, das von ihnen Funktionell abhängig ist, für die wirkende Bedeutung der Vibration zu halten.

Literatur.

The joint acceptance function describes the coupling between an excitation pressure field and a structure represented by its normal vibration modes, and is conventionally defined as:

$$j_r^2(\omega) = \frac{1}{A^2 S_{pX_0}(\omega)} \int \int \frac{Cp(\bar{x}, \bar{x}'; \omega)}{\bar{x} \bar{x}'} \psi^r(\bar{x}) \psi^r(\bar{x}') \, d\bar{x} d\bar{x}'$$

where $A$ is the panel area, $S_{pX_0}(\omega)$ is the pressure power spectral density at $X_0$, $Cp(\bar{x}, \bar{x}'; \omega)$ is the co-spectrum of the pressure field, and $\psi^r(\bar{x})$ is the $r$th structural mode shape.

Joint acceptance functions for homogeneous excitations have been investigated in detail elsewhere. Here the joint acceptance for a non-homogeneous excitation is considered where the excitation is represented as a convecting pressure field with pressure amplitude decaying with distance from the peak level location $X_0$ and with spatially-decaying coherence. Exponential forms for both amplitude and coherence decays are modeled for simplicity. Thus,

$$Cp(\bar{x}, \bar{x}'; \omega) = e^{-a|x-\bar{x}|} e^{-a|x'-\bar{x}|} e^{-c|x'-\bar{x}|} \cos[k(x'-\bar{x})].$$

where $a$ is the amplitude decay rate, $c$ is the coherence decay rate and $k$ is the excitation wavenumber at frequency.

Figure 1 illustrates the one-dimensional system where a beam is driven in its $m$th mode shape by a non-homogeneous pressure field. $X_0$ is chosen for simplicity to correspond to the beam end, i.e., $X_0=0$, so that using sinusoidal mode shapes, the joint acceptance becomes

$$j_{mm}^2(\omega) = \frac{1}{L^2} \int_0^L e^{-ax} \sin k_m x \, dx \int_0^L e^{-ax'} e^{-c|x'|} \cos k \xi \sin k_m x' \, dx'$$

where $k_m$ is the beam wavenumber and $\xi = x' - x$. The exact solution is tedious but straightforward.

Figure 2 illustrates the effects of variations in $a_x$ on $j_{mm}^2$ for $m=4$ when $c_x=0$. Maximum and minimum values of $j_{mm}^2$ occur for $a_x$ and $c_x$ both zero, corresponding to plane waves propagating over the beam length with no amplitude decay. However, as $a_x$ takes on increasingly higher values, the joint acceptance depends less strongly on the matching between structural and acoustic wavelengths: the strong oscillations in $j^2$, as $k_x/k_m$ varies, are damped by the presence of significant spatial amplitude decay. Clearly, neglect of this spatial amplitude decay may lead to large errors in response predictions. This approach has been used to model the coupling of an airplane fuselage to propeller noise excitation.*

On the statistics of acoustic power flow through structures: tonal excitation
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In calculation of the acoustic power flow into a cavity through a transmitting structure, recourse is routinely had to statistical energy analysis. The accuracy of such predictions is then of interest. One approach for estimating the prediction error bounds for tonal noise transmission is presented here. At low frequencies tonal noise transmission depends intimately on precise modal details, as expressed by modal admittance functions for transmitting structure and receiving cavity. Now for acoustic modes with resonance frequencies $\omega_n$ and loss factors $\eta_n$, the modal admittance function for tonal excitation at frequency $\omega_b$ is

$$g(\omega_b, \omega_n) = [(1 - (\omega_b/\omega_n)^2)^2 + \eta_n]^{-1}$$

As the modal overlap increases, one seeks an estimate for the band-averaged modal admittance function

$$E[G(\omega_b, \omega_n)] = E \sum_{n \in \Delta \omega} g(\omega_b, \omega_n)$$

for fixed values of $\omega_n$ as the tone $\omega_b$ samples in a narrow frequency band $\Delta \omega$. This is equivalent to the expected value of $G$ for fixed $\omega_b$, where the $\omega_n$'s are distributed in frequency according to their distribution functions for an ensemble of slightly differing cavities. It can be shown that, for $\eta_n<1$,

$$E[G(\omega_b, \omega_n)] = N_n(\omega_b)\pi \omega_b/2 \bar{n}_n$$

where $\bar{n}_n$ is the band-averaged acoustic loss factor and $N_n$ the acoustic modal density. The variance of $G$ can be derived as

$$\sigma^2[G(\omega_b, \omega_n)] = N_n(\omega_b)\pi \omega_b^3/8 \bar{n}_n^3$$

Thus, the normalized standard error in band-averaged acoustic admittance is found as

$$\epsilon_r = \sigma[G]/E[G] = [2\pi \bar{n}_n N_n(\omega_b)\omega_b]^{-1/2}$$

where $\bar{n}_n N_n(\omega_b)\omega_b$ is the modal overlap term for the acoustic cavity; i.e., the standard error decreases as acoustic modal density and loss factor increase. Similar analysis can be applied to the response of the transmitting structure; then, for high modal overlaps, the normalized standard error in the acoustic power flow associated with both structural and acoustic modeling is

$$\epsilon_r = [2\pi \bar{n}_n N_p(\omega_b)\omega_b]^{-1/2} + [2\pi \bar{n}_n N_n(\omega_b)\omega_b]^{-1/2}$$

where $\bar{n}_n$ and $N_p(\omega_b)$ are the band-average loss factor and modal density of the transmitting structure at frequency $\omega_b$. The figure shows calculated values of upper and lower bound 2$\sigma$ limits for 95% confidence in calculations of tonal power flow into a closed cylinder at frequencies where several acoustic modes exist in a narrow frequency band. The error limits should be interpreted to mean that the actual value of the power flow is expected to be within the noted confidence limits 95% of the time such a check is made. It is recommended that an approach such as this be used to qualify predictions based on SEA techniques, particularly when less than 5 modes exist in the analysis band.
ACOUSTIC EMISSION SOURCE IDENTIFICATION

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PROBLEM

The phenomenon of stress wave release from microstructural alterations induced in solid materials, known as acoustic emission, has been the subject of an ever increasing number of scientific investigations and technological applications for 25 years. Nevertheless, acoustic emission monitoring has not optimally fulfilled its promise as a nondestructive testing technique since the precise characteristics of the stress waves emitted from specific defects remain unknown. Prior to recent research efforts by the present author, his colleagues at Johns Hopkins, and his associates at the U.S. National Bureau of Standards, no reliable method had been developed to solve the acoustic emission source characterization problem. In fact, numerous investigators have gone on record in the published literature as calling the problem "unsolvable".

SOLUTION

The main reason for such a pessimistic view is that there are several difficulties which must be overcome in order to reliably characterize acoustic emission signals. First, a detector must be used which is capable of sensing surface displacements due to both internal and surface acoustic emission sources and which does not modify the signal because of its own limitations. Although they have been used in the past, and are still the detectors most often used in current practice, conventional piezoelectric transducers are completely unsuited for such measurements since they have their own amplitude, frequency, and directional response. In addition, they "ring" at their resonance frequency and it is impossible to distinguish the amplitude variations actually characteristic of the acoustic emission source from the amplitude excursions caused by this "ringing". The second difficulty is to accurately determine the acoustical transfer characteristics of the workpiece, since even with an ideal detector of surface displacements, the question still remains as to how these surface displacements relate to the displacements of the material at the internal or distant surface source of the acoustic emission signals. The results presented in the present paper illustrate that the problem with conventional piezoelectric transducers can be completely overcome by the use of either capacitive or optical detectors. A superior high speed digital capture and signal processing system completely eliminated usage of the commonly used signal capture/processing system incorporating a video tape recorder. This superior system permits real time capture and storage of acoustic emission signals on a magnetic disc, from which the signal is transferred to a digital computer for processing. The present work also describes how proper choice of workpiece geometry coupled with precise analytical calculations yields new insight into the determination of the transfer function. Several other new features are reported, which coupled with those already mentioned, indicate that the acoustic emission source identification problem can be solved.
ACOUSTIC EMISSION DURING CONTROLLED CRACK GROWTH IN ALUMINIUM ALLOYS.

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Acoustic emission (AE) is a rapidly developing non-destructive inspection technique which is being increasingly used to characterize flaws in materials. In this paper, we describe AE generated during plastic deformation and controlled crack growth in aluminium alloys. Measurements were made using the double torsion specimen geometry and a range of AE parameters was monitored.

Meaningful interpretation of AE measurements for flaw characterization requires considerable understanding of the processes of generation, propagation and detection of elastic waves. Calibration procedures can be used to assist in the analysis of the wave propagation and detection processes. However, the generation of elastic waves may be associated with a variety of events, such as slip-band formation, particle rupture and crack growth. The relative significance of each event is related not only to the applied stress but also to the micro-structure of the material, and these factors are considered in the evaluation of the present results.
Acoustic Emission from Insulators prior to Electric Breakdown

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An acoustic emission from insulating materials under electric field is observed far prior to the electric breakdown. The breakdown testing arrangement is made in the usual manner, in which an insulating sheet under test is placed between the electrodes. The electrode on the higher voltage side is a circular evaporated cupper film with a guard ring or a spherical steel ball electrode with epoxi adhesive filling provided. The insulator is acoustically coupled to the piezoelectric transducer (resonant frequency: 200 KHz) through an aluminum rod waveguide by way of a thin polyethylene polyphthalate film. The transfered electrical signal is led to the electronic counter by way of the preamplifier and band-pass filter (10KHz - 3MHz). The counted digital signal is then converted into the analog by means of the D/A converter to record.

Fig.1 shows an observed example. Acoustic emission rate, ionization current rate, DC current and applied DC voltage are recorded. The insulator under test is bakelite (thickness:2mm, diameter of the circular electrode: 1.7cm). The applied voltage is increased at the rate of 400 V/min. The acoustic emission was an impulsive burst. The event began at 4 KV, observed on the oscilloscope. The range of the electronic counter is manually reset in the course of the experiment (the multiplication factor is shown in the figure). The so-called ionization current is seen associated, which is known to be utilized for the nondestructive testing. The present phenomenon can provide an alternative technique.
Acoustic emission in concrete and other materials has proved a useful tool in diagnosing and predicting overstressing and failure. This work describes a preliminary investigation of the emission from prestressed concrete beams. Measurements were made at the onset of stressing and followed the history of the beam during post stressing curing, storage, transport and emplacement in a viaduct. The levels of emission were compared with samples of the concrete under compressive loading.
VIBRATION ISOLATION EFFICIENCY LIMITATIONS IN PRACTICE DUE TO AIR VISCOSITY AND ASSOCIATED DYNAMIC IMPEDEANCES

KROLL

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The literature and untold numbers of installations abound in examples of vibration isolation. The classical application is straightforward: account for the driving unit's energy, mass and frequency characteristics in terms of acceptable noise and vibration levels in the driven unit, whether floor, wall or entire structure with respect to the associated mass and stiffness. The cookbook solution usually works but rarely, if ever, provides for the stiffness of the air column involved. We present a case where this factor was predominant and suggest that the air or other surrounding medium can often be a problem manifest in reduced isolation efficiency and how it should be considered.
OPTIMISATION ET MISE EN ŒUVRE DES REVÊTEMENTS ANTI-VIBRATOIRES (POUTRES ET PLAQUES).

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INTRODUCTION.

Pour atténuer les vibrations et diminuer ainsi le bruit émis, il est d'usage de recouvrir les structures vibrantes par un matériau visco-élastique. Les études fondamentales de l'effet de ces revêtements anti-vibratoires ont été faites sur des poutres et des plaques en vibration transversale. Le but de ce travail est de faire le point de l'optimisation de ces matériaux anti-vibratoires en comportement linéaire ou non.

COMPORTEMENT LINEAIRE.

Les études théoriques et expérimentales ont montré tout d'abord que le procédé "sandwich", qui consiste à insérer une plaque de matériau visco-élastique entre la plaque vibrante métallique et une autre plaque mince métallique était beaucoup plus performant que le procédé dit à "une simple couche". Il n'est pas nécessaire d'utiliser un matériau à très fort amortissement car l'effet est brusque et les épaisseurs de la couche visco-élastique et de la plaque supérieure jouent un très grand rôle dans l'amortissement global. Pour des raisons économiques, le traitement peut être seulement partiel et l'efficacité du matériau amortissant reste cependant très convaincante. Selon les conditions aux limites, l'endroit où l'on met le matériau visco-élastique est très important. Il est remarquable que l'efficacité ne se limite pas à une fréquence particulière donnée. Enfin, en revêtements partiel et total, il y a une valeur optimale du module de cisaillement du matériau par rapport au module d'élasticité de la structure.

COMPORTEMENT NON LINEAIRE.

Si le matériau a une loi de comportement non linéaire, un calcul, basé sur la méthode du petit paramètre, montre que l'amortissement global maximum est obtenu lorsque les coefficients de proportionnalité du carré de la vitesse de la déformation sont grands par rapport à ceux de la déformation et ceci pour les modes de vibration graves. Pour les modes supérieurs, c'est le produit de ces coefficients qui est prépondérant.

CONCLUSION.

Les études théoriques et expérimentales citées ci-dessus ont montré que les caractéristiques géométriques et mécaniques des matériaux anti-vibratoires jouaient un rôle important dans l'amortissement global espéré en utilisant des matériaux sandwich.

NOTA:

La place restreinte ne nous a pas permis de citer ici les nombreuses références bibliographiques relatives à ce travail.
Impact noise radiated by bridge structures for elevated rapid transit systems is causing community annoyance in many countries. In some cases, acoustical enclosures were built around the bridges to reduce the noise radiated from the structure.

A cost effective method will be described where theoretical prediction for partial damping treatment of steel box girder type elevated bridges with a concrete deck have been developed. The most efficient vibration damping method for this application was a shear type damping material which was applied in calculated optimum size rectangles. This partial damping treatment was applied to the areas of maximum bending for the different modes of vibration between 20 and 200 Hz., which were the most annoying noise frequency bands generated by the structure.

This report will describe the theoretical studies, scale model tests, and actual field tests on a one mile long steel box girder elevated structure, with and without damping treatment. Actual noise levels with steel wheels at different speeds will be given and the noise reduction illustrated for the damped structure.
Design of a Vibration Isolator for Portable Vibrating Tools and the Effect of the Human Hand on Its Attenuation

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With the prevalence of portable vibrating tools such as a chipping-hammer and a grinder, incidence of vibration diseases (Raynaud's phenomenon) has roused the industrial health problems concerning vibration isolation. A simple low-pass filter (LPF) consisting of a mass and a capacitance was applied to a vibration isolator for portable vibrating tools. The characteristic of the isolator was analysed by the transmission circuit theory using mass-capacitance corresponding calculating by a mini-computer. The portable vibrating tools as the vibration sources were classified into two types such as the constant force source and the constant velocity source. The former is, for instance, a grinder and the latter is a rock-drill. The tool was simulated into the mass and the mechanical resistance.

Then, to simulate the human hand into the equivalent electric circuit, the mechanical impedance was measured separately for the compression and the shearing vibration on the aluminum bar. Two kinds of hand impedances were derived from both vibrations. To both vibrations, the same simple electric model could be induced except that the numerical values of its elements were different for the two hand impedances. After the LPF designed above was terminated by the equivalent circuits of the tool and the human hand, the insertion losses of the system were calculated for both vibration sources and both hand impedances.

The equation of insertion loss includes three terms which mean the attenuation, the reflection of mismatch, and the interaction of reflection. They were first calculated on the LPF without the mechanical resistance for both hand impedances. On the constant vibration force source ($i_1 = $ constant), the reflection of mismatch shows infinity at $\Omega = 1$, while the interaction of reflection does infinitesimal at $\Omega = 1$ in the attenuation curve. The singularity in the total insertion loss, therefore, is cancelled out, but the resonance (notch of the attenuation curve) appear in $\Omega > 1$ depending upon the hand impedance. The tendency of the insertion loss curve for both hand impedances is similar with each other. For the constant vibration velocity source ($e_1 = $ constant), the insertion loss was also calculated by the same procedure. The reflection of mismatch and the interaction of reflection only show the finite values at near $\Omega = 0.7$ instead of $\Omega = 1$ in $i_1 = $ constant. The total insertion loss indicates the resonance (notch in the attenuation curve) at about $\Omega = 0.7$ different from the hand impedances. The attenuation slope above $\Omega > 2$ is similar in both cases of keeping $e_1$ or $i_1$ constant regardless of the hand impedances.

When the spring of the LPF has the mechanical resistance, the insertion loss curves denote the resonance in $\Omega < 1$ in both cases of $e_1$ and $i_1$ constant. The attenuation slope gradually decreases in proportion to increase of the numerical value of the mechanical resistance included in the spring of the LPF as we know in the basic vibration theory. Finally, the human hand can be simulated by the equivalent dynamic mass of the hand in the low frequency range below $\Omega > 2$.

These results are helpful to design the isolator for the portable vibrating tools.
NOISE REDUCTION AND SOUND RADIATION EFFICIENCY OF A RECTANGULAR ENCLOSURE

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Statistical energy analysis is used to predict the internal sound pressure and the external of a rectangular enclosure with five flexible panels, when the enclosure is excited by the structure-borne sound. Then the measured values of the spatial average of the time-mean-square velocity on the panels are used to obtain those sound pressures.

As for the external sound pressure of the enclosure, it is shown that the external sound pressure near the panels can be computed as the sum of the radiation due to the flexural vibration and non-resonant transmission of the panel, if the reception room is assumed the free space or the anechoic room.

The noise reduction produced by a rectangular enclosure can be obtained from the ratio of the internal sound pressure to the external.

We compute the ratio of the non-resonant transmission power from the test panel to the radiation power from the rigid panel (such as the piston) with the same area, and introduce the apparent sound radiation efficiency, that is, the sum of the sound radiation efficiency and this ratio.

For the noise reduction and the apparent radiation efficiency, experimental results are compared with theoretical predictions with generally good agreement.

Figure shows noise reduction produced by a rectangular box with five flexible panels in exciting by the structure-borne sound. This test box has outside dimensions 900x600x600 mm, and is made from 0.8 mm thick five steel panels bolted at their edges to angle steels of dimensions 30x30x3 mm, with a bolt spacing of 150 mm.