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PHYSIO ET PSYCHO-ACOUSTIQUE
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PHYSIOLOGISCHE
UND PSYCHOLOGISCHE AKUSTIK
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Divers
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ACOUSTIC TRAFFIC SIGNAL FOR BLIND PEDESTRIANS

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INTRODUCTION

For many years acoustic traffic signals have been used to facilitate the situation for blind pedestrians. In Scandinavia, Sweden is the most well-equipped country as more than 1000 crossings are provided with an acoustic device. The device, which is often called "the Swedish ticker", gives a fast ticking during the WALK interval and a slow ticking during the WAIT interval. The present paper describes a device which may be taken as a modern substitute for the ticker. The acoustic traffic signal consists of a weak attention signal which is audible only in a distance of 3–5 m from the traffic light pole. This weak signal may be combined with a louder guiding signal which guides the blind across the crossing.

GENERAL REQUIREMENTS

Blind pedestrians are faced with various problems in connection with pedestrian-crossings:
- locate the crossing
- locate a possible pushbutton
- determine the direction of the crossing while waiting for WALK.
  (Many crossings are not perpendicular to the pavement)
- identify the beginning of the WALK interval
- keep the direction during crossing
- get informed about and locate a possible island (refuge).

The sound from the acoustic traffic signal shall preferably comply with the following requirements:
- easily localized
- easily distinguished from other sounds in the environment, especially traffic noise
- easily heard also by elderly people with a hearing impairment
- minimal annoying effect on other pedestrians, occupants, shop-keepers etc.
- highly attenuated by windows.

Furthermore the system should be reliable and self-explanatory to the user. Various investigations concerning acoustic traffic signals are described in ref. 1.
POULSEN, T.: Acoustic Traffic Signal

WEAK ATTENTION SIGNAL

The basic signal in the system is a 880 Hz square wave (or sawtooth) which is switched on and off in different rhythms for WAIT and WALK, see fig. 1.

![Diagram of WALK and WAIT signals](image1)

The WALK signal consists of a 200 ms pulse repeated each 400 ms. The WAIT signal is a 400 ms pulse repeated every 2 second.

The background noise levels controls an automatic volume control which keeps the signal at a level just above the background noise. In this way the signal will remain audible, but at the same time as soft as possible. The range for the automatic volume control is minimum 50 dB.

The reaction time for the automatic volume control is approx. 1 s for increase in level and 3-6 s for decrease in level. This reaction time is relatively fast, but listening tests have shown that it is important to be able to follow, for example, the noise from the engine of a starting bus. If a reaction time of several minutes was used, the signal would be either inaudible in such critical situations or too loud in more quiet periods.

![Diagram of sound pulse sequence](image2)

Measuring 1.5 m above ground, 0.3 m from the pole equipped with the acoustic device, the A-weighted sound pressure level (time weighting F) shall increase 10-12 dB above the background noise during WAIT.

The direction of the crossing is felt by touching a 10-15 cm long bar placed at the top of the acoustic device. See fig. 3.

A knob at the end of the bar points towards the crossing. If the crossing is divided into two by an island in the middle of the street, this will be indicated by two knobs.
FIELD TEST OF WEAK ATTENTION SIGNAL

The usability of the weak attention signal has been investigated by twenty-six blind people. They were asked to walk along the right-hand pavement towards the crossing, locate the crossing, cross the heavily used street (with an island in the middle), turn to the right and cross the side street and continue one the left-hand side pavement until they were stopped by the investigation. Thereafter they returned along the same route. The blind persons were observed during their walk and were interviewed both before and after the test walk. The main results are:

- the WALK and WAIT sounds are easily distinguished from each other
- some guidance was achieved during crossing from the attention signal at the far end
- tactile information regarding 'next stop' at the island or pavement was sufficient
- no guidance effect of the direction marking could be demonstrated.

The weak attention signal is being standardised in Denmark.

LOUD GUIDING SIGNAL

The main objective of this signal is to help the blind pedestrian to keep a proper walking direction during crossing. The sound - which is louder than the attention signal - is transmitted from the far end of the crossing after request (pushbutton). The guiding signal is essentially the same sound as the weak attention signal. In order to enable pedestrians to use the guiding signal in both directions at the same time it has been suggested to transmit the sound alternately from either end of the crossing. See fig. 4.
As for the attention signal the level of the guiding signal shall be controlled by an automatic volume control. In order to minimize the annoying effect on the environment the guiding signal should last only for the time usually used for crossing. Thereafter the level should decrease to the level of the attention signal. See fig. 5.

Fig. 5
Level change after request.
A: Level of attention signal
G: Level of guiding signal

FIELD TEST OF GUIDING SIGNAL

A full-scale test of the guiding signal has been performed at a difficult crossing in Copenhagen. 22 blind individuals participated in the test. The main results are

- the guidance effect is convincing for the users who reported a feeling of increased safety
- the alternating sound seems to be confusing for some users especially due to the loud sound from behind. Sound from the far end only is preferred
- despite the automatic volume control the level in the night-time seems to be too high
- the reaction time of the automatic volume control for the guiding signal is probably too fast. When the level decrease, some users feel it as if the sound source moves away. An increased time constant or a constant level may be better.

CONCLUSION

The weak attention signal has proven to be satisfactory. The system is being standardised in Denmark. The loud guiding signal is satisfactory according to the guidance effect. The alternating sound transmission may be given up seen in relation to the likelihood of two blind people using the crossing in opposite directions at the same time. More full-scale tests are necessary for the guiding signal.

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3.1

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Schallübertragung - Schallwandler Neurophysiologie
Klaus Genuit  Analytic Description of Average ...

ANALYTIC DESCRIPTION OF AVERAGE OUTER EAR TRANSFER FUNCTIONS IN DEPENDENCY ON DIRECTION OF SOUND INCIDENCE

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1. Introduction

Measuring the outer ear transfer functions on persons differences are recognizable by the amplitude and phase. The cause for these declining are an incorrect position of the probe microphone in the earcanal and the individual incorrelate deviations of the acoustical relevant structure of persons. But special significant structures are measured on all person in cause by the similar exterior geometrical arrangement. A tentative procedure to describe averaged outer ear transfer functions by a special structure average method /1/ causes a conception to describe the principle structure by an analysis on those parameters influencing the human outer ear transfer functions. The aim is to simulate the outer ear in a relative simple model on the premises of free field sound propagation and frequencies below 10 kHz and to get averaged transfer functions which consider the typical structure.

2. Analysis of outer ear transfer functions

On the development of a new artificial head /2/ significant influences on the directional characteristic could be demonstrated by the shoulder and the upper part of the body. So the term "outer ear" means a summary of all acoustical relevant parameters being composed of the upper part of the body, shoulder, head, pinna, earcanal and eardrum impedance. Fig. 1 shows the amplitude of the transfer function from the left human ear at incidence of sound from the median plane in front of the person, measured with a special probe microphone 4 mm inwards the earcanal (reference plane BE in fig. 2). The curve in fig.1 demonstrate the different acoustical efficient structures of the outer ear. In accordance with the geometrical dimensions it is possible to correlate each part of the transfer function with one part of the outer ear. The reinforcement of 3 dB on low frequencies (300 Hz) arises from a damped reflection caused by the upper part of the body. The shoulder causes the reinforcement of 3-4 dB at 800 Hz and to the attenuation of 3 dB at
1060 Hz. The influence of body and shoulder ranges from 100 Hz to about 2 kHz. The dependency of sound approach direction is not so important in the horizontal plane but especially in the median plane. Here the sound pressure at the earcanal entrance consists of the direct wave and the delayed damped reflection by the shoulder surface. This causes the reinforcement and attenuation in the transfer function. In dependency on the elevation of the sound source the time delay between the direct and reflected wave increased in such a manner that the influence of the shoulder shifts to lower frequencies. The pinna causes a wide reinforcement of the spectrum from 1.2 kHz to 14 kHz with a theoretical maximum of 19 dB at about 4.2 kHz and some resonances at 2 kHz and 5.3 kHz evoked by changes of diameters (look at Fig. 2). The complex design of the human pinna in Fig. 2 is not so important, especially at frequencies below 10 kHz - in this range the wave length is great in comparison with the cavum conchae. Therefore it is possible to approximate the transfer function by an easier geometrical model consisting of three pure cylinders with different diameters and length. The transfer function measured at such a simple pinna model and a mathematical estimation give a good agreement with the transfer function measured at an exact replica of a ear (Fig. 3). But it is important for the directional characteristic in the median plane that these cylinders are unsymmetrical mounted into one another and the earcanal entrance is positioned not centric. This and the opening of the pinna to the frontal plane and its inclination angle are the important facts for localisation of sound sources in the median plane. Especially a reflection inside the cavum conchae causes a significant dependency of direction. With sound source in front of a person the addition of direct incident sound with the delayed reflected sound results to a minimum at 8.6 kHz. This minimum corresponds to the "direction determinative bands" /3/ (the frequency range about 8.2 kHz is an "up-band"). While elevating the sound source this minimum moves to higher frequencies (approximately 12.5 kHz). In addition to the shoulder reflection this cavum conchae reflection is very important for the localisation of sound sources in the median plane and it is well-known that for localisation of sound sources wide-band high-frequency signals are needed /3/. The earcanal-resonances complete the structure of the outer ear transfer function in Fig. 1 Since the eardrum impedance is complex and a function of frequency the minima at 3.2 kHz, 10.6 kHz and 15 kHz do not correspond exact to the \((2n+1)\lambda/4\) order. But a mathematical verify with an averaged eardrum impedance gives a good agreement. The wide attenuation of the transfer function at about 8 to 11 kHz results from a superposition of the cavum conchae influence (reflection) and the \(3\lambda/4\) earcanal resonance. The earcanal and the eardrum impedance have no influence of the outer ear directional characteristic. The structure of the directional characteristic in the horizontal plane is remarkable determined by the head. On the side turned to the sound source the head surface works as a reflector in dependency on direction and fre-
Klaus Genuit  Analytic Description of Average

frequency. Since the ear position is a little bit displaced behind the centre only a 6-8 dB (instead of 10 dB) reinforcement is measured about the frequency range from 1 kHz to 6 kHz (the virtual sound source from the surface is located in the centre), on higher frequency the amplitude of the transfer function decreases (measured on an artificial head without pinna). On the side turned away the sound source the head works like a 6 dB/octav low-pass filter with a cut off frequency at 1.8 kHz and a "comb" filter caused by the time delay between both sound waves around the head. In consequence of the excentric ear position the character of the "comb" filter differs typical between sound incidence from the front and from behind.

3. Outer ear simulation

The outer ear transfer function can be approximated in a good approach with a model like fig. 4 at frequencies below 10 kHz and one sound source in the free field. This model considers the structures of the human outer ear. The main time delay $T_0(\phi)$ simulates the different distances of the both ears to the sound source in dependency of the azimuth angle $\phi$. The acoustical effect of the head is simulated with an high-pass $(H_{HP}(f,\phi); f_g \geq 800$ Hz) / low-pass $(H_{LP}(f,\phi); 1.9$ kHz $< f_g < 10$ kHz) combination for the reflection and a time-delay ($T_1(\phi)$) and another low-pass $(H_{TP2}(f,\phi); f_g \geq 1.8$ kHz) for the diffraction. The time delays $T_2(\delta)$ and $T_3(\delta)$ represented the influence of the shoulder and the cavum conchae reflection in dependency on the elevation angle $\delta$. The transfer functions $H_1(f)$ (similar a band-pass) and $H_2(f)$ (similar a high-pass) approximate the measured reflections from the shoulder and cavum conchae in dependency on the frequency. The pinna is realized by a band-pass system $(BP(f,\phi))$ with three ranges of reinforcement whose band-with and amplification depend on the azimuth angle $\phi$. The high-pass $(H_{HP}(f,\phi,\delta); f_g > 3.5$ kHz) simulates the damping effect of the pinna. The influence of the ear canal and eardrum impedance is approximated by the time delay ($T(1) = \text{carcanal length}$) and $H(f,T_{EAR}(f))$ which considers the measured transfer function of the ear canal. The simulation of the upper part of the body could be omitted since its influence upon the outer ear transfer functions is negligible. The outputs of the outer ear simulation approximate the averaged sound pressure $p_l(t)$ and $p_r(t)$ in the reference plane (BE in fig. 2) of the left and right ear canal entrance.

4. Remarks

There are three significant characteristics for spatial hearing: 1. the left/right discrimination results from the interaural differences: time delay $0< t < 0.71$ ms, amplitude differences caused by reinforcement of the spectrum of the one side and low-pass damped on the other; 2. the front/behind discrimination results from the not centric ear position; 3. the elevation results from the form and inclination of the pinna and the shoulder- and cavum conchae reflections. These structures - described in fig. 4 - exist at all persons. The measured transfer functions of indi-
Klaus Genuit Analytic Description of Average...

Individual human ears can be approximated with a computer considering the geometrical dimensions like: head-diameter, ear position and inclination, cavum conchae dimensions, ear canal length and distance ear canal entrance/shoulder. The average above these parameters in common with the averaged eardrum impedance is the base for a special structure average method of outer ear transfer functions.

1/ Genuit, Platte: Überlegungen zur Substitution des natürlichen Außenhores durch elektroakustische Mittel. DAGA München 1980


**Fig. 1** Free field outer ear transfer function
direction of sound incidence $\phi = 0^\circ$, $\delta = 0^\circ$

**Fig. 2** Low part pinna

**Fig. 3** Transfer function at BE:
- --- replica of a human ear
- - - approximation with cylinders
- - - mathematical approximation

BE reference plane

**Fig. 4** Chemical diagram, outer ear simulator, $s(t)$ = input signal, $p_l(t)$ and $p_r(t)$ = sound pressure at the ear canal entrance of the left and right ear
UN MODELE NUMERIQUE DU REFLEXE STAPEDIEN

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INTRODUCTION

Le but de cet exposé est de présenter un modèle mathématique simple de l'oreille moyenne et du réflexe stapédienn dans le cadre du fonctionnement d'ensemble de l'organe auditif. Le muscle stapédienn, fixé à l'étirier et à la paroi de l'oreille moyenne se contracte par réaction nerveuse lorsque le stimulus sonore est supérieur à environ 75 dB. Cette contraction atténue les oscillations de l'étirier, diminuant de ce fait la sensibilité de l'organe auditif pour les basses fréquences. L'atténuation maximale atteint environ 12 dB lorsque le stimulus sonore est de l'ordre de 110 dB. Au-delà de 2 kHz, une légère amplification est notée, comme l'illustre la Fig. 1b. En présence d'un son intense apparaissant soudainement, la contraction maximale n'est atteinte qu'après un certain délai variant de 50 à 300 milliseconds. Cet effet correspond approximativement à un filtrage passe-bas avec une fréquence de coupure de quelques Hertz.

1. DESCRIPTION DU MODELE

Le réflexe stapédienn se présente chez l'homme comme un système à contre-réaction englobant l'oreille interne et les cellules nerveuses (Fig. 2). L'ensemble du mécanisme comprend trois éléments : deux dans la chaîne de contre-réaction : un comparateur de seuil et un filtre passe-bas et le troisième, le muscle stapédienn, dans la chaîne directe.

Le modèle mathématique utilisé pour l'oreille moyenne (Flanagan) est décrit par l'E.q. (1) sous forme d'une fonction de transfert spectrale (mouvement de l'étirier rapporté à la pression au tympan) :

\[ G(a) = \frac{C}{(1+a/a)(a^2 + 2a_s + \alpha^2)} \]

Dans l'Eq. (1), s est la variable complexe de Laplace; le rapport des fréquences de coupure \( a/\alpha \) est égal à \( 5^{-1} \). Ce rapport est aussi égal au coefficient de résonance du trinôme. Le trinôme de l'Eq. (1) représente exactement la fonction de transfert d'un ressort. On est donc conduit à décrire le mouvement des osselets par l'Eq. (2) :

\[ \frac{d^2x(t)}{dt^2} + F \frac{dx(t)}{dt} + K x(t) = C p(t) \]
L'Éq. (2) est écrite pour l'unité de masse : $x(t)$ est le mouvement de l'articulation, $F$ le coefficient de frottement, $K$ la tension du muscle stapédien, tandis que le terme $Cp(t)$ représente la force directement appliquée par le tympan au système d’osselets vibrants. Une atténuation de 12 dB peut être obtenue simplement en quadruplant la tension du muscle stapédien, c'est-à-dire en doublant la fréquence de coupure $f_\alpha = \alpha / 2\pi$, dont la valeur au repos est voisine de 1,5 kHz. Le coefficient de frottement $F$ (c'est-à-dire la pulsation $\alpha$) est supposé constant, cet effet entraîne une augmentation de 6 dB de la résonance sur une fréquence voisine de 3 kHz. L'équation différentielle complète décrivant l'action de l'oreille moyenne, muscle stapédien inclus, est obtenue en rendant la tension $K (= \alpha^2)$ de l'Éq.(2) variable avec le temps.

(a) type "ressort" pour une atténuation de 0, 6 et 12 dB;

(b) réponse de l'oreille moyenne obtenue à partir de la Fig. 1 $a$; en pointillés : courbe expérimentale relevée par Moller$^1$.

Figure 1 : Réponses spectrales

Figure 2 : Schéma fonctionnel du réflexe stapédien
2. RESOLUTION NUMERIQUE ET RESULTATS

Pour simuler la détection nerveuse au sein de l'oreille interne, on a choisi le modèle simple de Flanagan\textsuperscript{2}. La résolution numérique de l'Eq. (2) a été effectuée au moyen de la méthode classique de Runge-Kutta d'ordre 4. Les constantes numériques du modèle ont été ajustées pour un son pur à 1000 Hz de sorte que la fenêtre de détection corresponde aux niveaux de pression sonore compris entre 80 et 106 dB à l'entrée du canal auditif. La concordance du modèle avec les résultats expérimentaux est bonne. La Fig. 3 présente la réponse de l'étier pour quelques stimuli sonores caractéristiques.

A. EFFET SUR LES BRUITS IMPULSIFS PROVOQUES PAR LES ARMES

Le modèle proposé a été appliqué à l'ensemble des impulsions ayant servi d'échantillon pour l'analyse statistique de la gêne auditive dans Stevin\textsuperscript{2}. L'introduction du réflexe stapédién entraîne une réduction moyenne de l'ordre de 1 dB de la sonorité calculée, par rapport aux résultats ne faisant pas intervenir cet effet; la différence n'excédant pas 3 dB pour les impulsions les plus longues. Cette atténuation est faible par rapport aux 12 dB que l'on pourrait attendre. Cela provient du fait que les impulsions analysées sont très courtes (au maximum une dizaine de millisecondes) et contiennent d'importantes composantes spectrales au-dessus de 2 kHz; elles ne sont donc que très partiellement atténuées par le réflexe stapédién.

(a) voyelle "o" sans et avec réflexe maximum;
(b) rafale de 3 coups avec fusil G1;
sons purs (c) : 250 Hz; (d) : 2 kHz et (e) : 3 kHz

Figure 3 : Réponse de l'étier
B. EFFET SUR LE TIR EN RAFALE

Les méthodes classiques d’analyse des bruits provoqués par les rafales d’armes à feu ne tiennent compte que de la première impulsion de la rafale; on admet implicitement que les impulsions sonores provoquées par les coups suivants sont suffisamment atténuées par la réaction stapédienne et qu’elles peuvent de ce fait être négligées. Cette règle n’est évidemment applicable que pour des rafales dont la cadence n’excède pas environ 600 coups par minute (c’est-à-dire moins d’un coup par 100 millisecondes). Certaines armes récentes présentent des cadences de tir nettement plus élevées. C’est le cas des armes antiaériennes multitudes, (tirant 600 coups par minute par tube) tels le "Vulcan" à 6 tubes, les mitrailleuses quadruples ou les bitubes de 30 ou 35 mm. C’est le cas également du nouveau fusil allemand G11 tirant en rafale de 3 coups à la cadence de 2000 coups par minute grâce à l’emploi de munitions sans étui. Pour chacune de ces armes, on a simulé une rafale de trois coups et procédé au calcul théorique de la sonorité. Pour les multitudes, on trouve respectivement : + 2,5 dB (6 tubes), + 2 dB (4 tubes) et + 1,5 dB (bitube), tandis que la rafale du fusil G11 (Fig. 3b) conduit à une augmentation de sonorité de 1,7 dB. Des valeurs aussi élevées ne peuvent en aucun cas être négligées. Ces écarts sont dus au fait que la cadence de ces rafales est très rapide; les impulsions surviennent alors que le réflexe stapédien n’est pas encore complètement installé.

En conséquence, on peut dire que les armes modernes à très grande cadence de tir sont nettement plus dangereuses lors du tir en rafale que ne le laisse prévoir les méthodes classiques d’évaluation.

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FLUID PARTICLE TRAJECTORIES IN THE COCHLEA

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Introduction
In the past analyses of wave motion in the cochlea have been restricted mainly to descriptions of the traveling wave along the basilar membrane (BM), with little regard given to the fluid particle trajectories (see Lighthill(1981) for a comprehensive literature survey). If considered, it was simply asserted that the trajectories are either elliptical or circular, depending on the product of scala depth and wavenumber, as is well known from the classical theory of small amplitude surface waves. However, we show that when damping is included in the cochlear partition (CP), the elliptical trajectories are tilted relative to the unloaded BM, by an angle which is proportional to the distance of the mean particle position from the CP. It will be shown that this tilting implies that the wavefronts are not propagated in the same direction as the traveling wave along the BM, as was once presumed.

Model
The following sets of assumptions are made:
1) Scala tympani (ST) and scala vestibuli (SV) are identical rectangular chambers surrounded by rigid walls containing fluids with identical properties.
2) The xx-plane coincides with the bottom of SV and the equilibrium position of the middle surface of the CP is located at y=h.
3) The fluid is incompressible and irrotational, and therefore there exists a velocity potential \( \Phi \), defined such that its gradient is the fluid velocity \( \vec{v} \) (\( \vec{v} = \text{grad}\Phi \)), which satisfies Laplace's equation,
\[
\nabla^2 \Phi = 0.
\]
(1)
4) The fluid particle displacement is small compared to the wavelength of the traveling wave and the scala dimensions.
5) The fluid is not subjected to conservative force fields (eg gravity).
6) The fluid pressure \( P \) is continuous across the CP.
For an irrotational fluid, assumptions 4), 5), and 6) imply that
\[
\rho = -\rho \frac{\partial \Phi}{\partial t}, \quad (y = h),
\]
(2)
where \( \rho \) is the fluid density.
7) The relative velocity between fluid and all surfaces is zero in the
direction normal to the surface; that is,
\[ v_y = \frac{\partial \phi}{\partial y} = \begin{cases} 0, & (y = 0, 2h) \\ \frac{\partial \eta}{\partial t}, & (y = h), \end{cases} \]
where \( \eta \) is the vertical displacement of the CP from its equilibrium position at \( y = h \).

8) The CP is infinitely thin and contains no longitudinal coupling.
9) The physical characteristics of the CP are described by the impedance parameters of surface mass \( m \) (kg/m²), volumetric stiffness \( \tau \) (N/m³) and volumetric damping \( r \) (N/sec/m³), all of which are independent of the stimulus.

Assumptions 8) and 9) imply the equilibrium equation
\[ m \frac{\partial^2 \eta}{\partial t^2} + r \frac{\partial \eta}{\partial t} + \tau \eta = 2p. \] (4)

10) The CP has an infinite extent; that is, the whole width of the CP vibrates with uniform amplitude and the CP is spatially invariant.

Thus, this model is a simplified version of the Lesser and Berkley model (Lesser and Berkley, 1972).

Results

For small displacement amplitudes the trajectory of a fluid particle within the scala e is described by the cartesian components \((u_x, u_y)\) of a displacement vector \(\mathbf{u}\) located at the mean particle position \((x, y)\). This vector is defined as the temporal integral of the velocity \(\mathbf{v}\) in a eulerian frame of reference, which in turn, is defined by the spatial gradient of the real part of the velocity potential \(\phi\).

For steady state sinusoidal stimulation of radial frequency \(\omega\), simultaneous solution of Eqs.1)-(4) gives the real part of \(\phi\) as
\[ \text{Re}\{\phi\} = A \exp(-\alpha x) \left\{ \cos^2 \gamma \alpha y - \sin^2 \gamma \alpha y \right\}^{1/2} \cos \left\{ \omega x - \omega t + \tan^{-1} \left( \tan \gamma \alpha y \tan \beta y \right) \right\}, \] (5)
where \(A\) is an arbitrary constant and \(\alpha\) and \(\beta\) are, respectively, the real and imaginary components of the wavenumber \(k\) \((- \alpha + j\beta)\).

After performing the aforesaid manipulations, it is found that the particle rotates with frequency \(\omega\) in an elliptical trajectory with a major axis which is tilted relative to the \(x\)-axis by an angle
\[ \xi = \beta y, \] (6)
as shown in Fig.1. The parametric equations of the ellipse are
\[ u_x' = a' \cos (\alpha x - \omega t + \tan^{-1} \beta/\alpha) \]
\[ u_y' = b' \sin (\alpha x - \omega t + \tan^{-1} \beta/\alpha), \] (7)
where the lengths of the major and minor axes are, respectively,

Fig.1: Fluid particle trajectory.
\[ a' = \eta_m \exp(-\beta x) \cosh(y) (\sinh^2 \alpha + \sin^2 \beta h)^{-1/2} \]
\[ b' = \eta_m \exp(-\beta x) \sinh(y) (\sinh^2 \alpha + \sin^2 \beta h)^{-1/2} \]
where \( \eta_m \) is the amplitude of \( \eta \) at \( x=0 \).

**Discussion**

For the lossless condition one can show that the wavenumber is real-valued \((\beta=0)\). Inspection of Eqs.(6)-(8) shows that when \( \beta=0 \), the spatial variation of the elliptical trajectories is identical to that resulting from small-amplitude surface wave phenomena (Lamb, 1932). Thus, the major axes of the ellipses are parallel to the \( x \)-axis and when the product \( \alpha h \) is much greater than unity the trajectories degenerate to circles (deep fluid wave condition).

For the lossy condition Eq(8) indicates an exponential amplitude decrease with distance \( x \), as one would expect for a damped membrane, together with a reduction (relative to the case \( \beta=0 \)) of the spatial change of amplitude with distance \( y \) (because \( \alpha \) is smaller when \( \beta \) is nonzero). More importantly, inspection of Eqs.(6) & (7) indicates that at any instant a particle in a lossy trajectory is phase shifted relative to one in a lossless trajectory about the same mean position. There exists two types of phase shift. Firstly, the major semi-axis is rotated anticlockwise relative to the horizontal major semi-axis for the lossless condition. The angle of rotation is proportional to the distance of the mean particle position from the bottom of the wave, where the constant of proportionality is simply the imaginary part of the wavenumber. Thus, this angle is the same for all particles located at the same mean depth. Secondly, within the new frame of reference \((u_x, u_y)\), the particle is further rotated in an anticlockwise direction by an angle whose tangent is \( \beta/\alpha \). The magnitude of this rotation is independent of position in the fluid and therefore is, for present purposes, of secondary importance compared to the semi-axis rotation.

The existence of a depth dependent rotation of the major semi-axis implies that particles with mean positions along the same vertical line rotate out of phase. Therefore, the wavefronts (lines of constant phase) no longer form vertical lines propagating in the \( x \)-direction, as would have been the case for the lossless condition. Instead, according to Eq(5), the shape of the wavefronts at any instant is implicitly defined by

\[ \alpha x + \tan^{-1}(\tanh y \tan \beta y) = \text{constant}, \]

where the constant is proportional to time. Since the wavefronts travel a distance of \( 2\pi/\alpha \) in the \( x \)-direction in a period of \( 2\pi/\omega \), the wave is propagated in the \( x \)-direction with the phase velocity of \( \omega/\alpha \). Fig.2 shows wavefronts separated horizontally by a phase of \( 2\pi \) for \( \beta/\alpha=0,1,2 \). The wavefronts are orthogonal to the bottom surface for every \( \beta \) because assumption 7) requires that all vertical motion vanishes in that region. The wavefronts lag further behind as the surface is approached or as the losses are increased. In the case of deep fluid waves \((\alpha h>>1)\), the curves degenerate to lines of slope \(-\omega/\beta\).

The physical significance of the wavefront shape can be evaluated by analyzing the distribution of power throughout the fluid. One can readily show that the average power density vector (average of \( py; \text{Stoker, 1957} \)) at any point in the fluid is directed parallel to the normal vector to the
wavefront at that point. That is, energy flows in a direction normal to the wavefront, whereas the wavefront is propagated in the x-direction. The two directions are coincident only when there are no losses in the CP.

![Diagram of wavefronts](image)

**Fig. 2:** Wavefronts separated in the x-direction by a phase difference $\beta$. 

Therefore physically, a curved wavefront occurs because not only is there a longitudinal transfer of energy by means of fluid particle collision, but also there is a vertical flow of energy into the CP damping mechanism, yielding a net flow of energy in a direction which is no longer coincident with the propagation direction of the traveling wave. Intuitively, particles which are closer to the CP will be more involved in the transport of energy to the damping mechanism than those which are located further below the surface. Therefore, the wavefront will lag further behind as the surface is approached.

The results of this analysis are important in view of the possibility that cochlear tuning is mediated by an active damping process (Kim et al., 1980), in which case the particle trajectories and the flow of energy between CP and fluid are of fundamental importance.

References


ETUDE DES PHÉNOMÈNES HYDROMÉCANIQUES COCHLÉAIRES AUX TRES BASSES FRÉQUENCES CHEZ LE COBAYE.

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Introduction

Si l'on considère la cochlée d'une espèce donnée, on peut déterminer une fréquence de stimulation pour laquelle la longueur de l'onde propageée s'inscrit exactement dans la longueur de la cochlée. Par exemple chez le cobaye, cette fréquence est de l'ordre de 450 Hz. Pour des fréquences bien inférieures à 450 Hz, il faut donc admettre que les phénomènes de propagation deviennent négligeables et que les différents points de la membrane basilaire se déplacent pratiquement en phase. Le but du présent travail est d'analyser le fonctionnement particulier de la cochlée dans ce domaine des très basses fréquences, ainsi que de préciser la transition avec le domaine classiquement connu des fréquences audibles (mécanisme de l'onde propageée).

Moyens

Le modèle animal qui a été retenu est le cobaye, dont la cochlée présente un accès facile.

La stimulation acoustique (son pur) est réalisée en circuit fermé et à bulle tympanique ouverte, à l'aide d'un microphonie à condensateur utilisée comme source. L'intérêt de ce mode de stimulation réside dans l'obtention d'un déplacement volumique quasi constant, tout au moins aux fréquences inférieures à 1500 Hz. Ces conditions de stimulation sont proches des conditions physiologiques "normales", à pression acoustique constante devant le tympan et à bulle fermée.

On réalise des mesures de pression acoustique dans le premier tour de la cochlée à l'aide de deux méthodes complémentaires :

a) une méthode directe utilisant un capteur de pression piézorésistif équipé d'une sonde remplie de liquide silicone

b) une méthode indirecte basée sur l'enregistrement du potentiel microphonique différentiel, qui dans des conditions bien définies, est directement lié à la différence de pression agissant sur la cloison cochléaire.

Résultats

La fig. 1 montre en fonction de la fréquence, l'amplitude et la phase de la pression acoustique dans la rampe vestibulaire (p_{vl}) et dans la rampe tympanique (p_{t1}) ainsi que celles du potentiel microphonique diffé-
Franke et Dancer : Hydromécanique cochléaire aux très basses fréquences

Fig. 1. Amplitude et phase de CM₁, $P_{vl}$ et $P_{t1}$ en fonction de la fréquence. Référence de phase : le déplacement volumique constant. Le niveau devant le tympan correspond à 78 dB SPL (à bulle fermée).

Amplitude and phase of CM₁, $P_{vl}$ and $P_{t1}$ versus frequency. Phase reference: constant volume displacement. The level in front of the tympanum corresponds to 78 dB SPL (closed bulla condition).

rentiel ($CM₁$) recueillies dans le premier tour.

Aux fréquences inférieures à 30 Hz, les amplitudes de $P_{vl}$ et $P_{t1}$ sont identiques et en phase avec la stimulation (déplacement volumique). Elles sont plus faibles de 18 dB par rapport au niveau de stimulation devant le tympan (à bulle fermée). Dans la rampe tympanique, la pression acoustique reste constante et en phase avec la stimulation jusqu'au delà de 200 Hz. Au dessus de 400 Hz, elle augmente rapidement à raison d'environ 15 dB/oct. et sa phase est en avance.

Les courbes d'amplitude de $P_{vl}$ et CM₁ ont des caractéristiques semblables : la pente de l'ordre de 6 dB/oct. est interrompue au voisinage de 100 Hz par un petit palier. Par contre on observe une différence importante aux fréquences inférieures à 30 Hz : la pente de 6 dB/oct. et l'avance de phase sont conservées en ce qui concerne CM₁.

Nous avons ensuite procédé à diverses modifications de la configuration mécanique de la cochlée, et observé les changements intervenus au niveau des pressions acoustiques intracochléaires et de CM₁. La Fig. 2 montre que lorsque l'on obture le 4ème tour, on obtient une amplitude constante de $P_{vl}$ aux fréquences inférieures à 140 Hz. Aux fréquences plus élevées, la courbe rejoint la courbe originale après quelques oscillations d'amplitude dé -

Fig. 2. Réponse typique de $P_{vl}$ en fonction de la fréquence :

- --- cochée intacte
- --- après obturation du 4ème tour
- --- après perçement d'un trou dans le 1er tour de la rampe tympanique (diam.: 0,4 mm).

Typical $P_{vl}$ response versus frequency :

- --- intact cochlea
- --- after sealing of 4th turn
- --- after drilling a hole into the 1st turn of scala tympani (diam.: 0,4 mm).
croissante. Conjointement, l'avance de phase disparaît aux basses fréquences. Lorsqu'un trou est percé dans la rampe tympanique du 1er tour, l'amplitude du plateau décroît de 5 à 6 dB.

Lorsque (fig. 3) un trou est percé dans la rampe tympanique du 1er tour, CM1 reste inchangé. L'obturation du 4ème tour provoque comme pour pM1, une amplitude sensiblement constante aux basses fréquences, avec un retard de phase de l'ordre de 90° par rapport à la réponse originale dans le même domaine de fréquence. Comme pour pM1, aux fréquences plus élevées, la courbe d'amplitude rejoint la courbe originale après quelques oscillations d'amplitude décroissante.

Fig. 3. Réponse typique de CM1 en fonction de la fréquence :
- cochée intacte
- après percément d'un trou dans le 1er tour de la rampe tympanique (diam.: 0,6 mm)
- après obturation du 4ème tour

Typical CM1 response versus frequency :
- intact cochlea
- after drilling a hole into the 1st turn of scala tympani (diam.: 0,6 mm)
- after sealing of 4th turn.

Fig. 4. Schéma de la stimulation acoust. du récepteur auditif a et du modèle électrique EF b.

V, tention alternative appliquée au microphone; kM, constante du microphone; Z0, impédance acoust. des osselets; ZHM, impédance acoust. de l'hélicotréme; CBFM, compliance de la membrane basilaire; CRW, compliance de la fenêtre ronde; x, déplacement volumique; pT1, pression acoust. dans le 1er tour de la RV; pT2, pression acoust. dans le 1er tour de la RT; CM2, potentiel microphonique différentiel dans le 1er tour; k, constante.

Simplified diagram of the acoustic stimulation of the auditory receptor a and the LF electric analog b; V, alternate voltage applied to the microphone; kM, constant factor of the microphone; Z0, acoustic impedance of the ossicles; ZHM, acoustic impedance of the helicotrema; CBFM, compliance of the basilar membrane; CRW, compliance of the round window; x, volume displacement; pT1, sound pressure in the 1st turn of SV; pT2, sound pressure in the 1st turn of ST; CM2, differential cochlear microphonic in the 1st turn; k, constant factor.
Mesures directes complémentaires

Pour préciser sur le plan quantitatif les résultats obtenus, on a procédé à la mesure directe de certains éléments de la cochlée:

a) Les compliances acoustiques de la membrane basilaire (C_{BM}) et de la fenêtre ronde (C_{RW}) ont été mesurées en injectant des volumes connus de liquide dans la cochlée, et en mesurant simultanément les variations de pression produites. Valeurs trouvées: C_{BM} 0,9 \times 10^{-13} \text{m}^2 \text{N}^{-1};
C_{RW} 1,4 \times 10^{-13} \text{m}^2 \text{N}^{-1}.

b) La résistance acoustique de l'hélicotroîme (R_{H}) (y compris la contribution des parties apicales des rampes vestibulaires et tympaniques) a été évaluée à partir de la mesure du temps d'écoulement de quantités données de péridyme artificiel par cet orifice. Valeur trouvée: R_{H} 5 \times 10^{10} \text{Nsm}^{-2}.

Discussion

Les résultats obtenus peuvent être expliqués aisément en considérant le modèle électrique simplifié du récepteur auditif excité par une source à déplacement volumique constant (fig. 4).

Supposons d'abord l'hélicotroîme obturé. Le déplacement volumique \( x \) de la fenêtre ovale étant constant, \( p_t 1 = x/CRW = Cte. \) De la même manière,
\( p_v 1 = k \cdot CM = x/CBM = Cte. \) On en déduit \( p_v 1 = x/CRW + x/CBM = Cte. \)

Lorsque l'hélicotroîme est fonctionnel, \( p_t 1 \) conserve la valeur précédemment définie. Par contre, \( k \cdot CM \) ne la conserve qu'aux fréquences élevées, car en deçà d'une certaine fréquence \( f_2 \), l'hélicotroîme \( Z_H \) opposera une impédance moindre que \( CM \) et si l'on suppose que \( Z_H = RH \) (résistance pure), \( k \cdot CM = RH \cdot \omega = j \omega \cdot R_H \) (pente de 6 dB/oct.). Dans ce domaine de fréquence, on a
\( p_v 1 = x/CRW + j \omega \cdot R_H. \)

Aux fréquences très basses (\( f < f_1 \)), la décroissance de \( p_v 1 \) est donc limitée par la valeur constante de \( p_t 1 \).

Ce modèle ne tient pas compte des phénomènes de propagation qui existent aux fréquences élevées (\( f > f_3 \)). Lorsque cette propagation a lieu, on sait que l'entrée de la cochlée est purement résistive et que l'amplitude de \( k \cdot CM \) et de \( p_v 1 \) augmente avec la fréquence avec une pente de 6 dB/oct. La fig. 5 représente les courbes calculées de \( p_v 1 \), \( p_v 2 \) et \( k \cdot CM \) selon le modèle décrit, en tenant compte du terme réactif de \( RH \) et en ajustant les différents paramètres.

Fig. 5. Réponses calculées à l'aide du modèle électrique, responses of the electric analog:
- \( p_v 1 \), cochlée intacte,
- intact cochlea
- \( p_v 1 \), apex obturé,
- sealed apex
- \( k \cdot CM \), cochlée intacte,
- intact cochlea
- \( k \cdot CM \), apex obturé,
- sealed apex
- \( p_t 1 \), cochlée intacte,
- intact cochlea
- \( p_v 1 \) & \( CM \) en régime onde propagée, with traveling wave.
THE SIGNIFICANCE OF THE SHARPLY TUNED BASILAR MEMBRANE RESPONSE IN THE CAT COCHLEA

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ABSTRACT

A sharply tuned mechanical response is seen at the basilar membrane. This response is nonlinear and highly susceptible to trauma. A comparison with hair cell and nerve fiber tuning curves indicates that the sharply tuned nonlinear response originates at the hair cell.

The resonant properties of the stereocilia bundles of individual hair cells determine the shape of tuning seen in the hair cell and single nerve fiber tuning curves in the auditory system. The tuning frequency of each hair cell seems to be determined by the mechanical parameters of its own stereocilia bundle and the associated tectorial tissue.

SPL required at the tympanic membrane (TM) to produce a vibration amplitude of $10^{-8}$ cm in the basal region of the basilar membrane (BM) was measured over a wide frequency range in cat cochleas (1). These low amplitude measurements were made possible by the use of laser interferometer (2). The cochlea was found to be highly susceptible to trauma and heroic efforts were required to minimize it. Under these conditions sharp tuning of the BM was observed in at least five experiments.

Trauma reduces the sharply tuned portion of the response leaving a BM response with the characteristics of a low pass filter. This low pass filter type of response is dependent mainly on the mechanical properties of the BM. It has been studied extensively in the past and is well understood, (for a review see 3,4,5,6). Insight into the origin of the sharply tuned response can be obtained by comparing the BM (1), hair cell (7) and nerve fiber tuning (8,9) curves with the same
center frequency. They are equally sharply tuned. The slopes of the tuning curve and Q10 values are quite similar. The height of the sharp tip for the basilar membrane is much smaller (35dB) as compared to that for the hair cell (80dB) (10). Nonlinear response characteristics are seen mainly in the sharply tuned tip region of both the hair cell (7) and the basilar membrane (11, 12) response. The nonlinearity in the basilar membrane response however disappears with trauma (3). The sharply tuned portion of the BM response is much more susceptible to trauma as compared to hair cells. Under similar degrees of surgical trauma sharply tuned responses are seen at the hair cell level but not at the basilar membrane. These observations indicate that the sharp tuning originates at the hair cell (13). It is seen at the BM due to the mechanical coupling between the two. It has been shown that the mechanical properties of the stereocilia bundle are nonlinear in nature (14). These stereocilia nonlinearities will also be seen in the BM response as a mechanical non-linearity.

The principle that the tuning in the ear is due to mechanical resonant properties of the stereocilia may be quite general in nature, because in over twenty five species of animals studied so far the physical dimensions of the stereocilia are systematically organized in such a way that the hair cells tuned to higher frequencies have the stiffer ciliary tufts (shorter and most numerous cilia) (15,16,17).

There are a large number of animal species (amphibians) in which the auditory organs lack a basilar membrane (18). The hair cells rest on stationary supporting structures. These hair cells can only be stimulated through the vibration of their ciliary tufts. The sharpness of tuning observed in the auditory nerve fibers in these ears is comparable to that observed in the mammalian ears (19). The frequency selectivity in the amphibian ears can only be due to the mechanical properties of the stereocilia and the attached tectorial tissue.

The mechanical properties of the stereocilia are based on their internal microstructure, i.e. the bundles of actin filaments and their cross bonds (20,21). In the presence of trauma the stereocilia stiffness is reduced as the actin depolymerizes and the cross bonds between actin filaments are broken (22).

Trauma reduces the stiffness of the tallest stereocilia
and decouples them from the tectorial membrane. Loss of stiffness lowers the resonant frequency of the stereocilia bundle and loss of coupling their sensitivity. The loss of coupling also prevents the sharply tuned nonlinear response of the stereocilia from appearing at the basilar membrane.

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MODIFICATION DES PROPRIÉTÉS DE TRANSDUCTION EN FRÉQUENCES DE LA MEMBRANE BASILAIRE AU COURS DU DÉVELOPPEMENT : DÉMONSTRATION PAR L’ÉTUDE DU TRAUMA ACoustique CHEZ LE POUSsIN

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Des expériences électrophysiologiques récentes ont montré que si l’on soumettait de jeunes poussins à un son pur intense de fréquence donnée les audiogrammes enregistrés sur ces animaux 10 jours plus tard montraient une augmentation de seuil maximale légèrement décalée en fréquence par rapport au son traumatique.

Afin d’expliquer cette différence, une étude anatomo-fonctionnelle plus précise du trauma acoustique a été menée pour déterminer l’influence du temps de survie post-traumatique sur l’évolution des dommages dans le récepteur auditif du poussin pendant son développement.

**Matériel et méthode**

Des poussins ont été soumis pendant 12 heures consécutives, à un son pur de 1 500 Hz et de 125 dB SPL, dans les 24 heures suivant leur éclosion. Ces animaux ont ensuite été placés dans les mêmes conditions d'élevage qu'un lot d'animaux témoins n'ayant pas été soumis au trauma acoustique. 10 jours plus tard, 25 % des poussins traumatisés ont été sacrifiés après perfusion de fixateur (glutaraldéhyde + paraformaldéhyde) dans l’oreille interne. Les papilles basilaire (BP) ont ensuite été prélevées, postfixées à l’acide osmique, déshydratées et incluses dans une résine. Des coupes histologiques sérées ont alors été effectuées et les cellules ciliées endommagées ont été comptées à chaque niveau de la BP.

Le potentiel global du VIIIème nerf enregistré par une électrode monopolaire en argent placée sur la fenêtre et les seuils de ces potentiels ont été mesurés à 10 fréquences comprises entre 250 et 4 000 Hz. 30 jours plus tard, les mêmes protocoles histologiques et physiologiques ont été appliqués aux animaux restants.

**Résultats**

*Après 10 jours de survie :

Des dommages anatomiques sont observés tout au long de la BP mais ils présentent un maximum très net à 1,53 mm de la base de la BP. Ceci
correspondant à 30 % de sa longueur comptée à partir de la base.

Ces audiogrammes enregistrés sur le lot d'animaux correspondant montrent eux aussi une augmentation des seuils auditifs sur toute la gamme des fréquences testées avec un maximum à 2 000 Hz.

Après 30 jours de survie :
On observe des dommages anatomiques de même type mais dont le maximum est situé à 1,82 mm de la base de la BP (32 % de sa longueur).

Les audiogrammes montrent une augmentation de seuil maximale à 3 000 Hz.

Discussion et conclusions

- Le déplacement observé aussi bien en ce qui concerne les dommages anatomiques qu'en ce qui concerne les augmentations de seuil est en parfaite concordance avec les résultats de Ryals et Rubel qui ont déterminé une courbe de régression qui permet de faire correspondre à chaque portion de la BP la fréquence la mieux codée à cet endroit.

Si l'on considère nos résultats anatomiques et qu'on se réfère à cette courbe, on constate que les dommages observés à 10 jours, qui sont situés à 1,53 mm de la base de la BP, devraient provoquer une augmentation maximale du seuil à 1 939 Hz et que ceux situés à 1,82 mm de la base (30 jours de survie) devraient provoquer une augmentation de seuil à 3 186 Hz.

- Le fait qu'un son traumatique de 1 500 Hz produise une augmentation de seuil maximale à 2 000 Hz après 10 jours et de 3 000 Hz après 30 jours permet de montrer :
1/ que les cellules ciliées peuvent coder des fréquences différentes et que seule leur position sur la BP est importante pour les fréquences qu'elles codent ;
2/ que les variations de position relative des dommages anatomiques et la variation d'un octave de l'augmentation de seuil par rapport au son traumatique après 30 jours de survie indiquent une modification de la mécanique de la BP. Cette hypothèse est d'autant plus vraisemblable que bien que les seuils normaux soient de type adulte dès le premier jour suivant l'éclosion, la longueur de la BP augmente jusqu'à environ 30 jours.

Référence
ANATOMO- ET PHYSIOPATHOLOGIE COCHLEAIRES CHEZ DEUX SOURIS MUTANTES (shaker-1 et bronx waltzer)

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Les anomalies héréditaires de l'oreille interne peuvent être utilisées comme des outils particulièrement précieux pour aborder des problèmes physiologiques encore non résolus chez l'animal normal. Certaines mutations interfèrent aussi avec le développement, ce qui leur attribue un intérêt supplémentaire dans l'étude du développement pathologique. Nous avons récemment entrepris des investigations anatomo-fonctionnelles sur 2 souris mutantes. La première est la "shaker-1" qui pourrait être un excellent modèle d'approche des systèmes efférents car elle semble présenter un déficit à ce niveau. La seconde est la "bronx waltzer" qui présente une anomalie semble-t-elle spécifique des cellules ciliées internes.

1 Le mutant "shaker-1" (Sh1/Sh1)

L'expérimentation entreprise avait pour but de compléter les connaissances anatomo-fonctionnelles sur ce mutant qui est l'un des premiers à avoir été décrits (Lord and Gates, 1920). Les souris homozygotes n'entendent semble-t-il, que très mal et pendant une très courte période (12ème au 18ème j. postnatal ; ccf. Mikaelian et Rubel, 1964 ; Deol, 1968). Ces premières études suggéraient que la surdité était due à un développement anormal et à une dégénérescence précoce des structures cochléaires. Kikuchi et Hilding (1965) utilisant la microscopie électronique observaient les premiers signes de dégénérescence à 12 jours et ils en rendaient responsable le système efférent "pratiquement absent". Il semblait nécessaire de reprendre et compléter cette étude à la lumière des connaissances actuelles sur les structures neuro-sensorielles de la cochlée.

26 souris homozygotes pour le gène Sh-1 ont été utilisées à des âges de 3, 6, 10, 12, 18 et 30 jours après la naissance. Des souris hétérozygotes et des souris C57Bl/6J de même âge ont servi de témoins. Après un rapide contrôle électrocochléographique (pour les animaux âgés de 10 jours ou plus), les cochlées étaient prélevées et traitées selon les techniques classiquement utilisées pour l'observation en microscopie électronique à transmission chez cette espèce (Shnerson et al., 1982).

A 3 jours, la cellule ciliée interne (CCI) et son innervation afférente et efférente présentent un aspect normal pour cet âge. A 6 jours cependant les premières anomalies cytoplasmatiques (vacuoles et lysosomes) apparaissent dans la CCI. Ces anomalies s'accentuent avec l'âge. A 10 jours les terminaisons
afférentes au contact de la CCI apparaissent anormales ; elles vont dégénérer totalement entre le 18ème et 30ème jour. Par contre, dans ces derniers stades étudiés, on rencontre une surabondance de terminaisons efférentes bien préservées.

Dans les cellules ciliées externes (CCE) les vacuoles et lysosomes sont notés dès le 1er stade (3 j.). Dans les stades suivants ces anomalies cytoplasmiques s'accroissent et sont accompagnées d'un retard global du développement de la CCE qui n'acquiert jamais sa morphologie de type adulte. L'innervation des CCE est très fortement anormale : dégénérescence précoce des afférences et retard de développement des efférences. À 18 j., seules quelques synapses efférentes sont constituées avec un faible nombre de CCE. En outre, au niveau de ces synapses, les microvésicules présynaptiques ont un aspect anormal et la membrane postsynaptique est incomplète. À 30 j., la plupart des éléments sensoriels et nerveux, dans cette région des CCE, ont totalement dégénéré.

Des anomalies peuvent être notées dès 3 et 6 jours dans les neurones du ganglion spiral. Les cellules ganglionnaires sont très lâchement engainées par les cellules gliales. Des fibres nerveuses font souvent contact directement avec le corps cellulaire et des images synaptiques sont fréquentes. À 10 jours on ne voit encore aucune trace de myéline autour des neurones ganglionnaires qui contiennent souvent un matériel fibrillaire très abondant.

En conclusion, l'apparition simultanée et très précoce, dans les cellules sensorielles et ganglionnaires d'anomalies cytoplasmiques peut être reliée avec les très faibles performances auditives que certains auteurs ont notées, entre le 12ème et 18ème jour postnatal. Pour notre part, aucun potentiel d'action notable n'a pu être enregistré au niveau de la fenêtre ronde, à quelque âge que ce soit. La surdité totale, où en tout cas très précoce, est à corrélérer en particulier avec les anomalies relevées au niveau des CCI et de leur innervation afférente, ainsi qu'au niveau ganglionnaire. Le développement anormal et sélectif du système efférent lié aux CCE est un nouvel argument en faveur de l'hypothèse des 2 systèmes efférents. Dans le cas des "shaker-1", seul le système relié aux CCE, de développement plus tardif, serait touché par le déficit génétique. Enfin, un problème particulier est posé par les contacts synaptiques anormaux rencontrés dans le ganglion de Corti.

2 Le mutant "bronx waltzer" (Bv/Bv)

C'est l'un des derniers mutants de l'oreille interne répertoriés chez la souris. Décrit par Deol and Gluecksohn-Waelsch (1979), il présente, d'après une étude en microscopie optique une anomalie semble-t-il spécifique des CCI. Une exploration physiologique des reliquats auditifs de cet animal a débuté (Bock, communication personnelle). Nous avons pour notre part entrepris une investigation en microscopie électronique à balayage (SEM) et à transmission (TEM).

Les premiers résultats, en SEM, confirment les données de la microscopie optique ; il reste apparemment 1 CCI intacte sur 7 alors que les CCE ne semblent pas présenter d'anomalies.

Des précisions importantes sont apportées par les premiers examens en
TEM sur des animaux âgés de 35 jours. La plupart des CCI ont dégénéré ou sont en cours ; de plus, même au niveau des quelques CCI encore intactes, l'innervation afférente est très peu abondante et dans un état de dégénérès- cence très avancé. La cytologie et l'innervation des CCE sont quasi normales avec un système efférent bien développé. Le ganglion de Corti est profondément affecté par le déficit héréditaire : perte considérable de cellules et dégénérès- cence de beaucoup d'autres, ce qui est en accord avec les dégâts relevés au niveau des CCI.

Ces observations préliminaires confortent l'idée que la "bronz waltzer" est un mutant particulièrement précieux pour confirmer et préciser le rôle des 2 types de récepteurs cochléaires et de leurs innervations respectives.

Références


EFFETS DE SONS PURS SUR LE POTENTIEL ENDOCHELIAIRE ET SUR LE POTASSIUM ENDOlyMPHATIQUE DU COBAYE.

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Introduction

L'effet du bruit sur le potentiel endolymphatique et sur le contenu ionique de l'oreille interne est encore l'objet de controverses.

SUGA et al (1969) montrent les premiers, à l'aide de microélectrodes de verre spécifique, que les concentrations ioniques endolymphatiques de sodium et de potassium sont affectées par l'asphyxie. En 1970, les mêmes auteurs étudient les effets d'une stimulation en sons purs appliquées pendant deux minutes, sur les concentrations ioniques endolymphatiques. Ils observent une augmentation du sodium endolymphatique associée à une diminution du potassium de la même rampe cochléaire. Les espaces périlymphatiques par contre, ne s'avèrent pas affectés par une stimulation sonore intense.

MELICHAR et al (1980) reportent que lors d'une stimulation par un bruit intense, le potentiel endocochléaire et le potassium endolymphatique décroissent jusqu'à atteindre de très faibles valeurs. Le retour aux niveaux de départ s'effectue après quelques jours. Ils signalent également une légère augmentation de la concentration de potassium périlymphatique.

KONISHI et al (1979) exposent des cobayes pendant sept jours à des niveaux acoustiques compris entre 95 et 105 dBA. Ils observent une augmentation significative du potassium endolymphatique ainsi que du chlorure. Ces mesures faites par photométrie après microprélèvements s'accompagnent d'une diminution du sodium endolymphatique. Ils n'observent pas de modifications des concentrations ioniques des espaces périlymphatiques.

Au contraire, SALT et KONISHI (1979), à l'aide de microélectrodes de verre à liquides échangeurs d'ions spécifiques trouvent après stimulation, une augmentation du potentiel endolymphatique associé à une croissance de la concentration du potassium endolymphatique. Cependant, après une exposition prolongée ils notent une légère diminution des valeurs du potentiel endocochléaire et du potassium endolymphatique.

Récemment KONISHI et al (1982) suggèrent que l'exposition au bruit n'altère pas la perméabilité au potassium de la membrane de Reissner mais
diminue la perméabilité de la barrière située entre la rampe médiane et la rampe tympanique.

Le but de notre expérience préliminaire est d'étudier les modifications éventuelles des liquides cochléaires du second et du troisième tour de la cochlée du cobaye soumis à des stimulations en sons purs de diverses fréquences et de divers niveaux d'intensité acoustique.

Méthodologie

Vingt trois cobayes tricolores, d'un poids compris entre deux cent cinquante et trois cent grammes, présentant un réflexe de Preyer normal ont été utilisés pour cette étude. Les animaux sont anesthésiés par une injection intramusculaire de lévopromazine (2,5 mg) et de sulfate d'atropine au 1/1000 (0,125 cc) suivie vingt minute plus tard par une injection intrapéritonéale de chlorhydrate de kétamine (150 mg/Kg).

La température interne des animaux est maintenue constante à l'aide d'une couverture chauffante reliée à une sonde de température rectale. (37 ± 0,5 °C).

La veine jugulaire externe est canulée afin de permettre l'injection de drogues. La cochlée est mise en évidence par une approche submandibulaire. Une petite fenêtre est réalisée au dessus de la strie vasculaire du deuxième ou du troisième tour de la cochlée afin d'insérer les microélectrodes. Les concentrations ioniques cochléaires ainsi que les potentiels endolymphatiques sont mesurés à l'aide de microélectrodes de verre doubles dont l'un des canaux est empli de liquide échangeur d'ion spécifique. La préparation des microélectrodes est identique à celle développée par MORGENSTERN (Düsseldorf). Les microélectrodes sont calibrées suivant la technique de BOSHER (1979).

La stimulation en sons purs est réalisée en circuit fermé par l'intermédiaire d'un haut parleur adapté directement sur une barre d'oreille creuse. Le niveau de bruit est mesuré juste devant le tympan. (FRANKE et DANCER 1983).

Après l'exposition au bruit et les diverses mesures, une injection d'acide éthacrinique (60 mg/Kg) est effectuée.

Résultats

Nous présentons uniquement les résultats préliminaires obtenus chez des animaux dont nous avons pu mesurer le potentiel endocochléal, les concentrations ioniques en potassium endolymphatique et péribulymatique ainsi que les modifications résultant de l'injection d'acide éthacrinique.

Les valeurs de départ du potentiel endolymphatique et des concentrations endolymphatique et péribulymatique de potassium sont conformes à celles des autres auteurs. Nous ne trouvons pas de différence significative entre le second et le troisième tour de la cochlée.
Valeurs moyennes obtenues chez les témoins. N±SEM nombre d'animaux.

<table>
<thead>
<tr>
<th>K_p ST</th>
<th>8.5 ± 0.9 meq.l⁻¹</th>
</tr>
</thead>
<tbody>
<tr>
<td>K_e</td>
<td>158 ± 6 meq.l⁻¹</td>
</tr>
<tr>
<td>EP</td>
<td>78.9 ± 6.8 mV</td>
</tr>
</tbody>
</table>

Modifications observées dans le 2ème tour après exposition à 1800 Hz, 104 dB SPL, pendant 1 heure.

Moyenne de 3 cobayes.

Modifications maximales observées après 1 heure d'exposition au bruits dans la rampe médiane. n=nombre d'animaux.

<table>
<thead>
<tr>
<th>Pure Tone 1 hour</th>
<th>500 Hz 114 dB SPL</th>
<th>1,6 KHz 104 dB SPL</th>
<th>3 KHz 118 dB SPL</th>
<th>5 KHz 119 dB SPL</th>
</tr>
</thead>
<tbody>
<tr>
<td>K_e meq.l⁻¹</td>
<td>30 ± 10</td>
<td>65 ± 2</td>
<td>12 ± 10</td>
<td>10 ± 5</td>
</tr>
<tr>
<td>EP mV</td>
<td>15 ± 3</td>
<td>49 ± 12</td>
<td>5 ± 3</td>
<td>48 ± 13</td>
</tr>
</tbody>
</table>

2ème tour

3ème tour

Variations observées dans la rampe médiane après injection d'acide éthacrinique (60mg/Kg). L'injection a lieu après complète récupération des niveaux de base pour les cobayes stimulés.

<table>
<thead>
<tr>
<th>Control</th>
<th>2nd Turn</th>
<th>3rd Turn</th>
</tr>
</thead>
<tbody>
<tr>
<td>K_e meq.l⁻¹</td>
<td>47 ± 7</td>
<td>55 ± 19</td>
</tr>
<tr>
<td>EP mV</td>
<td>88 ± 10</td>
<td>82 ± 15</td>
</tr>
</tbody>
</table>
Discussion.

Les résultats préliminaires que nous présentons montrent que la stimulation en sons purs de forts niveaux engendre une décroissance du potentiel endocochléaire associée à une diminution de la concentration du potassium endolymphatique. Le maximum de ces variations est observé après un laps de temps de vingt minutes en ce qui concerne le potentiel endocochléaire et de trente minutes pour le potassium de la rampe médiane. La récupération des niveaux de base est totale deux heures après l'arrêt de la stimulation pour le potentiel endocochléaire, celle-ci n'est que partielle pour le potassium. Aucune modification n'a été mise en évidence dans les rampes péristympathiques, cela exclut l'hypothèse du passage du potassium de la rampe médiane vers les espaces péristympathiques.

L'amplitude des modifications des concentrations de potassium endolymphatique et à un moindre niveau du potentiel endocochléaire semble dépendre de la fréquence de la stimulation acoustique. En effet, le maximum de décroissance est obtenu pour une stimulation de 1800 Hz dans le second tour et par une stimulation de 500 Hz dans le troisième tour de la cochlée. Dans cette expérience, la décroissance du potentiel endocochléaire est moins reproductible du fait des variations interindividuelles qui sont très importantes. Il est possible de penser que la stimulation acoustique en sons purs pourrait avoir un effet tonotopique relié aux mouvements de la membrane basilaire provoqués par l'onde voyageante.

Les modifications mises en évidence après l'injection d'acide éthacrine, que ce soit chez les animaux témoins ou chez les animaux stimulés, sont de même amplitude. D'autre part, aucune différence significative n'est visible entre les divers tours étudiés. Ce type d'injection ne nous servira à l'avenir uniquement de contrôle.

Ces expériences seront poursuivies afin de disposer d'un nombre suffisant d'animaux d'expérience pour vérifier l'hypothèse de l'effet tonotopique de la stimulation acoustique en sons purs.

Bibliographie.

ÉTUDE DU POTENTIEL D'ACTION DU NERF AUDITIF DÉCLENCHÉ PAR TRANSITOIRE ÉLECTRIQUE

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Introduction :

Le potentiel d'action composite du nerf auditif (PAC) déclenché par un transitoire acoustique est communément enregistrée sur la fenêtre ronde de la cochlée, selon la technique de l'électro-cochléographie (E. Coch. G.). Comme l'ont montré Goldstein et Xiang (1958), ce potentiel présente une forme complexe dépendant à la fois de l'analyse en fréquence cochléaire et de la forme de la réponse unitaire d'une fibre du nerf auditif (convolution). Pour séparer les fonctions cochléaire et nerveuse (déconvolution), il est nécessaire de connaître la forme de la réponse unitaire, mais la faible amplitude de cette réponse la rend difficile à enregistrer. La stimulation électrique (SE) du nerf auditif permet d’exciter simultanément un grand nombre de fibres nerveuses sans passer par l’intermédiaire de la cochlée (Charlet de Sauvage et coll. 1980) et d’éviter ainsi le phénomène de convolution. Le potentiel d’action résultant, semblable à celui d’une seule fibre mais de beaucoup plus grande amplitude, est donc facilement enregistrable. L’objet de ce travail est l’analyse des caractéristiques de cette réponse (PAE).

Méthode :

Dix cobsyes d’audition normale ont été implantés de manière chronique à l’aide d’une électrode sur la fenêtre ronde de la cochlée et d’une électrode indifférente sur le vertex, permettant d’enregistrer les potentiels E. Coch. G. selon la méthode décrite par Aran et Erre (1979). Le contrôle des potentiels auditifs corticaux se fait simultanément. La proximité du nerf auditif fait de la fenêtre ronde l’un des emplacements les mieux adaptés aussi bien à l’enregistrement des réponses du nerf qu’à la SE de ses fibres. Stimulation et enregistrement s’effectuent donc sur le même jeu d’électrodes, ce qui simplifie l’abord chirurgical, mais exige en contre partie une technique particulière d’élimination de l’artéfact de stimulation.

La SE consiste en des impulsions de courant négatif de durée 300 μs, et de fréquence de récurrence 25/s. Le signal préamplifié est échantillonné sur 12 bits avec une période de 10 μs/point, dans une fenêtre de 5,12 ms. Le PAE étant superposé à un artéfact de stimulation d’amplitude voisine du vol, on évite les saturations en utilisant un gain très faible (1 à 5). Il en résulte des difficultés pratiques de conversion analogique/numérique, résolues en faisant usage d’un bruit aléatoire additionnel réalisant un codage "statistique" du signal (Charlet de Sauvage et coll. 1983).
La réponse nerveuse doit être séparée de l'artéfact électrique qui la masque totalement (Fig. 1). Pour cela, on enregistre séparément les signaux S1, réponse contaminée par l'artéfact, et S2, forme d'onde de l'artéfact seul (moyennes de 100 à 1000 réponses). On obtient S2 par masquage du PAE à l'aide d'un bruit acoustique à large bande. Les fibres auditives, stimulées par le bruit ne peuvent plus répondre à la SE et la réponse nerveuse disparaît, l'artéfact restant inchangé. En effectuant la différence S1-S2, l'artéfact s'élimine totalement, laissant subsister le PAE seul (Fig. 1c).

Résultats :

1) Variations du PAE en fonction de l'intensité de SE :

Une série de PAE enregistrés à intensité de SE croissante et niveau de masquage constant, est présentée sur la Fig. 2a. La forme invariable des réponses indique que les décharges unitaires des fibres sont étroitement synchronisées entre elles. La croissance moyenne des PAE est de 25 μV/μA. Le seuil des réponses corticales (Fig. 2b) est sensiblement équivalent à celui des PAE. La dynamique de la SE (seuil des réactions faciales - seuil de détection visuelle des PAE) varie de 6 à 12 dB, valeurs proches de celles mesurées en clinique (6 à 20 dB chez l'homme).

2) Variations du PAE en fonction de l'intervalle entre stimulations :

L'adaptation du PAE a été explorée au moyen de trains de deux stimulations électriques espacées d'intervalles variables de 1 à 4 ms (Fig. 3). Ces tracés mettent en évidence la période réfractaire des fibres : 4 ms pour la période relative, moins de 1 ms pour la période absolue. Cette dernière se situe dans les limites normales (0,5 à 1 ms) observées pour les fibres sensorielles de ce diamètre (Bishop et Heinbocker, 1930).

3) Variations du PAE en fonction du masquage :

A intensité de SE constante, une série de PAE a été extraite à l'aide de bruits d'intensité croissante. La même invariance de forme et de latence s'observe, indépendamment du niveau de masquage.

La spécificité en fréquence du PAE a été examinée selon la technique du masquage passe-haut proposée par Teas et coll. (1962). Le bruit utilisé pour extraire le PAE est tronqué par filtrage passe-haut avant d'être appliqué à l'oreille. La pente de coupure du filtre étant très abrupte (96 dB/octave), le bruit filtré masque une région très bien délimitée de la cochlée (Von Békésy, 1960), ce qui permet d'évaluer de façon précise la contribution de cette zone. La figure 4 présente une série de réponses de ce type en fonction de la fréquence de coupure du filtre. Par différence entre 2 réponses voisines, on obtient la contribution des fibres dans la zone de fréquence correspondante. On constate alors que l'efficacité de la SE, prédominante pour les bandes de fréquences aiguës (base de la cochlée), décroît rapidement pour s'annuler aux fréquences moyennes de l'ordre de 5 kHz (fin du ler tour de la cochlée), les contributions restant faibles pour les fréquences les plus basses. Le même phénomène s'observe pour toutes les intensités de SE.

Discussion :

La méthode de stimulation électrique et d'extraction du PAE par masquage permet d'établir l'origine auditive de ces réponses chez l'animal normal. La latence du PAE, 0,16 ms en moyenne, est inférieure au délai de transmission synaptique entre cellule ciliée et fibre nerveuse ce qui suggère une action
directe de la SE sur le nerf auditif. La période réfractaire absolue, voisine de 1 ms, confirme cette hypothèse. La technique du masquage passe-haut montre que les contributions sont irrégulières, et que l'action de la SE est prépondérante sur la base de la cochée : les calculs indiquent que, pour les SE les plus fortes, environ 10000 fibres sont stimulées simultanément, ce qui correspond grossièrement à une stimulation de l'ensemble du ler tour basal de la cochée.

La présente technique permet de relier l'emplacement de l'électrode et la forme de la SE choisie au nombre et à la nature des fibres stimulées. Plusieurs applications importantes peuvent en résulter : étude de l'efficacité en fréquence d'une SE et de son optimisation éventuelle, détermination objective de la fonctionnalité résiduelle du nerf auditif chez le sourd profond (stimulation auditive artificielle). Des expérimentations animales indiquent en effet que l'on peut extraire les réponses du tronc cérébral déclenchées par SE à l'aide d'une méthode de masquage non acoustique, d'où les possibilités d'application au cas de surdités profondes d'origine cochliésaire. Par ailleurs, cet enregistrement précis de la réponse unitaire doit apporter une contribution sensible à la technique de déconvolution, permettant l'analyse fine du fonctionnement fréquentiel et dynamique de la cochée, complément essentiel aux examens E.Coch.G., dont l'analyse de forme est jusqu'à présent restée limitée à l'observation visuelle.

Remerciements :

Ce travail a pu être réalisé grâce au concours financier de l'INSERM, CRL 81-6-037, et à la collaboration technique de J.-P. ERRE, que nous tenons à remercier ici.

Bibliographie


Légendes des figures

**Fig. 1** : Principe d'extration du PAE. (a) moyenne de l'ensemble réponse + artéfact en l'absence de bruit masquant. (b) même procédure que (a), mais en présence d'un bruit blanc à large bande destiné à masquer la réponse nerveuse. (c) différence (a)−(b). La réponse nerveuse (PAE) subsiste seule. Noter le rapport d'amplitude entre PAE et artéfact.

**Fig. 2** : Influence de l'intensité de SE sur la forme et l'amplitude du PAE (a) et sur les réponses évoquées corticales (b). Noter la stabilité de la forme des PAE et leur croissance régulière en fonction de l'intensité de la SE.

**Fig. 3** : Adaptation du PAE à des trains de deux SE pour divers intervalles entre stimulations (IES).

**Fig. 4** : Influence du spectre du bruit masquant sur le PAE. Les réponses sont extraites à l'aide de bruits passe-haut à diverses fréquences de coupure.
RECEPTION ACOUSTIQUE SACULAIRE CHEZ LE COBAYE

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A l'occasion d'une étude chez le Cobaye sur l'ototoxicité d'un antibiotique aminoglycosidique, l'Amikacine, des phénomènes paradoxaux ont été observés: des réponses électrophysiologiques évoquées acoustiquement pouvaient être enregistrées alors que les contrôles histologiques révélaient une destruction complète des cellules sensorielles de la cochlée (figure 1).

Une série d'études a alors été entreprise afin de déterminer les caractéristiques propres de ces réponses, leur origine et les structures du système nerveux central mises en jeu.

Les réponses obtenues à la périphérie, sur la fenêtre ronde, présentent une morphologie tout à fait différente non seulement des réponses normales mais aussi des divers modèles observés dans des cas de pathologie variés (figure 2). Diverses propriétés indiquent que ces réponses sont de type nerveux et non de type microphonique. En effet elles ne s'inversent pas lorsqu'on change la stimulation acoustique de raréfaction à condensation. Elles peuvent être masquées totalement par un bruit blanc, ce sont les composantes de fréquence moyenne qui semblent efficaces. Enfin une anoxie provoque la disparition totale de la réponse en deux minutes. On a établi sans équivoque ultérieurement que ces réponses reflètent une activation du nerf auditif puisqu'on a pu enregistrer des réponses évoquées au niveau du tronc cérébral et du cortex. La particularité majeure de ces réponses périphériques est leur très courte constante de temps. Elles apparaissent avec une latence de 0,3 millisecondes beaucoup plus courte que celle de 0,7 milliseconde des réponses les plus précoces observées normalement. De plus, ces potentiels ne présentent pas d'adaptation pour des rythmes aussi rapides que 200 stimulations par seconde alors que normalement il existe une très forte adaptation des 100 stimulations par seconde. Ces données mises en relation avec des observations anatomiques suggèrent que des synapses électriques sont mises en jeu.
Figure 1 :

Cytocochléogrammes présentant les résultats des comptages cellulaires des deux catégories de cellules ciliées sensorielles, externes et internes, ainsi que des cellules nerveuses du ganglion spiral chez un cobaye normal et un traité à l'Amikacine. Les données de cette figure et des suivantes proviennent des deux mêmes animaux.

La détermination de l'endroit où s'effectue cette transduction acoustico-nervueuse a pu être faite en comparant les résultats obtenus avec d'autres traitements ototoxiques détruisant de façon totale ou partielle les receptraires vestibulaires. Dans le modèle original avec l'Amikacine on avait pu observer que le vestibule semblait fonctionner normalement puisque des réponses nystagmiqques normales étaient obtenues, de même au contrôle histologique tous les récepteurs vestibulaires paraissaient intacts. Cela poussait à penser qu'il s'agissait d'une réception acoustique vestibulaire. Par administration locale de Sisomicine on a produit une destruction totale cochléaire et vestibulaire, aucune réponse évoquée acoustique ne pouvait alors être obtenue. Par injection de Gentamicine on a pu produire une destruction cochléaire totale, une destruction très importante des ampoules et de l'utricule et très faible du saccule, dans ces cas les réponses particulières étaient observées. Ces résultats indiquent que très certainement ces réponses de latence très courte proviennent du saccule. On sait par des observations ultrastructurales que les cellules ciliées de type I pourraient avoir des synapses électriques avec les fibres nerveuses, on pense que ces réponses sont la démonstration de leur mise en jeu.

On a enregistré par électrode de surface les réponses évoquées du tronc cérébral dans ce modèle expérimental. Les potentiels évoqués observés étaient tout à fait différents de ceux normalement obtenus. En effet ils ne présentaient que trois ondes successives dans les 3 milliseconds suivant la stimulation au lieu de la série d'ondes normale allant jusqu'à 5 ou 6 milliseconds. Une étude anatomofonctionnelle par déoxyglucose marqué au carbone 14 semble indiquer une activation de quelques noyaux de la voie auditive. Des enregistrements en montage différentiel ont permis de localiser une activation corticale spécifique au niveau du lobe temporal dans la région même de l'aire auditive primaire (figure 3). Cette activation du cortex auditif suggère qu'il existe une sensation consciente et auditive.
L'analyse des performances de cette réception acoustique sacculaire la fait apparaître aux réceptions auditives et non d'équilibraîon. En effet, on a pu montrer que cette réception s'effectuait pour toutes les fréquences de la gamme auditive normale du cobaye et aussi que des propriétés de filtre de type passe-bas pouvaient être mises en évidence. De plus, on a montré que la sensibilité différentielle en intensité était tout à fait comparable à celle du normal, c'est-à-dire de l'ordre de 1 à 2 décibels environ. Cette excellente capacité discriminative renforce l'idée que cette réception est bien fonctionnelle.

D'un point de vue phylogénétique il n'est pas surprenant que le saccule puisse avoir une fonction auditive chez les mammifères. On sait que le saccule a certainement une fonction d'équilibraîon mais plusieurs travaux expérimentaux indiquent aussi une sensibilité acoustique tout à fait comparable à celle que nous avons étudiée. Chez un petit nombre de sujets humains des enregistrements par électrococchléographie trans tympanique indiquent des réponses anormalement courtes. Les stimulations acoustiques qui les produisent ont été décrites comme sonores par ces divers sujets.

Figure 2 :

Potentiels évoqués acoustiques mont par un clic à 80 dB au-dessus du seuil normal enregistrés à la fenêtre ronde chez un cobaye normal et un traité à l'Amikacine.

Figure 3 :

Potentiels évoqués acoustiques obtenus au cortex auditif chez un cobaye normal et un traité à l'Amikacine. Stimulation par clic à 80 db au-dessus du seuil normal.
REFERENCES


INPUT-OUTPUT CHARACTERISTICS FOR EVOKEO OTOACOUSTIC EMISSIONS

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Introduction

A miniature microphone into the ear canal of humans and of several other
mammals (1-6, 10-13) reveals the presence of an acoustical response follow-
ing acoustical stimulation. Although the origin of these emitted sounds is
still unclear, many indications suggest that otoacoustic emissions are ge-
erated within the cochlea, probably at the level of the mechano-reception
(1, 5-6, 10-11). Responses to short duration stimuli last some tens of mil-
iseconds and take the form of decaying oscillatory waveforms.

The responses evoked by clicks are dominated by one frequency (f_d all -
Throughout the paper) seldom by two or more frequencies (12). Intersubjec-
tive variability of response waveforms and frequency is very high; by contrast,
responses from a given subject are highly repetitive and remain stable from
session to session.

In a previous study (3), otoacoustic emissions evoked by clicks and 1kHz
bursts have been analysed in detail. Aim of the present researches was to
extend the analysis to emissions evoked by tone bursts at frequencies approx-
imately tuned to 1/2 f_d, f_d and 2f_d.

Materials and Methods

A probe containing a miniature earphone (Knowles BK-2606) and a conden-
sor microphone (Knowles BK-1751) was sealed into the ear canal (for more de-
tails, see (2-3)).

The sound source was driven with electrical pulses of 0.1 ms duration
(rarefaction clicks) and tone bursts at 0.5, 1 and 2kHz with a plateau dur-
ation of 3 ms and linear rise and decay (0.5 ms). Figure 1 shows the power
spectra of the acoustical responses within the time window 0-5ms. Stimuli
were delivered to the test ear at a constant rate (13/s). Responses were am-
plified, bandpass filtered 200-3500 Hz (Butterworth, 1 zero-2 poles), sampl-
ed over 512 points within a time window of 30.72 ms, averaged over 1500-2000
repetitions and stored on floppy disks for offline analysis. Further proces-
sing included zero phase shift bandpass digital filtering (Butterworth,
500-2500 Hz, 48 dB/oct). Stimulus generation and data acquisition were under
the control of an AMPLIAD MK6 system.

The subjects were young adults, aging 22-24 years, with normal hearing.
The subjects were lying on a bed in a quiet laboratory room and instructed
to remain relaxed and not to move the head nor swallow during the recording
session.

To reduce the number of all the combinations emission "frequency"-burst frequency, the analysis was restricted to a group of 6 subjects with click responses "tuned" to a frequency \( f_d \simeq 1 \text{ kHz} \) (as determined from computation of the rate of zero crossings or by analysis of the Fourier transforms).

Results

A typical example of responses to tone bursts at 0.5, 1 and 2 kHz, from one representative subject, is given in Fig. 2. Stimulus level was set at 50 dB SPL (re 20 \text{ \mu Pa}). For comparison, the response to clicks, at the same peak-to-peak level, is shown on the top of Fig. 2. There are some similarities among the four responses, for duration, overall waveshape and frequency content. To stress these similarities, the responses of Fig. 2 have been shifted along the time scale, with respect to the click response. The shift was 0.7, 2.1 and 4.04 ms, for the responses to 2, 1 and 0.5 kHz, respectively. These shifts have been determined from analysis of the crosscorrelation -grams between any given burst response and the click response (see also Discussion). Means and standard deviations from the group of subjects were: 0.75 \pm 0.09; 2.09 \pm 0.10 and 4.0 \pm 0.5, for 2, 1 and 0.5 kHz bursts, respectively. This behaviour was typical for all the subjects, at any intensity level. The responses to 0.5 kHz bursts, at moderate to high intensities, approximately above 50 dB SPL, show a 0.5 kHz oscillation superimposed to the component at \( f_d \) (see also the recording of Fig. 2).

Stimulus amplitude-response magnitude relationships (means and standard deviations) are given in Fig. 3. Magnitudes have been calculated as the peak-to-peak values of the response waves falling within the time window indicated in each panel. Dashed line indicates a linear input-output relationship (constant gain). Deviations from linearity are maximal for the responses at 2 kHz; Fig. 2 shows that the responses to 1 kHz are linearly related to the input intensity at the lowest intensity levels (see also 3, 8, 10, 13).

Analysis in the frequency domain reveals other interesting properties of the emissions evoked by clicks and tone bursts. As an example, Fig. 4 shows the first 256 points of the power spectra of click and 1 kHz evoked responses (time window 10-30 ms, cosine tapering on 20 points). The example was chosen among the responses of subjects whose click evoked emissions did not show a relatively pure frequency content; however, in this case also, all the relevant spectral maxima for 1 kHz and click emissions are clustered around the same frequency values. In addition, the slope of the spectrum, beyond the maximum, is always greater for 1 kHz evoked emissions than for clicks (see also Fig. 4).

Discussion

Responses to clicks and tone bursts can be interpreted, at a first approximation, by assuming the presence, within the cochlea, of a selective filter tuned to \( f_d \). Consider, in fact, a filter with a frequency response characterized by a high degree of tuning; it will produce a transient oscillatory response whose spectral maxima are clustered around \( f_d \), irrespective of the frequency of the burst. Analysis of temporal waveforms and their spectra seems to confirm this view (see also (3)). As reported to comment data of Fig. 2, responses to tone bursts show a "group delay" which is a (decreasing) function of the burst frequency. There is no ready explanation
for that. A possible interpretation is that emissions can be generated by any cochlear place; each "local generator" is activated by the stimulus with a delay which is a function of the travel time along the cochlea: the higher the stimulus frequency, the shorter will be the delay of the response of the "local generators". The frequency characteristics of all the "local generators", for any individual subject, are approximately the same at any cochlear location and could be a reflection of some transversal mode of vibration of the cochlear structures. It is however clear from a growing body of evidence that the behaviour of otoacoustic emissions suggests the presence of a nonlinear (intensity-dependent) time-varying frequency selective mechanism (2-3) which appears to be similar, in principle, to that revealed by basilar membrane data (7,9).

References

GRANDORI - Otoacoustic emissions

Figure 1

Figure 2

Figure 3

Figure 4
DISTORTION PRODUCTS DUE TO TWO-TONE STIMULUS ON NONLINEAR MODEL FOR THE INNER EAR

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Introduction:
A nonlinear phenomenological model which represents the motion of a point on the basilar membrane in the inner ear after Kim/1/ was re-examined /2/. It has proved its validity as a useful representation of events in the cochlea.

The object of this paper is to study the response of the model to a two-tone stimulus and to investigate the features of the resulting combination tones, which are known as distortion products (DPs). The effects of varying the amplitude as well as the frequency spacing of the input signal on the distortion products are also discussed.

Two-tone stimulus on the model:
The nonlinear model for the inner ear consists of ten cascaded unilaterally coupled elements, each of which is defined by a second order nonlinear differential equation:

\[ \ddot{x}_i(t) + 2D_{1}(1 + \eta_1^2(t))\dot{x}_i(t) + \omega_0^2x_i(t) = Bx_{i-1}(t) \]

where \( i=1,2,...,10 \), \( x_i(t) \) is the input function and \( x_{10}(t) \) is the output function from the model. \( D_1, \eta_1 \) and \( B \) are constants.
\( \omega_0 \) is the resonant angular frequency of the \( i^{th} \) element.

A two-tone stimulus of the form \((A \sin w_1 t + B \sin w_2 t)\) was introduced into this nonlinear model. The model was solved in the time domain first using the RKF 45 subroutine based on Runge-Kutta Formula; the 6th order; method /3/, then the steady state was analysed using the FFT algorithm to get the magnitude of the different resulting frequency components. Then each component was defined by its relation to \( f_1 \) and \( f_2 \). This procedure was repeated for several points on the basilar membrane having different characteristic frequencies. Table (1) shows the relation between the characteristic frequency and the distance from the stapes as given by Zwicker /4/.

Results:
The solution of the model has been done for eight points
on the basilar membrane by taking $A=C=48\ \text{dB} (\text{re } 2^{-19}\ \text{according to Kim/1/}) = -66\ \text{dB} (\text{re } 1\ \text{according to Hall/5/})$. This was repeated by increasing the amplitudes of the input signal to $94\ \text{dB} (\text{re } 2^{-19})$ i.e. $-22\ \text{dB (re 1)}$ while $f_1$ and $f_2$ were remained constant at $1\ \text{KHz}$ and $1.56\ \text{KHz}$ respectively.

The prominent distortion product $(2f_1-f_2)$ was obtained as well as a large number of another distortion products such as $(2f_2-f_1)$, $(2f_1+f_2)$, $(2f_2+f_1)$, $3f_1$, $3f_2$, $(3f_1-2f_2)$, $(3f_2-2f_1)$, $(4f_2-f_1)$, $(4f_1-f_2)$, $(4f_2-3f_1)$.

Figures 1, 2 and 3 show the spatial distribution for three different types of these distortion products. The quantity $(K)$ in the figures represents the ratio between the output amplitude and the input amplitude in dB.

Figures 4, 5 and 6 represent the effect of varying the frequency spacing between the input frequencies on the spatial distribution of the same three distortion products. These are obtained for $A=C=48\ \text{dB} (\text{re } 2^{-19}) = -22\ \text{dB (re 1)}$ and the frequencies are changed to $f_1 = 1\ \text{KHz}$ and $f_2 = 1.27\ \text{KHz}$.

Discussion and Conclusion:
From figs. 1, 2 and 3, which represent the effect of varying the amplitude of the input signal on the spatial distribution curves of some distortion products with considerable amplitudes, it is noticed that as the input level increases, the output to input ratio of the displacement patterns decreases. We notice also that some curves were basally transferred and others were apically transferred. We can conclude that the transfer of the curves is related to the changes which are happened in the curves of $f_1$ and $f_2$ due to the increase of the input level.

Figures 4, 5 and 6 represent the effect of varying the frequency spacing between the frequencies of the input signal on the spatial distribution of the same distortion products as in figures 1, 2 and 3. We notice that the variation in the relative level of the distortion products was not so apparent as by those in the case of varying the amplitude. We notice also that below $1\ \text{KHz}$ and above $2\ \text{KHz}$ the two curves at different frequency spacing for each distortion product (DP.) nearly coincide. Therefore, we can conclude that the nonlinear effects are not at all pronounced away from the region whose characteristic frequency lies around the frequency by which we stimulated the model.

References:


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Table 1: Relation between the characteristic frequency of a certain point on the basilar membrane and its distance from stapes.

Fig. (1) Spatial distribution of DP. \( (2f_1 - f_0) \)

Fig. (2) Spatial distribution of DP. \( (3f_1) \)
ABDEL ALIM & SHAAT
Nonlinear model for the inner ear

Fig. (3) Spatial distribution of DP. \((3f_2 - 2f_1)\)

Fig. (4) Spatial distribution of DP. \((2f_2 - f_1)\)

Fig. (5) Spatial distribution of DP. \((3f_1)\)

Fig. (6) Spatial distribution of DP. \((3f_2 - 2f_1)\)
AN IDEAL LATERAL INHIBITION FUNCTION IN HEARING

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INTRODUCTION

It is a widely accepted fact that the broadly tuned mechanical system of the basilar membrane of the inner ear cannot alone account for the observed frequency resolution in hearing. Some additional mechanism is required to further improve the frequency selectivity of hearing and make it comparable with the frequency discrimination observed in electrophysiological and psychoacoustical experiments. This frequency sharpening mechanism is often referred to as the "second filter". One of the possible candidates for such a mechanism is lateral inhibition (LI), a common feature of all sensory organs (Békésy, 1967). LI enhances contrast of the perceived stimulus patterns by sharpening the edges and peaks of the neural excitation distribution as a result of lateral inhibitory interconnections between sensory neurons. Nomoto, Suga, and Katsuki (1964), Sachs and Kiang (1968), Liff and Goldstein (1970), Arthur, Pfeifer, and Suga (1971) offer further electrophysiological, and Houtgast (1972, 1974) psychophysical, evidence for LI in hearing.

Several questions emerge in investigation of the limits of sharpening of the LI process. What is the shape of a LI function which would completely remove the spreading effect of the basilar membrane prefiltering? If such an ideal LI function were obtained, we should ask to what degree it can be realized by the known or at least possible mechanical and neural interactions along the auditory channel? This study offers an answer to the first question, irrespective of any criteria of physiological plausibility.

MODEL DESIGN

The computation of the optimal LI function is based on matrix algebra, verified on a digital computer. The following simplifying assumptions are made: due to the discrete nature of this model all functions are defined only for the sample points and the frequency resolution is limited by the size of the sampling interval Δx; only stationary signals are considered; LI is assumed as instant acting, with neither memory elements nor synaptic or propagation delays.
incorporated into the model; LI is regarded as a single stage operation; the model is treated as a noise-free linear time-invariant system with no saturation of the neural activity; only one type of receptor is considered and their output is assumed to be proportional to the envelope of the displacement of the basilar membrane; output activity of the neurons is treated as being proportional to the sum of excitatory and inhibitory input activities; the model neurons have both excitatory and inhibitory axon terminals; spontaneous neural activity is considered as a dc bias allowing transmission of negative excitation values within the neural network; efferent neural pathways are not represented in the model.

For our model the forward type of LI was selected as it requires only afferent pathways and precludes potentially unstable feedback loops and undesirable spreading of the inhibitory effect (Keirchardt and Mac Ginitie, 1962) and simplifies considerably our mathematical model. The neural network is a simplified version of a LI network presented by Békésy (1967, Fig.23). The tonotopic relationship between frequency and place is preserved in ascending neural pathways running in parallel. The input row of receptors represents an array of hair cells located on the basilar membrane. The input signal is a pure tone which can be represented by a spectrum containing one spectral line. Since the basilar membrane of the inner ear performs a frequency to place transformation, the spectrum of the input signal can be more conveniently regarded as $S(x)$, i.e. as a function of the spatial coordinate $x$ along the length of the basilar membrane.

The displacement of the basilar membrane represents the input to the sensory cells. These are regarded as linear transducers and their output neural activity $Y(x)$ as identical to their input. Due to mechanical and hydrodynamical couplings within the cochlea this stimulation $Y(x)$ is wider than the spectrum of the input signal $S(x)$. This function is given as

$$Y(x) = S(x) \ast H(x) = \int_{-\infty}^{\infty} S(\xi) H(x-\xi) \, d\xi. \quad (1)$$

In this convolution $\xi$ is the place shift along the basilar membrane and $H(x)$ the line response of the basilar membrane, i.e. its vibration envelope response to pure tone stimulation.

The LI process is assumed to reduce the spreading effect of $H(x)$ by the action of lateral inhibitory branches to the next higher layer of neurons. The purpose of these interconnections is to suppress neural activity in the neighbouring channels. The output $Z(x)$ of the LI operation,

$$Z(x) = Y(x) \ast L(x) = \int_{-\infty}^{\infty} Y(\xi) L(x-\xi) \, d\xi, \quad (2)$$

is given as a convolution between $Y(x)$ and the LI function $L(x)$ describing the polarity and weighting factors of the lateral couplings. Generally, $Z(x)$ is narrower than $Y(x)$. In an ideal case $Z(x)$ would be identical to or at least indistinguishable from $S(x)$. 

A.J. Rozsypal: An Ideal Lateral Inhibition Function in Hearing

The LI function is usually presented in the literature as having one positive peak corresponding to the excitation area surrounded by one negative peak on each side, representing the inhibitory areas. Such functions offer sharpening at the most by a factor of about 2.7 (Deutsch, 1977; Rozsypal, 1980).

IDEAL LATERAL INHIBITION FUNCTION

For the discrete solution attempted here Eq. (1) and (2) can be rewritten in matrix form:

\[
\{Y\} = [H] \{S\} \quad \text{and} \quad \{Z\} = [L] \{Y\}.
\]

(3) (4)

In these equations expressions in braces are column vectors of \(n\) elements representing the values of corresponding spectra uniformly sampled in \(n\) points along the space coordinate \(x\). Symbols in square brackets for convolution operators represent \(n\)-dimensional square matrices. Each row of the \([H]\) matrix represents one version of the normalized line response with its peak positioned at the main diagonal. Similarly in matrix \([L]\), each row represents the normalized LI function shifted along the main diagonal so that its principal positive peak is always centered at the main diagonal element. The inner product between the square matrix and column vector in Eq. (3) and (4) represents a discrete convolution.

In the ideal case the second convolution, Eq. (2) and (4), should completely remove the spreading effect of \(H(x)\) given by Eq. (1) and (3). Merging Eq. (3) and (4) produces

\[
\{Z\} = [L] [H] \{S\}
\]

(5)

An identity between \(Z(x)\) and \(S(x)\) can be obtained if in Eq. (5)

\[
[L] [H] = [I]
\]

(6)

i.e. if the above product is equal to the identity matrix in which all main diagonal elements are equal to one and the remaining elements are zero. This is the case if \([L]\) is an inverse matrix of \([H]\). Such a matrix represents an ideal LI operator. Each of its rows represents an ideal LI function centered around the main diagonal element.

Fig. 1 represents six cases of ideal LI functions (solid points) corresponding to Gaussian envelopes \(Y(x)\) of different frequency bandwidth and skewness (dashed lines). In all cases LI results in a single nonzero point located at the peak of input stimulation.

DISCUSSION

It would require a neural network of considerable complexity to realize the ideal LI functions obtained in this study. Easier to visualize would be a multilayer inhibitory operation (Weston, 1974) with a relatively simple individual LI function at each stage. The combined effect of these stages can produce the same effect as a single-stage operation with the ideal LI function presented here.
Fig. 1. Ideal LI functions (solid points) corresponding to stimulation (dashed lines) of varied bandwidth and skewness.

Spatial frequency of the ideal LI function, i.e. the width of the alternating excitatory and inhibitory areas, is determined by the required resolution characterized by $\Delta x$. This frequency is independent of the stimulation bandwidth. Skewed excitation functions generally require simpler LI functions.

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COMPUTING VIRTUAL DIPOLAR SOURCES OF BRAINSTEM EVOKED RESPONSES FROM SURFACE INFORMATION

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Introduction

Electrophysiological investigations of auditory functions with surface electrodes are primarily aimed to extract information about the neural sources from the measurement of electrical field generated on the scalp. In common practice, a unique potential difference is recorded; peaks and valleys of the response are usually taken as a measure of the underlying neural activity. Although analysis of the waveforms allows, at least under certain circumstances, the definition of criteria for normalcy or abnormalcy of the response, little is known about the relationship between the potentials on the surface and the neural sources of the auditory evoked brainstem response.

In the present paper, the electric fields generated on the scalp by intracranial sources are studied by means of a first approximation model of the head. This approach may provide a basis for the explanation of some salient features of the response; in particular, the model will be used to investigate upon the factors influencing the intersubjective variability of the response amplitude.

The volume-conducted activity of intracranial sources

For spontaneous EEG signals and some sensory evoked potentials, the relationship between electric fields on the scalp and their sources has been studied in several theoretical and experimental papers (see, e.g., Geisler and Gerstein, 1961; Rush and Driscoll, 1969; Schneider, 1972; Henderson et al., 1975; Kavanagh et al., 1978). Attempts at modeling the fields produced by the auditory brainstem nuclei were recently proposed by the present author (Grandori, 1982 a,b).

The calculation of potentials on the scalp, given certain assumptions for the sources (dipoles, multipoles, active layers), the geometry of the head and the properties of the media embodied within outer surface, does not offer serious theoretical difficulties. Restricting the analysis to a spherical shape, several analytical solutions have been proposed, for the homogeneous (see, e.g., Wilson and Bayley, 1950; Brody et al., 1973) and the inhomogeneous case (see, e.g., Rush and Driscoll, 1969; Arthur and Geselowitz, 1970; Kavanagh et al., 1978), with various kind of sources.

In the present paper, the head has been modeled as a concentric shell structure consisting of a homogeneous sphere of neural tissue (radius r_1 and
and conductivity $\sigma_1$) surrounded by two concentric spherical shells (outer radii $r_1$ and $R$, respectively) of differing conductivities ($\sigma_2$ and $\sigma_3$) representing the skull and the scalp (see fig. 1). Note that reactive effects are known to be negligible for frequencies lower than about 10 kHz (Plotney, 1969). The coordinate system was chosen with the origin at the center of the sphere that best approximates the subject's head, under visual estimation. The positive $z$ axis passes through $C_2$; on an average head, the positive $x$ axis extends approximately through $C_2^-$ (the negative passing through $F_3$), while the positive $y$ axis passes approximately above the right ear. $P_z$

The potential on the three-sphere model can be computed in resolving a boundary value problem. The method of calculation is the same as in Arthur and Geselowitz's study (1970) for cardiac generators; the same method was used also by Ary et al. (1981).

Consider a dipole (see fig. 1) with radial and tangential components $m_r$ and $m_t$ embedded in the innermost sphere and located at a distance $Z$ from the center of the sphere, on the $z$ axis; the dipole moment is in the positive $xz$ plane. Any other dipole position and orientation can be obtained by rotation. Given a point $A(R,\alpha,\beta)$ on the surface of the outermost sphere, let us put successively:

$$b = \frac{\sigma_2}{r_1}$$

$$\xi = \frac{\sigma_2}{\sigma_3}$$

$$\beta_1 = r_1/R; \quad \beta_2 = r_2/R.$$ The potential in $A$, $V(R,\alpha,\beta)$ is given by

$$V(R,\alpha,\beta) = \frac{1}{4\pi \sigma_3} \sum_{n=1}^{\infty} \sum_{m=-n}^{n} \frac{(2n+1)b^{n-1}}{(2n+1)^{2/3}} \cdot P_n^{(1)}(\cos \beta) A_n$$

with $A_n = \left\{ \begin{array}{ll}
\frac{P_n(\cos \alpha)}{(n+1)!} & \text{for } m = m_r \\
(1/n) \cdot \cos \beta \cdot P_n^{(1)}(\cos \alpha) & \text{for } m = m_t
\end{array} \right.$

where $P_n(\cos \alpha)$ and $P_n^{(1)}(\cos \alpha)$ are Legendre and associated Legendre polynomials, and

$$F_n = \xi(2n+1)^2/[d_n \cdot (n+1)]$$

$$d_n = \left[(n+1)\xi+n\right]^{1/2} \cdot \left[n+1\right]^{1/2} + \left[1+(1-\xi)\left(n+1\right)\xi+n\right]^{1/2} \cdot \left[f_1^2-f_2^2\right] \cdot \left[1-(1-\xi)^2f_1^2/f_2^2\right]^{1/2}$$

Rush and Driscoll (1969) have estimated radii and conductivities for the three-sphere model to reproduce an "average" adult human head. They suggest the following values: $r_1 = 8.0$ cm; $r_2 = 8.5$ cm; $R = 9.2$ cm; $\xi = 0.0125$; $\sigma = 54.5 \cdot 10^{-3}$ (n cm)$^{-1}$.

Effects of parameters changes on the "vertex-mastoid" potential difference

Amplitude of the response waves are known to show a rather large inter-subjective variability, in particular if compared with the small test-re-test variability from a given subject. This aspect can be easily investigated with the aid of the present model, with the following procedure:
GRANDORI - Evoked electrical fields

i) auditory brainstem potentials were recorded in response to clicks of 100 dB SPL (peak) from a group of seven normal hearing subjects; the peak-to-peak amplitude for wave I, II and III have been computed from the vertex positive peak to the following negative peak. Means and standard deviations were (in uV): wave I: 156±88; wave II: 143±70; wave III: 396±130. These data are consistent with the results obtained under similar recording procedures (see, e.g., Thornton, 1976);

ii) assuming the generators for wave I, II and III to be represented by dipolar sources, their parameters (moment and location) have been separately estimated from the potential fields measured on the scalp of a given subject (for details of the method, see (Grandori, 1982 a,b); see also (Schneider, 1972; Kavanagh et al., 1978));

iii) with these dipoles as sources, the three-sphere model was used to compute the potential difference between the "vertex" and the "mastoid" of the synthetic head (in rectangular coordinates (cm): (0,0,9.2) and (2.16, 8.33; -3.33), respectively);

iv) then the following parameters have been systematically varied: coordinates of the "vertex" and "mastoid", location of the dipole centre, dipole orientation, radii of the shells, conductivity σ and ratio σp/σ.

Each parameter was singularly perturbed to determine the range of variation producing the entire experimental standard deviations for the amplitude.

Studies on the sensitivity of the model parameters are still in progress; the most significant results obtained are the following. Dipoles and electrodes locations were among the most critical parameters; changes of ±1.3 - ±1.9 cm of relative dipole-electrode distance (in particular for the mastoid for wave I and II) were able to produce the observed changes of amplitude. Less critical is the situation for changes of σp/σ and σ; at a first approximation, variations of σ are linearly related (inverse proportionality) to the variations of the potential amplitude.

References
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Fig. 1 - Coordinate system and parameters adopted in the 3-sphere model
THE DATA OF MECHANICAL IMPEDANCE ON HUMAN HEAD

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A miniature excitor vibrates with sinusoidal signal at frequency range of 100 - 1000 Hz driven from B.F.O. It excites a impedance head which both with force and acceleration output terminals. Convert acceleration into velocity and keep it constant. Three mechanical impedance curves were recorded from mastoids and forehead for every subject. 444 persons were measured and their ages ranged from 5 to 80 years old. In order to compare with other reports, the measuring system was calibrated as pure mass controlled [1 - 8].

RESULTS

1. The Relations Between Impedance And Age

The measuring results show that there is correlation between impedance value and age at frequency range 100 - 1000 Hz and age range 5 - 65 Y. as given below:

\[ Z_{100} = 37.1 - 0.10 \ Y \]
\[ Z_{150} = 35.2 - 0.10 \ Y \]
\[ Z_{200} = 33.4 - 0.11 \ Y \]
\[ Z_{300} = 31.0 - 0.12 \ Y \]
\[ Z_{400} = 29.0 - 0.12 \ Y \]
\[ Z_{500} = 27.3 - 0.12 \ Y \]
\[ Z_{600} = 25.9 - 0.12 \ Y \]
\[ Z_{700} = 24.7 - 0.12 \ Y \]
\[ Z_{800} = 23.4 - 0.12 \ Y \]
\[ Z_{900} = 22.5 - 0.12 \, Y \]
\[ Z_{1000} = 21.4 - 0.12 \, Y \]

2. The Relation Between Resonance Frequency of Impedance Curve And Age

Another characteristic of impedance curve from mastoid is that the resonance frequency shifts toward low frequency when the age of subject increases. And a statistical equation is approximately given below:

\[ F \, (\text{Resona. freq. Hz}) = 3.1 - 0.17 \, Y^{1/2} \, (\text{Age, Y}) \]

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3.2

Psychoacoustique - Psychophysiology
Psychoacoustics - Psychophysiology
Psycho-Akustik - Psychophysiology
MESSAGES ELEMENTAIRES DE SONS PURES, MASQUAGES ET BATTEMENTS.

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Il y a plus de douze ans que T.S. Korn (décédé en 1980) a répondu à une des questions essentielles de la psychoacoustique : Quelle est la forme temporelle du stimulus acoustique le plus bref susceptible de créer une sensation unique de son pur, de durée subjectivement nulle ?

Korn a montré que le stimulus \( t^\alpha \exp(-\alpha t) \cos(\omega t + \varphi) \) répond à cette condition et possède les propriétés résumées ci-après :
- quelle que soit la valeur de \( \alpha \), on ne perçoit pas de transitoire d'attaque distinct, ce qui n'est pas le cas de stimulus dont l'enveloppe est différente de \( t^\alpha \exp(-\alpha t) \) (même en \( t \) ou en \( t^3 \), et, a fortiori, la forme gaussienne);
- lorsque pour une fréquence donnée, le facteur d'amortissement est suffisamment faible pour que le stimulus ait une durée perceptible, la sensation auditive qu'il provoque est celle d'un son pur, c'est-à-dire que la hauteur et le timbre perçus sont identiques à ceux d'un son sinusoidal de même fréquence;
- lorsque l'on augmente la valeur de \( \alpha \), la durée physique du stimulus diminue, de même que sa durée apparente, mais celle-ci devient subjectivement nulle pour une valeur de \( \alpha \) située, pour chaque individu, dans une plage assez étroite : on entend alors un "pip" toujours pur. Le stimulus correspondant à la valeur médiane \( \alpha \) ainsi définie, a été dénommé par Korn message élémentaire de son pur (MESP, elementary message of discrete frequency);
- pour des valeurs de \( \alpha \) supérieures à \( \alpha \), le timbre perçu s'altère progressivement, ces stimulus produisant de plus en plus l'effet audible de bruits plus ou moins "colorés" (toujours sans durée perceptible), c'est-à-dire dont la hauteur tonale reste, selon les individus, plus ou moins longtemps reconnaissable.

Ce n'est donc que pour des valeurs de \( \alpha \) comprises entre des limites très étroites que la sensation présente un caractère d'unité auditivement indivisible, de hauteur définie et de durée subjective nulle.

On voit que le MESP est le signal le plus court qui apporte une information audible sur la hauteur et la pureté d'un son, identiques à celles d'un signal stationnaire de durée infinie.
Des valeurs expérimentales de $\alpha$ ont été déterminées, depuis 1973 au laboratoire d'acoustique de la Faculté des Sciences Appliquées de l'Université de Bruxelles ainsi qu'au laboratoire de la Radio-Télévision Belge.

L'existence, pour chaque fréquence audible (à l'exception, jusqu'à présent, des fréquences très graves $< 65$ Hz et très aiguës $> 5$ kHz) d'un MESP a deux conséquences importantes :

- le fait que la durée subjective d'un MESP est nulle alors que sa durée physique est finie, constitue un effet de masque temporel, du raisemblablement à la nature et à l'évolution des contacts entre les cellules ciliées et la membrane tectoriale ;

- au niveau des fibres nerveuses, nous ignorons quel est, parmi l'ensemble de spectres successifs dus à un MESP, celui sur lequel s'exerce l'innervation latérale de G. von Békésy, mais une chose est certaine (en vertu de la propriété générale d'indétermination $\Delta f \Delta t$ qui affecte tout dispositif réalisable d'analyse spectrale) c'est que ce spectre désigné par $\Sigma(f)$ doit nécessairement être largement que le spectre $V(f)$ produit par un son pur stationnaire.

Ce spectre $\Sigma$ - tout au moins dans le voisinage de son maximum - paraît constituer dès lors la frontière, dans le plan amplitudes-fréquences, entre les spectres qui sont perçus comme des sons purs et ceux qui sont perçus comme des bruits ou des sons composés.

Dans cette hypothèse, considérons le spectre énergétique $< V(f) >$ d'un son pur de fréquence $f_0$. Si l'on additionne à cette énergie celle $< M(f) >$ d'un son indépendant de fréquence $f$, ce dernier sera masqué si la somme $< V(f) > + < M(f) >$ est inférieure à la limite constituée par $\Sigma(f)$ du son masquant de fréquence $f$. Se pose alors la question du rapport à établir entre le maximum spectral du son stationnaire $< V(f) >$ et celui du spectre $\Sigma(f)$.

Les résultats de Riesz, publiés il y a 55 ans, devraient permettre semble-t-il de répondre à cette question. Le but de cet auteur était la mesure du seuil différentiel audible d'intensité; on sais qu'il utilisait, à cette fin, les battements lents entre un son de fréquence $f_0$ et d'intensité fixe et un son de fréquence $f$ (très proche de $f_0$) et d'intensité juste suffisante pour que les battements de fréquence $f_0 - f$ soient audibles. On sais qu'il a constaté que les seuils les plus bas étaient obtenus pour $1,65 < f_H < 3,5 f$, d'où son choix de $f_H = 3 f$. En 1982, j'ai effectué quelques essais avec des moyens limites, sans mesure précise des niveaux absolu - confirment cependant ainsi des résultats obtenus par Riesz : lorsque l'intensité du son masquant de 1 kHz est maintenue à une valeur constante, les seuils d'audibilité des battements se relèvent :

- de - 34 dB à - 23 dB lorsque $1000,25 f < f < 1003$ Hz

(Riesz : - 27,5 dB à - 19,7 dB " $1000,2 f < f < 1003$ Hz)

- de - 29 dB à - 23 dB " $995,5 f < f < 1000$ Hz.

On peut tenter d'expliquer ces faits comme suit :

Le stimulus peut être représenté par la formule ci-après, où $a$ désigne l'amplitude relative du deuxième son :
\( s(t) = A(\cos 2\pi f_0 t + a \cos 2\pi f t) \)

ou encore
\( s(t) = A B(t) \cos \left[ 2\pi (f_0 + f) t/2 - \varphi \right] \)

où l'amplitude relative \( B(t) \) du battement est donnée par
\[
B(t) = \sqrt{1 + a^2 + 2 a \cos 2\pi (f - f_0) t}
\]

On montre aisément que la valeur absolue maximum de la dérivée par rapport au temps de l'amplitude relative vaut
\[
|B| = \max |\frac{dB}{dt}| = 2\pi a |f - f_0|
\]

Tout se passe donc comme si, pour les faibles valeurs de \( |f - f_0| \), cette grandeur \( |B| \) constituait une sorte de critère limite d'audibilité des battements.

On constate cependant en outre que lorsque l'on modifie le niveau du son masquant, les niveaux des sons masqués se modifient dans le même sens. On se trouve donc devant un comportement assez complexe que l'on ne peut que constater. Le rapport entre les maximums spectraux de \(<\sqrt{\alpha}\>\) et de \(<\Sigma^2>\) n'est donc pas très simple.

On sait également que la méthode de Riesz a été critiquée quant au but que cet auteur poursuivait. Quelque fondée que soit cette critique, des essais similaires ne semblent dès lors pas avoir été reproduits par qui que ce soit. Ils présentent cependant, à un point de vue différent, un grand intérêt, car ils font apparaître des seuils de perception qui, en fait, constituent une extension non discontinue des seuils de masquage dans cette zone de battements. Depuis Wegel et Lane (1924), on considère que, lors des mesures de masquage, les battements constituent un phénomène sinon gênant, tout au moins différent. Cette considération procède d'une idée "a priori" concernant la forme des courbes de masque, illustrée par les portions de courbes en traits interrompus tracées par les auteurs cités, dans le but de remplacer les deux maximums expérimentaux par un maximum au droit de la fréquence masquante. Or, il est évident et je le répète depuis des années, qu'un son ne peut se masquer lui-même, puisque deux sons de fréquence identique s'additionnent simplement en fonction de leur phase relative. Il en résulte qu'il est normal et logique que les spectres de masque entre sons purs présentent deux maximums de part et d'autre de la fréquence du son masquant, et que les spectres d'iso-réponse présentent deux minimums de part et d'autre de la fréquence du son test, comme cela apparaît clairement dans toutes les publications, de Wegel et Lane à Vogten (1974).

Quoiqu'il en soit, ces constatations ne modifient en rien les conclusions générales que Korn a déduites de la réalité des effets de masque temporel et fréquentiel - conséquences de l'existence des MESP : -

C'est grâce à ces effets que la fonction auditive se montre capable d'absorber en temps réel les débits d'information acoustique qui nous parviennent dans la vie de tous les jours. Sans cet effet de masque temporel et sans l'inhibition latérale dont l'effet de masque fréquentiel est la conséquence directe, la parole et la musique ne seraient que des bruits indistincts.
Pour conclure, je désire formuler trois voeux :

1. Que des laboratoires bien équipés reprennent l'étude du MESP (détermination du critère), en faisant varier la fréquence et l'intensité de crête. Je suis à leur disposition pour leur fournir tous les éléments en ma possession :

   Les stimulus de Korn sont aisés à réaliser :
   - soit par voie analogique :
     - trois circuits en cascade, ou encore :
     - multiplication d'une fonction enveloppe par une sinusoïde ;
   - soit par une méthode digitale.

2. Que l'on reprenne les expériences de Riesz qui ne peuvent qu'apporter plus d'information concernant le comportement de l'oreille au voisinage immédiat d'un son masquant et le ou les mécanismes encore mystérieux de l'inhibition latérale.

3. Que des spécialistes de la mécanique de la membrane basilaire s'efforcent d'en prédire le comportement en réponse à un MESP, soit par le calcul, soit par un modèle analogique fiable.

Références :


MASKING, A PERIPHERAL EFFECT

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INTRODUCTION

Masking is one of the most important properties of the ear. Tone-on-tone masking of continuous tones was systematically studied as early as 1924 by Wegel and Lane. In "classical" masking experiments narrow band noises are often used as maskers, in order to avoid beats. These masking patterns (see for example Zwicker 1963) illustrate heat the ear's frequency selectivity if masked threshold is plotted as a function of the critical-band rate. These patterns exhibit strong nonlinearities, especially in the frequency region above that of the masker. The ear's combination of frequency selectivity and nonlinearity raises the question: "is masking produced peripherally in the cochlea or more centrally through neural interactions?" (question a).

There are, however, additional aspects to the masking properties of the ear. Strong temporal effects have been measured in the cases of pre-, simultaneous, and post-masking (Fastl, 1977, 1979). Masking-period patterns (Zwicker, 1976a) have provided additional information, especially concerning the masking of periodic sounds such as very-low- and low-frequency tones, i.e. tones with frequencies below 1 kHz. Excitation-critical band rate patterns are an important aid to the understanding of many of the psychoacoustical properties of continuous sounds (see for example Zwicker 1970). These patterns are based on masking patterns measured with continuous test tones. An awareness of the importance of temporal effects leads to a second question (b): "How much influence do temporal effects have on the masking or excitation patterns produced by maskers with frequencies below 1 kHz?"

RESULTS

An attempt was made to answer both question (a) and question (b) using classical and temporal masking patterns on the one hand and so-called suppression-period patterns (Zwicker, 1981, Zwicker and Manley 1981) of delayed oto-acoustic emissions on the other.

The author, served as subject for all measurements. The methods and apparatus used are described in detail in the above-mentioned literature.

The top part (a) of Fig. 1 shows the stimulus configuration for the measurement of a masking-period pattern, with a 20 Hz-111 dB tone as masker and a tone burst composed of 1.5 ms-1350 Hz impulses as test sound. The threshold of the burst was measured as a function of its temporal po-
sition $\Delta t$ within the period $T$ of the masker. The masking-period pattern obtained in this way is drawn in Fig. 1b using open circles. The threshold in quiet for the test tone burst was measured before and after the series of measurements and is indicated by closed circles to the left and to the right of Fig. 1b. The thresholds of the continuous 1350-Hz tone in quiet (closed rhombus) and masked by the 20-Hz tone (open rhombus) are also indicated.

The sound pressure $p_{OAE}$ of the delayed oto-acoustic emissions evoked by the same test tone burst with a sensation level $SL = 20$ dB is plotted in Fig. 1c. At left and right, the $p_{OAE}$ without the suppressing 20-Hz tone is given, while the inner curves belong to conditions for which $p_{OAE}$ is suppressed by the 20-Hz tone at the corresponding temporal position $\Delta t$ within its period. The effective value of $p_{OAE}$ measured for the window within the two indicated dashed lines is given in Fig. 1d on a logarithmic scale (normalized to an arbitrary value $p_n$) as a function of $\Delta t$. The curves in Fig. 1b and Fig. 1d are virtually exact mirror images of each other.

More data of the same kind are given in Fig. 2 but for masker frequencies of 40 Hz, 80 Hz, and 160 Hz, as indicated. Masker levels are chosen so that an $SL$ of about 40 dB is reached in the masking pattern’s maximum. In Fig. 1 (for 20 Hz masker frequency), the temporal resolution was chosen corresponding to 1/8 of the period, while it was reduced to 1/4 of the period in Fig. 2 for all three masker frequencies. Thresholds of the continuous test tone in quiet, as well as masked by the low-frequency tones, are again indicated as open and closed rhombi respectively. Fig. 2 shows that the masking-period patterns become shallower with increasing
masker frequency. The same holds for the suppression-period patterns.

**DISCUSSION AND CONCLUSIONS**

The very close mirror relationship between curves (a) and (d) in Fig. 1 for 20-Hz masker frequency is found also for 40, 80, and 160 Hz masker frequency, as shown in Fig. 2. This relationship confirms the assumption expressed earlier (Zwicker 1979, 1983) that masking is produced peripherally, i.e. within the cochlea, at least for threshold shifts up to about 20 dB, the upper limit of linear rise of oto-acoustic emissions.

Masking produced by low-frequency maskers is not a steady state effect but varies with the period of the masker. Masking is most effective at the point of minimum pressure, i.e. maximum rarefaction, as indicated in the masking-period pattern (for example Fig. 1b). The differences between the maxima and the minima of the masking-period patterns decrease with increasing masker frequency from 19, to 19, 15 and finally 6 dB for frequencies of 20, 40, 80 and 160 Hz, respectively.
Continuous tones are masked by low frequency maskers in the following way. The continuous tone becomes audible during that temporal part of the masking-period pattern during which it remains at low levels (see Zwicker 1976b). Since the minimal value \( L_T \) reached within one period increases with increasing masker frequency from 21 to 23, 25, and finally 32 dB, the masked threshold \( L_T \) for continuous tones also increases in this case from 8 to 12, 15 and finally 21 dB. This latter increment should not, however, be used as an indication of increased maximal excitation, which is correlated to the maximum within the masking-period pattern and is specified here by \( L_T \) 40 dB.

In summary, the above two questions may be answered as follows: (a) masking is produced very peripherally within the cochlea, at least for threshold shifts up to 20 dB. (b) Temporal effects strongly influence the excitation pattern produced by low frequency sounds. Masked threshold of continuous tones is not directly related to the excitation maximum which has to be extracted from masking-period patterns produced by low-frequency maskers.

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REFERENCES


ON MASKING AND TTS CAUSED BY BONE-CONDUCTED ULTRASOUND

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1. INTRODUCTION

Usually, we can not hear the ultrasound above 20 kHz by air-borne conduction. It is well known, however, that some auditory sensation by means of bone conduction exists as to hearing of ultrasound. The ultrasound may induce very high frequency vibration on the cranial bone and the internal ear, and so there is a possibility that this vibration causes the hearing damage and other physiological effects on man. Under these circumstances, ultrasound is applied to some washing machine and used for medical use. In this study, we discuss the actual effects of ultrasound which seem to cause hearing damage in audio frequency range, from the viewpoint of masking and TTS.

2. EXPERIMENTAL PROCEDURE

1) Transmitter
Magnetostriuctive transducer was used as a transmitter of ultrasound. Its resonant frequency was 47.5kHz and the diameter of radiation surface was 14mm. The frequency response of the transmitter is shown in Fig.1. The response curve was obtained in the water tank (150 litters) so as to simulate the impedance of the forehead. This method was used after the Corso's experiment.

2) Presentation of ultrasound
The magnitude of ultrasound perception varies widely depending upon the contact point on the body and the contact pressure. In order to keep these condition constant, we made an equipment shown in Fig.2. This equipment was based on Bekesy's idea. The transmitter was attached to the forehead and then the contact pressure was determined by the weight (300g).

3) Testing procedure
The block diagram of experimental equipment is shown in Fig.3. Only half of this diagram was used at the time of two experiments. The subjects who participated in the experiments were four adult men with normal hearing.
3. AUDITORY THRESHOLD FOR ULTRASOUND

As basic data, the auditory threshold for ultrasound was measured for four subjects at fourteen points of frequency, every 5kHz from 15kHz to 80 kHz. Method of limits was used and the measurements were repeated five times at each frequency.

The results are shown in Fig.4. In this figure, the sound pressure level of the longitudinal axis is based on the pressure shown in Fig.1.

The frequency characteristic of the threshold shows a great resemblance to reported results, though there are some differences in their absolute values.

4. MASKING EFFECTS OF ULTRASOUND ON AUDIO FREQUENCY

At the first stage of experiments, we measured the masking effects of ultrasound on audio frequency. The frequency of ultrasound as masker was settled at 25kHz and its pressure level was 10 dB higher than the threshold level in each subject. To be more precise, the absolute levels of masker varied from 128 dB to 139 dBSPL according to the subjects. Several frequencies of maskee were chosen between 1kHz and 12.5kHz. Threshold levels at each frequency were determined under the conditions with and without masker by using the method of limits.

The results for each subject are shown in Fig.5 and the masking audiograms are shown in Fig.6.

Though the shapes of masking audiograms are different widely from subject to subject, the masking effect of ultrasound is distinguished at over 8kHz for all subjects. According to this result, the exposure to ultrasound seems to cause the vibration around the root of basilar membrane.

5. MEASUREMENT OF TTS (Temporary Threshold Shift) FOR THE AUDIO FREQUENCY CAUSED BY ULTRASOUND

Before and after the exposure of 60 seconds to ultrasound on the same condition as in the former experiment, hearing thresholds at 4 kHz and 8 kHz were measured. And TTS were obtained at 30 seconds and 2 minutes after the exposure to ultrasound.

Time pattern of a session of experiment is shown in Fig.7 and the results of each subject are shown in Table 1. As a result of the statistical test, it can be said that TTS is caused by the exposure to ultrasound, even though the amount of TTS varies with frequency, the time after the exposure and subjects.

Incidentally, all subjects said that they always had a sensation of some sound similar to white noise for a while after the exposure to ultrasound. The hint of resolving the mechanism of TTS seems to lie in this fact.

6. Discussion

From the two experiments, masking and TTS occurred in the audio frequency region after the exposure to ultrasound at the level just 10 dB higher than threshold. According to those facts, when we perceive the ultrasound from some equipment or instrument, it is important to beware of
the damage to hearing. Hereupon if some equipments generate not only ultrasound but also high frequency sound in audio region, then we should be able to know the influence of each sound on hearing. Moreover we will have to know the relation between the amount of hearing loss and the intensity of ultrasound.

Reference

Fig.1 Frequency response of the transmitter in the water tank at the point of 1cm apart from the hydrophone. (Input Level : LV)

Fig.2 The manner of supporting a transmitter for application to the head at constant pressure.

Fig.3 Block diagram of the experimental equipment.
**Fig. 4** The threshold curve for bone-conducted tones in the ultrasound.

**Fig. 5** The frequency response of threshold level of maskee (audio frequency).
- Fat line: with ultrasound (25kHz)
- Fine line: without ultrasound

**Table 1.** TTS for audio frequency caused by ultrasound (25 kHz).

<table>
<thead>
<tr>
<th></th>
<th>4 kHz</th>
<th>8 kHz</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>30sec</td>
<td>2min</td>
</tr>
<tr>
<td>Sub.1</td>
<td>2.9*</td>
<td>2.7*</td>
</tr>
<tr>
<td>Sub.2</td>
<td>2.0**</td>
<td>1.2**</td>
</tr>
<tr>
<td>Sub.3</td>
<td>2.5*</td>
<td>0.8*</td>
</tr>
<tr>
<td>Sub.4</td>
<td>3.6**</td>
<td>3.7**</td>
</tr>
<tr>
<td>Total</td>
<td>2.7**</td>
<td>2.1**</td>
</tr>
</tbody>
</table>

**Fig. 6** The frequency response of masking.

**Fig. 7** The time pattern of one session for the measurement of TTS.
COMPARISON OF A LISTENER TO AN IDEAL DETECTOR OF SIGNAL IN NOISE

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INTRODUCTION

From the moment when Peterson, Birdsall and Fox (1) presented their mathematical models of signal in noise ideal detectors, attempts have been made to compare a human observer with them. Although a quarter of a century already has passed there is no agreement about the most appropriate model of auditory detection. A great deal of results obtained in solving this problem are presented in the monograph by Green and Swets (2).

The only way to determine which model is the best, is to compare, for given model, predicted measures of signal detection with results obtained in psychoacoustical experiments. In presented paper the results of one more comparison of a listener to an ideal detector are given.

SIGNAL AND NOISE

In investigations a sine signal has been detected in three types of noise. Particular noise has been obtained by passing the narrow-band Gaussian noise through nonlinear device with a transfer function

\[ y = k|x|^a \text{sgn}(x) \quad (1) \]

and a narrow bandpass filter with the same center frequency as the input noise. Setting the index "a" in the transfer function (1) to a=0.5, a=1 and a=2 three types of noise, denoted NO.5, N1 and N2, have been obtained. All these noises
A Podrez, Comparison of a listener to an ideal detector

have equal power spectral densities but different probability distributions. Measurements of autocorrelation functions performed by means of HF 3721A correlator as well as frequency analysis by B&K heterodyne analyser confirmed equality of spectrums.

PSYCHOACOUSTICAL MEASUREMENTS

In psychoacoustical measurements percent of correct responses P(C) by listeners has been measured in 2AFC procedure experiments. The measurements have been carried out in two steps. In both of them one signal (1000 Hz tone) has been detected in a continuous noise N0.5, N1 or N2 by eight listeners. In the first step signal to noise ratio was set to S/N = -9dB. For given signal-noise pair every listener had to make decision in 1600 trials. Obtained results are presented in a table 1. In the second step of experiments psychometric functions have been determined. Averaged functions are plotted in fig.1.

Table 1. Results of a first step of psychoacoustical measurements

<table>
<thead>
<tr>
<th>listener number</th>
<th>P(C) in %</th>
<th>N0.5</th>
<th>N1</th>
<th>N2</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>79.1</td>
<td>73.9</td>
<td>96.3</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>85.3</td>
<td>57.6</td>
<td>99</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>83.5</td>
<td>82.8</td>
<td>98</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>87.7</td>
<td>71.4</td>
<td>99</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>92.6</td>
<td>64</td>
<td>94.7</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>65.7</td>
<td>51.3</td>
<td>97</td>
<td></td>
</tr>
<tr>
<td>7</td>
<td>93.5</td>
<td>71.9</td>
<td>94.3</td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>83</td>
<td>68.7</td>
<td>96</td>
<td></td>
</tr>
<tr>
<td>mean</td>
<td>83.8</td>
<td>67.7</td>
<td>94</td>
<td></td>
</tr>
</tbody>
</table>

Fig.1. Psychometric functions obtained in second step of measurements.

It can be easily noticed, that in the both steps of psychoacoustical measurements the signal was best detected in a background of noise N2 and worst detected in the noise N1. It can be denoted as a detection sequence N2-N0.5-N1.
PREDICTION OF DETECTION

To decide which ideal detector model is appropriate, it is necessary to calculate percent of correct decisions $P(C)$ in 2APC procedure, taking into account noise and signal plus noise probability distributions. Then model for which sequence $N2-N0.5-N1$ is obtained, can be accepted.

Due to the fact, that all used types of noise have equal power spectral density functions, energy detector would have equal $P(C)$ for all three noises. Energy detector isn't a good model of a listener. This same can be said about correlation detector. The noises have equal autocorrelation functions, so also crosscorrelation functions of signal and noise are equal. But in the case of nongaussian noise ideal SKE detector does not need to be a correlator, and after excluding the correlator this model still should be considered as a possible model of auditory detection. There is similar situation with SKS detector and an amplitude detector as well as more general model should be taken into account.

To obtain the detectability measure $P(C)$ it is necessary to determine probability distributions of observed waveform when noise alone and noise plus signal are presented, then plot ROC curve and calculate the area under the curve. Because determination of probability distribution on the output of nonlinear inertial network is very difficult, author has decided to measure it by means of HP3721A correlator and then make suitable computations. It was possible to consider only one-dimensional probability distributions. This simplification corresponds to assumption, that during the observation interval to detector is given only one sample of a waveform. Computed detectability $P(C)$ for particular models is given in a table 2.

As it was expected, computed $P(C)$ values are much less then obtained in psychoacoustical measurements. Accuracy of predicted detectability was tested by computing the $P(C)$ of zero signal. In the worst case the accuracy was better then
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0.4671%. Only for model of detector for which signal is known except for phase, the sequence of predicted detectabilities is this same as obtained in the psychoacoustical measurements.

Table 2. Predicted detectability $P(C)$ of the sine signal in three types of noise.

<table>
<thead>
<tr>
<th>ideal detector type</th>
<th>N0.5</th>
<th>N1</th>
<th>N2</th>
<th>S/N = - 9dB sequence</th>
</tr>
</thead>
<tbody>
<tr>
<td>SKE</td>
<td>52.7982</td>
<td>53.2413</td>
<td>54.6158</td>
<td>N2-N1-N0.5</td>
</tr>
<tr>
<td>amplitude</td>
<td>52.2762</td>
<td>51.5964</td>
<td>51.4578</td>
<td>N0.5-N1-N2</td>
</tr>
<tr>
<td>SKS</td>
<td>50.7337</td>
<td>50.4541</td>
<td>51.7977</td>
<td>N2-N0.5-N1</td>
</tr>
</tbody>
</table>

CONCLUSIONS

It has been proved that neither correlation detector nor energy detector are not proper models of human auditory detection. The most likely model seems to be an ideal detector for which signal is known except for phase. Only for that model detection sequence obtained in the psychoacoustical measurements is in agreement with predicted one.

REFERENCES

LA SONIE. INTERACTIONS ENTRE FLUX SONORES SIMULTANÉS.

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Quelques travaux récents ont montré qu'un son continu monaural dont le niveau physique est constant diminue subjectivement d'intensité quand on fait entendre simultanément un autre son de façon intermittente soit dans l'oreille opposée (BOTTE, CANEVET et SCHARF, 1982), soit dans la même oreille (CANEVET, SCHARF et BOTTE, sous presse). Ces auteurs ont encore constaté l'existence du même phénomène en écoute binaurale, lorsque le son continu et le son intermittent proviennent de haut-parleurs différents. La réduction de la sonie du son continu est progressive, elle atteint son maximum en 3 minutes environ. En écoute dichotique, avec un son continu (1000 Hz ; 60 dB) présenté à l'oreille droite et un son intermittent (1000 Hz ; 60 dB) à l'oreille gauche pendant 500 ms toutes les 1000 ms, au bout de 3 minutes, la sonie a diminué de moitié, en moyenne. La sonie du son intermittent est, par contre, augmentée dans le même temps (BOTTE et al., 1982).

Lorsque plusieurs flux sonores atteignent simultanément et de façon prolongée le système auditif, il peut donc s'établir des modifications à long terme dans la perception de chacun d'eux, en particulier pour ce qui concerne leurs sonies respectives.

Les expériences présentées ont porté uniquement sur l'adaptation induite de la sonie en écoute dichotique. Les deux flux sonores (l'un continu et l'autre intermittent) étaient présentés par écouteurs, l'un à l'oreille droite et l'autre à l'oreille gauche. Le but était de rechercher comment :

1) le son inducteur cause une diminution de la sonie de l'autre son (Expérience I);

2) la diminution de la sonie évolue au cours du temps (Expérience II)

Dans ces expériences, la sonie était mesurée au moyen d'estimations directes de la grandeur. Chaque fois qu'une lampe s'allumait, le sujet devait estimer par un nombre la sonie du son continu présenté à l'oreille droite. Les demandes d'estimation se produisaient en général toutes les 8 secondes. Le sujet devait donc, à chaque test, donner plusieurs estimations successives. Chacune de ces estimations chiffrées (E) était transformée en
taux d'adaptation à l'instant $t$ ($TA_t$) selon la formule :

$$TA_t = \frac{E_i - E_t}{E_i}$$

où $E_i$ = estimation initiale
$E_t$ = estimation à l'instant $t$.

**EXPERIENCE I**

Dans la première expérience, on a recherché l'effet d'un son inducteur de durée limitée (10 secondes) sur la sonie d'un son beaucoup plus long. Le déroulement d'un test est schématisé sur la figure 1.

![Schéma d'un test](image)

Fig. 1 : Schéma d'un test

Le son sur l'oreille droite était toujours un son de 1000 Hz à 60 dB; le son sur l'oreille gauche était toujours à 60 dB, mais sa fréquence a varié selon les tests ; 7 fréquences différentes, comprises entre 1000 et 4050 Hz, ont été testées. Dans une condition contrôle, les sujets estimaient la sonie du son continu en l'absence de son inducteur à droite. 10 sujets audiométriquement normaux pour toutes les fréquences testées ont participé aux mesures.

Le son inducteur (OG), à son apparition, est toujours latéralisé à gauche, mais :

1) si sa fréquence est très différente du son déjà présent à droite, 2 sons continuent à être perçus, l'un à droite et l'autre à gauche

2) si sa fréquence est identique ou proche de celle du son présent à droite, il y a fusion en un seul son, latéralisé d'abord à gauche et qui, progressivement, en quelques secondes, devient médian.

La figure 2 présente les valeurs moyennes de taux d'adaptation du son de 1000 Hz à droite, pour la condition contrôle et pour les 7 conditions différentes de fréquence sur l'oreille gauche.
Ces résultats montrent que :

1) la différence de fréquence entre son inducteur et son adapté est cruciale pour le taux d'adaptation. Plus les fréquences respectives des 2 sons sont proches, plus l'effet du son inducteur est important. Comme le confirment des données ultérieures, l'adaptation est significativement plus forte quand les fréquences des 2 sons appartiennent à la même bande critique.

2) l'effet du son inducteur sur la sonie du son adapté se partage en deux temps :

- pendant le son inducteur, il y a une forte diminution de sonie. Cette diminution est progressive; elle atteint son maximum en 10 à 20 secondes

- après l'arrêt du son inducteur, la sonie ne retrouve pas sa valeur initiale il demeure une adaptation résiduelle. L'adaptation résiduelle est d'autant plus marquée que la diminution de sonie en présence du son inducteur avait été plus importante, autrement dit que la différence de fréquence était plus faible. Quand la fréquence du son inducteur est assez éloignée de celle du son adapté (1320, 1640 ou 4050 Hz), l'adaptation résiduelle est à peu près nulle. Pour les fréquences du son inducteur qui s'accompagnent d'une adaptation résiduelle significative, il y a une certaine récupération de la sonie au cours de la période qui suit le son inducteur, mais il apparaît que cette récupération n'est pas complète en 40 secondes.

Fig. 2 : Adaptation induite contralatéralement pour un son continu sur OD (1000 Hz, 60 dB SPL) par un son inducteur sur OG. Moyennes de 10 sujets.

Ces résultats nous ont amené à faire l'hypothèse que la progression observée dans l'adaptation induite de la sonie au cours d'interactions de plus longue durée (3 minutes) provient du cumul d'effets successifs du type de celui que met en évidence l'Expérience I. On peut supposer que si un deuxième son inducteur apparaît pendant la période où l'adaptation résiduelle est présente, son effet s'ajoutera, en partie au moins, à l'effet du premier son inducteur.
Fig. 3 : Adaptation induite contralatéralement pour un son continu sur OD par un son intermittent sur OG. Les 2 sons : 1000 Hz ; 60 dB SPL. Moyennes de 10 sujets.

(adaptation résiduelle). Alors que l'adaptation induite pendant la présence du son inducteur est pratiquement constante au cours des 3 minutes, l'adaptation résiduelle augmente à chaque nouvelle occurrence du son inducteur.

L'ensemble de ces expériences permet de mieux comprendre le mécanisme de l'adaptation induite en écoute dichotique. Elles mettent en évidence l'effet de la fréquence et de l'intermittence du son inducteur.


ESTIMATION DE LA SONIE EN CHAMP LIBRE ET SEMI-REVERBERANT

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La relation entre l'intensité physique d'un son et son intensité subjective (sonie) a été déterminée le plus fréquemment à l'aide d'écouteurs, en cabine audiométrique. Pour contrôler sa validité dans le cas d'un environnement réel, nous avons élargi les conditions expérimentales au champ libre et aux salles, les sons étant alors présentés par haut-parleurs. De plus, les estimations ont été faites, non plus par des individus isolés mais par des groupes de sujets, en général de 18 à 25 ans, dont la plupart n'avait jamais participé à de telles expériences auparavant. Au total, nous avons ainsi testé 395 personnes réparties en 29 groupes.

Sonie en champ libre.
Tout d'abord, nous présentons les résultats d'une expérience effectuée en chambre sourde (5mx5mx5m) avec le concours de 77 étudiants.

Procédure expérimentale.
Les sujets ont été testés par ensembles de 4 à 11 personnes, qui se regroupaient face à un haut parleur situé à environ 3 mètres d'eux. L'étalonnage des niveaux sonores se faisait au début du test, à l'aide d'un micro- phone disposé au centre de l'emplacement qu'occuperait le groupe par la suite. Le test commençait par une mesure du seuil de détection sur des séries de 3 trains d'onde de 500 ms à 1000 Hz, présentés à différents niveaux, décroissant d'abord puis croissant ensuite selon une méthode des limites. En réponse à chaque présentation les sujets indiquaient, en inscrivant le symbole + ou − sur une feuille qui leur était distribuée, qu'ils avaient entendu ou non le signal. Ensuite, venaient les mesures de sonie sur les mêmes trains d'onde, présentés à 10, 20, 30, 40, 50 et 60 dB dans un ordre aléatoire et différent pour chaque groupe.
Pour mesurer la sonie, nous avons utilisé la méthode de Stevens (1975), selon laquelle les sujets évaluent la sonie en la caractérisant par un nombre qui lui est proportionnel, sans échelle ni référence imposée. La présentation des 6 niveaux était répétée trois fois, et nous prenions en considération les estimations des deux dernières séries.

Résultats.
Etant donné la diversité des échelles individuelles nous avons procédé à une sorte de normalisation des données : chaque estimation d'un sujet a été divisée par la moyenne géométrique de l'ensemble de ses estimations.
La figure 1 représente les moyennes géométriques et écart types des données ainsi normalisées de 70 sujets, les 7 autres n'ayant pu être retenus, soit parce qu'ils avaient des seuils trop élevés, soit pour cause d'erreurs dans la transcrition de leurs estimations. On peut observer que, malgré l'inévitable imprécision des mesures, les estimations s'alignent selon la courbe standard ISO (1959), indiquant une relation de type puissance ($S = K I^n$) entre sonie et pression sonore établie initialement par Stevens (en 1955). Dans le cas présent, nous avons $K = 55$ et $n = 0.55$ comme valeurs moyennes. La dispersion des exposants individuels est illustrée par l'histogramme de la figure 1, donnant les effectifs en ordonnées et les pentes en abscisse.

La loi de Stevens rend bien compte des résultats expérimentaux au-dessus de 30 dB. Aux niveaux plus faibles, la sonie ne décroît plus linéairement, sur cette échelle doublement logarithmique, en fonction du niveau. Ceci avait d'ailleurs déjà été signalé par Scharf et Stevens (1959) et bien d'autres auteurs depuis. Plusieurs interprétations de ce phénomène ont alors été proposées. Elles ont conduit à des formulations plus élabores de la fonction de sonie, que Marks et JC Stevens (1968) ont analysées en détail. Elles apparaissent toutefois comme de simples variantes de la
fonction de Stevens et la variabilité des résultats expérimentaux ne permet pas encore de sélectionner la meilleure formulation. Dans tous les cas on pourra constater l'accord remarquable entre nos données et celles recueillies par Hellman et Zwilocki (1961) à partir de publications de divers laboratoires.

Sonie en champ semi-réverberant.
S'il est important de mesurer la fonction de sonie dans des conditions expérimentales bien contrôlées, ce que permet l'expérience en chambre sourde ou en cabine audiométrique, il est également important de connaître les perturbations éventuelles que peut produire un environnement réel, tel qu'une salle par exemple. Les données que nous présentons ici montrent que les mesures faites en salles de cours concordent très bien avec celle du champ libre.

- Figure 2 -

Procédure.
Ces mesures ont été faites dans différents pays surtout dans des salles de dimensions moyennes, et sur des groupes de sujets allant de 6 à 42 personnes. La procédure expérimentale était la même que ci-dessus, excepté que l'on n'a pas mesuré le seuil des sujets. La sonie a été mesurée par la méthode d'estimation directe de S.S. Stevens. Suivant le cas, il y avait 2 ou 3 présentations des stimuli et l'on prenait en général la moyenne de
2 jugements. Les niveaux étaient mesurés en un point de la salle et n'ont donc qu'une valeur relative.

Résultats.
Nous avons regroupé toutes les données sur un même graphique indiquant, pour chaque groupe, le nombre de sujets testés et la pente de la droite donnant le meilleur ajustement au sens des moindres carrés. Pour éviter de charger la figure nous n'avons pas reporté les paramètres de dispersion mais ils sont du même ordre que ceux de la figure 1. Il est clair que dans tous les cas la tendance qui apparaît est que la fonction de sonie est analogue à celle du champ libre, ce qui rejoint une publication antérieure de JC Stevens et Tulving (1957). La variabilité de l'exposant de cette fonction est également tout à fait comparable, avec une moyenne, tous groupes confondus de 0,597.

Conclusion.
L'ensemble de ces expériences montre donc l'universalité de la loi de puissance pour la sonie, quels que soient l'environnement et les sujets testés, en même temps que l'utilité de la méthode d'estimation de grandeur. De plus, il est clair que l'exposant de 0,6 retenu pour la normalisation correspond bien à la limite vers laquelle converge la moyenne de grands nombres de données obtenues dans des conditions expérimentales très diverses.

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7/ STEVENS S.S.
ON THE EVALUATION OF LOUDNESS OF IMPACT SOUND

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1. INTRODUCTION

Several psycho-acoustical experiments on loudness of impact sounds were carried out, in which the model impact sounds determined on the basis of a wide investigation of actual ones were used as the stimuli.1)-4)

Those stimuli were synthesized by amplitude modulation of sinusoidal carrier (500, 1000 or 2000 Hz), pink noise or white noise. The point of subjective equality (PSE) for loudness of impact sound was obtained in an anechoic room by the method of adjustment. Test and comparison stimuli were presented through a loudspeaker.

Two examples of the results are shown in Fig.1(2),3) and Fig.2(4). As seen from the figures, the loudness of impact sound cannot be said to be entirely dependent on the sound energy of stimulus. The influence of peak sound pressure level (SPL) of stimulus is rather dominant. Under the same peak SPL condition, however, the PSE increases along with the increase in sound energy, which is caused by lengthening the stimulus duration and the rise of repetition rate. Accordingly the effect of the peak SPL and other factors concerned should be included in the method of evaluating loudness of impact sound.

We propose a method for measuring the quantity here, which shows a good agreement with loudness of impact sound. An instrumentation for it may include a squaring circuit and an exponential averaging circuit with a certain time constant, and it can measure a steady sound in the same way as the sound level meter prescribed by JIS C 1502 (the Japanese Industrial Standards) does.

2. EXPONENTIAL AVERAGING CIRCUIT OF A SOUND LEVEL METER FOR MEASURING AN IMPACT SOUND

It is desirable that the impact sound level meter is able to use not only for impact sound but also for steady sound as well without any modification or discrimination.

With reference to experimental results of a comparison of loudness of impact sound with that of quasi-steady sound, it can be said that PSE's for loudness of both sounds are nearly equal to each other in most cases as far as the sound pressure levels of those sounds measured through *p* response circuit of a sound level meter (SPL_125) indicate the same
value. The loudness of impact sound is underestimated in the case of a very short sound, if we adopt the SPL_{125}. This error is considered to be caused by the influence of steep onset of stimulus on neural activity, i.e. the firing rate of neural spikes increases in a stimulus with time duration shorter than a certain degree. Some compensation should be contrived, therefore, in order to cover this area by the same method of measurement.

Fig. 3 is a block diagram of a measurement system to compensate the imperfection of SPL_{125} for evaluating an impact sound with shorter duration. We indicate, hereafter, the SPL_{T1/T2} as the instantaneous sound

Fig. 1
Loudness of a single burst of impact sound. Relation between calculated sound energy level and PSE is shown here. The rise time and the decay time of test stimuli are 4/60, 4/250 or 4/1000 ms (without steady part), and the rise time, the steady duration and the decay time are 30/0.3/30, 30/14/30 or 30/68/30 ms (with steady part). Those of the comparison stimulus are 60/200/60 ms. The rise time and the decay time are defined here as the time required for 60 dB change in sound pressure level.

Fig. 2
Loudness of repeated impact sound. The rise time of each sound is 4 ms, and its decay time is 60, 120 or 250 ms. Those sounds are presented repeatedly as test stimuli. The number of repetition is 1, 2, 4 or 8.
Kunagai

**LOUDNESS OF IMPACT SOUND**

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**Fig. 3** The block diagram of a measuring system for impact and steady sounds.

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**Fig. 4**

Loudness of a single burst of impact sound. Relation between calculated SPL and PSE is shown here. SPL is calculated for the system as shown in Fig. 3, and PSE is taken from the experimental result shown in Fig. 1. The solid line in this figure denotes the estimated PSE for quasi-steady sound with 200 ms steady duration.

---

**Fig. 5**

Loudness of repeated impact sound. PSE is an experimental result shown in Fig. 2. Other conditions are the same as in Fig. 4.
pressure level through an exponential averaging circuit with a rise time constant $T_1$ ms and a decay time constant $T_2$ ms.

The $\text{SPL}_{5/250}$ is always greater than $\text{SPL}_{125/250}$. The difference between these two levels attains 10 dB for the stimulus with very short duration, so that the output voltage from the adder circuit is mainly determined by $\text{SPL}_{5/250}$. This does not suit our purpose. Then the magnitude of $\text{SPL}_{5/250}$ must be attenuated before feeding it to the adder circuit as shown in Fig.3.

It is desirable for us to evaluate the loudness of both impact and steady sounds by the same method using such a sound level meter as mentioned above. If the PSE for loudness of impact sound and that of steady sound are equal, therefore, the SPL of impact sound is required to be equal to the SPL of steady sound. The optimum attenuation for $\text{SPL}_{5/250}$ to minimize the deviation of SPL of impact sound from that of quasi-steady sound, which has the same PSE as the former, is found to be 16 dB.

Fig.4 shows the relation between the resultant SPL and PSE, where the SPL is calculated on the attenuation for $\text{SPL}_{5/250}$ is 16 dB and PSE is taken from the experimental results shown in Fig.1. Fig.5 shows the relation between SPL and PSE for loudness of repeated impact sound, where PSE is the values taken from the experimental results in Fig.2. The solid lines in Fig.4 and Fig.5 denote the estimated PSE for quasi-steady sound with 200 ms steady part.

From Fig.5, it can be said that the method of measuring SPL through 16 dB attenuation is appropriate for the evaluation of the loudness of repeated impact sound as well as a single burst of impact sound.

The quasi-steady sound with duration of 200 ms is treated as steady sound in our discussion on the comparison of loudness of impact sound with steady sound, because the quasi-steady sound with a duration of 200 ms is regarded as the steady sound, referring to the literatures concerning the critical duration of sound from a point of loudness.

3. CONCLUSION

A sound level meter with a circuit as shown in Fig.3 indicates the SPL value which corresponds well to the loudness of both impact and steady sounds. The remaining part of this sound level meter is just the same as the usual sound level meter.

The SPL of impact sound measured by this sound level meter, whether it is a single burst or a repeated impact sound, shows nearly the same SPL as steady sound which has a loudness equal to the impact sound. This sound level meter is useful for evaluating the loudness of impact sound and steady sound by the same method.

Furthermore, the difference of SPL between a quasi-steady sound with duration of 200 ms and a steady sound is about 1.0 dB on the sound level meter mentioned here, and so it meets the requirement of the IEC Standard, Pub. 651 "FM" Type 1, 2 and 3.

REFERENCES

DETECTION THRESHOLDS IN PULSATION MEASUREMENT

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Introduction

In pulsation threshold measurement, the listener is presented with an alternation of a pulsator and a probe stimulus, that often is a sine wave. The listener is asked in most cases to adjust the level of the probe to the threshold between continuous and pulsating perception of the probe.

Listeners do not consider the task difficult as long as the probe frequency is close to a frequency component of the pulsator and the level of this component is sufficiently high. If these conditions are not met it is very difficult to tell whether a weak probe is perceived as if it sounds pulsating or as if it were continuous. Raising the probe's level by a few decibels will result in a clearly pulsating probe tone.

The described phenomenon suggests two different criteria in pulsation measurement. One is based on continuity perception and the other on the difficulty of detecting short interruptions in a tone presented just above its threshold. It is conceivable that the two criteria, pulsation and masked threshold, are based on different physiological processes (Schreiner, 1979) and thus might reveal different properties of the auditory system. We became aware of the different criteria when we measured pulsation patterns for listeners with a cochlear impairment. It could be shown that their patterns were largely determined by the detection threshold and not by continuity perception.

It is the aim of the present paper to investigate the detection threshold in normally hearing listeners and to see what parts of the pulsation patterns are based on masked thresholds and what parts on continuity thresholds.

Methods

All reported pulsation and masked threshold patterns were determined for the same stimulation conditions. The difference between the two measurements is the task given to the listener. It is the adjustment of the probe level to the highest value at which the probe is perceived as not pulsating or at which the probe cannot be detected.

The pulsator/masker and probe are both presented as signals with a duration of 125 ms of which the transients are smoothed by gaussian amplitude envelopes with time constants of 3.55 ms (Verschuure et al., 1976).
Thresholds are measured with the aid of a computer controlled Bakesy tracking method for fixed frequencies. The threshold is defined as the averaged level of the last four reversals out of a total number of seven. The step size of the level change is 0.5 dB and the levels are changed at a rate of one step per second.

Three experienced listeners and a number of untrained students of medicine participated. The results obtained from the latter group do not differ from those of the former group except for a difference in test-retest variability. The presented data are all from experienced listeners.

Results

Pulsation patterns have been reported extensively by Verschuure (1981a) and the reader is referred to that publication for details. The patterns can be described in general terms as a triangular pattern of which the slopes depend on level:

- The low-frequency edge to the pattern gets steeper at higher pulsator levels.
- The high-frequency edge to the pattern gets less steep at higher pulsator levels.
- Pulsator and probe levels are almost equal for equal pulsator and probe frequencies.

The effects and their consequences for the interpretation of pulsation patterns have been discussed by Verschuure (1981b).

![Graph of masked threshold patterns](image)

**Fig. 1.** Masked threshold patterns of obs. HvdO at 2 kHz determined under pulsation conditions.

Masked threshold patterns were measured for comparison. An example is shown in fig. 1. The dependence of these patterns on level can be described as:

- The low-frequency edge to the pattern coincides for different pulsator levels, except for levels lower than about 40 dB.
- The high-frequency edge to the pattern gets less steep for higher pulsator levels, in line with the finding for pulsation patterns.
- The summit of the pattern tends to shift towards higher frequencies for pulsator levels over about 60 dB SPL for some listeners. Others show only a change in slope with level.

The dependence of the masked threshold under pulsation conditions on the pulsator/masker level can be studied more directly by measuring the threshold level as a function of pulsator level for various probe frequencies. An example is given in fig. 2.

![Graph showing masked threshold level as a function of pulsator level for various probe frequencies.](image)

**Fig. 2.** Masked threshold level as a function of pulsator level at a number of probe frequencies. Pulsator frequency is 2.0 kHz.

We show only curves for probe frequencies above the pulsator frequency, because the curves for probe frequencies below the pulsator are all flat. The curves of fig. 2 show a flat part for low pulsator levels. The first flat part represents the absolute threshold at the particular frequency and it remains flat as long as the activity of the pulsator does not become apparent in the auditory channels through which the information on the probe is carried.

For medium pulsator levels we observe a strong dependence of the masked threshold on pulsator level, but this dependence seems to level off for high pulsator levels if the probe frequency is close to the pulsator frequency.

The dependence was studied by fitting through the data points of curves measured for equal probe and pulsator frequency a function of the form:

\[ L(\text{probe}) - L(\text{thresh}) = a(1 - \exp(-L(\text{puls}) - L(\text{thresh}))) / b \]

with L(\text{thresh}) is the absolute threshold level and a, b the parameters of the function.

We found the parameter a to be 25 dB for three observers at 1 kHz and for one observer at 0.5, 1.0 and 2.0 kHz.

We also measured similar plots for the pulsation criterion. Those figures show that the rising part of the pulsation threshold starts at about the same or a somewhat lower level and that the slope of the rising part is somewhat steeper in pulsation than in masking. It suggests a masked threshold
pattern contained within the pulsation pattern. This suggestion was verified in a direct experiment of which a result is shown in Fig. 3.

![Graph showing comparison of pulsation and masked threshold patterns.](image)

**Fig. 3.** Comparison of pulsation ans masked threshold pattern.

The data show that the masked threshold pattern is contained within the pulsation pattern. The masked threshold pattern reaches the absolute threshold for a frequency somewhat closer to the pulsator than the pulsator pattern does.

**Discussion and conclusion**

Masked threshold patterns are strongly nonlinear. Their dynamic range covers only 25 dB and the slope of the patterns depend on level in very much the same way as pulsation patterns do.

Comparison of pulsation and masked threshold patterns show that the pulsation pattern is determined by a transition from continuity perception towards pulsation, even in conditions of weak pulsator components or remote probe frequencies. We conclude that the entire pulsation pattern is based on continuity perception as long as we deal with normally hearing listeners. It is remarkable that for low-level pulsators there is only a small difference between the masked pattern and the pulsation pattern, while for high-level pulsators there is quite a range of levels where continuity is perceived. This might well explain why listeners find it difficult to measure low-level pulsation patterns.

I will be interesting to look for a physiological foundation of the masked threshold. In terms of the working hypothesis of Verschuure (1981b) it could well be that continuity is perceived as long as the absence of the probe during pulsator presentation cannot be detected. The masked threshold could be based on the amount of activity present during probe presentation and might thus reveal the adaptation of the auditory nerve under tone stimulation.

In view of our experiences with hearing-impaired listeners it will be interesting to repeat the study for such listeners.
MODULATION THRESHOLDS OF NARROW NOISE BANDS

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Introduction
Temporal variations of the amplitude of a sound signal are important carriers of acoustic information. This is one reason why amplitude modulation thresholds, i.e., the limits set for the ear in perceiving such variations, have been the object of psychoacoustic investigations for a long time /1,2/. Moreover, the question has been asked which mechanisms could be the basis of the perceptibility of amplitude variations /3,4/.

Experiments
As temporal effects are to be considered, the spectra of the sound signals used must not exceed the critical bandwidth /1/. Since the critical bandwidth is large at high frequency regions, sound signals with frequencies and center frequencies, respectively, of 5 kHz have been chosen for the investigations described in the following. The sound pressure level was always 80 dB re 20 μPa. Four subjects took part in the experiments; for further details see /4/.

All amplitude modulation thresholds resulting from earlier investigations /1-4/ have always clearly revealed the low-pass characteristic of the ear. Some typical thresholds \( m_{th} \) (\( m_{th} \) being the degree of the impressed amplitude modulation yielding a 50% discrimination) for the frequency region 5 kHz are plotted as a function of the modulation frequency \( f_{mod} \) in Fig. 1. For the sinusoidal tone (ST, •) as well as for narrow band

![Graph showing modulation thresholds](image)
noise (NBN) of the bandwidths \( \Delta f = 530 \text{ Hz} \) (o) and 314 Hz (A), respectively, the thresholds consonantly have a similar, increasing characteristic. In addition, another feature can be taken from Fig. 1: For one and the same modulation frequency, the threshold of a noise signal unambiguously depends on the bandwidth \( \Delta f \). In the case of a larger bandwidth it is always lower than in the case of a smaller one. If the amplitude modulation thresholds of NBN exhibit these two features, they shall be called "regular".

A striking deviation from this "bandwidth rule", however, can be observed in the case of the ST: In spite of its extremely narrow bandwidth, the ST does not show a higher, but always a lower threshold than the noise signals considered. The problem, which is to be dealt with now, is how the modulation threshold behaves when the bandwidth of the noise signal is more and more diminished.

For this reason, further experiments were carried out. They reveal "irregular" behavior of the thresholds as soon as the modulation frequency exceeds half the bandwidth of the unmodulated noise band \( f_{\text{mod}} > \Delta f/2 \). This "irregularity" means:

1. With increasing modulation frequency, the threshold is no longer increasing, but has a decreasing tendency.

2. With increasing bandwidth, the threshold is no longer decreasing, but has an increasing tendency.

The "irregularity" clearly shows up in the case of noise signals of very small bandwidths. As an example, the amplitude modulation threshold \( m_{\text{th}} \) for NBN at 5 kHz with the bandwidth \( \Delta f = 3 \text{ Hz} \) (o) is presented in Fig. 2 versus modulation frequency \( f_{\text{mod}} \). The "irregularly" decreasing tendency of the threshold in the whole range of low and medium modulation frequencies can be clearly made out. Additionally, also the threshold for a ST (●) at 5 kHz is given in Fig. 2. As can be seen, both curves get closer together with increasing modulation frequency. For high modulation frequencies \( f_{\text{mod}} > 10 \cdot \Delta f ... 20 \Delta f \), the modulation thresholds for NBN and ST finally agree to a large extent.

Modelling
The laws that manifest themselves in the results of the experiments are now to be summarized. For this purpose, already known and reliable conceptions are used, which have to be appropriately expanded. Our own model 4 is based on the assumption that NBN can be interpreted as a tone the amplitude of which is sto-
Fleischer, H.: Modulation thresholds of narrow noise bands

The degree of the inherent fluctuation acting in the ear depends on the bandwidth of the noise; it can be calculated with the aid of the model. An additionally impressed modulation is perceived as soon as it increases the inherent fluctuation by a certain factor. As pointed out earlier /4/, this model fully applies to slow modulations.

For faster modulations, however, it has to be expanded. If the modulation frequency becomes \( f_{mod} > \Delta f/2 \), the effect of the inherent fluctuation does not cease all of a sudden, but lessens gradually. Obviously, there is some kind of "cross talk" which, as a result, causes an impairment of the perceptibility of an impressed modulation, even when it fluctuates faster than the inherent fluctuation. The disturbing influence of the inherent fluctuation gradually recedes with increasing modulation frequency. It has practically disappeared as soon as the modulation frequency exceeds the 10...20fold of the bandwidth of the unmodulated noise signal.

Fig. 3 shows the amplitude modulation threshold \( m_{th} \) for NBN at 5 kHz with the bandwidth \( \Delta f = 10 \) Hz (see the small sketch) as a function of the modulation frequency \( f_{mod} \). The experimental data (o) are compared with the values resulting from the model. It can be seen that in the case of slow modulations (\( f_{mod} \leq \Delta f/2 = 5 \) Hz), the threshold is fairly well described by the already existing model (---). For higher modulation frequencies, however, the expansion has to be taken into account. The curve (---) is based on the assumption that the influence of the inherent fluctuation decreases with 16 dB per decade of the modulation frequency, starting at half the bandwidth of the unmodulated NBN. The "irregular" decrease of the modulation threshold is thus given correctly. For very fast modulations (\( f_{mod} > 200 \) Hz in the example), threshold values would be expected from the model which are smaller than those for a ST. But as demonstrated in Fig. 2, this cannot be the case; the threshold for NBN then coincides with that for the 5 kHz-ST (---) to a large extent. Thus, this example shows that the amplitude modulation threshold for a noise signal of a very small bandwidth is adequately described by the expanded model.

In addition to the actual field of the model, modulation thresholds of other carrier signals exhibiting an inherent amplitude modulation can also be calculated. In Fig. 4, the modulation threshold \( m_{th} \) of a 5 kHz-ST is indicated, that is sinusoidally pre-modula-
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ted with a constant modulation frequency \( f_{pm} = 5 \text{ Hz} \) (see the small sketch). The degree of the pre-modulation is 0.4; it has been chosen in such a way that the inherent fluctuation acting in the ear, according to /4/, corresponds to that of NBN with the bandwidth \( \Delta f = 10 \text{ Hz} \). This means that the model yields the same curve as in Fig. 3. As can be seen from the experimental data (o), the "irregular" decrease of the threshold of the impressed amplitude modulation is slightly steeper than in the case of the comparable NBN. Its overall characteristic, however, is very well reflected by the model for this type of sound signal, too.

Conclusions
From the evaluation of psychoacoustic experiments follows that amplitude modulation thresholds of NBN can no longer be described by existing models once the modulation frequency exceeds half the bandwidth of the unmodulated carrier signal. In order to give a description of the "irregular" gain of the auditory modulation sensitivity observed then, an expansion of these models becomes necessary. This expansion comprises the assumption of a "cross talk" of the inherent fluctuation of the carrier signal beyond the range of the modulation frequencies contained in the carrier signal. The disturbing influence of the inherent fluctuation is diminishing more and more the faster the impressed modulation is getting compared to the inherent fluctuation. Assuming a numeric value of 16 dB per decade for this decrease, the model works quite well with the experimental data with regard to quality and quantity. As indicated by other investigations not presented here, this does not only hold for high but also for medium and low frequency regions, provided that the spectrum of the modulated signal does not exceed a critical band. These ideas can be successfully applied to carrier signals with a stochastically fluctuating amplitude. Apart from this, as demonstrated by an example, they are also suited for the description of experimental results obtained with carrier signals, which are sinusoidally pre-modulated.

Literature
Florentine - Temporal Acuity as a Function of Level and Frequency

TEMPORAL ACUITY AS A FUNCTION OF LEVEL AND FREQUENCY

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Introduction

The purpose of the present paper is to examine the detection of a temporal pause in a band of noise as a function of level and frequency. We refer to the pause as a temporal gap and the minimal detectable gap as the MDG.

Fitzgibbons (1979) showed that the MDG decreases with increasing frequency of the signal. This result can be expected on the basis of our knowledge of the auditory filter. For frequencies above 500 Hz, the bandwidth of the auditory filter is approximately proportional to frequency (Zwicker & Feldtkeller, 1967; Schorff, 1970). The response of a filter decays with a time constant which is inversely proportional to the bandwidth. Therefore, we might expect that the MDG is inversely proportional to the center frequency of the signal or, at least, decreases as frequency increases. On the basis of measurements at two levels and three frequencies, Fitzgibbons suggested that the MDG decreased linearly with frequency. To determine whether the decrease is linear or inversely proportional to frequency, as we would expect from the consideration of the auditory filter, we measured the MDG over a wide range of levels and frequencies.

Method

The MDG in an octave band of noise was measured as a function of frequency and level in three normal listeners. The gap was produced by turning off and on the signal with fall and rise times of 1 ms. To eliminate the effect of spectral splatter, each octave band of noise was presented with its complementary band-stop masker. The masker was continuous and its spectrum level was equal to that of the signal. For comparison, the MDG in a wide-band noise (low-pass filtered white noise at 7 kHz) was also measured. The listener's task was to identify the interval containing the brief gap in the otherwise continuous signal. Each listener was tested on each condition three times using an adaptive, two-interval, two-alternative forced-choice paradigm with feedback. Each trial was initiated by the listener and consisted of two observation intervals marked by lights.
Results and Discussion

As shown in Figure 1, the MDG for the wide-band noise decreases with increasing level up to about 40 dB SPL, reaches a minimum at 60 to 70 dB SPL, above which it increases slightly with increasing level. This trend is also apparent in the data of Plomp (1964) and the data of Irwin et al. (1981). Performance for the octave-bands of noise improves with increasing level up to approximately 60 or 70 dB SPL, after which it either remains constant or becomes slightly worse. The MDG is smaller for the wide-band condition than any of the octave-band conditions. Probably information across the entire frequency range is combined to optimize the detection of the gap.

The effect of frequency is readily apparent: the MDG decreases by a factor 10 as the frequency increases from .25 to 8 kHz. For example, at a fixed level of 63 dB SPL, the MDG is approximately 54 ms at the .25-kHz center frequency and 5 ms at the 8-kHz center frequency. The effect of center frequency can be better seen in Figure 2. This figure contains the same data plotted in the previous figure but the MDG is plotted as a function of center frequency with level as the parameter. For levels of 33 and 43 dB SPL, the MDG decreases with increasing center frequency up to 4 kHz, after which it increases. For 63 dB SPL, the MDG decreases up to 8 kHz. The rise above 4 or 8 kHz probably is due to the normal increase in threshold towards higher frequencies, so that the signal was either inaudible or at a very low SL. For 83 dB SPL, the MDG continuously decreases with increasing center frequency. At the lowest level, the gap is almost inversely proportional to the frequency up to 4 kHz. At the higher levels, the functions tend to be flatter—meaning that the MDG decreases slower than inversely proportional to the frequency. Especially, above 4 kHz, the functions are quite flat showing almost no further decrease in MDG. The effect of frequency and the combination of information across frequencies appear to be at least part of the reason why listeners with high-frequency sensorineural impairments have enlarged MDGs (Boothroyd, 1973; Fitzgibbons, 1979, Irwin et al., 1981; Florentine & Buus, 1982).

Model

In an effort to understand our results we have investigated a simple model which is basically an energy-detector with a filter at the input. We have assumed that the ringing of the filter has an exponential decay. The time constant is assumed to be the inverse of the standard critical bandwidth as tabulated by Zwicker and Feldtkeller (1967). The output of the filter is converted into instantaneous power. To simulate the presence of an absolute threshold, we add a small DC component. The resulting function is then integrated and the energy, which is the output of the integrator, is led to a very simple detector. The rule for detection of the gap is that the energy output must decrease some fixed proportion of the steady-state output that was present before the onset of the gap. To account for the variability of the short-term energy of a narrow-band noise, we have let the criterion depend on the bandwidth of the filter and the integration time of the exponential integrator.
Florentine - Temporal Acuity as a Function of Level and Frequency

Our calculations show that the model predicts the general trends in the data: the MDG decreases as frequency increases, and for a given frequency it decreases with increasing level for the first 15 or 20 dB after which it stays relatively constant. However, there are three shortcomings in the model's predictions: First, the frequency dependence at high levels is only correct between 1 and 8 kHz. The predicted performance for 500 Hz is about 25% better than the measured performance, and at 250 Hz it is about 60% better. Second, the model predicts better performance at 14 kHz than at 8 kHz, whereas the data indicate no difference or even worse performance at 14 kHz than at 8 kHz. However, the difference between the model's predictions and our data at the extreme frequencies may be due to the rapid increase in threshold at these frequencies which renders the effective bandwidth of the signal less than one octave. Third, the predictions as a function of level are not entirely correct. The predicted increase of the MDG at low levels appears slightly more severe than that indicated by our data and the model predicts constant performance above some level, whereas the data consistently show that the MDG increases above 60 or 70 dB SPL. Although small, we believe the increase at the high levels is real because it is apparent in all our data and is also evident in the data of Plomp (1964) and Irwin, et al. (1981). Despite these problems, we find the model helpful in understanding the data. It suggests that the peripheral filter plays an important role in determining the temporal resolution of the auditory system. We are currently investigating the properties of this general type of model and attempting to refine it to yield predictions in better agreement with the data.

Acknowledgements

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References

Figure 1. The minimum detectable gap for various bands of noise is plotted as a function of SPL (bottom abscissa) and spectrum level (top abscissa). Each data point is the geometric mean from three normal listeners. The vertical bars show the standard deviations.

Figure 2. The minimum detectable gap at various SPLs is plotted as a function of center frequency. Each point is replotted from Figure 1.
SPEEDED DISCRIMINATION OF FREQUENCY AND INTENSITY DIFFERENCES

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1. Introduction

According to the classical model of Zwicker (1970), the neural correlate of a just noticeable frequency difference between two pure tones is of the same nature as the neural correlate of a just noticeable intensity difference. Recently, however, it has been claimed on various grounds that separate neural information is actually used for the discrimination of these two types of difference. The present contribution to this issue consists of two experiments in which the discriminability of differences between pure tones was measured in one subject (the author) within the framework of Schouten and Bekker's (1967) "forced reaction time" paradigm.

In both experiments, successive pairs of 40 ms pure tones were presented to the subject's right ear. The first tone of each pair was fixed (1500 Hz, 70 dB SL) and the second, occurring 500 ms later, could differ from it or not; the subject thus had to classify the pairs in two categories, "AA" and "AB". The pairs were grouped in sequences which comprised 20 "AA" pairs and 20 "AB" pairs, randomly ordered and separated by 1.5 sec. intervals. Each pair was followed by a click. The subject's task was to give responses in temporal coincidence with this click, by pressing one button with the left forefinger for "AA" responses and another button with the right forefinger for "AB" responses. In addition to the responses themselves, the actual response delays or "reaction times" (RT's) were recorded.

2. Experiment 1

The aim of this experiment was to compare three speed-accuracy tradeoff functions, each expressing the discriminability for various RT's of a particular difference B-A between the tones A and B: a positive intensity difference in the first case, a positive frequency difference in the second, a negative intensity difference in the third. For three differences having the same discriminability at a given mean RT, it seemed interesting to see whether or not there is one and the same speed-accuracy tradeoff.

The three difference values adopted in the experiment, I+ = +1.18 dB, F+ = +8.3 Hz, and I- = -1.09 dB, were selected on the basis of psychometric
functions (d' vs. B-A) obtained while the click delay, i.e., the interval between the second tone of each pair and the following click, was fixed at 750 ms. I+, F+, and I- were the difference values for which, at this "long" delay, d' was estimated at 2.5. In the main phase of the experiment, the click delay was set successively at 750, 450, 350, and 275 ms. In each condition of difference (I+, F+, I-) and delay, the subject was presented with 36 practice sequences and 60 test sequences. The 60 test sequences (2400 test pairs) were arranged in 5 successive blocks of 12 sequences and a separate d' was computed for each block.

Table 1

<table>
<thead>
<tr>
<th>Click delay</th>
<th>750 ms</th>
<th>450 ms</th>
<th>350 ms</th>
<th>275 ms</th>
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<td>I+ = +1.18 dB</td>
<td>2.46 (.16)</td>
<td>2.36 (.19)</td>
<td>1.92 (.08)</td>
<td>1.09 (.13)</td>
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<tr>
<td>mean d' (s.d.)</td>
<td>702 (31)</td>
<td>425 (24)</td>
<td>334 (24)</td>
<td>267 (23)</td>
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<tr>
<td>mean RT (s.d.)</td>
<td>2.42 (.18)</td>
<td>2.40 (.12)</td>
<td>2.01 (.18)</td>
<td>1.10 (.16)</td>
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<tr>
<td>F+ = +8.3 Hz</td>
<td>703 (27)</td>
<td>426 (25)</td>
<td>337 (24)</td>
<td>269 (27)</td>
</tr>
<tr>
<td>mean d' (s.d.)</td>
<td>2.51 (.24)</td>
<td>2.35 (.15)</td>
<td>1.96 (.07)</td>
<td>1.03 (.21)</td>
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<tr>
<td>mean RT (s.d.)</td>
<td>699 (26)</td>
<td>419 (23)</td>
<td>331 (28)</td>
<td>270 (29)</td>
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</tbody>
</table>

Table 1 displays the mean and standard deviation (s.d.) of the 5 d's and 2400 RT's obtained in each of the 12 test conditions. A clear-cut conclusion can be drawn from these results: in spite of the large number of data collected, there is no significant difference between the three speed-accuracy tradeoff functions.

3. Experiment 2

The aim of Experiment 2 was to compare "redundancy gains" resulting from the combination of a frequency difference f with an intensity difference i. In studies where response delays were not strictly constrained (Harris et al., 1958; Pollack, 1961; Zagorski, 1975), it has been shown that a two-dimensional difference f+i is appreciably easier to discriminate than each of its components f and i. If some response delay is imposed, a similar gain in discriminability may be expected when the delay is long, but what will happen when this delay is very short?

Experiment 2 comprised three successive conditions (see the legend of Table 2). Each condition consisted in 21 blocks of 12 sequences, the first three blocks being considered as practice runs. Within each block, B-A was respectively f, i, and f+i in four sequences. For each type of difference in a given condition, 72 test sequences (18 test blocks x 4 sequences) were thus presented. They were considered as six successive groups of 12 sequences and a separate d' was computed for each group.

In Condition 1 (click delay: 750 ms) and Condition 2 (click delay: 275 ms), the selected values of f (+8.6 Hz) and i (-1.09 dB) were about equal to the values of F+ and I- in Experiment 1. For these two unidimen-
sional differences, mean d'\(a\) equivalent to those obtained in the previous experiment were thus expected. It can be seen in Table 2 that this expectation was satisfied. In the case of Condition 3, \(f\) and \(i\) were determined on the basis of the mean d'\(a\) obtained in Conditions 1 and 2 for +8.6 Hz and -1.09 dB. It was my intention to choose in Condition 3 values of \(f\) and \(i\) for which mean d'\(a\) equal to those obtained in Condition 1 should have been observed if for a click delay of 275 ms, d'\(a\) was the same function of B-A as in a condition of long click delay or free response delay; the corresponding theoretical values were +8.6 Hz x 2.43/1.16 = +18.0 Hz and -1.09 dB x 2.43/1.09 = -2.43 dB. An examination of Table 2 reveals that with such difference values, the mean d'\(a\) actually obtained were larger than in Condition 1; this is a noteworthy finding given that the mean RT's are not larger in Condition 3 than in Condition 2.

Table 2

<table>
<thead>
<tr>
<th>Condition</th>
<th>(f) (RT)</th>
<th>(i) (RT)</th>
<th>(f') (RT)</th>
<th>Gain (s.d.)</th>
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<tr>
<td>1</td>
<td>2.43 (699)</td>
<td>2.43 (699)</td>
<td>2.97 (701)</td>
<td>.224 (.077)</td>
</tr>
<tr>
<td>2</td>
<td>1.16 (275)</td>
<td>1.09 (271)</td>
<td>1.23 (272)</td>
<td>.098 (.073)</td>
</tr>
<tr>
<td>3</td>
<td>2.79 (268)</td>
<td>2.99 (265)</td>
<td>3.17 (266)</td>
<td>.096 (.054)</td>
</tr>
</tbody>
</table>

Since in all conditions, very similar mean RT's were observed for sequences involving \(f\), \(i\), and \(f'\) (cf. Table 2), a fair comparison between discriminality gains was possible. The last column of Table 2 presents, for each condition, the mean and standard deviation of six gain values; these gain values were derived from the three d'\(a\) (\(f\), \(i\), \(f'\)) observed for each group of three successive test blocks and computed according to the formula:

\[
\text{Gain} = \left(\frac{2d'_{f'}}{(d'_{f} + d'_{i})}\right) - 1.
\]

Of the three mean gains presented in Table 2, two are very similar (Conditions 2 and 3) and the third (Condition 1) is about twice as large. Two-tailed \(t\) tests were used to compare the mean gain of Condition 1 with the other two mean gains: according to these tests, the differences are significant at the .02 level.

4. Discussion

Insofar as they concern only one subject, the data reported here need to be replicated in a more extensive research. Nevertheless, they have in themselves clear implications with respect to Zwicker's model, and more generally to any theory assuming that the detectability of differences in frequency and in intensity between pure tones depends on a single neural process. Theories of this type are compatible with the results of Experiment 1, but not with those of Experiment 2. In Experiment 2, it was observed that, at least when 1 < d' < 3, the psychometric functions of unidimensional differences (\(f\) or \(i\)) have a steeper slope for a short click
delay (275 ms) than for a long click delay (750 ms). In such conditions, the single process hypothesis predicts that redundancy gains will be larger when the click delay is short. What was actually found is just the opposite. The fact that smaller gains were obtained at the short click delay does not only support the notion that separate neural information is used for the detection of frequency and intensity differences; it also seems to indicate that frequency and intensity differences cannot be detected simultaneously.

Note that in the above statements, the terms "frequency" and "intensity" could be respectively replaced by "pitch" and "loudness": For a standard tone such as the one used in this investigation (1500 Hz, 70 dB SL), small frequency changes have an effect on pitch but not on loudness and conversely, small intensity changes affect loudness but not pitch. It should be pointed out in this respect that the main data supporting Zwicker's model were obtained in experimental situations where this independence of pitch and loudness did not exist (see Coninx, 1978).

Finally, it is interesting to consider the relationship between the present results and the findings of Pollack (1961) and Zagorski (1975). The conclusion at which these two authors arrived is quite compatible with the one formulated here. There seems to be, however, a discrepancy between the data: The mean gain observed in Condition 1 of Experiment 2 (long click delay) is definitely smaller than the gains described by Pollack and Zagorski. A factor of tone duration might be invoked to account for this effect; the exact source of the discrepancy, however, is not clear as yet.

References

PITCH SHIFT OF A TONE BURST ON THE PRESENCE OF PRECEDING TONE

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1. Introduction

The pitch is one of the important attributes of tones and it is primarily determined by their frequency. Many investigations have been made that the pitch is changeable by means of some other factors such as intensity, duration, masking stimuli and so on.

S.S. Stevens (1935) reported that the pitch of pure tones systematically depends on their intensity; by increasing intensity, lower tones become slightly lower, high tones slightly higher. Later, many reports have published concerning with the pitch-intensity relationship. However, these results differ among authors and sometimes they are in conflict with one another, so that the effect of intensity on pitch is considered to be little.

It is said that the effect of duration on pitch is small, too, at least for the tone whose duration is so long that the tone-pitch can be heard.

There are some reports on the effect of masking, adaptation or fatigue on the pitch. Békésy investigated the pitch shift of a tone by prior stimulation in the frequency region of the tone. According to his reports, the pitch shift is upward for test frequencies above the fatiguing tone and downward for test frequencies below the fatiguing tone. He explained this phenomenon as follows; After fatigue, a great number of hair cells on one side of the place of stimulation for test tone, is switched out and the point of maximal stimulation for the test tone moves from that place toward the region which is less fatigued. Later, Larkin has studied this subject and got almost the same result.

Numerous investigations on pitch shifts of tones due to masking have been made. Békésy investigated that the pitch of a tone burst can be raised or lowered by simultaneous masking. The results showed that the addition of the low-frequency band of noise to a tone always resulted in a shift in pitch of the tone to a lower frequency. When a high-frequency band of white noise is used, the pitch of the masked tone rises with increasing cutoff frequency. He gave the possible explanation of pitch shifts in masking conditions by the plane theory of hearing.

Rakowski and Hirsh (1980) measured the pitch shift owing to temporal masking. Their results showed the shift opposite to the Békésy's, that is; The pitch shifts of a test tone were downward for frequencies above the preceding tone(masker) and upward for frequencies below the preceding tone.
The preceding tone used by them was one with the duration of 500 msec and was presented at 70 dB SPL so that it could not be a fatigue tone if it was presented singly. They referred this to an adaptation phenomenon, while their explanation is as the same as one by means of fatigue.

In present experiment, the pitch shift of tone bursts was obtained using shorter preceding tones in order that the effect of the auditory fatigue were excepted. Moreover, measurements were performed in an individual session approximately five minutes each, in order not to fatigue the ear. The effect of frequency separation of the preceding and the test tone and time interval between them on pitch shift was investigated. We also investigated the mechanism of the pitch shift and estimated the shift by means of the time window.

2. Methods

Each of the stimuli was composed of a preceding tone, a standard tone and a comparison tone as shown in Fig.1. The standard tone and the comparison one were 20 msec in duration and the preceding tone was a 200-msec tone burst. All tones had 3.5 msec rise-fall time. Sound pressure level of stimuli was to be approximately 70 dB SPL at the headphone(TWATSU DR-305) and stimuli were presented diotically.

A two-alternative forced-choice procedure were used. The standard tone was presented at 1000 Hz and the frequency of the comparison one differed from 1000 Hz by \( \Delta f = \pm n \cdot df, n = 1, 2, 3, 4 \) or 5. To avoid the absolute judgment, roving standard procedures(Harris, 1952) were used; that is, a frequency jitter(within \( \pm 2 \text{df Hz} \)) was imposed upon both tones. The higher-frequency signal was presented first in half of the trials according to a random schedule. A single experimental session included 103 trials where the first 3 trials were always discarded as those for practice. The results of five sessions or more were combined to determine the pitch shift for each subject. More than 50 trials were done for each value of \( \Delta f \). The resultant psychometric functions were determined based on 500 trials or more. The frequency correspond to 50 % correct judgment was determined by the least-square solution using the Müller-Urban weights. The frequency difference between this frequency and one of the standard tone was used as a measure of pitch shift. The time interval between the preceding tone and the standard tone \( \Delta t \), and the frequency of the preceding tone were held constant within each set of trials. The frequencies of the preceding tone were varied from 900 Hz to 1100 Hz. The subject's task was to select the higher-pitched signal and press an appropriate response button.

Subjects were seven young adults with normal hearing and their ages ranged from 20 to 23. They were trained for a few days or more before the experiments were begun.

3. Results

The results are shown in Fig.2 on the average of seven subjects. The abscissa represents the frequency difference between the preceding tone.
and the standard one \((f_p - f_s)\) and the ordinate shows the pitch shift in Hz. The parameter in the figure is the time interval \((\Delta t)\) between the preceding tone and the standard tone. As can be seen in this figure, the pitches of the standard tone shift upward when the frequency of the preceding tone is higher than one of the standard, on the contrary, the pitch shifts are downward for preceding-tone frequencies below the standard tone. The direction of the shift is the same as one in simultaneous masking after Békésy, while they are contradictory to the results obtained by Rakowski et al. The difference in the experimental conditions between ours and theirs, lies in the duration of the preceding tone and the psychophysical method used (we used the constant method, while they used the method of adjustment).

![Pitch shift diagram](image)

**Fig. 2** Poststimulatory pitch shift for short tone bursts as a function of frequency difference between the preceding tone and the standard one. The parameter is time interval.

Then, in order to investigate the effect of the duration of the preceding tone on pitch, we made another experiment as follows; All of the experimental procedures were identical to those of the previous experiment, except the duration of the preceding tone. \(\Delta t\) was set 30 msec (constant) and the duration of the preceding tone was changed as a parameter. Fig. 3 represents the results. The abscissa shows the duration of the preceding tone and the ordinate shows the pitch shift. As is evident from this figure, the direction of pitch shifts is changed with the increase of the duration. The same results can be seen in Fig. 2 of Rakowski (1980). It is speculated that in condition of longer preceding tone than 300 msec, the effect of adaptation is produced.

**Fig. 3** Pitch shift of short tones as a function of the duration of preceding tone.

\[ h_i \]

**Discussion**

We assume that the auditory system is the frequency analyser with a certain time window. The pitch shift due to the preceding tone is explained by means of a suitable time window. The preceding tone as well as the standard tone are included in a time window for small \(\Delta t\). The pitch of the standard tone shifts to one of the preceding, for the same reason as the pitch shift owing to simultaneous masking mentioned above.
Suppose that the time window is
\[ h(t) = \exp(-\alpha t) \quad \text{for } t \geq 0. \]
Spectral patterns obtained by means of this window are shown in Fig. 4. Moreover, suppose that the pitch of brieftones is given by the averaged frequency which is weighted by the energy involved in the critical band. Fig. 5 shows the pitch shift calculated in this way. The calculated values correspond comparatively well to the measured one (Fig. 2). This means the time window works properly in modeling frequency analysis of hearing.

![Spectral patterns of short tones observed through the time window.](image)

Fig. 4 Spectral patterns of short tones observed through the time window.

![Pitch shift estimated by the model.](image)

Fig. 5 Pitch shift estimated by the model.

References
LOCALIZED PRINCIPAL PITCH OF FM-AM TONES AS A FUNCTION OF
PHASE DIFFERENCE BETWEEN FREQUENCY MODULATION AND AMPLITUDE
MODULATION

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Introduction
When we listen to frequency modulated tones like vibrato
tones, we perceive not only frequency fluctuation but also
somewhat steady pitch. This pitch sensation of vibrato
tones is defined as "principal pitch".1) According to
recent studies, the principal pitch is located around the
carrier frequency (middle of the extent of frequency
modulation) when the modulation wave is symmetrical.2)–4)
These studies were done when only frequency is modulated.
Tiffin studied where the principal pitch is located also
when frequency and amplitude are modulated simultaneously.5)
But, as the experimental conditions were not defined
clearly, only qualitative tendency is presented.

In this study, we set physical properties related to
localized principal pitch of FM-AM tones (vibrato tones
whose frequency and amplitude are modulated by same
modulation wave simultaneously) definitely, and studied
where the principal pitch is located as a function of these
properties. And, we provided the psychoacoustical model to
perceive principal pitch of FM-AM tones.

Experimental method
The experiment was done by a method of adjustment.
Subjects were suggested to match pitch of a matching pure
tone to principal pitch of a FM-AM tone, while they compared
two signals alternatively. Signals were presented
diotically through a headphone (Stax SR-X/MK-3). Matching
pure tones were presented at a level of 70 dB SPL, and the
time average of amplitude of FM-AM tones was set equal to
the amplitude of matching pure tones. The carrier waveform
is sinusoidal, and the carrier frequency is 440 Hz.

The experiment was divided into three cases. In Case 1,
the degree of amplitude modulation (Am) was set a constant
value of 1.00. And, we studied where the principal pitch of
FM-AM tones is located as a function of the extent of
frequency modulation (E) when the phase difference between
frequency modulation (FM) and amplitude modulation (AM) is in-phase and anti-phase. "In-phase" means that when frequency becomes higher, amplitude becomes larger, and when frequency becomes lower, amplitude becomes smaller. Oppositely, in the case of "anti-phase", when frequency becomes higher, amplitude becomes smaller. In Case 2, E was set a constant value of 100 cents, and the similar experiment to Case 1 was done as a function of Am. In Case 1 and 2, the modulation waveform is triangular and the modulation rate is 6Hz.

In Case 3, the phase difference between FM and AM is varied 0° to 315° at intervals of 45° when E is 50 cents and Am is 1.00. The modulation waveform is sinusoidal, and the modulation rate is 7 Hz.

Five subjects with normal hearing, aged between 21 and 29 years, participated.

Results
Results are represented in Fig.1. The principal pitch of FM-AM tones is shifted higher than the carrier frequency when the phase difference between FM and AM is in-phase, and lower when it is anti-phase in Case 1 and 2. The deviation from the carrier frequency to the localized principal pitch in cents (P) increases proportionally to E in Case 1. Solid lines in Case 1 are regression lines between P and E.

In Case 2, regression lines between P and Am² are represented. As Am² is set equal interval along the abscissa in Case 2, these relationships are represented as lines. There are Tiffin's data 5), too, in Case 2. They presented similar tendency to our data. But, we cannot compare our data with his quantitatively as he did not define the value of Am. His data are only these.

In Case 3, the principal pitch of FM-AM tones is shifted as a function of the phase difference between FM and AM.

When the carrier frequency was set 800 and 1500 Hz, the similar tendency was obtained from the similar experiment, too.

Discussion
From experimental results, there seems to be the pitch averaging mechanism weighted loudness fluctuation in the perceived process of principal pitch of FM-AM tones. As the power law 6) is able to be applied to the relationship between loudness and amplitude, this model is formulated as:

$$P = \frac{\int_0^T C(t) \cdot A(t)^a \, dt}{\int_0^T A(t)^a \, dt} \quad \text{(cents)} \quad (1)$$

Where C(t) is FM function in cents, referred to the carrier frequency with respect to time t, A(t) is AM function, T is the modulation period, a is the exponent.

C(t) also must be transformed into pitch modulation function. But, as the extent of FM is small value within 100 cents, we omit this transformation.
First, we discuss when the phase difference between FM and AM is in-phase and the modulation waveform is triangular, as is in Case 1 and 2. Eq.(1) becomes

$$P = \frac{E \cdot (1+Am)\alpha + 2 - (1-Am)\alpha + 2}{2Am \cdot (1+Am)\alpha + 1 - (1-Am)\alpha + 1 - 1}. \quad (2)$$

We substitute Am=1.00 into Eq.(2) to compare with the experimental results in Case 1. And, as the experiment was done under diotic listening condition, $\alpha=0.6$ 6) is substituted. So, Eq.(2) becomes

$$P = 0.12E. \quad (3)$$

Similarly, in anti-phase condition, we obtain:

$$P = -0.12E. \quad (4)$$

Eq.(3) and (4) almost correspond with the regression lines in Case 1. And, the prediction value of $P$ is represented by the dashed line in Case 3 of Fig.3 when experimental conditions in Case 3 are substituted into Eq.(1). The experimental results are slightly lower than the prediction values. But, the relative shift process of $P$ corresponds with each other as a function of the phase difference between FM and AM.

But, we cannot derive the relationship that $P$ is proportional to $Am^2$ in Case 2 from Eq.(3). As an example to study the relation between subjective modulation depth of AM tones and Am, there is "Study on roughness sensation of AM tones",7) In this study, "roughness" is defined as a scale of subjective fluctuation of AM tones whose modulation rate is about 50 Hz. And, roughness is supposed to be proportional to $Am^2$. In our study, as the modulation rate is 6 Hz, roughness is not able to be perceived. But, as $P$ is proportional to $Am^2$, there seems to be similar auditory mechanism to perceive roughness in the process to perceive principal pitch of FM-AM tones.

Conclusions

The principal pitch of FM-AM tones is shifted as a function of the phase difference between FM and AM. We suppose that the process of principal pitch shift is accomplished by the pitch averaging mechanism weighted with loudness modulation function, combined with the similar mechanism to perceive roughness.

References
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**Case 1** Degree of AM is 1.00.
**Case 2** Extent of FM is 100 cents.

Prediction by pitch averaging mechanism

- : Average value
- : Standard deviation
* : Anti-phase between FM and AM (Others are in-phase in Case 1 and 2)

**Case 3** Degree of AM is 1.00, and Extent of FM is 100 cents.

Fig.1 Localized principal pitch of FM-AM tones in cents, referred to the carrier frequency of 440 Hz in each case.
HARMONIC AND ANHARMONIC DICHOTIC PITCHES

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Introduction

If there exists a more or less harmonic relation between the frequency components of a complex sound, a low pitch arises that can be described on the basis of the internal spectral representation of that sound. Modern pitch theories (Goldstein, 1973; Terhardt, 1972; Wightman, 1973) all adopted the concept of a spectral pattern processing for pitch perception. In this, signals having an anharmonic spectrum occupy a special place. Ritsma (1976) introduced the idea of spectral dominance to account for the ambiguous pitch that appears for anharmonic signals. For instance, the pitch of cosine noise, i.e., noise with a harmonic cosine-shaped power spectrum that is generated by adding white gaussian noise to its over $\tau$ delayed replica, corresponds to $1/\tau$ (compare Fig.1a). Do we perform a subtraction of white noise and the same noise delayed over $\tau$ we obtain an anharmonic powerspectrum like it is plotted in Fig.1a (dashed line). The corresponding pitch is ambiguous and given by $0.87/\tau$ and $1.14/\tau$ (Bilsen, 1966). This is in agreement with a spectral dominance for monaural pitch perception of the fourth harmonic.

The existence of dichotic pitch has played an important role in the development of pitch theories. So Bilsen and Goldstein (1974) applied noise and the same noise delayed, not added monaurally, but offered dichotically instead, i.e., the noise in one ear and the delayed noise in the other ear. The evoked (faint) pitch sensation (dichotic repetition pitch, DRP) could be explained by assuming the generation of a central spectrum due to binaural interaction. This spectrum, with the same characteristics as the spectrum of monaural cosine noise, is plotted in Fig.1a in an idealized form.

It has been argued (Bilsen and Goldstein, 1974; Bilsen, 1977) that the central spectrum of dichotic pitch configurations in general is processed by the pitch extractor in the same way as assumed for monaural signals. This "central spectrum" model also includes the spectral dominance concept, though in this case the dominant spectral region consists of one fixed frequency region around 600 Hz. Lateralization measurements (Bilsen and Raartgever, 1973; Raartgever, 1976, 1980a) showed that for lateralization of low-frequency signals due to interaural time- or phase-differences this very frequency region is dominant in the broad-band behaviour. It is this frequency region around 500-600 Hz also, that is optimal with respect to many other binaural phenomena.
Here we intend to report dichotic pitch matchings for harmonic and anharmonic central spectrum configurations in order to test the spectral dominance concept and to develop further the model of binaural interaction mentioned in the next section.

Model

The central spectrum theory mentioned before has been extended into a model of binaural interaction that describes dichotic pitch phenomena, lateralization and binaural unmasking (BMUD's) in a consistent way (Raatgever and Bilsen, 1977; Raatgever, 1980 a,b). The basic elements of the model are a sharp peripheral frequency analysis and, dependent on an internal (neural) delay, a coincidence of the signals from corresponding peripheral filters at both ears under preservation of their time structures. As Raatgever and Bilsen described this leads to the generation of internally projected activity patterns in which the activity (or power) is a function of frequency and internal delay. These patterns are being scanned by the central processor in such a way that salient spectral patterns are selected. This selection constitutes the lateralization process. Spectral information such as harmonicity and modulation depth is the important clue in this selection. This is illustrated in Fig. 1 for an idealized MPS configuration.

MPS-noise is a dichotic noise configuration that results in a rather strong dichotic pitch sensation (Bilsen, 1976). Anharmonic MPS is the result of feeding white gaussian noise into a delay-line with inverted input and a delay $\tau$. A fraction $g$ of the thus inverted and delayed signal at the output is fed back to the input. A negative fraction $-g/(1-g)$ of the noise at the input is added to the output. It can easily be shown that the resulting noise signal has a flat power spectrum and sharp phase transitions of $2\pi$ at anharmonic frequencies: $(n+1)/\tau$ (n=0,1,2,...) in relation to the signal at the other ear which is the original noise balanced for equal power. Were we to add the signals at both ears together, a comb-like spectrum would arise as plotted in Fig.1b (dashed line). The harmonic case, also plotted in Fig.1b, is the result of an analogous operation, however, without the inverter at the input of the delay-line.

Such a dichotic signal configuration gives rise to a complicated internal activity pattern in the model. A cross section at a certain power level is shown in Fig.2a. The hatched areas indicate the intersection of the ridges in the pattern. The open areas indicate that the pattern has less power there. It is clear that at an internal delay of 0 ms one encounters
sharp ridges that are perpendicular to the frequency axis at the anharmonic frequencies concerned. The internal delay corresponds to the place of the percept. The percept that is lateralized in the centre has a spectrum with sharp equidistant peaks at these frequencies (Fig.2b, upper graph) leading to a dichotic pitch sensation. The power spectrum in the centre can be considered, in first order approximation, to be the result of an addition of the signals at both ears. The same reasoning holds for harmonic MPS. The corresponding idealized power spectra of MPS are plotted in Fig.1b.

In a similar way the origin of other dichotic pitch phenomena like fourcin pitch (FP), see next section) can be described.

Experiments and discussion

Bilsen and Goldstein (1974) showed pitch matchings for harmonic and anharmonic dichotic pitch configurations. They measured dichotic repetition pitch (DRP) and fourcin pitch (FP). DRP has already been introduced in the former sections. FP is the result of a signal configuration consisting of two uncorrelated dichotically delayed noises (Fourcin, 1962). One noise, like DRP, has an interaural delay \( \tau \) between about 2 to 10 ms, while the second noise superimposed on the first is interaurally delayed less than about 1 ms. This second noise can be understood in the model to have the function of masking irrelevant information in the central pattern due to the first and thus favouring the corresponding pitch information (Raatgever, 1980b). The dichotic pitch sensation of FP is therefore stronger. Under certain conditions the internal power spec-

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**Fig.2** Central activity pattern of anharmonic MPS (\( \tau = 5 \) ms).
- a. cross section at level a (equal power contours).
- b. power spectra at internal delays 0 ms and 8 ms respectively.

**Fig.3** Pitch matchings of fourcin pitch (FP). From: Bilsen and Goldstein (1974).
trum responsible for the pitch sensation is identical with the spectrum for DRP (compare Fig.1a). The results of Bilsen and Goldstein as far as the results of an anharmonic FP configuration is concerned are presented in Fig.3 for two subjects. From these and similar results for DRP they conclude for the pitch to be ambiguous and given by: \(1/\tau \pm 0.0008\). This corresponds to dominance of the 600 Hz frequency region.

Our observations concerning MPS pitch showed that for harmonic MPS the pitch relation \(1/\tau\) is very well matched. MPS-measurements for the anharmonic MPS situation will be presented at the meeting. The pitch matchings were carried out, like those for harmonic MPS, by using a match or reference stimulus that was made to sound as much like MPS as possible. It consisted of a periodic pulse of given repetition rate added to a noise with carefully balanced level. This signal was presented diotically and had to be matched with the MPS signal in a computer-controlled pitch match procedure. Preliminary results lead to the conclusion that ambiguous low pitch arises only if sufficient spectral information is present in the dominant region around 600 Hz, i.e. for \(\tau > 3\) to 4 ms. For \(\tau < 3\) ms the pitch matches the lowest (anharmonic) component in the spectrum.

Whether the, for monaural signals, postulated relative dominant frequency band around the fourth harmonic also plays a role in the perception of dichotic pitch is still subject of investigation. However if it does, its role is obscured by the dominance of the 500 to 600 Hz frequency region in many binaural phenomena.

References


FLUCTUATION STRENGTH OF FM-TONES

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INTRODUCTION

Sinusoidally frequency-modulated pure tones (FM-tones) elicit one of two different kinds of auditory sensation, depending on the speed of modulation. In the case of modulation frequencies above about 20 Hz roughness is perceived (Kemp, 1982). Lower modulation frequencies lead to an auditory sensation called fluctuation strength. While fluctuation strength of amplitude-modulated tones has been described in several papers (e.g. Terhardt, 1968, Schöne, 1979, Fastl, 1982a), fluctuation strength of FM-tones as yet is largely unexplored. This paper describes the dependence of fluctuation strength of FM-tones on modulation frequency, frequency deviation, center frequency, and sound pressure level.

METHOD

Seven subjects with normal hearing, 26-37 years of age, took part in the experiments, which took place in a sound-isolated booth. The FM-tones were presented monaurally through an electrodynamic earphone (Beyer DT 48) with a free-field equalizer (Zwicker and Feldtkeller, 1967, p. 40). A method of magnitude estimation was applied, involving comparisons between pairs of FM-tones. The tones were switched on and off at minima of the frequency modulation, i.e. at the lowest actual frequency within a cycle. To avoid audible clicks, the amplitude of the FM-tones was switched on and off using a Gaussian-shaped gating signal with 10 ms rise/fall time. The FM-tones had a duration of 4 s and were separated by an interstimulus interval of 800 ms. The pauses between pairs lasted 3 s. The first FM-tone of a pair was assigned a number (e.g. 100) representing the magnitude of its fluctuation strength. Relative to this standard, the subjects had to scale the fluctuation strength of the second FM-tone within each pair. The subjects were asked to base their judgements only on fluctuation strength and to ignore differences in timbre, pitch, loudness, and in particular roughness. For each stimulus parameter, two sets of experiments were performed: one with a standard of large fluctuation strength (assigned 100) and another with a standard of small fluctuation strength (assigned 10). During a session, each combination of standard and comparison was presented four times in random order. The corresponding four numbers, assigned to identical stimuli by each subject, generally differed by less than ± 10, indicating only small intra-individual differences.
For each comparison the responses of the seven subjects were compiled, leading to a total of 28 datapoints, for which medians and interquartile ranges were calculated. For each standard, the correlated medians and interquartiles were normalized relative to the maximal median value, which was set to 100% relative fluctuation strength. In the figures, the standards are indicated by filled symbols.

RESULTS AND DISCUSSION

Fig. 1 shows the dependence of fluctuation strength of FM-tones on modulation frequency. The fluctuation strength F relative to its maximal value $F_{\text{max}}$ is plotted as a function of modulation frequency $f_{\text{mod}}$. The center frequency of the FM-tones was $f_m = 1500$ Hz, the frequency deviation $\Delta f = 700$ Hz and the sound pressure level $L = 70$ dB.

![Fig. 1. Fluctuation strength of sinusoidally frequency modulated pure tones as a function of modulation frequency. $f_m = 1500$ Hz, $\Delta f = 700$ Hz, $L = 70$ dB. Circles: standard with $f_{\text{mod}} = 4$ Hz; squares: standard with $f_{\text{mod}} = 0.5$ Hz.](image)

The results depicted in Fig. 1 indicate that fluctuation strength of FM-tones shows a bandpass characteristic with a maximum at $f_{\text{mod}} = 4$ Hz. The two standards yielded almost the same results, except at $f_{\text{mod}} = 8$ Hz where large interquartiles show up due to inter-individual differences. Fluctuation strength of amplitude-modulated tones and broadband noise also exhibits a bandpass characteristic as a function of modulation frequency, with a maximum around 4 Hz (see Fastl, 1983). Even roughness of FM-tones shows a bandpass characteristic, however, with a maximum near 70 Hz modulation frequency (Kemp, 1982). It diminishes for lower modulation frequencies and becomes almost negligible around 16 Hz, where fluctuation strength begins to take over as the dominant sensation.

Fig. 2 shows the dependence of fluctuation strength of FM-tones on frequency deviation. For FM-tones with $f_m = 1500$ Hz, $f_{\text{mod}} = 4$ Hz and $L = 70$ dB, fluctuation strength starts to be perceived at about $\Delta f = 20$ Hz and increases approximately linearly with the logarithm of frequency deviation. Again, both standards lead to almost the same results. Significant values of fluctuation strength ($F/F_{\text{max}} = 10\%$) are reached if the frequency deviation exceeds about 10 JNMDs (for details see Fastl, 1983). For roughness of FM-tones, Kemp (1982) found an increase slightly faster than linear with the logarithm of frequency deviation.
Fig. 2. Fluctuation strength of sinusoidally frequency modulated pure tones as a function of frequency deviation.

$f_m = 1500$ Hz, $f_{mod} = 4$ Hz, $L = 70$ dB. Circles: standard with $\Delta f = 700$ Hz; squares: standard with $\Delta f = 32$ Hz.

Fig. 3 shows the dependence of fluctuation strength of FM-tones on center frequency. A frequency deviation of 200 Hz was used throughout. Up to a center frequency of about 1000 Hz, fluctuation strength is constant and decreases approximately linearly with the logarithm of $f_m$ towards higher frequencies. This decrease can be understood in terms of the number of critical bands encompassed by FM-tones at different center frequencies but with a constant frequency deviation of $\Delta f = 200$ Hz. For example, the FM-tone at 500 Hz sweeps between 300 Hz and 700 Hz, i.e. between critical band rates of 3 Bark and 6.5 Bark, respectively. At 8000 Hz the modulation occurs between frequencies of 7800 Hz and 8200 Hz, corresponding to 21.1 Bark and 21.3 Bark. The critical band interval in the second case has decreased from 3.5 Bark to 0.2 Bark, i.e. by a factor of 17.5. As shown in Fig. 3, the same factor is found for the difference in fluctuation strength at 500 Hz and 8000 Hz. However, it should be realized that only the maximum and the minimum of the actual frequency of the FM-tone are taken into account in this example. A more realistic description has to be based on the correlated masking patterns and is given in another paper.

Fig. 3. Fluctuation strength of sinusoidally frequency modulated pure tones as a function of center frequency.

$f_{mod} = 4$ Hz, $\Delta f = 200$ Hz, $L = 70$ dB. Circles: standard with $f_m = 500$ Hz; squares: standard with $f_m = 8000$ Hz.
Fig. 4. Fluctuation strength of sinusoidally frequency modulated pure tones as a function of sound pressure level.

\[ f_\text{m} = 1500 \text{ Hz}, \ \Delta f = 700 \text{ Hz} \]

Circles: standard with \( L = 70 \text{ dB} \); squares: standard with \( L = 40 \text{ dB} \).

Fig. 4 shows the dependence of fluctuation strength of FM-tones on sound pressure level. For an increase in level of 40 dB, fluctuation strength increases on the average by a factor of about 1.7. The results clearly depend on the standard used. The large interquartile ranges for the 70 dB-standard (circles) are almost entirely due to large inter-individual differences. For amplitude-modulated tones and broadband noise, fluctuation strength increases by a factor of about 3 for an increase in level of 40 dB (Terhardt, 1968, Fastl, 1982b, 1983). This larger increase with level holds also for the roughness of FM-tones (Kemp, 1982).

**SUMMARY**

The fluctuation strength of sinusoidally frequency modulated pure tones, as a function of modulation frequency, shows a bandpass characteristic with a maximum at 4 Hz. For frequency deviations larger than about 20 Hz, fluctuation strength increases approximately linearly with the logarithm of frequency deviation. When the frequency deviation is held constant, the fluctuation strength of FM-tones is found to be independent of center frequency for low tones, and to decrease with increasing center frequency. The latter effect can be related to the number of critical bands encompassed by the FM-tones. For an increase in level of 40 dB, fluctuation strength of FM-tones increases by a factor of about 1.7.

**REFERENCES**


ACOUSTIC CUES CONTRIBUTING TO SPECTRAL FUSION

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Introduction.
Spectral fusion is a process by which elements of sound are grouped into a more or less unified auditory source "image". The grouping processes use acoustic cues such as the coherence of sub-audio frequency modulation (FM) among spectral components and the relative constancy of a spectral envelope. It also appears that there may be mechanisms which signal the presence of multiple sources that are separate from mechanisms involved in the actual formation of source images.

Experiment on Coherence of Frequency Modulation.
In most forced-vibration systems such as bowed strings, wind instruments and voice, random or periodic perturbations of the driving, or fundamental, frequency (F₀) are imparted proportionally to its higher partials. Thus, with both jitter (random FM) and vibrato (periodic FM), partials retain their frequency ratio relations. Partials that derive from separate sources and fall adjacent to one another would be expected to have random FM waveforms that were temporally independent. This would lead to an incoherence of FM across these components that may be used to signal the presence of multiple sources. This experiment investigates the effects of modulation incoherence in adjacent partials on the perception of source multiplicity.

Stimuli of 1.5 sec. duration were synthesized with 16 equal-amplitude partials. In coherent tones, all partials were modulated by a jitter waveform as in Eq. 1, maintaining the frequency ratios (even if these were inharmonic ratios):

\[ f(n) = R(n) \times F_0 \times (1 + A \times \text{Mod}(t)), \]  

(Eq. 1)

where \( f(n) \) is the instantaneous frequency of partial \( n \), \( R(n) \) is its frequency ratio relative to \( F_0 \), \( A \) is the rms deviation of the modulation, and \( \text{Mod}(t) \) is the modulating waveform. In "incoherent" tones, one of the odd-numbered partials was modulated by a separate jitter waveform of similar statistical characteristics, while the other 15 partials were coherently modulated by the original jitter at the same rms deviation. Five values of rms deviation were chosen for each of 8 stimuli with an incoherent partial. These tones were presented in one of 3 conditions: 1) partials were harmonics of 220 Hz presented at 75 dB(A) (H75), 2) same stimulus as (1) presented at 50 dB(A) (H50), and 3) partial frequencies were randomly
changed from their harmonic frequencies by between +/- 5 cents, which keeps roughly the same inter-partial distance but disturbs significantly the phase synchrony among partials (I75). These 3 conditions were presented in separate experimental blocks. Subjects were presented coherent and incoherent tones in randomly ordered pairs with the same rms deviation. They were to decide which tone had more sources. 30 repetitions of each stimulus were presented and the % of times the incoherent tone was chosen was calculated.

Results: When this % measure is plotted as a function of rms deviation for each partial number and condition, almost all curves are monotone ascending, i.e. an increase in rms deviation is accompanied by an increase in the proportion of times the incoherent tone was chosen as having more sources. These data points were fitted with a cubic spline and the 70.7% point is estimated. This value was chosen as a measure of the threshold of the effect of modulation incoherence on the perception of source multiplicity. This measure reflects the rms deviation necessary to just barely produce a multiple source image with the 2 independent jitters used.

Figure 1 shows a comparison between the thresholds for H75 and those for H50 and I75, respectively. Results are averaged over 3 subjects, who all performed very similarly in this task. For harmonic tones there is generally a decrease in the source multiplicity threshold (SMT) with increasing harmonic number. For the inharmonic tone, there is a decrease in SMT to partials 5 and 7 and an increase for higher partials.

Effect of change in intensity: a decrease in SPL from 75 to 50 dB is accompanied by higher SMTs, except for harmonics 7 and 9. The increase is substantial for harmonics 1, 3 and 5 (for H50, harmonics 1 and 3 never reached 70.7% with the rms deviations used) and is much smaller for harmonics 11, 13 and 15. The disappearance of the effect at harmonics 7 and 9 may be due to the slight increase in SMT for the H75 stimuli at these harmonics.

Effect of change in harmonicity (phase synchrony of adjacent partials): an incoherently modulated PV creates a multiple source percept at much smaller deviations for an inharmonic complex than for a harmonic one. For the rest of the partials, the SMT is greater for inharmonic tones by at least 1 cent. The difference decreases from partial 3 to partials 5 and 7 and then increases again for higher partials. For stimuli where the incoherent partial is less than a critical band away from an adjacent partial, incoherence becomes more noticeable with increasing partial number for harmonic stimuli, but becomes less noticeable with inharmonic stimuli.

Relation of data to modulation detection thresholds of coherent, harmo-
ic stimuli: SMTs are below modulation detection thresholds for harmonic numbers above \( f_1 \) for H75, and above \( f_3 \) (\( f_5 \) for S3) for H50. For incoherently modulated partials in a region where adjacent partials are within a critical band, the SMT is much less than modulation detection threshold. These data suggest the possibility that information concerning the presence of multiple sound sources in the environment may be already encoded in the auditory nerve. This may be encoded as a lack of periodicity or synchrony of firing patterns in local population of auditory nerve fibers being stimulated by more than one partial, i.e. whose excitation patterns overlap. Decreasing excitation pattern overlap in lower harmonics by decreasing intensity causes an increase in the rms deviation necessary to signal source multiplicity. Disturbing the phase synchrony of adjacent partials by making them inharmonic also requires much greater rms deviations to detect incoherence of modulation (as measured by the SMT). For harmonic tones, the slightest incoherence (\(< 1\) cent rms deviation) among partials within a critical band is accompanied by increased judgments of source multiplicity.

Experiment on Spectral Envelope Constancy and Global Fusion Mechanisms: Local mechanisms detecting incoherence of adjacent partials as a cue for the presence of multiple sources do not suffice to explain how the various spectral components dispersed and embedded within a complex spectrum are grouped into a unit which gives rise to the qualities associated with that source image (e.g. pitch and phonemic identity). The nature of formant peaks in vocal sounds are known to provide the cues for vowel identification. With frequency modulation of the source of excitation, the resonance structure induces a coupled modulation in amplitude of the frequency components. In effect, these amplitudes "trace out" the spectral envelope describing the resonance structure. This may give the auditory system information which reduces the ambiguity about the nature of the source's identity. This experiment investigates the role of the coupling between FM and AM in sung vowels for the formation and distinction of multiple source images.

Three vowels (male voice) were used: /a/ (father), /o/ (spoke), /i/ (spree). A pre-test determined that all subjects could identify each of these vowels in isolation at 3 pitches and either with or without a combined vibrato/jitter modulation. In the experiment, chords of the 3 vowels at 3 pitches (C2: 131 Hz; F2: 175 Hz; Bb2: 239 Hz) were constructed from the vowel sounds in the pre-test. All six configurations of 3 vowels at 3 pitches were used. In one set of conditions ("ground steady", labeled "S") one or none of the 3 vowels in each configuration was selected to have modulation (labeled N = none, A, 0, I). In a second set ("ground modulating", labeled "V"), the other vowels were modulated with a separate, independent modulating function. Subjects listened to each of these 48 stimuli repeating continually and were asked to judge the prominence or their certainty of presence of each vowel within the tone complex. The judgments were made on a continuous scale between 0 and 100, and one judgment was made for each vowel with each stimulus. The Ss were not informed that all 3 vowels were always present in some form. Each stimulus was presented 5 times and the judgments of prominence for each vowel were averaged over these 5 estimates.

Figure 2 shows results averaged over 10 Ss, 6 configurations and 5 repetitions. There were some important effects due to configuration that show that masking effects can affect the spectral/temporal resolution of multi-
ple sources, but space does not allow their description here. Bars representing judgments on the vowel that was selected as "figure" are hashed. One notices for the S condition that there is an increase in prominence judgments compared with the NS condition for the vowel that is "figure" (AS-NS = 28% for /a/; OS-NS = 24% for /o/; IS-NS = 10% for /i/). (The vertical lines represent 1 standard deviation of the mean across Ss and configurations). No such effect obtains in the V conditions. Judgments for any vowel in any V condition are the same as for that vowel being figure in the S conditions. Thus, the relative coherence of modulation seems to have no effect on perceived prominence in this situation. One notes that the 3 vowels differ in their relative prominence. The vowel /a/ tends to be the most prominent, perhaps due to the concentration of energy in lower formants which makes them more noticeable. The main effect to notice here is that when the frequency components are modulated and this modulation is coupled with an AM due to the vocal resonance, there is an increase in the ability to identify the vowel as present in a complex spectrum. This implies that some mechanism is capable of grouping the source's components across the complex spectrum by detecting their coherent modulation at which point the spectral envelope is available as a cue for identifying the resonance structure. In the NS condition this task was much more difficult.

Conclusions: The auditory system can use coherent FM across a set of spectral components to group them into a source image. It may also use the detection of local incoherence to signal the presence of multiple sources. If FM is coupled with an AM describing the resonance structure of a source this may reduce the ambiguity of the nature of the source's resonance structure and aid identification. There are implications here that the system first groups components (forms sources) and then derives the qualities of the source from the nature of the subset of acoustic components.
APPAREILS DE SIMULATION DE L'ECCOUTE D'ORDRE PROCHE ET DE L'ECCOUTE D'ORDRE LOINTAIN.

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- Les paramètres acoustiques perceptibles par le récepteur humain étant le niveau, la hauteur et la durée, auxquels correspondent respectivement les grandeurs physiques : intensité, fréquence et temps, un message sonore peut être représenté par projection sur l'un des trois plans : spectral, dynamique ou mélodique. La grande sensibilité du système auditif à la fréquence et au temps explique que la représentation la plus adéquate d'un message sonore destiné à un récepteur humain soit la projection de ce message sur le plan mélodique : ceci a été amplement démontré par E. LEIPP et M. CASTELLENGO.

- Tout message sonore transporte simultanément de l'information sémantique, évidemment prépondérante dans les messages phonétiques, et de l'information esthétique prédominante dans les messages musicaux par suite des degrés de liberté que permettent le déchiffrement et l'exécution instrumentale. Certains procédés de traitement des messages enregistrés, tels que la compression dynamique ou la lecture inversée, permettent de dire que l'information sémantique est très étroitement liée au découpage temporel du message et donc à la perception du temps, l'information esthétique résidant surtout dans la coloration liée à la perception fréquentielle.

- Une des caractéristiques fondamentales du récepteur humain est l'existence d'une limite maximale du débit d'informations perceptibles. Pour expliquer le comportement de ce récepteur, A. MOLES a proposé une synthèse entre la théorie de l'exploration suggérée par la psychophysiology expérimentale et la théorie de l'intégration dérivée de la notion de "forme" : selon la complexité du message qu'il reçoit, l'intérêt qu'il porte à ce message et sa propre capacité perceptive, le récepteur humain procède à une analyse plus ou moins fine en hauteur et en durée - ou bien il appréhende globalement, en fonction de son expérience - des paquets d'information en tant que formes. Ces deux attitudes possibles montrent l'importance de la perception du temps que l'on peut envisager à différents niveaux : le "délai de reconnaissance" indispensable à la perception du timbre, la "durée de présence" ou profondeur de la mémoire dite phosphorescente nécessaire à la perception des formes dans un message trop complexe pour être instantanément saisi, et enfin la mémoire proprement dite, où l'on distingue d'une part la mémoire des faits récents qui a une profondeur de quelques heures et suppose motivation et attention,
d'autre part la mémoire consolidée qui permet à l'individu de réagir en fonction de son passé.

- L'expérience montre que le système auditif humain peut pratiquer deux types d'écoute. L'écoute d'ordre proche consiste en une analyse fine du contenu de la mémoire phosphorescente : c'est l'écoute sélective pratiquée par les preneurs de son et simulée par le Sonagraph. L'écoute d'ordre lointain fait appel à la mémoire proprement dite pour formuler un jugement global sur la coloration d'une pièce musicale, le timbre d'une voix ou d'un instrument de musique, la sonorité d'une salle ... à l'aide d'un nombre limité de qualificatifs connus de tous. Ce type d'écoute, familier aux musiciens, relève manifestement d'un processus d'intégration pendant lequel le système auditif s'intéresse à ce qu'il perçoit à l'intérieur d'un certain nombre de bandes fréquentielles.

Ces bandes, dites "bandes sensibles" ont été définies par E. LEIPPP à la suite de tests portant sur un grand nombre de sujets entraînés à l'écoute. Bornées par les fréquences 50, 200, 400, 800, 1 200, 1 800, 3 000, 6 000 et 15 000 Hz, elles sont appelées : basse, grave, bas-medium, medium, haut-medium, aigüe, suraigüe, strident. Le nombre de huit bandes sensibles est à rapprocher des sept échelons d'intensité utilisés par les musiciens ou des six principales couleurs qu'un observateur normal perçoit dans le spectre visible ; il est en accord avec ce que disent les psychologues sur la manière dont nous traitons les données sensorielles par paquets d'informations en nombre toujours voisin de sept pour les données où n'interviennent qu'un paramètre.

- D'après les données actuelles de la neurophysiologie sensorielle, l'hémisphère cérébral gauche gouverne la pensée abstraite et les activités liées à la parole, l'information y étant analysée de façon continue et linéaire, alors que l'hémisphère droit contrôle la pensée concrète et les images mentales relatives aux activités artistiques, l'information y étant appréhendée de manière globale et simultanée. L'individu accorde un rôle dominant à l'un ou l'autre hémisphère selon la situation : l'information sémantique, la perception analytique et l'écoute d'ordre proche relèvent de l'hémisphère gauche, l'information esthétique, la perception globale et l'écoute d'ordre lointain étant du ressort de l'hémisphère droit.

- C'est pour simuler l'écoute d'ordre lointain que nous avons mis au point l'intégrateur de densité spectrale à bandes sensibles ou I.D.S. Cet appareil comporte huit voies de mesure dont chacune comprend un filtre passe-bande, un redresseur et un intégrateur. Les fréquences de coupure des filtres passe-bande du second ordre se recoupent aux frontières des huit bandes sensibles. Les redresseurs "double alternance" sont linéaires. Quant aux intégrateurs, de type "longue durée", leurs sorties V/ représentent l'intégrale du spectre d'amplitude de la pression acoustique en bandes sensibles. Pour obtenir des résultats indépendants du niveau de la prise de son et de la durée d'intégration, un calculateur fournit les huit nombres sans dimension V/ qui caractérisent la balance spectrale moyenne du phénomène sonore analysé.

L'I.D.S. a déjà été utilisé : en Acoustique musicale, pour comparer les sonorités d'instruments de même nature mais de factures différentes - en
Acoustique des salles, pour tester la sonorité d'un lieu d'écoute - enfin dans le domaine encore peu exploré des Paysages sonores, pour caractériser la coloration d'une ambiance sonore en milieu rural ou urbain.

- En conservant le découpage du domaine fréquentiel en bandes sensibles, nous avons réalisé un sonoscope en temps réel (S.T.R.). Un premier mode de présentation de l'analyse du message, dit "en amplitude", consiste à visualiser sur écran cathodique à mémoire l'ensemble des sorties redressées des huit filtres en fonction du temps (moitié supérieure de la figure 1). L'image ainsi constituée de huit courbes amplitude-temps occupe la moitié supérieure de l'écran : on peut ainsi comparer deux événements sonores, par exemple deux éditions du même message émis ou capté dans des conditions différentes.

Un autre mode de présentation est la visualisation "en tout-ou-rien" des sorties redressées des huit filtres selon que celles-ci sont supérieures ou inférieures à un certain seuil dont la valeur ajustable est asservie au niveau moyen du signal d'entrée. L'image, très largement indépendante du niveau d'entrée, est alors formée de huit bandes horizontales, dont les portions lumineuses matérialisent les intervalles de temps pendant lesquels la sortie correspondante a été supérieure au seuil de visualisation choisi ; la présentation par bandes adjacentes en contact fournit des formes simples plus faciles à apprécier que si les bandes sont séparées (moitié inférieure de la figure 1). Comme dans le mode précédent, deux analyses successives occupent une moitié, puis l'autre de l'écran.

Le mode de présentation en tout-ou-rien au-dessus d'un certain seuil respecte le découpage temporel du message analysé, équivaut à une compression de la dynamique ainsi qu'à une condensation de l'information fréquentielle : on peut donc s'attendre à ce que le S.T.R. permette l'extraction de l'esquelette sémantique d'un message phonétique (figures 2 et 3) et trouve ses premières applications en Reconnaissance de la parole d'une part, et en Orthophonie d'autre part, comme aide à l'éducation des jeunes sourds par contrôle vidéo-phonatoire.

**Figure 1**

Les deux modes de visualisation de l'analyse du message.

En haut : "en amplitude".
En bas : "en tout-ou-rien".
La même phrase :
"Vois-tu le baobab ?"
prononcée par 4 locuteurs
différents (3 hommes et 1
femme), le seuil de visuali-
sation étant réglé au mieux
pour faire émerger le sque-
lette sémantique de la phrase.

Figure 3
La même phrase prononcée par
le même locuteur en voix
normale et en voix criée.
ON THE PERCEPTION OF SPACIOUSNESS WITH ARTIFICIAL HEAD

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Introduction

"Spaciousness" (in German: Raumlichkeit) is an important subjective attribute of the complex sound field in a concert hall. Inspecting the literature (e.g. Kuhl, 1978, for a review) one encounters similar notions, like "spatial impression" and "spatial responsiveness" which to a large extent all seem to express the same subjective experience. This seems best described by the following two aspects: (a) the sensation of broadening of the sound source, and (b) the sensation of being enveloped (involved) in the sound.

From the physical point of view, evidence has been provided (e.g. Siebrasse, 1973; Cottlob, 1973) that spaciousness is inversely related to interaural signal correlation, and secondly, that spaciousness is dependent on sound pressure level (Marshall, 1968).

From the roomacoustical point of view, it has been pointed out (e.g. Marshall, 1967; Barron, 1971) that spaciousness is positively related to the occurrence of early lateral reflections in a concert hall.

In a former paper (Bilsen, 1980), we dealt with the notion spaciousness from a pure psychophysical point of view by investigating "the subjective extensiveness" (= perceived width) of white noise dichotically presented by headphones to a listener. The left- and right-ear signal were derived from two independent white (or pink) noise sources by appropriate (partially) mixing of the two noises; thus, the interaural degree of correlation could be varied in small steps from 0 to 1. It turned out that (a) subjective extensiveness increases linearly with decreasing interaural correlation (being defined as the value of the normalized cross-correlation function for \( \tau = 0 \)), (b) a linear relationship on a log-log scale exists between subjective extensiveness and sound pressure, and (c) low frequencies in the stimulus are more effective than higher ones with respect to subjective extensiveness. An optimum is found for the frequency band around about 500 Hz, in agreement with other binaural phenomena. A shift towards lower frequencies was found when pink noise was used instead of white noise.

In the present paper, stimulus presentation again is by means of headphones. Contrary to the former experiments, the left- and right-ear signal are obtained from an artificial head placed in the middle of a hall simulator producing a sound field similar to that of a real concert hall. It was the purpose of this study to investigate whether such a stimulus configura-
Bilsen - Spaciousness with artificial head

tion would be able to evoke a sensation of spaciousness comparable to the more artificial one of the former study. The basic signals were pink noise or "dry" orchestral music. In particular, we were interested in finding the threshold of just noticeable increment in spaciousness for the lateral reflections (LR-component) and for the reverberation (RV-component) in the sound field. Results from psychophysical tests will be compared with physical quantities like the interaural correlation coefficient.

Signal description and experimental procedure

The original signal - pink noise or a repetitive one-second fragment of dry orchestral music (Mozart's 41st Symphony, BBC-tape) - is fed to a hall simulator (Han, 1977) which essentially consists of a tapped delay line with 12 independent outputs, each of which can be adjusted with respect to delay time and amplitude. The output signals are fed into a mixing console, which provides the signals for 9 loudspeakers properly positioned in an anechoic chamber. Reverberation is produced by feeding the original signal (delayed) into a 200 m$^3$-reverberation room with a reduced reverberation time of about 2 sec and recording the sound field at two different (incoherent) places.

Data on loudspeaker position, signal mixing, signal delay and amplitude are given in Fig.1 and Table 1. The values were chosen such that a realistic simulation of a concert hall was obtained.

![Diagram of loudspeaker configuration](image)

**Table 1. Reflection pattern.**
Delay (ms), attenuation (dB) and grouping across loudspeakers (A......J);
notation: e.g. J: (ms,dB).

<table>
<thead>
<tr>
<th>PRIMARY SIGNAL (PS)</th>
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<tbody>
<tr>
<td>F: (0,0),(33,10)</td>
<td></td>
</tr>
<tr>
<td>C: (53,10),(89,13)</td>
<td></td>
</tr>
<tr>
<td>I: (61,16),(89,18)</td>
<td></td>
</tr>
<tr>
<td>G: (65,16),(71,18)</td>
<td></td>
</tr>
<tr>
<td>D: (65,13)</td>
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<table>
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<tr>
<th>LATERAL REFLECTIONS (LR)</th>
</tr>
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<tbody>
<tr>
<td>E: (32,0),(33,3)</td>
</tr>
<tr>
<td>J: (30,0),(34,3)</td>
</tr>
<tr>
<td>B: (61,1),(71,3)</td>
</tr>
<tr>
<td>A: (53,1),(89,3)</td>
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<table>
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<tr>
<th>REVERBERATION (RV)</th>
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<tr>
<td>delay=100 ms; $T_{60}=2$ sec</td>
</tr>
<tr>
<td>&quot;left&quot; &quot;right&quot; &quot;left+right&quot;</td>
</tr>
<tr>
<td>B</td>
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<td>E</td>
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Using an artificial head (Boone et al., 1977) the following recordings were made on an 8-track Ampex tape recorder: Track 1: the original signal; Track 2: synchronization signals; Track 3 and 4: The "primary signal (PS)". the left- and right-ear signal from the artificial head, using only the corresponding field components (c.q. loudspeakers) as given in Table 1; Track 5 and 6: the "lateral reflections (LR)"; Track 7 and 8: the "reverberation (RV)".

During the listening tests the stimuli were composed of the three (dichotic) signal components from the Ampex recorder, PS, LR, and RV, as described in the foregoing paragraph. Only the three component levels were varied.

Subjects were presented with pairs of stimuli in random order. One stimulus, the "reference stimulus", consisted of PS at 0 dB attenuation plus the "fixed parameter" being either LR or RV at either -10 dB or -100 dB. The other stimulus, the "test stimulus", consisted of PS at 0 dB plus the fixed parameter at either -10 dB or -100 dB, plus the "variable parameter" being either RV or LR at the test level X dB.

The test level X was controlled by a computer in steps of 1 dB such that 8 presentations at 10 different X-values (thus 80 presentations in total) were given in one session, in random order. The computer also controlled the a-priori chance of 30% of the reference stimulus (thus also the test stimulus) being the first in a pair.

The subjects, seated in a sound-proof booth and listening by headphones (Telephonics TDH 39), were instructed to indicate by pressing one of two knobs which stimulus had larger spaciousness (subjective extensiveness). By definition a correct response corresponds to the decision "test stimulus more spacious than reference stimulus". From the psychometric functions thus determined, the 75%-correct response level was calculated. This level may be called a "threshold of just-noticeable increment in spaciousness"; it is indicated by the symbol L.

Results

Using the three signal components as defined in the foregoing section and the two original signals (music or pink noise), 8 different signal configurations have been investigated by four subjects. The results of the listening tests are presented in Fig. 2. Here, the threshold level L of the variable parameter (RV or LR) is given for each subject separately. The vertical bars indicate the standard deviation. The horizontal dashed lines indicate the average level for the four subjects.

Fig. 2. Results of listening tests by four subjects. For explanation see text.
for each signal configuration. For example, the signal configuration "music, LR -100 dB" means that music is the original signal and that LR is the fixed parameter at -100 dB (i.e., below the noise level); RV is the variable parameter and its level.

It can be noticed that both LR and RV have a positive influence on spaciousness. Further, it is remarkable, though not unexpected, that the subjects are less in accordance with each other for music than for noise.

Discussion and conclusions

The artificial head appears to be able to communicate the sensation of spaciousness (subjective extensiveness). For the noise signal, in particular, the experienced subjects described the percept to be similar to the percept of two independent and partially mixed noise signals presented by headphones (Bilsen, 1980).

The perceived spaciousness can be increased by adding either lateral reflections (LR) or reverberation (RV). This is understandable from the interaural correlation coefficients as measured for the signal components at the left and right ear of the artificial head, viz for PS 0.73 and 0.72, for LR 0.0 and 0.18, and for RV 0.17 and 0.08 for music and pink noise resp.

The influence of the variable parameter (LR or RV) appears to be greater, thus its threshold level L lower by about 5 dB on the average, as the fixed parameter (RV or LR) has a lower level (-100 dB instead of -10 dB). This is not unexpected because the fixed parameter has a very low interaural correlation coefficient also.

The finding that adding reverberation (RV) also increases spaciousness, for music even more than adding lateral reflections (LR), seems to contradict earlier conclusions in the literature. One has to realize, however, that the RV-signal in the present experiments has a very low interaural correlation. This is due, of course, to the incoherence of the two signals from the reverberation chamber and to the angle of incidence (for a great deal from the side loudspeakers).

Further details and other results will be discussed during the presentation of the paper.

Acknowledgement. We thank our colleagues F. v.d. Berg, L. Han, J. Raatgever, and D. de Vries for experimental assistance and stimulating discussions.

References

LOCALIZATION OF SOUND IN ROOMS-THE EFFECT OF VISUAL FIXATION

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Introduction

It is well established that visual cues can significantly bias the localization of auditory stimuli. This is especially true if the observer expects that the sources of visual and auditory sensations are one and the same, in which case the bias is termed a ventriloquism effect [1].

We have recently performed a study of human localization of sound sources in a concert hall situation [2]. Noting that previous studies of visual bias on auditory localization have been confined to small test beds or anechoic rooms we thought it worthwhile to search for visual bias in the larger scale geometry.

Experiment

Subjects: The subjects for these experiments, lettered A-J, were 10 subjects from the experiments of ref. 2, numbers 2, 3, 4, 5, 6, 7, 8, 9, 10 and 13 respectively. They ranged in age from 8 to 42. Subjects E and F were female. Subject D was left handed; subjects B and F had been switched from left handed to right handed in childhood. All subjects reported normal hearing and normal or completely corrected vision. Before participating in the experiments described here subjects had performed at least five runs of the auditory localization task.

Method: The auditory stimulus and experimental geometry were identical to those of ref. 2. The signal was an 80 dB 500 Hz sine tone pulse, gated on and off with a hard edge. The pulse had a duration of 50 ms. The sine pulse was sent to one of 8 small loudspeakers located on an arc 12 meters from the subject and spaced 4 degrees apart. The speakers were 1 meter from the floor. The experiments were performed in the Espace de Projection at IRCAM, a variable acoustics concert hall with dimensions 24x15.5 m by 11.5 m high. The wall and ceiling surfaces of the room were chosen to be highly absorbing, providing a reverberation time of 1 s at 500 Hz. The subject's task was to decide which loudspeaker had sounded. An experimental run consisted of 10 pulses from each of the speakers, presented in random order, requiring a total of 80 judgments.
The new feature of the experiments described here is that the subject's gaze was directed. This was done by placing two small pilot lamps at the center of loudspeaker number 1, the leftmost, or at the center of loudspeaker number 8, the rightmost. Prior to the acoustic pulse either one or both of the pilot lamps was illuminated for about ½ second. The subject was required to declare first how many lamps had been lit and then which speaker had sounded the pulse.

The two lamps were separated by 0.4 degrees. When the lamps were on speaker number 1, for example, it was easy for the subject to count the illuminated lamps if he was looking at speaker 1, but impossible to count them if he was looking to the right of speaker 4. If on a given trial a subject incorrectly reported the number of lamps his localization judgement for that trial was not accepted as data. No record was kept of the incorrect responses to the lamps; we noted only that the number was almost nil. The purpose of the visual task was merely to fix the subject's gaze near speaker 1 or near speaker 8.

Although the visual task directed the subject's gaze, we did not want the subject's head to be oriented towards the extreme left or right. The subject was required to maintain his head in a forward orientation, along the line which passed between speakers 4 and 5, and to move only his eyes to left or right to observe the lamps on speakers 1 or 8. In order to monitor the head position we placed a television camera directly in front of the subject, well below his line of sight to the speakers. If a subject's head was not oriented properly the experimenter halted the experiment briefly to remind the subject to correct his head position.

Each subject completed three runs, one with his gaze directed to the left, one with his gaze directed to the right, and a control run with no visual requirement.

Results

The most important statistical measure of performance is \( E(k) \), the average localization error in degrees given that the sound source was speaker \( k \) \((1 \leq k \leq 8)\). It is defined as \( 4[r(k)-k] \), where \( r(k) \) is the subject's numerical response given source number \( k \), averaged over the 10 trials of a run. Figure 1 shows \( E(k) \), averaged over the 10 subjects for the three conditions: leftward gaze (1), rightward gaze (8) and undirected gaze, the control (c). The figure shows that there is a significant bias of auditory localization towards the direction of the subject's gaze. The amount of the bias can be determined by subtracting \( E(k) \) in the control condition from \( E(k) \) in the directed gaze conditions 1 and 8. It shows no particular dependence upon the angular discrepancy between the visual target and the sound source, which ranged from -28 to +28 degrees. The errors, averaged over subjects, trials and source location are \( E_c(k) = -0.6(\pm1.3) \) degrees, \( E_l(k) = -2.3(\pm1.3) \) degrees, \( E_r(k) = 1.5(\pm1.6) \) degrees. The average bias then is 1.9 degrees towards the gaze.

Figure 2 shows the standard deviation of subject judgements. There is a small effect supporting the conclusions of Platt and Warren [3] and Berman and Welch [4] that the variability in auditory localization is reduced when the eyes are directed towards the target.
Fig. 1 Localization error, averaged over subjects, vs correct source number, k.

Fig. 2 Standard deviation of the localization judgements, averaged over subjects, vs correct source number, k. Symbols are the same as in Fig. 1.

Discussion

Besides the large-scale geometry, our experiment differed in several other ways from other work on visual bias. Unlike most previous studies our visual cue was not synchronous with the auditory stimulus. There was a gap of approximately 250 ms between the offset of the lamp(s) and the onset of the acoustic signal. Although previous workers have tried to separate ventriloquism from bias by using lamps (not normally associated with auditory cues) it appears that mere synchrony may lead to the perception of a unified source [5]. In this respect our experiment was similar to one by Weerts and Thurlow [6] who also obtained a visual bias of 2 degrees. We note also that when Bertelson and Radeau [7] counted only trials in which subjects reported no source unity the bias effect was independent of the visual-auditory angular discrepancy, a result similar to ours. By contrast their bias increased with increasing discrepancy for those trials in which subjects reported source unity.
Further, most previous work has employed pointing responses. Given the effects of visual bias on proprioception [8] one might suspect pointing responses to include an unwanted bias. Such an effect was avoided by our source identification procedure.

Conclusion

Visual bias effects on auditory localization, previously found in the laboratory, occur as well in concert hall environments. Bias effects of directed gaze agree in sign, magnitude and in their independence of visual-acoustical discrepancy angle with those of previous experiments which have excluded ventriloquism effects.

References


SOUND LOCALIZATION ON THE UPPER HEMISPHERE

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1. INTRODUCTION

All past investigations on sound localization have been confirmed to horizontal-plane and median-plane localizations. Few researcher has dealt with localization of a sound at any point on the upper hemisphere, except Wallach's report\(^1\) on the role of head movements in sound localization.

We hypothesize that if a direction of a sound on the upper hemisphere is expressed by two angles \(\alpha\) and \(\beta\), instead of the azimuth angle \(\psi\) and the elevation angle \(\theta\) as shown in Fig.1, the direction is perceived by two mutual independent cues; one is binaural disparity cues which determines the angle \(\alpha\) and another is spectral cues which determines the angle \(\beta\). The angle \(\alpha\) is the angle between the aural axis and a straight line connecting a sound with the center of a subject's head. The angle \(\beta\) is the angle between the horizontal plane and the perpendicular from a sound on the aural axis. In this paper, two localization tests are conducted to verify our hypothesis. Subjects are four males.

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2. LOCALIZATION TEST I

Seven loudspeakers were located at 15 degrees in the right half quadrant of the traverse plane on the upper hemisphere with a radius of 1.5m.

\[\text{Fig. 1. Definition of head-related coordinate system.}\]
Stimuli were three kinds of band limited noise; (1) the wide-band (300-13600 Hz), (2) the high-pass (4800-13600 Hz) and (3) the low-pass (300-4800 Hz).

Table 1 shows the localization errors $E^2$ in regard to the angle $\alpha$ and the angle $\beta$, separately. The errors of the angle $\alpha$ for any stimulus are small. On the other hand, the errors of the angle $\beta$ for the wide-band and the high-pass noises are also small, but the error for the low-pass noise is much larger than the other stimuli.

Fig. 2 shows the detailed behavior of responses to a sound at $\alpha=45^\circ$, $\beta=90^\circ$ on the $\alpha$-$\beta$ plane, as an example. For the wide-band and the high-pass noises, the perceived directions agree with the source direction. For the low-pass noise, however, though the angle $\alpha$ is perceived correctly, the angle $\beta$ is perceived far from the traverse plane ($\beta=90^\circ$) and most sound images shift to the horizontal plane ($\beta=0^\circ$ or $180^\circ$). This tendency is evident for all source directions.

Fig. 2. Responses to (a) wide-band, (b) high-pass and (c) low-pass noises in the traverse plane. Source direction, $\alpha=45^\circ$, $\beta=90^\circ$. 
Table 1. Average error $E$ (in degree) of wide-band, high-pass and low-pass noises in traverse plane localisation.

<table>
<thead>
<tr>
<th>Angle</th>
<th>Wide-band</th>
<th>High-pass</th>
<th>Low-pass</th>
</tr>
</thead>
<tbody>
<tr>
<td>$\alpha$</td>
<td>5</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>$\beta$</td>
<td>3</td>
<td>3</td>
<td>66</td>
</tr>
</tbody>
</table>

3. LOCALIZATION TEST II

Nine loudspeakers were located at the positions defined by combining one of three angles $\alpha=30^\circ$, $60^\circ$ and $90^\circ$ with one of three angles $\beta=0^\circ$, $90^\circ$ and $180^\circ$ on the right quadrant of the sphere. Stimuli were six 1/3 oct. band noises with the center frequencies of 1.0, 3.15, 5.0, 6.3, 8.0 and 10.0kHz.

Table 2 shows average values and standard deviations of perceived angle $\alpha$, independent of source angle $\beta$. The perceived angle $\alpha$ for any stimulus agrees well with the source angle $\alpha$. The perceived angle $\beta$ were compared with Blauert's "directional band", since they have a tendency to appear at a certain biased angle which corresponds to the center frequency.

Fig.3 shows the relative frequencies of "front", "above" and "rear" judgements. The directional band of each judgement occurs at the same frequencies as Blauert. "All over patterns" of the relative frequency of each judgement as a function of the center frequency are very similar to each other in the three planes defined by the angle $\alpha=90^\circ$, $60^\circ$ and $30^\circ$, and also to Blauert.

4. DISCUSSION AND CONCLUSION

It is wellknown that binaural disparity cues are not influenced by the frequency range of the stimulus, and that spectral cues are considerably influenced by it.\(^3,^4,^5\)

Table 2. Average value and standard deviation (in degree) of perceived angle $\alpha$ of 1/3 oct. band noise.

<table>
<thead>
<tr>
<th>Source angle $\alpha$</th>
<th>Center frequency of 1/3 oct. band noise (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>1.0</td>
</tr>
<tr>
<td>30°</td>
<td></td>
</tr>
<tr>
<td>Av.</td>
<td>28</td>
</tr>
<tr>
<td>SD</td>
<td>11</td>
</tr>
<tr>
<td>60°</td>
<td></td>
</tr>
<tr>
<td>Av.</td>
<td>52</td>
</tr>
<tr>
<td>SD</td>
<td>9</td>
</tr>
<tr>
<td>90°</td>
<td></td>
</tr>
<tr>
<td>Av.</td>
<td>89</td>
</tr>
<tr>
<td>SD</td>
<td>4</td>
</tr>
</tbody>
</table>
The results that the perceived angles $\alpha$ for any stimulus are correct mean that the angle $\alpha$ is determined by binaural disparity cues. On the other hand, the results that the perceived angles $\beta$ for the wide-band and the high-pass noises are correct, but those for the low-pass and 1/3 oct. band noises are not, mean that the angle $\beta$ is determined by spectral cues. And also, the results that the perceived angles $\alpha$ for the low-pass and 1/3 oct. band noises are correct, in spite that their angles $\beta$ are not at all, mean that the angle $\alpha$ and the angle $\beta$ are determined independently each other. Consequently, those results verify our hypothesis. Furthermore, it is made clear that the same "directional band" as Blauert occurs in any plane parallel to the median plane.

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Fig. 3. Relative frequency of each judgement. Source angle $\alpha$: 30°(open circle), 60°(triangle), 90°(closed circle). Dotted line is Blauert's data.
VERBESSERTE WIEDERGABE VON PHANTOMQUELLEN

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1. Phantom- und Realschallquellen

1.1 Unterschiedliche Höreindrücke
Ein Zuhörer, der sich in der üblichen Stereoabhörposition gleichweit von beiden Lautsprechern entfernt befindet (Fig. 1a), hört dann eine Schallquelle in der Mitte vor sich, wenn beide mit dem gleichen Signal gespeist werden. Diese Tatsache, daß zwischen zwei mehr oder weniger kohärent betriebenen Lautsprechern eine einheitliche 'Phantomquelle' geortet wird, ist Grundvoraussetzung für die stereophonische Wiedergabe. Der Höreindruck der Phantomschallquelle ist jedoch alles andere als natürlich. Von einer realen Schallquelle unterscheidet sich die Phantomquelle in bemerkenswerter Weise:

a) Eine Phantomquelle wird - geometrieabhängig - nahe gehört.

b) Eine Phantomquelle erscheint ausgedehnt, diffus und im Verhältnis zu ihrer Ausdehnung flach und meist deutlich über der Ebene eleviert, die von den Lautsprechern und dem Kopf des Zuhörers (VP) aufgespannt wird /1/.

c) Die Lokalisation der Phantomquelle hängt besonders im Hinblick auf die Elevation stark vom Kopf und seiner Haltung ab. Aus dem Zusammenwirken der beiden letzten Punkte b) und c) ergibt sich eine geometrische Undeutlichkeit, die die Ortbarkeit erschwert und bei Stereophonie zum 'Loch in der Mitte' führt und überdies auch einen Mangel an zeitlicher Auflösung zu verursachen scheint. Die große Nähe der Lokalisation (a) bewirkt zusammen mit den Einflüssen (b, c), daß die Quelle als nicht vom Kopf separiert empfunden wird, daß der Kopf nicht 'frei ist'.

1.2 Physikalische Unterschiede
Physikalisch liegt der einzige Unterschied der Phantombeschallung gegenüber der Beschallung durch eine nahe Einzelquelle darin, daß die Wellenfronten den Kopf des Zuhörers einhüllen, Fig.1a. Um die 'Form' der Wellenfront zu erfassen, muß zusätzlich eine Auswertung über einen dritten Weg angenommen werden. Es ist naheliegend, die Ursache für die hörbare Andersartigkeit der Phantombeschallung in der unterschiedlichen Körperschallanregung des Schädels zu suchen. Körperschalleinflüsse auf die Lokalisation stellten bereits die Verfasser von /2/ fest.
In Experimenten zur Vorne-Hinten-Ortung konnte der Verfasser /3/ nachweisen, daß die Polarität der Schädelshwingung im Schallfeld gegenüber der Anregung der Trommelfelle darüber entscheidet, ob man eine Schallquelle vorne oder hinten ortet. Durch Abnahme der relativen Körperschallanregung gegenüber der Anregung der Trommelfelle rückt eine Schallquelle näher. Dies erklärt, warum die Phantomschallquelle - abhängig vom Einstrahlwinkel α - näher gehört wird, warum man Kopfhörersignale normalerweise im Kopf wahrmimmt und wieso die Zweiebenennoidphonie in der Lage ist, die Entfernung einer Schallquelle zu übertragen /4/. Fig.1a läßt vermuten, daß auch die Einleitung vom Körperschall in den Schädel bei Phantombeschallung geringer ist. Die Arbeitshypothese lautet demnach, daß ausreichende Körperschalleinleitung in den Schädel die Punktförmigkeit der Schallquellenwahrnehmung befördert.


2. Subjektive Verbesserung der Phantomschallwiedergabe

In /3/ wurde vermutet, daß zur V-H-Ortung die normale Luftschall- und die Körperschallanregung über das Innenohr multiplikativ
verknüpft sind. Ein Teil der Körperschallanregung erfolgt direkt über das Kopfinnere zum Innenohr, vgl. /6/. Andererseits gelangt auch ein Teil des Körperschalls vom Mittelohr aus zum Innenohr. In Fig.3 sind diese Wege skizziert. Das Übertragungsmaß (Dämpfung, Phasengang, Laufzeiten) ist dabei sicherlich unterschiedlich. Auf Weg I gelangt sozusagen die Körperschallanregung des Schädel's zum Innenohr, die für die Punktualität der nahen Quelle 'zuständig' ist. Die Frage ist, ob bei mangelnder Körperschallversorgung des Innenohrs über Weg I (z.B. bei Phantomquellenanregung) es möglich ist, dem Innenohr über Weg II das 'Fehlende nachzuliefern'. Es war demnach zu prüfen, ob ein Vor- oder Nachecho geeigneter Stärke, Phasenlage und zeitlicher Verschiebung in der Lage ist, die Phantomquellenwiedergabe zu verbessern, ähnlich, wie es durch Körperschallzusatz möglich ist. Tatsächlich war die Suche erfolgreich. Ein zeitlich um 0,5 bis max. 2 ms verzögertes Kurzzeitecho, dessen Amplitude 1/4 bis 1/5 des Hauptsignals ausmacht, erfüllt den Zweck. Im Gegen- satz zur Wahl der Zeit ist die Wahl der Amplitude (ca. -12 dB) kritisch. Fig.4 zeigt den zur Verbesserung der Phantomquellen- wiedergabe notwendigen Aufbau. Um den Hörindruck besser mit denjenigen nach Aufbau Fig.2a vergleichen zu können, wurden diese Hörversuche ebenfalls tiefpaß-gefiltert abgehort. In Ab- hängigkeit von der Art der Musik ist die Punktualisierung bzw. das Ausbleiben der Elevation zu hören, was im übrigen durch negativen Körperschallzusatz nach Fig.2a rückgängig gemacht werden kann. Die verbesserte Phantomquellenwiedergabe bleibt auch bestehen, wenn der durch die Reflexion bedingte Kammefiltereinfuß entzerrt wird. Bei Stereoaufnahmen, die mit eng benachbarten Stereomikrophonen aufgenommen wurden, ist sowohl die geometrische als auch zeitliche Auflösung für insbesonders leise Instrumente in der Mitte verbessert, besonders dann, wenn jeweils eine weitere Reflexion contralateral zugesetzt wird (-12 dB, 2 x 0,7 ms). Eidophonische Zweilebenenwiedergabe profitiert in noch stärkerem Maß von zusätzlichen Kurzzeitreflexionen, weil diese Technik in noch größerm Maß von der Phantomquelle Gebrauch macht. Hier fällt besonders die von der Zuhörerposition unabhängige Lage naher Quellen auf.

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1a Phantombeschallung
2a Phantombeschallung mit Körperschallanregung

1b Reale Schallquelle
2b Reale Schallquelle mit kompensierter Körperschallanregung

Fig. 3 Körperschallanregung des Innenohrs
Fig. 4 Einrichtung zur Erzeugung eines Signals mit Kurzzeitecho
LATERALIZATION OF SPECTRALY SIMPLE AM SIGNALS

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Introduction

Lateralization is one method for studying directional hearing. Auditory signals are presented binaurally over headphones and the input parameters may be independently controlled. If the signals to the two ears are identical, a single sound will be perceived in the midline within the head. An interaural onset delay of as little as 15 µs or an intensity difference in the order of 1 dB will result in a shift in the laterality of the image toward the ear leading in time or receiving the more intense input.

Our particular interest has been the clarification of the roles of envelope and phase cues for lateralization. In the first experiment of a series (Kunov and Abel, 1981), we noted that the region of interaural onset delays, corresponding to phase shifts of 180° to 360°, was particularly sensitive to conflicts in the use of envelope (i.e. onset) and phase (i.e. ongoing) cues for lateralization of a 1000 Hz pure tone. An unexpected finding was the relatively long rise/decay time (R/D) in the order of 200 to 500 ms needed to secure psychophysically steady-state signals, i.e. pure tones for which the onset has no perceptual effect.

A second study (Abel and Kunov, 1983) extended the work by investigating the effects of variation in frequency, intensity, shape of R/D and duration of peak amplitude of the signal, focusing in particular on signals interaurally delayed by 180°. The results indicated that the value of R/D needed to preclude envelope cues depends on the values of the other parameters. Within limits, decreasing intensity could be compensated for by decreasing R/D, suggesting the psychophysical importance of the initial segment of the signal as opposed to the precise time of onset (precedence effect). This observation was further corroborated by the finding that lateralization remained unchanged with variation in peak duration from 25 to 200 ms. For low-frequency pure tones of 650, 800, 1000 and 1250 Hz performance was sensitive to interaural phase shift and R/D, but was
largely independent of frequency. For higher frequencies of 1250 and 1500 Hz lateralization did not change appreciably with variation in either interaural phase or R/D between 50 and 200 ms.

The transition between the two trends observed for 1000 and 1250 Hz in the second study appeared to be abrupt. The third experiment in the series (Kunov and Abel, 1983) aimed specifically at studying this transition in detail in the range of 1000 to 2000 Hz. Discrimination of interaural phase was assessed for psychophysically steady-state pure tones (R/D=700 ms) nearly in-phase (θ=0°) and nearly out-of-phase (θ=180°). Plotting the slopes of the obtained psychometric functions against frequency, it was observed that the phase cue decreased linearly between 500 and 1500 Hz. There was some evidence that discrimination was better for binaural out-of-phase signals.

Rationale

All our previous studies described above related in one way or another to the effect of rise/decay time. On the basis of the results, we are confident that we can create psychophysically steady-state signals in which the stimulus envelope has no effect for lateralization, and are therefore in a position to study explicitly the effect of slowly varying envelope un-cumbered by rise and decay of the stimulus. In other words, our signals can be designed to be virtually spectrally constant throughout the presentation, and thus avoid the problems of the varying spectrum during the rising and falling parts of the function. Precedence effects will not be related to the onset of the signal.

The following experiments were designed to investigate signals with relatively simple spectra so that hypotheses relating to lateralization cues (e.g. envelope cue versus spectral cues) could be tested.

Method

Our method has been to delay the entire signal (envelope as well as phase) so that the only difference between the signals to the two ears is the time of arrival. For a given delay the leading signal is presented randomly to the right and left ears over a block of 100 trials and on each trial the laterality of the image is judged. Other stimulus parameters are constant within the block. Binaural signals are presented at the rate of one every 5 s. A detailed description of the apparatus appears in Kunov and Abel (1981).

Three experiments were performed with one highly practiced observer. In the first experiment eleven pure-tone frequencies ranging from 1000 to 2000 Hz, in steps of 100 Hz, were presented. Signals had R/D of 700 ms and duration of peak amplitude of 50 ms. For each frequency two values of delay, corresponding to interaural phase shifts of 180° and 360° were
investigated. For each combination of frequency and delay the modulation frequency \( f_{mod} \) was varied across blocks from 1 to 50 Hz. The second experiment was designed to evaluate the effect of locking the phase (\( \phi \)) of the modulating signal to the onset of the rise of the stimulus. Values of \( \phi \) investigated were 0°, 45°, 90°, and 135°. Two carrier frequencies, 1000 and 1600 Hz and two values of 9, 160° and 360° were used. \( f_{mod} \) ranged from 1 to 50 Hz. In the third experiment the pure-tone modulation signal was replaced by noise which had been passed through three cascaded low-pass filters with identical cutoff frequencies \( f_0 \). It can be shown that the modulated signal will have linearly weighted sidebands equal to pure tones with frequencies \( f_0 \pm f_{mod} \). Values of \( f_0 \) were chosen to allow for comparison with variation in \( f_{mod} \) for Experiment 1.

Results and Discussion

The results of Experiment 1 are shown in Fig.1 for binaural signals interaurally delayed by 180°. The envelope information expressed as the modulation frequency required for \( P(C)=0.80 \) is plotted against the carrier frequency. Shown for comparison are the data of a previous study (Kunov and Abel, 1983) on the effect of R/D. In order to use the same ordinate, R/D was converted to \( f_{mod} \) using the formula: \( f_{mod} = 1/2\sqrt{2\pi}(R/D) \). This sinusoidally modulated AM signal has an envelope with slope at zero crossing equal to the slope of the corresponding R/D. It appears from Fig.1 that pure tone AM modulation and R/D give comparable results between 1000 and 1800 Hz. In this region, then, the envelope cue at the initial segment of the rise is slightly better but comparable to an equivalent AM signal. Beyond 1800 Hz the ability to use envelope in the AM signal deteriorates drastically, as compared with R/D. The upper range corresponds to the range of frequencies where the subject cannot detect a pure interaural phase difference.

The results of AM noise modulation from Experiment 3 are also shown in Fig.1. \( f_{mod} \) is taken as the cutoff frequency as specified in Methods above. The values of \( f_{mod} \) for \( P(C)=0.80 \) are approximately a factor of five greater than the values obtained for pure tone AM for carrier frequencies below 1800 Hz, but tend to be the same above 1800Hz. This discrepancy suggests that the weighting function employed for the noise spectra to calculate the equivalent pure tone AM signal is over-emphasizing the higher sideband frequencies, at least below 1800 Hz.

In Fig.2 we show the effect of locking the phase of a pure tone AM signal to the onset of the rise of the carrier. This was done in order to investigate further the nature of precedence effects for lateralization. The ordinate is the same as in Fig.1. The abscissa represents the phase of the modulating signal at the time of the onset of the 700 ms rise. Data are shown for binaural signals both 180° and 360° out of phase, and at two carriers, 1000 Hz and 1600 Hz. Performance does not change as a function of phase locking, indicating that prece-
dence does not play a role in the initial segment of the signals used. Discrimination at 360° is more difficult than at 180°, and the psychometric functions suggest different processes.

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Fig.1: Effect of envelope modulation on pure tone laterali-
Fig.2: Effect of phase-locking AM envelope to onset of signal.
A NEURAL MODEL OF BINAURAL INTERACTION AND DOMINANCY FOR SOUND IMAGE SPACE

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Introduction

The localization of a sound in space depends on the interaural acoustic difference. When a brief stimulus is presented to both ears, the binaural image is located in dependence on arrival time and intensity. However, for sustained stimuli of low frequency (lower than 1,500 Hz), the binaural image moves following the interaural phase difference rather than the absolute time delay.  

In various relay nuclei, etc. of the mammalian auditory system, there have been found neurons sensitive to such interaural cues. Cassaday and Neff concluded after cat's lesion experiment that the superior olivary complex and trapezoid body are involved in the fundamental processing of binaural information, which is then projected to higher nuclei and the cortex. Anatomically, the two major masses in the superior olivary complex (SO), i.e., the lateral superior olive (LSO) and the medial superior olive (MSO) are recognized as the first relay nuclei form the bilateral cochlear nuclei (CN).

Békésy first presented a hypothetical model of binaural interaction. Van Bergeijk refined Bekesy's model to explain binaural interaction for bilateral MSOs driven by inhibitory output of the ipsilateral CN and excitatory output of the contralateral CN. The balance of the two MSOs in terms of firing rate determines the location of the image.  

In this paper, Bergeijk's model will be examined at the synaptic level to observe the effect of 'binaural fusion' as well as phase-intensity tradeoff.

Model

The present model consists of bilateral groups of four relay nuclei, i.e., CN; MNTB (medial nucleus of the trapezoid body); LSO; and IC (inferior colliculus). As shown in Fig.1, the CN, projected by the cochlear nerve, outputs to the ipsilateral LSO and contralateral MNTB, which is connected to the LSO on the same side. The LSO sends outputs to both ipsilateral and contralateral ICs. The system processes binaural information as follows.

1) Each CN receives a train of pulses with density analogous to the input wave.
2) Each neuron in CN responds transitorily as the pulse density of the CN input passes over its own threshold.

3) A burst of pulses of CN neurons is conducted to the ipsilateral LSO as well as to the contralateral MNTB, which acts as an inverter to discharge a burst of inhibitory pulses in the LSO on the same side.

4) These two types pulses from the ipsilateral CN and from the contralateral CN (via the MNTB) cause excitatory post-synaptic potentials (epsp) and inhibitory post-synaptic potentials (ipsp), respectively on LSO neurons.10

5) The LSO generates a burst of pulses corresponding to the positive peak amplitude of the combined positive epsp and negative ipsp.

6) The IC counts and compares the two bursts from the bilateral LSOs in reciprocal action to determine the image lateralization and the probability of fusion.

In the present model, the time course unit for isolated epsp ($v_e(t)$) or ipsp ($v_i(t)$) is simplified with reference to post-membrane equivalent circuit model composed of resting membrane potential, resistance, and capacitance.

$$v_e(t) = \begin{cases} \sin \frac{2\pi (T'/4)t}{\tau} & (0 \leq t < T') \\ \exp(-t/\tau) & (T' \leq t) \end{cases}$$

$$v_i(t) = -v_e(t)$$

where $T'$ is the duration of charging the membrane capacitance and $\tau$ is the time constant of discharging the capacitance. In repetitive mode,9

$$v_e(t) = \frac{\sin \frac{2\pi (T'/4)t}{\tau}}{(\exp(-\frac{(T'-T)/\tau}{\tau}) - \exp(-T/\tau))}$$

$$v_i(t) = -v_e(t)$$

where $T$ is repetition interval.

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Fig. 1. Block diagram of the binaural system. CN: cochlear nucleus; IC: inferior colliculus; LE: left ear input; LSO: lateral superior olive; MSO: medial superior olive; MNTB: medial superior olive; CN: cochlear nucleus of trapezoid body; RE: right ear input.
Simulation

Figure 2 shows the steady-state response of the system with $T' = T/3$ for binaural stimuli of the sinusoid ($f=1/T$) balanced bilaterally in amplitude when the interaural phase difference (IPD) is shifted from 0 to 360°. The laterality index $l(\text{IPD})$ and probability of fusion $u(\text{TPN})$ were defined as follows:

$$l = (L - R)/(L_0 + R_0)$$
$$u = 1 - (L + R)/(L_0 + R_0)$$

where $L$ and $R$ are left and right LSO outputs, and $L_0$ and $R_0$ are left and right LSO outputs when left or right ear is stimulated monaurally.

Figure 3 shows the steady-state response of the system with $T'=T/3$ for unbalanced binaural stimuli ((right input) = 0.9*(left input)) (A) and for balanced stimuli but with unbalanced LSO gains ((right G) = 0.9*(left G)) (B).

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Fig.2. Steady-state response of the system for binaural stimuli of the sinusoid ($f=1/T$) bilaterally balanced in amplitude in the case of psp charging duration $T'=T/3$. IPD: interaural phase difference; (A): epsp (excitatory post synaptic potential) & ipsp (inhibitory post synaptic potential); (B): combined potential of epsp & ipsp on LSO neurons; (C): outputs from the left (upward) and right (downward) LSOs; (D): laterality $l$ (1 div = 1/4 of the image position of a monaural stimulus); (E): probability of fusion $u$. 

Fig. 3. Steady-state response of the system for unbalanced binaural stimuli ($T'=T/3$). (A): (right input) = 0.9*(left input); (B): same as (A) but with unbalanced LSO gains ((right G) = 0.9*(left G)). 

The laterality index $l(\text{IPD})$ and probability of fusion $u(\text{TPN})$ were defined as follows:

$$l = (L - R)/(L_0 + R_0)$$
$$u = 1 - (L + R)/(L_0 + R_0)$$

where $L$ and $R$ are left and right LSO outputs, and $L_0$ and $R_0$ are left and right LSO outputs when left or right ear is stimulated monaurally.
Fig. 3. Two types of asymmetrical responses of the system. (A): for unbalanced binaural stimuli ((right ear input)=.9*(left ear input)) with balanced LSO gains. (B): for balanced input stimuli but with unbalanced LSO gains ((right LSO gain)=.9*(left LSO gain)).

Discussion and Conclusion

The phase-angle at the zero-crossing point of the lateralization curve in Fig.3(A) corresponds to the phase-advantage of the right ear necessary for centering the image of the binaural input with unbalanced amplitude, i.e., the 'phase-amplitude trade' when the epsp and the ipsp increase linearly as the input amplitude is strengthened. The lateralization curve in Fig.3(B) indicates a left dominancy of the image space, though the phase advantage is not necessary for centering the image at a zero interaural phase difference, unlike the case of an unbalanced input amplitude.

Further modelling of the LSO and MSO innervated by the visual system as well as the higher auditory system is interesting for analysing the mechanism of the interaction between sound localization and visual orientation in the egocentric perceptual field to explain the dominancy for binaural information processing as shown in the experimental data.

References
A COMPUTER MODEL OF BINAURAL LOCALIZATION

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INTRODUCTION

One of the objectives of stereophonic sound reproduction is to simulate the directivity of the original sound field so that a listener will perceive not only the sounds from the sources but also their location. Well defined tests are available for evaluating in physical terms the "fidelity" of the reproduced sounds, but assessing the "imaging" relies almost entirely on listening tests and subjective response. While a valid argument can be made for the use of listeners as the final proof for a given system it would nevertheless be very useful to have a repeatable physical testing procedure to evaluate imaging independent of the momentary perception and subjective judgement by listeners. Before an adequate test can be devised however, it is first necessary to understand and to model the processes by which a human with normal hearing perceives the position of sound sources. The objective of the present study was to create a computer model which simulates binaural localization.

Human perception and interpretation of sounds involves several very complex processes and the present model attempts to simulate only those segments of hearing that are important in localization. A sound field will generate many types of localization cues which can be divided into two primary categories, monaural and binaural, and further subdivided into temporal and spectral cues. For example, a binaural temporal cue will arise from the difference in time of arrival at the left and right ears of sound from a source according to the displacement of the source from the central plane of the head. A monaural temporal cue is created by the difference in time of arrival of the direct sound and the first reflection from the listener's shoulder. In addition spectral cues will be generated by frequency dependent diffraction around the head and by directivity of the ear flap.
From this it is apparent that the ability of human hearing to perceive spectral differences in sounds is important in localization. In other words the model must simulate the frequency-to-position transduction which takes place in the basilar membrane and hair cells. It is also apparent that the mechanism which detects small time differences for both impulsive and steady sounds must be modelled. Accurate localization using only these temporal and spectral cues can be achieved in an anechoic environment, but when such a model is exposed to direct and reflected sound the localization process is confused and unreliable. In human hearing the confusion is reduced by the precedence effect which suppresses response after the initial stimulus, and this characteristic should be included in the model.

LOCALIZATION MODEL

Figure 1 shows the block diagram for the proposed model. Sound from binaural microphones on a dummy head is digitized and analyzed by FFT. The two signals are then transmitted to the basilar membrane segment of the model where they are spectrally analyzed by fifty band-pass filters, simulating the tuning curves at equally spaced points along the basilar membrane. The signals are then reanalyzed by an inverse transform and sent to the hair cell segment of the model. This part of the model is fashioned after an equivalent circuit which uses the output of each basilar membrane filter to modulate a current which in turn is transformed into a neural firing rate. The hair cell model includes both rectification and high frequency roll-off as well as the spontaneous neural rate and the precedence effect. At this stage the signals in the model simulate the
actual pulsed signals in the auditory nerve fibres corresponding to the selected positions on the basilar membrane. Because these signals preserve the temporal patterns of the input sounds and because they are spectrally segregated they are used next to determine the interaural time delay. This is done by cross-correlation analysis, the first peak indicating the interaural time delay. Calculation of the interaural intensity ratio cannot be done easily on these neural signals because of the highly non-linear hair cell model. Instead this calculation is done on the spectrally analyzed analogue signal before the haircell transduction to neural pulses. This procedure is simply a calculation of the ratio of rms values of corresponding signals from the two ears.

At this point the model has sufficient information about the sound to make a localization judgement using interaural time delays and intensity ratios which are available for a large number of basilar membrane locations. The model for the decision-making process duplicates measurements in a controlled environment, using well defined sound originating from a known location. Time delays and intensity ratios were calculated for a wide variety of conditions and the model was devised. Typical relationships are shown in Figures 2 and 3. Ideally, localization decisions by time delay will always be the same as those by intensity ratio. Measurements have shown that time delay is a more reliable predictor at low frequencies and intensity ratio at high frequencies. The model uses a frequency dependent weighting function for the final decision.

MODEL VERIFICATION

For testing, the system has been exposed to a variety of sounds at many different positions relative to the dummy head. For example, one test used an offset cosine pulse from a source in the horizontal plane of the head but displaced 25 degrees to the left. For this measurement the output of one of the left side hair cell models had the neural rate pattern

![Figure 2 - Interaural Intensity Ratio (IIR) vs Angle of Incidence at 1.6 kHz](image-url)
shown in Figure 4. For the right ear the corresponding pattern was similar but at a lower level and delayed. Cross-correlation analysis revealed a time delay of .20 ms which translated into a displacement of 25.0 degrees when compared to Figure 3. The calculated interaural intensity ratio was .72 which from Figure 2, indicates an angle of 24.9 degrees. While the results do not always agree so well with each other or with the input information especially at very high or very low frequencies, present tests indicate close agreement over a large range of signal parameters and source positions. Although some problems with reflections and reverberation have yet to be resolved the performance of the model to date indicates that it can be used effectively to evaluate imaging.
EFFECT OF INTERAURAL TIME DELAY ON DEGREE
OF PERCEPTIBILITY OF REPETITION COLORATION

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1. Introduction

Degree of perceptibility of the subjective phenomenon repetition coloration of white noise, DRC, in an anechoic environment depends considerably upon the amount of time delay, type of presentation (i.e., directions from which the direct sound and its repetition arrive at the listener's ear), type of hearing (ditoic, dichotic and monaural), relative level of the repetition, group of subjects, type of posture (posture of the subject's head relative to the direction of facing the direct sound), etc. (1-3).

The repetition tone perceived, (2), is predominant and forms the majority of the overall sound impression especially with time delays of \( T = 5-15 \) ms (depending on the group of subjects: A, B or C), whilst it is very much fainter with \( T < 3 \) ms and \( T > 30 \) ms (2). The DRC-1 (1-3) for white noise perceived monaurally or diotically in free field is maximum with \( T = 5,10 \) or 15 ms for group (A), (B) or (C) of subjects, respectively, for frontal or overhead types of presentation (1,2). The DRC-1 for white noise perceived diotically in free field with \( T = 5-15 \) ms for lateral types of presentation is considerably less than those perceived monaurally for the same presentations and those perceived monaurally or diotically for frontal and overhead types of presentation (1-3). Furthermore, the characters (timbre and pitch) of the repetition tones perceived diotically for lateral types of presentation with \( T = 5,10 \) and 15 ms are entirely different from those perceived monaurally.

The characteristics of the repetition tones monaurally perceived are exactly the same as those perceived diotically or monaurally for frontal and overhead types of presentation. These differences are also appreciably different for the different lateral types of presentation (2,3). It is clear that the reason for these differences in character and degree of repetition coloration can only be due to the effect of interaural time delays of the direct sound and its delayed coherent repetition, arising due to the type of presentation and the posture of the subject's head. Therefore, our previous hypothesis was that neutralization and/or distortion of perceptibility of repetition coloration depend considerably upon the absolute value of the difference between the interaural time delays of the direct sound and its delayed repetition. No neutralization and/or distortion of perceptibility of repetition coloration occur when these interaural time delays are equal, i.e., when \( T_D = T_R \) as indicated in figure 1, (2). The aim of the present work is to further investigate the validity of this hypothesis.
2. Experimental technique

The direct sound and its delayed repetition were separately radiated via two loudspeakers, which were placed at an angle of azimuth of $90^\circ$ (i.e., LP: $90^\circ$) and at a distance of three metres from the centre of the subject's head in an anechoic environment. Both loudspeakers were exactly directed to the position of the centre of a subject's head. The direct sound and its delayed repetition were stereophonically recorded according to a suitable changeover design for a subjective experiment (2). The subject hears the colored stimuli when a certain switch is on position 1, whilst he can hear the uncolored presentation ($\tau=0$) by turning the switch to position 2. This kind of comparison could be made by the subject himself, at any time, as often and for as long as he might require throughout the experiment in order to give accurate judgements. Three types of posture of a subject's seating or head were employed, namely the $0^\circ$, $90^\circ$, and $135^\circ$ facing, which mean that the subject's facings are directed at angles of azimuth of $0^\circ$, $90^\circ$, and $135^\circ$, respectively, relative to the direction of the direct sound.

3. Results

Seven subjects participated in this experiment. Four of them were belonging to group (A), and three of them were belonging to group (B). Any value of DRC-1 for any value of time delay and type of posture shown in figures 2 and 3 represents the mean value of the means determined diotically by the subjects. Each mean value determined by every subject is the mean value of six judgements performed by him, according to the changeover subjective design (2), with that particular condition. The variations of DRC-1 with time delay for three types of seating posture determined by the subjects of group (A) or (B) are shown in figure 2. Figure 3 shows those variations determined by all subjects of groups (A) and (B) together.

4. Discussion

Figure 2 shows that the DRC-1 values determined by group (A) with $135^\circ$ facing are considerably higher than those determined with $0^\circ$- and $90^\circ$-facing, for all values of time delay, in particular, for $\tau=5-20$ ms. The percentage increases of DRC-1 values determined with $135^\circ$-facing relative to those determined with, e.g., $0^\circ$-facing are $119, 117, 117, 114, 67$ and $49\%$ for $\tau=5, 10, 15, 20, 30$ and $50$ ms, respectively. These increments are more or less the same as those relative to the DRC-1 values determined with $90^\circ$-facing. Figure 2 shows that the DRC-1 values determined by group (B) of the subjects with $135^\circ$-facing are considerably higher than those determined with $0^\circ$- and $90^\circ$-facing for all values of time delay, particularly at $\tau=5$ and $10$ ms. Further interesting observation is that the maximum DRC-1 response of group (B) with $0^\circ$- and $90^\circ$-facing is at $\tau=15$ ms, whilst it is at $10$ ms with $135^\circ$-facing. The percentage increase of DRC-1 values determined with $135^\circ$-facing relative to those determined with, e.g., $0^\circ$-facing are $291, 189, 65, 40$ and $56$ and $28\%$. Figure 3 shows that the DRC-1 values determined by all subjects with $135^\circ$-facing are considerably higher than those determined with $0^\circ$- and $90^\circ$-facing for all values of time delay, particularly at $\tau=5-20$ ms. Furthermore, the characters of repetition tones perceived diotically with $135^\circ$-facing are more or less the same as those perceived monaurally for all lateral types of presentation, or monaurally and diotically perceived for frontal and overhead types of presentation. On the other hand, the characters of repetition tones perceived with these conditions, particularly at $\tau=5$ and $10$ ms, are entirely different.
from those diotically perceived with $0^\circ$- and $90^\circ$-facing for the lateral type of presentation; LP:$90^\circ$. Moreover, the character and degree of repetition coloration perceived with $135^\circ$-facing for LP:$90^\circ$ presentation, particularly at $T=5$ and $10$ ms, change considerably with movements of the subject's head around its axis, as reported by all subjects. Therefore, each subject participated in the $135^\circ$-facing experiment required a little adjustment of his head's position until he perceived the required repetition coloration. These mean that the repetition pitch for $T=5$ and $10$ ms diotically perceived for lateral types of presentation does not correspond to the reciprocal value of time delay. This is in contrast to the hitherto accepted repetition pitch theory (4-7). These facts indicate clearly that the reason for these considerable changes in character and degree of repetition coloration can only be due to the effect of the absolute value of the difference between the interaural time delays of the direct sound and its delayed repetition, $T$. This difference is almost zero or in the order of 330-400$\mu$s (depending on the size of the subject's head) for $135^\circ$-facing or $0^\circ$- and $90^\circ$-facing, respectively. Therefore, we can certainly state that our present investigation proves indeed the validity of our previous hypothesis, as indicated above (2).

5. Conclusion

The character and the degree of perceptibility of repetition coloration diotically perceived for LP:$90^\circ$ with $135^\circ$-facing (approximately $T=0$) are more or less the same as those monaurally perceived for lateral types of presentation, and those monaurally and diotically perceived for frontal and overhead types of presentation. Character and degree of perceptibility of repetition coloration determined diotically for LP:$90^\circ$ with $0^\circ$- and $90^\circ$-facing ($T=330-400$ $\mu$s) are considerably different from those determined with the above-mentioned conditions. Therefore, we can conclude that our present investigation proves indeed the validity of our previous hypothesis: "the neutralization and/or distortion of perceptibility of repetition coloration depend considerably upon the absolute value of the difference between the interaural time delays of the direct sound and its delayed repetition. No neutralization and/or distortion of perceptibility of repetition coloration occur when these interaural time delays are equal" (2).

6. Refences

1. N.Y. AL-RAWAS 1978 IOA Spring Conf., Cambridge, England. Monaural or diotic perception of degree of Repetition Coloration with frontal or lateral presentation in free field.
AL - RAWAS, EFFECT OF INTERAURAL TIME DELAY ON REPETITION COLORATION

Fig.1. Interaural time delays of a direct sound, $D_1$, and its delayed repetition, $R_1$, for a certain lateral type of presentation.

Fig.2. Variation of DRC-1 with time delay determined by groups (A) & (B) for 0°-, 90°- and 135°-facing.

Fig.3. The variation of DRC-1 with time delay determined by all subjects for 0°-, 90°- and 135°-facing.
LOCALIZATION OF SOUND BY SUBJECTS WITH VARYING DEGREES AND TYPES OF DEAFNESS

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There have been few investigations delineating effects that deafness of various categories sensori-neural; conductive; mixed) and degrees (minimal to severe) might have on sound localization accuracy. As well, little is known about whether ameliorative techniques (whether surgical; medical; or particularly the use of hearing aids) restore "better" localization capability.

Data on such problems is somewhat unreliable due primarily to inconsistent categorizations of deafness, and non-systematic variations of the sources. What is known is that: when compared with normal hearers, deafness debilitates localization; especially azimuthal, 2) whether one type of deafness debilitates if more is unsettled, e.g. Roser (1966), Palmer (1966) say Conductive poorest; while Jongkees & Veer (1957) and Viewheg & Campbell (1960) say Conductives best; and Nordlund (1964) said the order of poorest to best localization was monaural, SN, conductive, CNS lesions and vestibular problems. 3) Loss levels' effects are not widely understood although the more asymmetric the loss in bilateral deafness, the more likely it is disruptive (Viewheg et al; Roser). What could have been the most relevant articles regarding clinical categories, loss levels, audiometric symmetry were Tonning's (1969-75). He presented masked or unmasked speech to monaurals and binaurally deaf under aided or non-aided conditions. Unfortunately, he largely ignored the clinical data in his studies and his definition of unilateral deafness was not strict. Nonetheless, he reported; 1) unilateral losses <33dB PTA, regardless of type (SN; C; M) disrupt speech localization 2) asymmetric loss is more debilitating 3) audiograms may not predict levels of localization difficulty, 4) masking of speech depends on azimuth of the source relative to the signal and ear positions. Gatehouse et al (1976; 79) using vigorous definitions of monaurality, and testing "simultaneous localization" (specification of signal elevation and azimuth) in full spherical space (most other studies used only frontal aural or median plane, or single elevation 360º azimuthal positions) found; no differences between SN and Conductives; that monaurals localize well above chance; that no simple relationships exist between various etiological factors and localizing ability.
LOCALIZATION OF SOUND BY SUBJECTS WITH VARYING DEGREES AND TYPES OF DEAFNESS

GATEHOUSE, R. WAYNE, Ph.D.

Recent evidence has suggested hearing aids, in meeting demands for amplification and frequency response, may not be meeting other psycho-acoustic dimensions. For example, since localization is extremely dependent on auricle transforms, use of in-ear molds might be disruptive. Orton & Prevees (1979) compared unaided, and monaurally in-ear and over-ear aids in mild to moderate losses under different S/N ratios. Aided localization was less accurate than unaided where some residual hearing remained but aidshelped where little hearing remained. The decreases under aided conditions were attributed to mold occlusion effects. Toning (op cit) using several aided conditions (aid type; Mon. vs Bin.), in general reported aided unmasked speech localization was better than unaided, but only to signals aimed directly at the ear. In no case was aided better than unaided localization for all the other conditions, but asymmetry of the audiogram was important. Finally, Nabelek et al (1980) reported aids increased errors, and that binaural aids actually increase asymmetries.

In summary, neither the effects of Loss Type or degree, nor the effects of aid use on localization are well-established. The object of this study was to look at these effects in full localization space.

METHOD
Subjects: Binaurally deaf (av. PTA across .25 to 8.0 kHz) in loss level ranges of 26-35, 36-45, 46>, of varied symmetry, diagnosed as sensori-neural or mixed deaf.
Procedure: Ss were seated in a height-adjustable pedestal chair, visually isolated by surrounding curtains from the rest of the apparatus. All wore a headband-mounted light which when non-aligned with a photocell embedded at 0° in the curtain, interrupted the signal delivery programmes. Head movements were thus controlled. Outside the curtains, a computerized boom-mast allowed speakers to be positioned at any of eight 45° azimuths (re 0°). The mast had 3 vertical speakers (0°±30°) permitting any level to be selected on any trial. Thus signals could randomly come from any of 24 positions (8 az. x 3 elevations). Three aided/or unaided 24-trial blocks were localized to each of WN, 1.0 and 2.5 kHz tones, and a speech segment. Ss' responses were via a hand-held device that recorded their estimate of the stimulus position in degrees. The computer translated these responses to error patterns.
LOCALIZATION OF SOUND BY SUBJECTS WITH VARYING DEGREES AND TYPES OF DEAFNESS

GATEHOUSE, R. WAYNE

RESULTS AND CONCLUSIONS

ANOVA's showed both loss levels and clinical categories (Types) were significant (p<0.05) for hits (correct simultaneous localization) and azimuthal, but not elevation errors. Fig. 1 gives mean percentage hits by unaided SN and M Se (left). Obviously significant type effects are only to tonal stimuli. Level effects (right) do make considerable difference to accuracy especially on WN and speech. Fig. 2 shows unaided average azimuthal errors (i.e. positions away from the source). Now the types differ. Mixed deaf are more accurate (smaller errors) on all stimuli. Generally, for Loss Levels, the less the loss, the less the error. Fig. 3 shows overall, that unaided accuracy is significantly better for WN and speech. In conclusion, audiometrically defined loss levels, and clinical types may be predictive of localization difficulties especially in azimuth. Unaided localization is superior for everyday sounds (speech; noise) possibly because they disrupt pinna transforms. Such data may be useful in both diagnosis and treatment of deafness, and is certainly suggestive that aid use might be disruptive rather than beneficial in some basic acoustic phenomena. Manufacturers should consider such factors as localization in their design of aids.

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LOCALIZATION OF SOUND BY SUBJECTS WITH VARYING DEGREES AND TYPES OF DEAFNESS

GATEHOUSE, R. WAYNE

Figure 1 Mean Percent Correct Simultaneous Localization by Clinical Category (left) and by Loss Levels (right)

Figure 2 Average Azimuthal Errors by Clinical Category (left) and Loss Levels (right)

Figure 3 Mean Percent Correct Simultaneous Localization of Four Stimuli Under Aided and Unaided Conditions
INFLUENCE OF AMBIENT NOISE ON LOCALIZATION OF TONE BURST

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1. INTRODUCTION

For a click to be lateralized at the center of the head, Raab et al. reported that if noise was presented to one ear, sound pressure level of the click (signal) at that ear should be increased as compared with that at the other ear.1) Burtseva et al. reported, on the other hand, that when tonal signals given to both ears through earphones are 180° out of phase with each other, a location of the image could move if homophonic noise signals are presented to both ears.2) However, no literature can be found in connection with the effect of noise on sound localization.

As a first step to study the effect of noise on sound localization, experiments on tonal signals of a real or a phantom image were carried out in the presence of noise radiated from a single loudspeaker.

2. EXPERIMENTAL PROCEDURE

Seventeen loudspeakers are arranged in an anechoic chamber within the angles from -40° (left to the front) to +40° (right to the front) every 5° on the circumference of a circle with a radius of 2 meters and at the same height as subject's ears. A subject sits at the center of the circle.

The pure tones with the frequencies of 500, 1k or 2k Hz were used as the signal. The real image signal was radiated from a loudspeaker located at every 10° within ±30° and the sound pressure levels were 50, 60 or 70 dB. The phantom image signals, on the other hand, were radiated from a couple of loudspeakers located at -30° (left channel) and +30° (right channel) respectively, and both signals were in phase. One of the two channels is taken as the standard channel here, and the sound level of the signal is indicated by the level in the standard channel (the standard level) and by the level in the other channel relative to the standard (the relative level). The standard level was always 60 dB SPL and the relative levels were changed between -18 dB and +18 dB with 3 dB step.

Interfering noise was the band-limited pink noise with the cut-off frequencies at 300 and 8k Hz. The level was either 50 dB SPL or 70 dB SPL, and its direction was taken as -30°, 0° or +30°. An experiment of noiseless condition was also carried out for the comparison.

The way of signal presentation is as follows: In the first place, the signal sound was presented in company with the interfering noise. In the second place, only a pure tone signal of the same sound level as the signal which appeared in the first stimulus was presented as a comparison.
stimulus. The comparison stimulus was always radiated from a single loudspeaker, and a subject could select one of the loudspeakers by turning a control knob at his hand within the limit of ±35° with 5° step. The subject adjusted a direction of a comparison stimulus so as to coincide with that of a signal in test stimulus. Then the result of his adjustment was read by a computer. Every stimulus condition was repeated five times, except for the noiseless condition in which the number of repetition was nine in order to test the accuracy of subject's judgment. Subjects were two males in their twenties and two females in their late teens with normal hearing acuity.

3. EXPERIMENTAL RESULTS

3.1 Sound localization for a signal to create a real image

Figures 1-3 show the results for the signal frequency of 500 Hz. The abscissa shows the direction of the loudspeaker from which a signal sound is radiated, and the ordinate indicates the direction of the loudspeaker selected by a subject for comparison stimulus. The broken line shows the line corresponding to the case when the judged direction is in accordance with the direction of test signal. Experimental values are the averages among four subjects, whose judgment showed similar results to each other.

Figs. 1-3 Perceived direction of real sound image vs. direction of the source. The signal frequency is 500Hz. Fig.1 shows the results for noiseless condition. In Figs.2-3, the signal level is 60dB SPL, and the noise levels are 50 and 70 dB SPL(A) respectively.
Figure 1 showed the result for noiseless condition. Judgements seem to be fairly accurate. In Fig.2 & 3 the results in the presence of noise are plotted. It can be seen from the figures that sound image for test signal moves from its actual position to the direction opposite to that of noise source. The smaller the directional separation between signal and noise sources is, the greater the perceived shift of signal source direction is. And the influence of noise on sound localization for the signal disappears if the separation of direction is about 30° or more.

For 1 kHz tone signal, the results show the same tendency as those for 500 Hz, though the amount of angle shift is smaller in 1 kHz than in 500 Hz. For 2 kHz the judgment varied very much from person to person. This fact means that the sound localization for 2 kHz tone is considerably disturbed by the presence of interfering noise.

3.2 Sound localization for a signal to create a phantom image

Figures 4–6 show the experimental results for 500 Hz. All subjects exhibited the same tendency, and accordingly the values shown in the figures are the averages among them. In these figures, the abscissa denotes the relative level of signal, and the ordinate shows the direction of resultant image reported by the subjects in terms of a comparison stimulus.

Figs. 4–6 Perceived direction of phantom sound image vs. the signal level relative to that of standard channel. Fig.4 shows the results for noiseless condition. In Figs.5 and 6, the signal level of the standard channel is 60dB SPL, and the noise levels are 50 and 70 dB SPL(A) respectively.
Fig. 4 shows the result for noiseless condition. The dashed line in this figure shows the calculated direction of phantom image given by \( \tan \alpha = (k-1) \tan \theta / (k+1) \), where \( \alpha \) is an angle of sound localization from the front, \( 2\theta \) is an angle between two loudspeakers and \( k \) is the ratio of the sound intensity of one channel to the other. 3)

As seen from the Figs. 5 & 6, the resultant image for test signal moves from its position in noiseless condition to the direction opposite to noise source. The smaller the directional separation between resultant images for signal and noise is, the greater the perceived shift of signal location is. These tendencies are the same as in the case of real image.

For 1 kHz the tendency of results varied among subjects, though the dispersion of perceived direction within a subject is not so great. Judgment of sound localization for 2 kHz signal was very difficult as in a real image.

4. DISCUSSION

In sound localization for both real and phantom images, all of the subjects reported similar results, that is if an interfering noise was given to them, the sound image shifted to the direction opposite to noise source as far as the signal frequency was 1 kHz or below. This fact may be explained by the difference in masking between both ears. Namely, if noise comes from the right, loudness of noise at right ear is generally greater than the opposite ear. This causes a reduction of loudness of the signal sound at the right ear and consequently the shift of sound image to the left. The sound image, however, may not simply be decided by the loudness difference between both ears, but the higher level of central nervous system may have a strong connection with auditory space perception. In this case, a contrast between signal and noise images in an auditory space may cause the deformation and shift of sound images.

For a phantom image with the signal frequency of 1 kHz, the direction of a signal reported by subjects was varied in its tendency with each other, in the presence of interfering noise. For the frequency of 2 kHz, furthermore, the subjects' judgments showed a quite large variance and a difference of tendency in the presence of noise. This means that it is very difficult to judge the sound source with confidence if the signal frequency is higher, and the critical frequency seems to lie around 1 kHz. Can the mode of the mechanism for sound localization change around this critical frequency?

5. CONCLUSION

It is clarified that when the signal frequency is about 1 kHz or below, the sound image shifts to the direction opposite to that of noise irrespective of the kind of sound image, real or phantom. When the signal frequency is higher than about 1 kHz, the sound localization for tone signal is disturbed very much by the presence of interfering noise.

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RELATIONS BETWEEN HEARING LOSS AND OTHER AUDIOMETRIC DATA

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Introduction

Often results of psychoacoustic measurements are presented as auditory functions giving properties of the auditory system as a function of some physical parameter of the stimulus. Because many of such functions can be measured in a single ear, they are just different cross-sections for the same auditory system. Therefore, coherence among these auditory functions is very likely. In an attempt to find relations among auditory functions 22 sensorineurally hearing-impaired listeners were subjected to a battery of auditory tests. Apart from the audiogram we selected some tests on frequency resolution and some on temporal resolution. In order to limit the number of measurement conditions, all tests were performed for a single probe-tone frequency (1000 Hz). Additionally, speech reception both in quiet and in noise was measured in order to study its relation with the auditory functions. For all subjects masker levels were kept in a narrow range of sound-pressure levels in order to avoid differences among subjects that are simply the result of differences in presentation level. The measurements were performed using an adaptive two-alternative-forced-choice (2AFC) procedure with visual feedback. Bias in the data resulting from differences in training and motivation among the subjects was eliminated in most of the tests because auditory properties were expressed by the difference between two threshold values. All tests were repeated once after a few days. From the correlations between test and retest, reliabilities were calculated which were above 0.9 for nearly all tests. In this paper we will focus on the relations with hearing loss. A more detailed report on this study is published elsewhere (Festen and Plomp, 1983).

Experiments

Pure-tone audiometry: For the selected subjects absence of conductive hearing loss was confirmed using standard audiometry. Further tests were conducted monaurally on the ear with the lower threshold averaged over 500, 1000, and 2000 Hz. In each of 4 sessions detailed threshold measurements were performed in the 1000-Hz region. Measurement frequencies were 630, 800, 1000, 1260, and 1590 Hz. In the analysis these data are converted to mean audiometric loss and mean audiometric slope covering together 92.9% of the variance. For the purpose of data presentation, in all figures subjects are divided in four subgroups on the basis of their mean loss. The two middle groups contain 6 subjects each and the other groups 5 subjects.
Frequency resolution: Three paradigms were used to study frequency resolution: auditory bandwidth derived from the masking difference between peak and trough of comb-filtered noise, psychophysical tuning curve, and critical ratio. The first two paradigms were used in simultaneous as well as in nonsimultaneous masking. For the bandwidth experiment results are given in Fig. 1. In normal hearing the bandwidth in nonsimultaneous masking is about half that in simultaneous masking (80 Hz and 160 Hz, respectively). This is presumably a consequence of lateral suppression (cf. Houtgast, 1974). However, for our group of hearing-impaired subjects these bandwidths are nearly similar. Only for the least hearing-impaired subjects the bandwidth in nonsimultaneous masking is the smaller one. Indications for reduced lateral suppression with sensorineural hearing loss were found before by Wightman et al. (1977). Following this reasoning, the high correlation (see Table I) between bandwidth in nonsimultaneous masking and hearing loss is mediated by suppression. In simultaneous masking, where the effect of suppression is not measured, the correlation is much lower. For the psychophysical tuning curves, slope values were calculated for the high and low-frequency edges. Slopes were much shallower than for normal hearing, but they did not show clear correlations with hearing loss.

Temporal resolution: The ability to resolve auditory events in the time domain was studied in three experiments. First, we measured the width of a time window in an experiment which is the time-domain analog of the bandwidth experiment. Further, the time course of forward and backward masking was determined. In all three experiments the probe signal was a 0.4-ms click octave filtered with a central frequency of 1000 Hz. For the hearing impaired the average time window is 53 ms, much wider than for normal-

![Diagram](image1.png)

**Fig. 1.** Threshold-level difference for a 1000-Hz probe tone between peak and trough of comb-filtered noise (20 db modulation) as a function of peak spacing. Panel (a) for simultaneous masking and panel (b) for nonsimultaneous masking. Various symbols represent subgroups with different loss. The smooth curves give calculated threshold differences for a Gaussian-shaped filter.

![Diagram](image2.png)

**Fig. 2.** Thresholds for an octave-filtered click probe (1000 Hz) as a function of the time between masker and probe. The masker is an octave band of noise centered at 1000 Hz and with a spectral density of 60 dB/Hz. The dashed lines represent average results for two normal-hearing subjects for masker levels of 20, 40, and 60 dB/Hz, respectively. Various symbols indicate subgroups with different mean loss.
hearing subjects at comparable sound-pressure level. Although at equal sensation levels the discrepancy is largely reduced, no correlation with hearing loss could be demonstrated. Clear relations with hearing loss were found for the temporal masking curves shown in Fig. 2. Slopes of forward and backward masking, calculated from straight line approximations to the data, show strong correlations with mean audiometric loss. Shallowest slopes are found for the most hearing-impaired subjects. The shape of the masking curves clearly differs from normal hearing. While for normal-hearing subjects masking drops sharply immediately before and after the masker and more gradually at greater delays, for impaired hearing a gradual decay is found over the whole range. Apparently long lasting components of masking are stronger for hearing-impaired subjects and dominate even at short delays.

Fig. 3. Speech-reception threshold as a function of background noise level. The dashed curve represents the average result for normal-hearing subjects. Various symbols are used for subgroups with different mean loss.

Speech reception: With 10 lists of short sentences speech-reception thresholds were measured at four levels of interfering noise and in quiet. Results are shown in Fig. 3. Following the model by Plomp (1978), the results for hearing-impaired subjects can be described with two parameters: (1) the D parameter representing hearing loss for speech in noise and interpreted as a distortion term; (2) the (A+D) parameter representing hearing loss for speech in quiet and interpreted as resulting from attenuation (A) and distortion (D) together. In quiet there is a close relation between hearing loss for speech and mean audiometric loss, but with interfering noise this relation is lost.

Relations and discussion

Table I only gives correlations with mean audiometric loss, but for all pairs of tests correlation coefficients can be calculated and collected in a matrix. After principal-components analysis, factor loadings on the first two dimensions are shown in Fig. 4. Because of the low subject-to-variable ratio in this analysis, the stability of the derived principal components was verified with separate analyses on the results of the test and the retest. These two analyses showed almost the same principal components as

| logB_s Cr logB_n Lf_s Hf_s Lf_n Hf_n τ Forw Backw Clth Clin A+D D |
|-----------------|-----------------|-----------------|-----------------|-----------------|-----------------|-----------------|-----------------|-----------------|
| 0.44            | 0.59            | 0.71            | 0.32            | 0.34            | 0.37            | 0.49–0.18       | 0.78            | 0.80            |
| 0.85            | 0.34            | 0.83            | 0.38            |

Table I. Correlations of test scores with mean audiometric loss. The sign of the scores is taken such that lower scores represent better hearing in all tests. Fully underlined values are significant at 1%; dashed underlinings are for the 5% level. Symbolic names are as listed in Fig. 4.
for the average results. In the factor loadings of panel (a) two distinct clusters can be seen. The upper cluster contains hearing loss for speech in noise and frequency-selectivity scores. The lower cluster contains scores related to the absolute threshold. Seen from the origin the two clusters are not in perpendicular directions, which means that there is at least some relationship. Critical ratio and both bandwidth and high-frequency edge of the PTC in nonsimultaneous masking take up positions between the two clusters. For the latter two this may be caused by the vulnerability of lateral suppression to hearing loss as discussed. Regarding the critical ratio, the threshold of a tone in noise is determined by two subject-dependent quantities: the width of the auditory filter and the efficiency of the system (cf. Patterson, 1976). It may be that this efficiency, not measured separately, also is a function of hearing loss. Tests situated close to the origin are poorly represented in the two-dimensional subspace of the first two factors and have only weak relations with the other tests.

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COURBES D'ACCORD ET DEFICIENTS AUDITIFS

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Dans les examens audiométriques, deux types d'épreuves sont habituelles : la recherche du seuil tonal et l'appréciation de la compréhension de la parole. Pour cerner le comportement auditif d'un sujet, il est nécessaire d'inclure des épreuves de sélectivité afin d'établir un diagnostic ou un pronostic, d'où trois types d'examens nécessaires étudiant :

1. la sensibilité, la sélectivité, la reconnaissance.

Bien qu'ayant une sensibilité liminaire considérée comme normale, certains sujets se plaignent de ne pas comprendre dans des ambiances bruyantes alors que dans le calme, les épreuves de reconnaissance phonétique sont normales. Il ressort que dans la plupart des cas, la sélectivité apparaît comme aménagée par des facteurs à découvrir. L'évaluation de la sélectivité s'effectue par la mesure des courbes d'accord psychoacoustiques (C.A.P.A.) ou PGT (Psychoacoustical Tuning Curves).

Moyens de mesure

Les épreuves s'effectuant au casque puis en champ frontal, le signal doit produire un minimum d'ondes stationnaires. La modulation en basse fréquence convient et permet des mesures fiables chez le déficient auditif appareillé. Un double générateur produit le son pulsé à masquer et le son masquant ayant les caractéristiques suivantes (ΔF / F = 10 %, vitesse de modulation 3 Hz, fréquence centrale réglable par 1/3 d'octave, temps de montée et de descente 20 ms, deux bouffées tonales par seconde).

Les méthodes du prémasquage ou du masquage simultané peuvent être utilisées, la seconde sera celle utilisée communément.
Application de l'étude de la sélectivité

De nombreux auteurs (1, 2, 3) se sont penchés depuis cinq ans sur les courbes d'accord psychoacoustiques chez le sujet normal. Pour les sujets présentant une perte auditive d'origine neurosensorielle, les études moins nombreuses (4, 5, 6, 7), montrent que l'intelligibilité et la forme des courbes d'accord sont liées car l'élargissement de celles-ci provoque une diminution de la reconnaissance.

Appliquée à la correction auditive, la mesure des courbes d'accord du sujet apparié donne la résultante du couplage aide auditive - oreille, lorsque le sujet est placé à deux mètres d'un haut-parleur. L'épreuve pratiquée pour les fréquences caractéristiques $F_c = 1$, 2 et 3 KHz est limitée par la courbe de réponse de l'aide auditive par l'efficacité du système contrôlant la dynamique (AGC, PC ...) ainsi que pour certains cas la réponse de l'oreille.

Il ressort que la mesure à 2 KHz permet d'attribuer ou d'infirmer la cause de confusions phonétiques à la forme des courbes d'accord. À 1 KHz, la forme est souvent plus normale alors que les aplatissements apparaissent à 2 KHz et au-delà dans les surdités neurosensorielles, soit dans la zone la plus atteinte pour les courbes plongeantes.

Sociallement, l'altération des courbes d'accord conduit à des difficultés surmontables dans le silence pour la compréhension d'un ou deux locuteurs, jusqu'à une inintelligibilité complète dans un bruit d'ambiance. La forme de la pointe et sa largeur $(Q_{10})$ et la "dynamique" de la courbe d'accord conditionnent l'extraction des traits phonétiques concourant à la reconnaissance. Ainsi verras-t-on des confusions /s/ $\rightarrow$ /ʃ/, /t/ $\rightarrow$ /k/, /s/ $\rightarrow$ /v/, /d/ $\rightarrow$ /g/ ... /ʒ/ $\rightarrow$ /z/.

La dispersion des résultats et les divers types de surdité rendent difficiles l'établissement de matrices de confusions.

Une étude a été faite (8) pour tenir compte de l'influence de l'âge d'un sujet normal (de 55 à 85 ans) sur la forme des courbes d'accord. Nous appellerons "dynamique" la différence d'ordonnée entre le niveau à la fréquence $F_0$ et le niveau pris à l'octave inférieure ou supérieure.

Statistiquement, nous obtenons :

Pour $F_c = 1000$ Hz
- dynamique grave : $y = 53,4 - 5,42 \log x$
- dynamique aigue : $y = 373 - 72,5 \log x$

Pour $F_c = 2000$ Hz
- dynamique grave : $y = 94,3 - 16,73 \log x$
- dynamique aigue : $y = 533 - 113,43 \log x$

 où $y$ représente la dynamique exprimée en dB relatif et $x$ l'âge du sujet (avec $x \geq 60$).
Conclusion

Les altérations de la sélectivité s'évaluent dans différents buts : dans le domaine de l'Audioprothèse, l'épreuve autorise un meilleur pronostic de la réhabilitation, en dissociant le gain quantitatif (essentiellement par amplification) du gain qualitatif, c'est-à-dire de l'amélioration de la reconnaissance obtenue par traitement du signal.

La sélectivité de l'oreille conditionne sa capacité à discerner dans une ambiance de bruit et à discerner les formants et leurs transitions. Les déficiences auditives d'origine cochléaire voient une modification des réponses des "filtres auditifs" qui tendent à perdre leur forme normale : la dynamique diminue plus les pointes s'élargissent.


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MELODY RECOGNITION BY HEARING-IMPAIRED LISTENERS

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1. INTRODUCTION

In previous research in our group (Dreschler & Plomp, 1980; Festen & Plomp, 1983) data on different auditory functions were correlated with the speech-reception threshold in quiet and in noise, both for normal-hearing and hearing-impaired listeners. In this paper we will report on experiments in which the streaming organization of the auditory capacity is studied: to which degree is the perception of a hearing-impaired subject disturbed by \textit{hearing two or more well-defined fluctuating signals at the same time}. From the speech-intelligibility experiments (see also Plomp & Mimpen, 1979) we know that hearing-impaired subjects need a 5 to 10 dB better signal-to-noise ratio than normal-hearing subjects to understand speech. The audibility of simultaneous signals was studied by means of musical melodies, with the frequency distance between three melodies required to perceive correctly the middle one as a criterion. Additionally to the melody recognition also the auditory bandwidth was measured (Houtgast, 1974). This parameter, together with hearing loss, appeared to be the most representative factor for the description of impaired hearing (Festen & Plomp, 1981).

2. EXPERIMENTS

Method

Firstly the pure tone threshold was measured for 11 frequencies (125 Hz, 250 Hz, 500 Hz, 630 Hz, 800 Hz, 1000 Hz, 1250 Hz, 1600 Hz, 2000 Hz, 4000 Hz and 8000 Hz). We used an adapted Békésy procedure which takes about ten minutes per ear.

Secondly the melody-recognition limit was investigated. The subject had three push buttons at his disposal, one corresponding to the example melody, the other two to the example or an alternative melody presented simultaneously with two masking melodies, see fig. 1. The subject is allowed to hear the three signals each two times after which he has to decide which of the two complex signals contains the example melody. The example melody was in the frequency region around 500 Hz; the alternative melody is the same except with the second and third notes in the reversed order. The two masking melodies were in a frequency region at a variable distance above and below the example melody. The melodies consisted of four pure tones. The
distance of the tones within one melody was never greater than three semitones, with no equal tones in one melody. The tone duration was 350 ms with silent intervals between the tones of 50 ms. The tones were gated with rise and fall times of 15 ms and were chosen according to the equal-tempered scale. There were no octave relations between tones of the example melody, the alternative melody and the masking melodies. The melodies were presented on an equal, most comfortable level. The minimum frequency distance required to perceive the middle melody was measured with a two-alternative forced-choice procedure. The starting value was 45 semitones distance between masking melodies and example melody. The step size was two semitones, the threshold distance was reached with a probability on a correct response of 79% for an individual trial.

Fig. 1. Example of the procedure for the melody-recognition experiment.

Thirdly the auditory bandwidth was measured with a 1000 Hz probe tone presented simultaneously with a comb-filtered noise (85 dB(A)). The detection threshold was measured both on the peak and in the valley of the noise signal, with ripple densities of 0.5, 1 and 2 ripples per 1000 Hz. We used an adapted two-alternative forced-choice procedure which takes about thirty minutes per ear.

Apparatus

The experiments were controlled by computer. The subjects were situated in an anechoic chamber and listened monaurally by headphone.

Subjects

The subjects were 20 pupils of a high-school for the hearing impaired. Their age varied from 13 to 17 years. Their hearing loss was sensorineural. As a reference group we used 10 pupils at the same education level as the hearing-impaired subjects.
3. RESULTS

In fig. 2 the results of the first experiment on the pure-tone threshold are plotted, with reference to the mean results of the normal-hearing subjects. As can be seen the group of 20 subjects had a great variability in slope in the audiogram, plotted along the x-axis, as well as in mean audiometric loss, plotted along the y-axis. In the experiments of Dreschler & Plomp (1980) these two parameters appeared to be the most important to describe the audiogram, so we have used them for correlation with the other data.

![Fig. 2. Mean slope and mean audiometric loss for the 20 hearing-impaired subjects.](image)

Fig. 3 represents the distance threshold of masked melodies. The mean threshold for the normal-hearing subjects is 5 semitones (standard deviation 3 semitones) and the mean threshold for the hearing-impaired subjects is 27 semitones (standard deviation 10 semitones). The test-retest reliability for both groups is very good, the coefficient is higher than 0.90. Apparent there is a great difference between both groups.

![Fig. 3. Histogram of the number of subjects with a melody-distance threshold as given by the abscissa.](image)

In fig. 4 the auditory-bandwidth data are combined with the results of the melody-recognition experiment. Since in both experiments the ear's frequency resolution is involved, one would expect a relation between the results of both tests. The correlation, however, is very low (the coefficient is 0.23) and not significant. The results of the melody-recognition test correlate not better with other parameters measured.
Fig. 4. Frequency distance between masking melodies and example melody as a function of the logarithm of the auditory bandwidth, both for the normal-hearing (under-cast symbols) and hearing-impaired (upper-cast symbols) subject.

It is apparent that some hearing-impaired subjects with wide auditory bandwidths are very good in recognizing melodies, and the reverse.

4. CONCLUSIONS

From these experiments the following conclusions can be drawn. The relation between melody recognition and auditory bandwidth is very weak, which means that an essential part of the process that governs melody recognition takes place at a higher level than the peripheral ear. We also checked the musical intelligence of all subjects, owing to daily listening to all sorts of music, and asked them to recognize musical instruments. There is no evidence that more connection with music improves the results of the melody-recognition test. Apart from the result that there is no correlation between melody recognition and auditory bandwidth, it is surprising that there is such a clear separation between the data of the normal-hearing and the hearing-impaired subjects.

Acknowledgement: This study is supported by the Netherlands Organization for the Advancement of Pure Research (ZWO).

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A NEW METHOD FOR RATING THE NOISINESS OF LONG-TERM INTENSITY-FLUCTUATING SOUNDS

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Introduction
In our constantly changing environments we are exposed to sounds of varying intensity and duration such as speech, music or road traffic noise. Subjectively, we tend to generate two kinds of impressions of such noise: the overall impression of noisiness and the short-term impressions at any given moment. In order to develop a clear view of what constitutes noisiness, it seems important to examine the relation between the overall noisiness impression and the short-term noisiness impressions. The author has developed a new method for rating the short-term impressions of a subject exposed to noise of long duration.

The purpose of the experiments described are to (1) confirm the usefulness of this method, (2) determine the most appropriate period that may be labeled 'short-term' in the context of the experiment, (3) confirm the appearance or lack of the so-called 'context effect' in regards to the general category judgments used in this method and (4) make clear the relation between short-term and overall impressions of noisiness.

Experiment 1
In the first experiment five subjects, whose ages were between twenty and thirty years old and with normal hearing ability, were individually exposed to stimuli consisting of artificially generated pink noise. Two kinds of pink-noise stimuli were used (Fig.1); Stimulus A: consisting of fluctuations of wide dynamic range (about 50dB): of 9 minutes total duration, and Stimulus B: consisting of narrow dynamic range fluctuations (about 20dB): of 10 minutes total duration (modulations of the intensity of the stimuli were modeled on previous measurements of road traffic noise\(^2\)). The subjects were asked to rate the overall noisiness of the stimuli. On a following day the subjects were again exposed to the same stimuli, however, simultaneous with the noise, the subjects were given a visual cue in the form of an LED (light-emitting diode) sig-
nalling the subject to judge the noisiness of the stimulus during the LED's lit period. Extinguishment of the LED signalled the subject to indicate his judgment by pressing one of 15 category buttons labeled 'very quiet' through 'very noisy'. A 2-channel tape recorder was used to provide both the visual and auditory stimuli and allowed for synchronization of both. As is shown in Fig.2, four different duty cycles were used to trigger the LED; in Condition 1 the LED is lit for 1 second and off for 3 seconds, in Cond.2 the LED is lit for 2 sec and off for 4 sec, in Cond.3 the LED is lit for 3 sec and off for 3 sec and in Cond.4 the LED is lit for 5 sec and off for 5 sec. In all conditions

![Graphs showing mean category rating vs physical value (Leq)]

$$r = \frac{\sum (x_i - \bar{x})(y_i - \bar{y})}{\sqrt{\sum (x_i - \bar{x})^2 \sum (y_i - \bar{y})^2}}$$

![Table 1]

<table>
<thead>
<tr>
<th>Stimulus A</th>
<th>1sec 2sec 3sec 5sec</th>
<th>Stimulus B</th>
<th>1sec 2sec 3sec 5sec</th>
</tr>
</thead>
<tbody>
<tr>
<td>Preceding $L_{eq(L_{i-1})}: R_i$</td>
<td>$0.941 \quad 0.916 \quad 0.865 \quad 0.716$</td>
<td>$0.489 \quad 0.400 \quad 0.383 \quad 0.437$</td>
<td></td>
</tr>
<tr>
<td>Present $L_{eq(L_i)}: R_i$</td>
<td>$0.963 \quad 0.969 \quad 0.976 \quad 0.975$</td>
<td>$0.759 \quad 0.814 \quad 0.753 \quad 0.743$</td>
<td></td>
</tr>
<tr>
<td>Following $L_{eq(L_{i+1})}: R_i$</td>
<td>$0.956 \quad 0.923 \quad 0.900 \quad 0.814$</td>
<td>$0.601 \quad 0.479 \quad 0.413 \quad 0.372$</td>
<td></td>
</tr>
</tbody>
</table>

Rating

$$R_i$$
(1, 2, 3 and 4) the duty cycles were repeated continuously until termination of the stimulus noise. In addition, each condition consisted of a number of trials wherein the lit cycle of the LED was shifted by 1 pulse width from time 'zero' in order to allow for a full coverage of the noise stimuli after an appropriate number of trials. Upon completing the series of trials in each condition, the subjects were again asked to give an overall (post-trial) rating to the stimuli and, as before, their responses were recorded.

Fig. 3 shows the results of Exp.1. Each plot represents the mean category rating of the five subjects as a function of the $L_{eq}$ of those parts of the stimulus that were judged. High correlation can be seen between the physical values and the psychological values measured using this method of rating. As shown in Table 1, the highest coefficients of correlation can be seen between $L_i$ ($L_{eq}$ of the sound while LED is on) and $R_i$ (noisiness rating). These data show that the subjects judged the noisiness of the part of stimulus indicated by the LED. Among the four conditions of LED duty cycle, Cond.3 (3 sec) in Stimulus A and Cond.2 (2 sec) in Stimulus B show the highest coefficients of correlation between the physical values and the psychological values recorded. It may be concluded that from 2 to 3 seconds is the most appropriate duration for evaluating short-term impressions of intensity-fluctuating sound, such as used in this experiment. These values agree with the value of the Psychological Present as demonstrated by P.Fraisse.

**Experiment 2**

Procedures of this experiment are the same as in Exp.1 except that only the 3 sec LED duty cycle condition was used in accordance with the result of Exp.1. The subjects (same as in Exp.1) were exposed to three kinds of stimuli;
Stimulus C shown in Fig.4 in addition to the Stimuli A and B used in Exp.1. These stimuli have three different intensity-fluctuation ranges each, shifting by 10dB: in the case of Stimulus C, the maximum peak level is 70, 80 or 90dBA. The level is 80, 90 or 100dBA for Stimulus A and 69, 79 or 89dBA for Stimulus B.

The results of Exp.2 are shown in Fig.5. Rating values shown in this figure follow a straight line over three ranges. No context effect can be found. The straight lines in the figure represent the regression lines for each intensity fluctuation range calculated by the least square method. If context effect had arisen, the three lines for each stimulus would not be co-linear but individually offset along the Leq axis. It could be said that having no context effect is one of the merits of this new method.

As an example, Fig.6 shows the distribution of the five subjects' averaged short-term impressions for every 3 sec fraction of Stimulus B-69. Fig.7 shows the relation between the short-term responses to the stimuli (as in Fig.6) and the overall (post-trial) judgments. R10, the upper-end value of the central 80% range of the short-term response distribution correlates well with the overall impression values. However, it can be seen that the average of the short-term impression values is less than the overall impression value even in the case of the post-trial overall impression, although conditioning of the subjects was observed, i.e., the psychological noise value of the post-trial overall impression was lower than the pre-trial overall impression (Fig.8). The author hopes that future experiments will clarify the deeper psychological meanings of the results of these present experiments and lead to a greater understanding of the human mind.

The author is grateful to Professor S.Namba for valuable discussions during this work.

References
AN EMPIRICAL MODEL OF NOISE ANNOYANCE REACTION

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1. Introduction: The concept of noise and annoyance contains subjective elements of evaluation and stimulus processing. These elements are not deducible from physical parameters of the environment. The simple physical relationship between stimulus energy and the elicited sensation is not valid for the complex field situation of environmental noise. Beside the acoustic properties of sound, further variables, such as the non-acoustic traits of source, the physical and social environment, the exposure situation, and especially the person, are to be viewed as independent determinants of the annoyance effect. Annoyance thereby becomes a multidimensional concept. The object of this paper is to test a cognitive-psychological concept of environmental sound processing (Lindvall & Radford). In this concept, annoyance is viewed as a feeling of discomfort, which is generated by expected or attributed negative effects of the agent on the individual. Besides the perceptual experience, the emphasis is laid on the cognitive processing of attributed causal relationships between agent and person.

2. Method: In this paper, sensory and cognitive parameters of annoyance are empirically derived from field data comparable to the Lindvall and Radford-conception; their usefulness will be tested by analysis of relationships to the noise level, and by analysis of the mediating influence of subjective cognitive structure on the different annoyance components and Ld. 274 residents of 8 streets were interviewed, and for the second study 359 residents in 15 samples, at varying distances to the motorways (20-200 m). Noise levels were measured as energy-equivalent soundlevels Ld. Values were 55-60 dB for the streets, and 50-72 dB for the motorways. Standardized questions about sensory experiences as well as the attribution of harmful effects were included in both surveys.

3. Results:
3.1. Empirical delineation of sensory and attributional components of annoyance reaction
The number of independent dimensions of annoyance was established through a factor analysis of the correlations of the items. Factor analysis reduces a number of covarying variables to underlying independent dimensions or aspects. In the street survey there were three of these independent aspects of annoyance. The first was termed K1, a sensory reaction to the irritant, in which the loudness, frequency of occurrence and degree of disturbance is described by the respondents. The second component K2 is defined as a subject-centred experience, in which the disturbance of every-day activities (social, emotional, and somatic) is attributed to the negative effects of the noise. These disturbances include the subjective feeling of anger and irritation, impairment of leisure activities, sleep and rest; the production of negative symptoms such as headache, ear complaints; lack of concentration, and fright reactions. The third component, also in some part subject-centred and attributive, describes disturbances of communication as well as the perceptions of acoustic vibrations and secondary noises e.g. shaking of windows. This splitting of the annoyance reaction into a stimulus-centred component K1 and attributive and subject-centred tendencies was also confirmed by the results of the motorway study. For both K1 and K2, cumulative scales were formed on the basis of the factor-analytical results and their interpretative identification as empirical indicators of the Lindvall & Radford-conception of annoyance. These scales quantified the extent of the component-reactions for each individual.

3.2. Dependency analysis results: The correlations and partial correlations of the annoyance reactions with each other and with L0 are shown in fig.1. High correlations, computed on an individual basis, are found for L0 K1 (0.85) and L0 K2 (0.62). If, however, the influence of the sensory component is eliminated by the computation of a partial correlation r K2 L0;K1, this relation between L0 and K2 approaches zero (r L0 R2 K1 = 0.06). This is not the case for the relationship of L0 to K1, upon which K2 has only very limited influence. This suggests a hierarchical model of the relationship, whereby the sensory experience mediates between the objective input and the evaluative function K2. The latter represents the core of the psychological condition of annoyance. Fig. 2 supplies a further impression of the role which sensory experience plays in the organisation of annoyance. Here each of the 8 samples are divided into two groups, with either above-average (Group A) or below-average (Group B) sensory experiences. The means of each group are represented as a function of L0 and above all show that Group B do not exhibit any substantial tendency towards an L0 level-dependent increase of attributional effect up to the level of 70 dB and above. Psychologically this means that Group B, despite their level-related perceptions, do not convert these into negative attributional cognitions up to the region of 70 dB. Group B therefore, according to Lindvall and Radford's model, is not affected, since they do not display this subjective assessment.
This has the further psychological interpretation that these individuals perceive the noise situation up to the above-mentioned level, but not as environmental conditions from which they could expect harmful effects upon themselves. This suggests the existence of very effective internal-psychological or external-physical control mechanisms, whereby the subjective availability of such mechanisms also leads to a subjective reduction of the negative effects.

3.3. Moderator analysis: In the field of empirical research into environmental noise, there are some well-known subjective moderators, which amplify or attenuate the individual's reaction to noise. These identified moderators include the affective attitude or response of the subjects to traffic noise in general as well as the subject's beliefs in the possible health risks produced by exposure to noise in general. For both of these moderators, individual scales were included in the interview. The data on the emotional reaction to traffic noise in general shows that this is a strong moderator of K 2, but not of K 1. The same is true for the frequently-examined concept of the supposed health-damaging effect of environmental noise. This concept additionally can be proved to be a strong stimulus-independent moderator of the attribution tendency, but not of the sensory experience.

4. Discussion: All in all, these observations suggest that the Lindvall & Radford-conception of the subjective or psychological structure of annoyance is empirically substantiated. Two distinct psychological tendencies seem to develop: K 1, which describes the sensory experience, and K 2, in which the sensory input is processed into the causal cognition of personal harmfulness. Since K 1 exerts an influence on K 2, three possible approaches to noise control can be proposed on the basis of this model:
1. Reduction of the objective noise
2. Measures which reduce the subjective input (eg. double-glazing)
3. Measures in which the subjective or cognitive meaning of sounds is altered through processing and interpretation (eg. traffic noise calming measures, in which traffic noise is relieved of its meaning as a stress-factor in the environment).

Fig. 1: Correlative Relations of $L_D$, $K_1$ and $K_2$ and their Partial correlations.

Fig. 2: Mean values of sensory experience $K_1$ and attributed negative personal effects $K_2$ for individuals with above-average sensory experience within samples (Group A) and below-average sensory experience (Group B).
DIGITALE SYNTHETISATOREN UND IHRE ANWENDUNG IN DER PSYCHOAKUSTIK

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1. Einleitung
Je nach Bedarf kann die Signalerzeugung, der Versuchsablauf sowie Ein- und Ausgabe der Testparameter durch den Prozessor gesteuert werden. Das ermöglicht einfache Versuchsdurchführung, reproduzierbare Versuchsbedingungen und Kontrolle aller Versuchsparameter.

2. Synthese auf digitalen Systemen
Aufgrund zeitkritischer Aspekte bei der digitalen Synthese wendet man zur Erzeugung von komplexen Spektrum im Audio-Bereich neben der additiven Synthese (Fourier Synthese) zumindest ein zusätzliches Modulationsverfahren (Frequenzmodulation, Amplitudendemodulation u.a.) an. Den Aufbau einer Synthesizer-Einheit, die bei Versuchen zur Verfügung stand, zeigt Abb. 1.

2.1 Additive Synthese
Bekanntlich beruht die additive Synthese auf der Tatsache, dass sich periodische Schwingungsvorgänge als Summen von Sinus- bzw. Kosinus-Funktionen darstellen lassen. Beschreibende Parameter für diese Technik sind:
- Frequenzen der einzelnen Spektralanteile sowie deren Zeitverhalten
- Amplituden der einzelnen Spektralanteile sowie deren Zeitverhalten.
Die additive Synthese ist ein wichtiges und vielfältig anwendbares Verfahren bei der digitalen Klangerzeugung. Der Aufbau von komplexen Spektrum in Echtzeit ist jedoch aufgrund der großen Menge notwendiger Oszillatoren begrenzt und kostenaufwendig.

Vorteil von Modulationstechniken dagegen ist es, daß trotz weniger zu kontrollierender Parameter komplexe Spektren in Echtzeit geschaffen werden können.

2.2 Frequenzmodulation

Die Zusammenhänge bei der Frequenzmodulation sind seit geraumer Zeit bekannt. Entsprechend nachfolgendem Ausdruck wird ein Signal \( y(n) \) durch Modulation eines Trägers der Frequenz \( f_T \) mit einem Modulator der Frequenz \( f_M \) erzeugt. Für sinusförmige Signale ergibt sich dann:

\[
y(n) = A(n) \sin \left[ 2\pi n \Delta t f_T + I(n) \sin 2\pi n \Delta t f_M \right]
\] (1)

wobei
- \( A(n) \) Amplitude des modulierten Signals
- \( I(n) \) Modulationsindex
- \( \Delta t \) Zeit zwischen zwei Achtastwerten.


Mittels dieser Methode können:
- harmonische Spektren \((f_M/f_T \text{ oder reziproker Wert ist ganze Zahl})\)
- inharmonische Spektren \((f_M/f_T \text{ ist irrationale Zahl})\)
3. Ein- und Ausgabe von Parametern

Abb. 2 zeigt den Aufbau des verwendeten Computersystems ABLE/40 einschließlich der benutzten Peripherie-Systeme.

Abb. 2: Aufbau des verwendeten Computersystems

Die Programmierung des dargestellten Rechners sowie der Ein- und Ausgabeinheiten erfolgte über die Programmiersprache XPL/4 und einen Superset dieser Sprache MAX. Die drei Teile von MAX erlauben dem Anwender:
1. Kontrolle des digitalen Synthesizers (Frequenz, Wellenform, Frequenzmodulation, Hüllkurven etc.)
2. Kontrolle über Ein- und Ausgabevorgänge der Keyboard-Kontrolleinheit (8x16 Bedienknöpfe, A/D-Wandler, D/A-Wandler, Display etc.)

Die interne Weiterverarbeitung der Daten und Parameter erfolgt durch Programmierung mittels XPL/4.

4. Anwendung

Oft ist es für rundfunktechnische oder musikalische Anwendungen wünschenswert, die Lokalisation von Quellen unterschiedlichen Charakters für den Hörer mehrkanalig zu simulieren, d.h. ein Modell zu schaffen, das sowohl die Klangcharakteristik der Quelle als auch die Volumenhüllkurve im Raum sowie die Frequenz-
verschiebung bei Annäherung bzw. Entfernung der Quelle vom Be-trachter beschreibt. Dieses Modell wurde für eine auf Abb. 3
dargestellte Lautsprecheranordnung realisiert.

Abb. 3: Lautsprecheranordnung bei vierkanaliger Bewegungssimu-
lation von Schallquellen

Folgende Informationen für den Hörer wurden nachgebildet:
- Intensitätsänderung durch Abstandsänderung der Quelle (6 dB
  Verminderung des Schalldruckpegels bei Abstandsverdopplung)
- Winkeländerung durch Änderung der Energieverhältnisse
  zwischen den Lautsprecherpaaren
- Frequenzänderung durch auftretenden Dopplereffekt.
Neben dem beschriebenen Beispiel sind weitere Einsatzmöglich-
keiten bei psychoakustischen Tests oder Simulationen denkbar, bei
denen komplizierte Signalverarbeitungsprozesse oder Versuchs-
verhältnisse realisiert werden müssen.

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3.3

Exploration fonctionnelle subjective et objective. Audiométrie

Subjective and objective functional investigation. Audiometry

Subjektive und objektive Funktionsforschung. Audiometrie
CRITICAL DISCUSSION OF THE NORMAL THRESHOLD OF HEARING BY BONE CONDUCTION

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ABSTRACT: Measurements of the threshold of hearing for stimuli presented via bone conduction have been made by a number of investigators with groups of young adults, in terms of force, displacement and acceleration. Results obtained for measurements of the same quantity differ among investigations. A large part of these differences may arise from physical limitations in the measurement methods. The physical system, i.e.: the head, the driver, and the mechanical coupling between them, can be a significant source of variability. The head is massive; the skull is stiff and curved; and the flesh padding overlying the skull varies greatly among individuals. This paper presents factors affecting measurements in terms both of subject selection and measuring apparatus. Given the problems observed, the force stimulus eliciting threshold sensation is probably the most reliable quantity to measure, and the simplest to calibrate.

APPROPRIATE STIMULUS: Hearing by air conduction can be defined in terms of sound pressure level at some reference plane in the ear canal. Measurement of the sound pressure stimulus can be made directly. In hearing by bone conduction, the stimulus that produces the hearing sensation is the vibration that reaches the otic capsule via the skull. These vibrations cannot be measured directly due to the intervening skin and the locus of the vibrator relative to the otic capsule. The vibration of the skull itself has to be found by some degree of inference. This paper discusses the relationship between external measurements and the actual vibration of the skull.

The situation can be represented by models such as those in Figure 1. The equivalent mechanical network describing transmission of the vibratory stimuli from the mechanical vibrator to the skull is presented in Fig. 1(A) which shows equivalent analogous mechanical and electrical circuits adapted from Corliess and Koidan (1955) and tabulates physical parameters and variances from Corliess, Smith and Magruder (1959). Figure 1(B), from Flottorp and Solberg (1976), approximates the human headbone by a three parameter model fitted to their data over the range from 125 to 6300 Hz, with the electrical equivalent model for their bone vibrator patterned after Weiss (1960).

The options for measuring the external stimulus correspond to measuring $F_0$, the net force applied by the driver at the contact with the skin, or to measuring quantities related to velocity, $v$, at the contact surface. The parameters in the networks, inferred from the respective headbone impe-
dance measurements, are entered in Figure 1. The stimulus from the vibrator
acts through a series element (m_s in 1(A)) representing the mass of flesh
and bone set into motion near the contact point. Figure 1(A) shows two
mechanical or electrical branches essentially in parallel. The branch com-
posed of the high impedance elements R_L and M_L becomes increasingly signi-
ficant as stimulus frequencies are decreased below 500 Hz. Above 500 Hz,
the significant impedance is the branch consisting of k and r. Both 1(A)
and 1(B) represent the appropriate stimulus in the branch containing the
stiffness, k, or its representation, C. However, k itself is a parallel
combination of the stiffness of the skull and padding by the skin. Although
both electrical and mechanical models can be redrawn with added elements,
the division of the signal between the skin compression and the deflection
of the temporal bone can only be inferred. No instrumentation now permits
measurement of that stimulus.

FIGURE 1: SCHEMATIC REPRESENTATION OF BONE CONDUCTION PARAMETERS

(A) Mechanical and electrical (Corliss et al, 1959)

MODAL VALUES

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Foreheads 33 subjects</th>
<th>Mastoids 23 subjects</th>
</tr>
</thead>
<tbody>
<tr>
<td>k, dyn/cm</td>
<td>4.2x10^6</td>
<td>3.8x10^6</td>
</tr>
<tr>
<td>Est. SD</td>
<td>30%</td>
<td>40%</td>
</tr>
<tr>
<td>m_s, gm</td>
<td>1.7</td>
<td>1.3</td>
</tr>
<tr>
<td>Est. SD</td>
<td>45%</td>
<td>45%</td>
</tr>
<tr>
<td>r, dyn-sec/cm</td>
<td>3.2x10^4</td>
<td>3.7x10^4</td>
</tr>
<tr>
<td>Est. SD</td>
<td>20%</td>
<td>20%</td>
</tr>
</tbody>
</table>

Heads, All Subjects

| Parameter | | |
|-----------|----------------|
| M_L, gm   | 2.4x10^3 |
| Est. SD   | 20%      |
| R_L, dyn-sec/cm | 3.4x10^5|
| Est. SD   | 25%      |

(B) Electrical, vibrator driving headbone (Flottorp and Solberg, 1976)
DRIVER AND HEADBONE IMPEDANCE: At frequencies below 500 Hz, the reactance magnitude of $m_0$ is very small compared to that of $k$ (Cf. 1(A).) so that the stiffness reactance dominates. The external force measured at the surface of the skin is related directly to the force $F_T$ applied at the surface of the skull. As shown in 1(A) $F_T$ is approximated by $F_T'$. At frequencies below 250 Hz the parallel resonance of $M$ and $k$, and the consequent circulating current, not considered in 1(B), will also affect the net force transmitted. This resonance further increases the mechanical impedance presented to the driver. This increase represents a significant constraint upon the internal impedance (Thévenin equivalent) of the driver, which ideally should exceed the head load by at least an order of magnitude.

These requirements led Corliss and Koidan (1955) to mount their rigid driver against a massive masonry pillar. Corliss, Smith and Magruder (1959) used a pendulum-mounted 100-Kg rigid driver. Khanna, Tonndorf and Queller (1976) used a massive and stiff magnetostrictive driver. At frequencies above 500 Hz the requirement for high driver impedance becomes less stringent, and ordinary audiometric and hearing-aid types of bone vibrators become practical sources.

The model in 1(B) indicates a second-order problem that can become significant at high frequencies. A force-measuring pickup inserted between the driver and the headbone also has an inherent mass that may prove comparable to the series high-frequency reactance presented by the mass of flesh and skull deflected by the driver. It is shown in 1(B) as the driver mass in series with $M$. To avoid the influence of this element, Corliss and Koidan and Corliss et al. used a balanced accelerometer for no-load compensation, and Queller and Khanna (1982) subtracted out the output of the unloaded pickup.

An additional problem can be seen from 1(A). At high frequencies, above 1500 Hz, the series mass $m_0$ becomes important because it decreases to some degree the net force driving the skull. Moreover, this mass resonates with the combined stiffness of the skull and flesh to produce a damped series resonance in the vicinity of 3 kHz. In this frequency region, neither velocity nor force alone at the contact point represents the applied stimulus with great accuracy. Above the series resonance frequency, the velocity is controlled by the increasing inertial reactance of $m_0$; $x$ becomes larger than the diminishing reactance of $k$; and the observed velocity is proportional to the force, $F_T$, imparted to the skull. Thus, at octaves above the 3 kHz resonance, at 6 kHz and above, velocity at the contact point of driver and skin will be more nearly representative of the stimulus applied at the skull.

Therefore, neither contact force nor functions of tip velocity alone give a truly representative measure of the internal stimulus presented via bone conduction. The discussion above assumed good contact between the driver and the head. There are wide variations among subjects in curvature of the head. This produces variability in the area of contact. Queller and Khanna studied the effect of contact area in detail. They found threshold forces to be nearly independent of contact area. Larger variations were found when threshold of hearing by bone conduction was measured by acceleration (Khanna et al.). A combined measurement that extracts the parameters for each individual over the test frequency range would permit closer inference of the internal stimulus. In fact, few direct measurements of even the externally applied stimulus have been made. In addition to those cited above, there have been recent studies by Whittle (1965) and
by Richter and Brinkmann (1981), and earlier studies by Barany, von Békésy, Hawley and Watson.

PRESENT STATUS: Most measurements of threshold norms have been indirect, as in ordinary bone conduction audiometry. In these, a bone-conduction vibrator is calibrated on an artificial mastoid designed to represent a prototype human head. Using this indirect threshold measurement, Dirks, Lybarger, Olsen and Billings (1976) carried out air- and bone-conduction audiometry on a group of normal subjects, relating their thresholds to the norms developed by ANST/ISO, relative to the American National Standards Institute Standard for Audiometers (ANSI S3.6-1969). Then they measured threshold elevations by both air- and bone-conduction for nearly two hundred persons clinically diagnosed as having pure sensorineural impairments. The threshold elevations as observed by air conduction in terms of coupler pressures and in terms of force calibrations of the bone vibrators were very closely correlated for the mean of the group over the range from 250 to 4000 Hz.

CONCLUSION: There is a variety of choices for measurement of hearing by bone conduction. The most precise is to ascertain as closely as possible the internal stimulus producing hearing sensation. This requires precise measurement simultaneously of force and motion thresholds, and extraction of mechanical parameters for each individual. The next lower level of precision is reached by direct measurement of the threshold force at the contact with the skin. The indirect force threshold (calibrated by an artificial mastoid) is still less precise. Least direct, and most limited in frequency range, is the acceleration at threshold for the contact tip.

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SPEECH LEVEL STANDARDISATION IN AUDIOMETRY

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Standard procedures for pure tone air and bone conduction audiometry are now well established, but there is still no standard procedure for conducting speech audiometry. Recent interest in the field has led to a number of methodological studies and to the development of new test procedures, yet the most basic problem encountered in the field is the one which has received the least attention. There is, at the moment, no standard definition of the speech level.

It is a fundamental assumption of all speech testing that it is possible to measure stimulus levels in a way which reflects their subjective impact. In both the clinical and research fields it is essential that a common basis be developed for the measurement of speech level. This is far from simple, however. Fig 1 shows a plot of amplitude against time for a typical target word, "SHIFT". The four phonemic elements of the word are clearly seen: the (f) at a moderate level and lasting 180ms; the vowel (i) with a very fast rise-time, lasting for 100 ms and dominating the overall level; the (f) - low in level but lasting for 300 ms; and then after a silent pause, the tiny burst of the final (t) which lasts for only 60 ms. The difficulty here is the very basic one of how to make an appropriate acoustical measurement of the signal, that is, how to find a measure which will reflect the different features of the stimuli and enable them to be equalised so as to reflect the perceptual thresholds of the normal-hearing person.

![Diagram of speech level vs time for the word "SHIFT".](image-url)
There are two perceptual thresholds commonly used for speech material - the speech detection threshold (SDT) and the speech reception threshold (SRT). As a first step towards the standardising of speech level in audiology, an investigation was made of the best way to set the levels for the SDT, defined as the level at which 50% of the test material can just be detected as being present without being understood. The SDT was chosen because it is the simpler of the two thresholds, being largely unaffected by the linguistic and phonetic features involved in the SRT (the threshold for the correct perception of the stimuli). Using normal hearing subjects our aims were as follows:

1) To establish the SDT for a sample of monosyllabic test words
2) To find the objective measure which gave the best prediction of the mean SDT for each word.

Speech material

The test words used were taken from a recording of the Medical Research Council speech audiology word lists made by a professional radio announcer (1). Two lists were used (3/1M1 and 3/1M2), each consisting of 25 monosyllabic words, giving a total of 50 different test words. New tape recordings were constructed in which each word was repeated at 0.5 s intervals for a period of 45 s, with a pause of 15 s between words. In this way four tapes were made of each of the lists, using different random orders of the words. A continuous 1 kHz calibration tone was added to each tape, together with two 45 s periods of 1 kHz pure tone pulsed twice per second, one before and the other after the word blocks.

Method

24 normal-hearing, audiometrically experienced subjects (10m, 14f) were tested in a room where the ambient noise was well below the normal free-field threshold of hearing. The tapes were replayed via a pair of TDH 39, MX-41/AR earphones in a standard headband with only the right ear energised. Subjects used a Bekesy technique, with the 45 s repeated presentation of each word as the signal and a 5 dB/s attenuation rate, to set the SDT for each word. Using a balanced design throughout, subjects made four visits, during each of which they were tested on one recording of each word list and completed a pure-tone hearing test. The pulsed 1 kHz tone on the tapes enabled the results from the recording to be linked to the pure-tone audiograms.

A wide range of objective measurements was taken, using the sound pressure developed by the earphone in an IEC reference coupler where applicable. These varied from standard instrumentation (V.U. meter, peak programme meter, rms (fast), maximum peak (fast) and impulse sound level) to measures computed from the digitized waveform (Lmax, Leq and the total energy measure Lx). In each case the relative levels of the words and the calibration tone were measured, giving different results for the different objective measures.

Results and discussion

Using the attenuator setting to establish the SDT, the subjective results showed that the two word lists were equivalent, the mean threshold setting being 13.8 dB for list 3/1M1 and 13.7 dB for list 3/1M2. However, the individual words in each list varied considerably in the difficulty
which subjects found in setting their threshold. Some words gave much better repeatability than others, with a range of 6 dB in the mean intra-subject standard deviation for the different words. The range of settings for the total sample of words was approximately 15 dB (as compared with a range of 13.6 db in the means of the pure-tone thresholds measured at 0.5, 1 and 2 kHz).

The mean SDT for all subjects and all replications was calculated for each of the 50 words and a linear regression was performed against each of the measures in turn. Some of the measures used are shown in the table. The most commonly used method of setting word levels is probably the V.U. meter and the results show that with a linear weighting, this gave a correlation of 0.7, with only marginal improvements as meters with faster response times were used.

<table>
<thead>
<tr>
<th>Weighting Network</th>
<th>Test Measure</th>
<th>Correlation</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Linear</strong></td>
<td>P.P.M.</td>
<td>0.64</td>
</tr>
<tr>
<td></td>
<td>Impulse (hold)</td>
<td>0.70</td>
</tr>
<tr>
<td></td>
<td>V.U.</td>
<td>0.70</td>
</tr>
<tr>
<td></td>
<td>RMS (fast)</td>
<td>0.72</td>
</tr>
<tr>
<td></td>
<td>L_{eq} (10 or 30 ms)</td>
<td>0.75</td>
</tr>
<tr>
<td></td>
<td>L_X (10 or 30 ms)</td>
<td>0.77</td>
</tr>
<tr>
<td><strong>A-weighted</strong></td>
<td>V.U.</td>
<td>0.92</td>
</tr>
<tr>
<td></td>
<td>RMS (fast)</td>
<td>0.93</td>
</tr>
<tr>
<td></td>
<td>L_{eq} (10 or 30 ms)</td>
<td>0.94</td>
</tr>
<tr>
<td></td>
<td>L_X (10 or 30 ms)</td>
<td>0.94</td>
</tr>
<tr>
<td></td>
<td>P.P.M.</td>
<td>0.94</td>
</tr>
<tr>
<td></td>
<td>Impulse (hold)</td>
<td>0.96</td>
</tr>
<tr>
<td><strong>Earphone response matched to threshold curve</strong></td>
<td>Impulse (hold)</td>
<td>0.98</td>
</tr>
</tbody>
</table>

Correlations between objective measures and subjective thresholds.

The use of A-weighting before the level measurement, however, produced correlations of better than 0.9. A further slight improvement was possible by using a special frequency weighting network which matched the response of the earphone to the RETSPL (reference equivalent threshold sound pressure level). This resulted in a correlation of 0.98.

The inclusion of the 1 kHz tone on the test tapes and 1 kHz as one of the audiometric frequencies made it possible to find the relationship between the SDT and the pure tone threshold. Using the A-weighted impulse hold measure of the level of the words, the mean SDT was 8.8 dB above the 1 kHz threshold. At 1 kHz the mean threshold was 4.1 dB SPL. Combining these two results gives a value of 12.7 dB SPL for the monaural speech detection threshold for this group of 24 subjects.

Fig 2 shows the result of plotting the mean SDT for each word against L_{AI} - measured, for ease of reading, with the 'hold' circuit. This measure
is the one which gave marginally the best results of the methods using standard instrumentation, but the advantages of using the hold facility may be outweighed by the response characteristics of the measure. All of the A-weighted measures shown in the table gave a very good agreement with the subjective response, and there is little to recommend one above the others. It is to be expected that the SDT will be dominated by the level of the vowel, and it is interesting to note in fig 2 that the words cluster according to the place of articulation of the vowels.

The problem of measuring the speech level in such a way that it reflects the SRT is far more complicated than that of the SDT, for the SRT relies on the correct perception of all the phonemic elements of the words. A simple measure which is dominated by the level of the vowel can only ensure that the stimuli are roughly equalised for the SRT. In addition to problems of measurement indicated by fig 1, the frequencies present in the words may be important. There is a tendency for formant frequencies to be affected by the following phonemes in complete words and for these changes to carry information. Thus simple level measurements, even if these are made on the separate elements of the words, are unlikely to be sufficient to predict the SRT well.

Fry (2) suggested that instead of trying to equalise the words acoustically, a trained speaker could produce the words with "equal vocal effort" and he used this method to some effect. As speakers, we automatically emphasise the more easily confusable elements of a message and it may well be that this is yet another field in which we shall find it very difficult to measure and reproduce what a person does naturally.

References

1. LYREGAARD, P.E., ROBINSON, D.W. and HINCHCLIFFE, R. 1976
A feasibility study of diagnostic speech audiometry. NPL Acoustics Report AC73, Teddington, UK.

OCCUPATIONAL HEARING CONSERVATION - COMPUTERIZED DATA ANALYSIS AND REPORTING SYSTEM

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A computerized hearing analysis program for occupational hearing conservation data is presented. The program is written to conform to the United States Occupational Safety and Health Act (OSHA), Hearing Conservation Amendment (29 CFR Part 1910).

Sections highlighted are; 1) computerized reporting of baseline and monitor audiometric threshold data; 2) identification of significant permanent threshold shifts (PTS); and 3) management follow-through program.
3.4

Effets du bruit, de l’environnement, des drogues, synergie
Prévention - Protection
Effets extra-auditifs - Coût social

Effects of noise, environment, drugs, synergy
Prevention - Protection
Extra-auditory effects - Social cost

Einwirkung von Lärm, Umwelt, Medikamenten, Synergie
Vorbeugung - Gehörschutz
Sonder-Höreffekte - Sozialkosten
APPROCHE DU COUT SOCIAL DU BRUIT EN MILIEU INDUSTRIEL

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Les motivations de l'étude

Les stratégies de lutte contre le bruit en milieu industriel sont aujourdhui bien connues, mais les réticences pour leur mise en oeuvre sont fréquentes. Dans la majorité des cas ce sont des considérations économiques qui s'opposent à la mise en oeuvre d'une politique réaliste de lutte contre le bruit, notamment cette dernière nécessite des investissements importants. En effet, la rentabilisation de ces investissements n'apparaît pas de façon claire comme c'est le cas par exemple pour des investissements en matière d'isolation thermique. Si cette rentabilisation n'apparaît pas c'est sans doute parce que le coût du bruit n'est pas immédiatement traduisible en termes monétaires mais aussi parce qu'une partie de ce coût est "externée" vis-à-vis de l'entreprise, en fait il s'agit d'un coût social qui n'a pas encore fait l'objet d'évaluations réalistes.

Les dommages et les coûts entraînés par le bruit

Une longue expérience du Laboratoire en matière de bruit dans les entreprises montre que lorsqu'en milieu industriel, des niveaux sonores Leq A = 85dB sont atteints ou dépassés, on peut généralement s'attendre aux conséquences suivantes :
- apparition de surdités professionnelles
- aggravation de l'absentéisme et du turn-over
- aggravation d'un certain nombre d'accidents non auditifs
- effets à long terme sur la santé des personnes exposées
- climat favorable à l'apparition de conflits sociaux
- diminution du rendement et/ou de la qualité du travail
- répercussions sur la vie individuelle et familiale.

Ces diverses conséquences du bruit entraînent un certain nombre de coûts qui sont diversement imputables, certains peuvent s'exprimer facilement en termes monétaires comme le coût de "réparation" de la maladie professionnelle, d'autres non, comme l'altération de la qualité de la vie ; enfin, certains coûts sont supportés par l'Entreprise, d'autres par la collectivité, certains le seront par l'individu.

C'est cet ensemble de conséquences que nous avons tenté d'évaluer en les incluant dans la notion de coût social.
La méthodologie d'approche

Il n'est pas facile d'appréhender point par point les conséquences évoquées ci-dessus, nous avons retenu deux axes d'investigation :
- l'un monétaire : le coût de la surdité professionnelle "réparée"
- l'autre non monétaire : l'évaluation des effets du bruit sur l'homme au travail à l'aide d'une enquête psychosociale.

Le premier axe de Recherche a été exploré à l'aide des dossiers de la Caisse Régionale d'Assurance Maladie de notre région sur une période de cinq ans. Le second axe a été exploré sur le terrain offert par des entreprises régionales qui ont bien voulu se prêter aussi bien à l'enquête psychosociale qu'à l'enquête sonométrique.

Le coût de réparation de la maladie professionnelle

L'échantillon est relatif à la période 1972-1977, il comportait 53 dossiers (uniquement des hommes), les caractéristiques de l'échantillon étaient les suivantes :
- âge moyen : 53 ans ; - âge moyen dans l'entreprise : 30 ans
- domaine d'activité : - travail des métaux : 68 % ; - textile 6 %
- divers connus : 13 % ; - inconnus : 15 %.
- déficit audiométrique moyen : 51 dB ; - taux moyen d'IPP : 25 %.

Les caractéristiques de cet échantillon appartiennent de nombreux commentaires et le nombre de maladies professionnelles "réparées" ne constitue à coup sûr qu'un faible pourcentage du nombre des surdités réelles professionnellement contractées.

Le calcul du coût monétaire est relativement aisé, il fait appel à l'âge moyen des déclarants, à l'espérance de vie dans leur catégorie socioprofessionnelle, au taux moyen d'IPP et au salaire de base, on ajoute le coût de diagnostic et de "suivi" médical, celui des journées perdues et celui d'une prothèse qui est généralement adoptée (et rarement retenue de façon définitive).

Exprimé en Francs au moment de l'étude (1977) ce coût de réparation oscille entre 100 000 et 120 000 F. Quelques années plus tard, notre estimation a été très bien confirmée par la Sécurité Sociale qui en avait réalisé le calcul moyen pour la même époque mais "à posteriori" et par une approche tout à fait différente.

L'enquête psychosociale

Elle avait pour but de connaître comment le bruit est perçu en tant que nuisance au poste de travail et quelles en sont les conséquences individuelles. Pour cela un questionnaire a été distribué à 2 000 exemplaires dans 18 établissements industriels de notre région appartenant à 14 sociétés. Le questionnaire était volontairement très simple destiné à être rempli sans le secours d'un enquêteur, il comportait 22 questions portant sur :
- l'identification du sujet (âge, sexe, qualification...etc)
- l'autoévaluation de la santé
- la perception du bruit parmi d'autres facteurs d'ambiance
- des éléments d'appréciation de l'ambiance au poste de travail
- des questions "masquées" destinées à éviter la focalisation du sujet sur le bruit.

L'anonymat du questionnaire a conduit à un taux élevé de réponses (982), par contre, il nous a empêché d'afecter ces réponses à des postes
Nous retiendrons les réponses à deux questions :
- qu'est-ce qui est pénible après un arrêt ?
- le travail posté : 10 % ; — la difficulté de la tâche : 3 % ;
- le bruit : 49 % ; — les odeurs : 4 % ; — la cadence : 8 % ;
- la répétition : 6 % ; — autres éléments : 4 % ; — réponses non
exploitables : 16 %.
- quel sont les facteurs qui vous paraissent dangereux ?
- chaleur : 9 % ; — poussière : 20 % ; — éclairage : 7 % ;
- bruit : 51 % ; — autres facteurs : 3 % ; — réponses non exploita-
bles : 10 %.

Un coût individuel important mais non monétaire apparaît clairement
à travers ces réponses.

Conclusion
Nous avons appréhendé une des composantes monétaires et une des com-
posantes non monétaires du coût social du bruit en milieu industriel, il
manque encore des composantes d'accès difficile : par exemple l'aggravation
des accidents non auditifs dus au bruit. De nouveaux travaux qui pourraient
intéresser ceux qui se penchent sur l'économie des conditions de travail,
pourraient nous apporter des éléments complémentaires pour justifier les
investissements nécessaires en vue de réduire l'exposition des travailleurs
au bruit.

Références
L'étude dont est tiré le résumé ci-dessus a fait l'objet d'un contrat
de Recherche n° 27301760013 avec le Ministre de l'Environnement qui dispose
du compte rendu complet. Mission Etudes et Recherches, 14 Bd du Général
Leclerc, 92524 NEUILLY SUR SEINE.
L'exposition au bruit des personnes visées par l'enquête.

A Effectif en nombre exposé à un niveau continu équivalent donné par classes de 3 dB (40 heures par semaines)

B Pourcentage de la population exposée en fonction du niveau sonore. 50 % de l'effectif était exposé à $L_{eq} > 85$ dB.

L'enquête sonométrique a été réalisée de façon discrète pour éviter toute sensibilisation au bruit qui se serait répercuté sur l'enquête.
A PROPOS D'UNE ETUDE EPIDEMIOLOGIQUE POUR L'EVALUATION COMPARATIVE DES EFFETS DU Bruit SPECIFIQUES ET EXTRA-AUDITIFS CHEZ DES TRAVAILLEURS EN MILIEU INDUSTRIEL.

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INTRODUCTION

Nous avons été appelés dans nos industries dès 1956 à évaluer la nuisance bruit de certaines postes de travail dans les centrales électriques thermiques et hydrauliques d'E.D.F. et dans les postes de compression et détente du gaz de G.D.F., de même que pour de rares ateliers de forge ou de chaudronnerie, ou sur des chantiers.

Les médecins du travail locaux et un médecin détaché au Comité d'Etudes des emplois actifs et insalubres participent à la mesure des niveaux d'insalubrité des postes, à partir des cartographies de bruit, en particulier, et à l'évaluation des temps équivalents expriment le dose de bruit reçu, à partir des temps de passage aux différents lieux de travail.

Parallèlement, la courbe audiométrique tonale aérienne des agents exposés au bruit, à partir de 85 dBA, a fait l'objet d'une surveillance régulière dès avant l'arrêté ministériel de 1976 qui nous en a fait obligation.

Nous avons mis au point au S.G.M.I. une classification des différents stades de perte audiométrique par traumatisme sonore à partir de la moyenne des seuils aux fréquences 500 - 1 000 - 2 000 Hz (déficit moyen défini dans la réglementation française au tableau 42 des maladies professionnelles).

CLASSIFICATION DES TYPES AUDIOMÉTRIQUES

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
<th>(IPA)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type I</td>
<td>Audition normale ou subnormale.</td>
<td>&lt; 10%</td>
</tr>
<tr>
<td>Type II</td>
<td>Surdité de transmission.</td>
<td>&lt; 22%</td>
</tr>
<tr>
<td>Type III A</td>
<td>Surdité de perception légère caractérisée par un déficit à 4 000 ou 6 000 Hz, type traumatisme sonore.</td>
<td>&lt; 10%</td>
</tr>
<tr>
<td>Type III B</td>
<td>Surdité de perception plus importante que III A, 2e stade du traumatisme sonore.</td>
<td>&lt; 10%</td>
</tr>
<tr>
<td>Type IV A</td>
<td>Surdité de perception profonde. Scotome élargi aux fréquences conversationnelles. Sujets à retirer du bruit autant que possible.</td>
<td>10% &lt; IPA &lt; 22%</td>
</tr>
<tr>
<td>Type IV B</td>
<td>Surdité mixte ou surdité de perception importante, zone des fréquences conversationnelles très atteinte, pouvant être considérée comme maladie professionnelle.</td>
<td>&gt; 22%</td>
</tr>
</tbody>
</table>

Un premier traitement statistique des données audiométriques tenant compte des données somatométriques et des antécédents médicaux des sujets a permis d'établir un profil audiométrique d'une population de référence par tranche d'âge.

La 1ère population de référence choisie pour base comparative de l'étude était donc non exposée au bruit.


Restait à compléter ce protocole par un questionnaire médical de surveillance des autres systèmes que celui de l'audition, c'est-à-dire le système - cardio-vasculaire,
- respiratoire,
- digestif,
- neurologique.

Et c'est ce qui a fait l'objet du protocole d'enquête épidémiologique tel qu'il a été accepté par le Ministère de l'Environnement.
PRINCIPE DU RECUEIL DES DONNÉES

Etant donné l'importance de la population de travailleurs à F.D.F.-G.D.F. (130 000 agents), nous allons recueillir plusieurs milliers de dossiers comportant :
- un examen audiométrique comprenant un questionnaire complet,
- une partie administrative visant à reconstituer, par le développement de carrière, l'exposition globale au bruit dans le temps, en fonction des postes de travail,
- une partie médicale recherchant tous les antécédents pathologiques, non seulement otologiques, mais des autres systèmes de l'organisme : cardio-vasculaire, respiratoire, neurologique qui peuvent être le siège de troubles non spécifiques dus à l'exposition au bruit, ou de troubles d'une autre origine, mais révélant une sensibilité individuelle particulière qui pourrait faire apprécier une sensibilité individuelle plus grande pour le bruit.

Cette enquête comporte deux stades et le questionnaire est double :
- 1 feuillet A -selon l'enquête transversale- de recueil de données.
- 1 feuillet B -sur le même plan- mais allégé, qui devra permettre de noter les signes d'évolution ou les nouveaux symptômes apparus, c'est-à-dire la 2ème partie longitudinale de l'enquête.

Ces données seront traitées selon deux modes :

1°) Pour la première enquête transversale, le traitement consiste à relier chacun des signes cliniques à une éventuelle modification de l'audiogramme et aux doses de bruit reçues.
En analysant variable par variable, selon une analyse factorielle des correspondances entre les deux groupes de variables audiométriques et cliniques.

2°) Pour le suivi des données longitudinales, le traitement est aussi de multivariables, mais cette fois des variations de la valeur de chacune des variables aux instants t1 - t2...

DIFFICULTÉS PRATIQUES DANS LE RECUEIL DES DONNÉES

La population de travailleurs intéressée par cette enquête comporte deux catégories.
La 1ère : celle des nouveaux embauchés.

Nous n’aurons au départ que peu d’informations -excepté les antécédents médicaux- puisqu’il s’agit de sujets jeunes dont l’exposition au bruit est en principe courte, mais les données de départ sonométriques, audiométriques et médicales pourront être précises. Les effets et donc les résultats seront plus longs à dégager.

La 2ème catégorie : celle des agents travaillant depuis de nombreuses années dans le bruit.

La reconstitution de carrière de ces agents est souvent difficile à établir et les doses de bruit reçues dès l’origine sont difficiles à apprécier ; certaines postes de travail n’existent plus.

Et enfin, il y a une interférence et une interaction de plusieurs facteurs d’origine des troubles extra-auditifs recherchés :

- travailleurs exposés au bruit et travaillant en horaire 3 × 8, par exemple.

Par contre, la perte auditive en fonction de l’âge pourra être évaluée et déduite, à partir de notre population de référence.

Une autre difficulté pratique réside dans le contrôle des questionnaires remplis. Ce travail s’avère lourd et complexe ; d’une part, nous nous attachons avec une très grande rigueur à ce qu’aucune identification des agents ne puisse être faite en dehors du service médical. D’autre part, l’expérience prouve que, malgré toutes les indications préparatoires fournies en même temps, comme un spécimen de questionnaire rempli, les erreurs, omissions et ombres (même de la part des sujets surveillés) restent fréquents.

Ces lacunes imposent un retour en arrière pour complément d’information, et par conséquent un freinage important dans le déroulement de l’enquête.

CONCLUSION

Sur les plus de 1 000 questionnaires déjà recueillis, un premier traitement informatique permet de laisser présumer une liaison probable entre les variables audiométriques et cliniques, mais le point actuel d’avancement des travaux ne permet pas encore d’en dégager des constantes précises, ni la proportionnalité.
PROTOCOLE D'ÉVALUATION DE LA PREVALENCE D'ATEINTES AUDITIVES DUES AU BRUIT A L'ÉCHELLE D'UN ÉTABLISSEMENT INDUSTRIEL

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Problématique

Un programme de prévention de la surdité professionnelle, lorsqu'il est développé dans une perspective de santé publique, comporte plusieurs phases et suppose l'application de plusieurs outils d'intervention tel que l'illustré la Figure 1. Ainsi, l'échantillonnage de l'environnement de travail et la quantification des risques d'atteinte à l'audition constituent la base des décisions importantes en matière d'implantation de mesures correctives.

Mais, on doit fréquemment recourir à une information complémentaire, soit l'évaluation rétrospective de l'état de l'audition de la population exposée au bruit. C'est le cas notamment lorsque la présence de bruits nocifs a été échantillonnée mais les informations obtenues ne suffisent pas à justifier l'élimination ou le contrôle des risques ; ou encore lorsque les dangers potentiels de certaines conditions d'exposition au bruit ne sont que très peu ou pas connus (e.g. présence de bruits d'impacts ou présence d'agents potentialisants).

C'est dans ce contexte qu'a été mis au point le Protocole d'Évaluation Rétrospective de l'audition d'une Population Exposée au Bruit Industriel. Il s'agit d'un ensemble complet de procédures d'exams, d'analyse numérique des données, de références médicales et professionnelles, élaboré dans la perspective d'améliorer, à l'échelle de l'entreprise, la salubrité du milieu de travail au plan des ambiances sonores.

Principes méthodologiques

Pour évaluer de façon rigoureuse la prévalence de pertes auditives dues à l'exposition professionnelle au bruit, la démarche suivante a été adoptée :
- une procédure d'examen audiométrique de dépistage hautement valide a d'abord été définie sur la base du contrôle rigoureux des différentes sources d'erreurs de mesure
- on recourt de plus à un test impédancémétrique de dépistage permettant d'identifier la contribution éventuelle de pathologies de l'oreille moyenne
- une histoire auditive du travailleur est dressée au moyen d'un questionnaire informatisé
- une grille informatisée d'analyse à la fois épidémiologique et clinique des résultats d'exams permet un traitement individuel et statistique des
Hétu, R. et al., Protocole d'évaluation de la prévalence d'atteintes auditives

données
- les résultats prennent enfin la forme de bilans individuels et d'un bilan collectif établi à l'échelle de l'entreprise cible.

La procédure d'analyse des résultats d'examens est fondée sur des principes d'exclusion des facteurs parasites dans l'évaluation de la relation bruit - surdité à l'échelle d'une population. Cette analyse procède en quatre étapes:

1° on vérifie d'abord la validité des résultats individuels d'après des critères pré-établis
2° puisqu'il est impossible, dans le contexte d'un examen de dépistage, de distinguer la contribution relative d'une maladie de l'oreille des effets de l'exposition au bruit industriel, on identifie au moyen d'une grille numérique les résultats d'examens qui témoignent de problèmes d'audition qui ne sont pas attribuables aux seuls effets du bruit industriel et du vieillissement normal. Ces résultats sont traités séparément et au besoin, donnent lieu à des références appropriées en cliniques spécialisées.
3° dans la population restante, on identifie ensuite la proportion d'individus qui montrent des pertes auditives dont l'ampleur est hautement improbable en termes d'effets du vieillissement normal (perte au-delà du 90e percentile de l'effet de l'âge à chaque fréquence audiométrique entre 500 et 6000 Hz). Cette proportion représente la prévalence de Pertes Auditives Significatives (P.A.S.) imputables aux effets du bruit industriel.
4° cette population fait alors l'objet d'une analyse au niveau de la relation entre ancienneté d'exposition au bruit de l'usine considérée et prévalence de P.A.S. Le nombre de fréquences audiométriques affectées de P.A.S. sert de critère de sévérité des atteintes à l'audition.

Exemple d'Application du Protocole

Une banque de données est déjà disponible, le protocole ayant été utilisé à ce jour auprès de plus de 2500 travailleurs provenant d'une quinzaine d'entreprises bruyantes.

À titre d'exemple, considérons les résultats obtenus dans une acierie. Parmi les 298 travailleurs examinés, 199 ont montré des résultats qui ne témoignaient pas des effets d'une maladie de l'oreille ni de ceux d'exposition au bruit dans un autre milieu. Comme le montre le tableau 1, près de 40% de cette population souffrait d'une P.A.S. à une ou à plusieurs fréquences audiométriques. Mais, seulement 3% de la population souffraient d'une perte suffisamment importante pour satisfaire la définition médico-légale de la surdité professionnelle en vigueur au Québec (i.e. 25 dB sur la moyenne des seuils à 500, 1000 et 2000 Hz).

D'autre part, comme le montre le Tableau 2, la prévalence d'atteintes auditives définies en termes de P.A.S. reflète bien l'effet de la dose cumulative de bruit telle qu'évaluée pour toute la durée de l'emploi à l'acierie (dose établie sur une base annuelle de 2000 heures d'exposition).

Ainsi, le protocole constitue un moyen de dresser un portrait valide des méfaits du bruit sur l'audition des travailleurs d'une entreprise donnée. Il offre une base rigoureuse de quantification des risques biologiques liés à l'exposition au bruit industriel et par conséquent, un outil d'intervention pour favoriser l'implantation de mesures correctives dans le milieu de travail. Le protocole peut également être avantageusement
mis à contribution dans l'identification de secteurs industriels prioritaires pour l'intervention en prévention de la surdité professionnelle.

Tableau 1. Répartition des travailleurs en fonction de la sévérité de la perte d'audition due au bruit défini en termes de P.A.S.

<table>
<thead>
<tr>
<th>Audition normale</th>
<th>P.A.S. à 1 ou 2 fréquences</th>
<th>P.A.S. à 3 fréquences</th>
<th>P.A.S. à 4 fréquences ou plus</th>
<th>Surdité indemnisable</th>
</tr>
</thead>
<tbody>
<tr>
<td>Nombre de travailleurs</td>
<td>123</td>
<td>46</td>
<td>14</td>
<td>10</td>
</tr>
<tr>
<td>Pourcentage de cette population</td>
<td>62</td>
<td>23</td>
<td>7</td>
<td>5</td>
</tr>
</tbody>
</table>

Tableau 2. Relation entre la dose cumulative de bruit (L_{eq}) et le pourcentage d'individus atteints de P.A.S. dans une acierie

<table>
<thead>
<tr>
<th>L_{eq} (dBA)</th>
<th>Nombre total de travailleurs examinés</th>
<th>Pourcentage atteints de P.A.S. (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>moins de 91</td>
<td>14</td>
<td>0</td>
</tr>
<tr>
<td>91 - 95</td>
<td>20</td>
<td>30</td>
</tr>
<tr>
<td>96 - 100</td>
<td>71</td>
<td>26</td>
</tr>
<tr>
<td>101 - 105</td>
<td>64</td>
<td>50</td>
</tr>
<tr>
<td>106 - 114</td>
<td>30</td>
<td>63</td>
</tr>
</tbody>
</table>
Figure 1. Organigramme démontrant les principales étapes de l'implantation d'un programme de prévention de la surdité professionnelle.
METHODOLOGIES D'ÉVALUATION DU NOMBRE DE TRAVAILLEURS EXPOSÉS AUX BRUITS INDUSTRIELS

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Différentes évaluations du nombre de travailleurs exposés aux bruits ont été effectuées dans le monde depuis 1970. Les méthodes utilisées pour 8 estimations réalisées aux U.S.A., aux Pays-Bas et en France sont présentées en distinguant quatre caractères principaux :

* l'objectif poursuivi : outre la connaissance du nombre de travailleurs exposés, il peut inclure - ou non - le recueil de données sur le contexte de l'exposition. Par ailleurs, cette évaluation peut s'insérer dans des recherches sur le coût social du bruit ou sur les conditions de travail ; elle peut être limitée à un type particulier de bruit.

* l'étendue du champ étudié : elle peut être vaste - toute l'industrie d'un pays par exemple - ou volontairement limitée - un secteur industriel ou une région.

* l'échantillonnage de la population : il est effectué par une méthode empirique ou par tirage aléatoire dans une base de sondage.

* les informations recueillies : elles proviennent de différentes sources - les travailleurs, les entreprises, des experts en acoustique - et sont de nature diverse : mesures d'exposition, avis subjectifs, données sur les machines bruyantes...

La prise en compte de ces caractères permet d'apprécier la validité des méthodes et des résultats des évaluations.

1 - Panorama de diverses évaluations

Un tableau récapitulatif contient les indications principales. Quelques précisions sur les méthodes ou les objectifs sont utiles :

* L'enquête du NIOSH (1971) a été réalisée par questionnaire envoyé aux entreprises.

* La première évaluation de BOLT, BERANEK and NEWMAN (BBN 1974) est basée uniquement sur les opinions d'experts en acoustique, alors que la seconde (1976) s'appuie sur des mesures d'exposition dans un échantillon d'entreprises.

* L'estimation du TNO résulte d'une transposition aux Pays-Bas de mesures systématiques d'exposition aux bruits des travailleurs autrichiens.

* L'enquête de l'Institut Français de Statistiques (INSEE 1978) a été réalisée par questionnaire auprès d'un échantillon de travailleurs, représentant la totalité des activités professionnelles. Elle concernait les conditions de travail, notamment l'exposition au bruit.
Les deux autres évaluations effectuées en France visent un champ volontairement restreint : l'enquête réalisée pour le Ministère de l'Environnement (MECV) est une première approche du coût social du bruit ; l'enquête réalisée par une CRAM et l'INRS avait pour but de tester une méthode de sondage aléatoire et d'estimer sa précision.

L'enquête du NIOSH (1980) vise l'exposition aux bruits impulsionnels uniquement : recenser leurs sources et estimer le nombre de travailleurs exposés à chaque source.

2 - Résultats

Ils sont indiqués brièvement dans le tableau récapitulatif. Leur lecture ne peut être dissociée de celle des caractères principaux des évaluations, en particulier du champ étudié. Malgré cela, d'un pays à l'autre, ils ne sont pas immédiatement comparables car la structure industrielle n'est pas identique et les branches d'activités ne sont pas définies avec les mêmes classifications.

Seuls peuvent être rapprochés les résultats de trois évaluations qui concernent l'industrie des U.S.A. : le pourcentage de travailleurs exposés à plus de 90 dB(A) varie de 14 à 26 % ; avec le seuil de 85 dB(A), il varie de 34 à 59 %.

3 - Discussion

3.1 - Précision des évaluations et type d'échantillonnage

Les diverses études concernant l'industrie des U.S.A. conduisent à des estimations qui varient du simple au double. Or l'évaluation de la précision des résultats obtenus a été négligée dans la majorité des travaux cités ici. L'étude CRAM/INRS montre que même s'il est réalisé pour un secteur professionnel restreint, l'échantillonnage est source d'erreur beaucoup plus que les informations recueillies pendant l'enquête (précision du sondage : 25 % ; précision des données : 5 %). Il faut noter, de plus, que la méthode d'échantillonnage par tirage au hasard autorise l'estimation de la précision du sondage. Mais ce n'est pas le cas pour les méthodes empiriques et on peut craindre l'existence de biais dans les trois évaluations ayant utilisé ce mode d'échantillonnage.

3.2 - Sources d'information et taille de l'échantillon sondé

Trois études citées s'appuient sur des mesures précises d'exposition : deux concernent des secteurs industriels particuliers, sondés par des échantillons de moins de 25 entreprises (MECV et CRAM/INRS) ; la troisième s'étend à toute l'industrie, représentée par un échantillon de 68 entreprises (BBN 1976).

Or, pour obtenir dans toute l'industrie une évaluation de précision analogue à celle de l'enquête CRAM/INRS, nous estimons que la taille de l'échantillon devrait être voisine de 200 entreprises. La réalisation de mesures d'exposition aux bruits devient, avec de tels échantillons, très lourde, voire impossible.

Utilisant d'autres sources d'information, les enquêtes par questionnaire autorisent le sondage de très grands échantillons : 341 entreprises (NIOSH 1971), 18 500 travailleurs (INSEE). Ceci garantit une bonne représentation des secteurs d'activités et permet leur classement selon l'importance du "problème bruit". Mais ces sources d'information, subjectives et sujettes à des biais nombreux, ne permettent pas de quantifier précisément l'exposition.
3.3 - Connaissance du contexte de l'exposition

Deux évaluations réalisées par enquête en entreprise (BBN 1976 et NIOSH 1980) ont inclus dans le recueil d'information des données sur le contexte de l'exposition aux bruits : les machines bruyantes, les réalisations en matière d'insonorisation. L'enquête du NIOSH recense les sources de bruits impulsionnels, en même temps que la population des travailleurs exposés à chaque source, grâce à l'utilisation de statistiques disponibles auprès d'organismes professionnels, telles que la taille de parc machine ou la diffusion de technologies bruyantes. Ce type d'information permet l'approche statistique du contexte de l'exposition au bruit.

4 - Conclusion

L'analyse des travaux cités révèle que, pour la plupart d'entre eux, la recherche de la précision n'a pas été un critère de choix des méthodes utilisées pour estimer le nombre de travailleurs exposés au bruit. Accorder plus de poids à ce point conduit à choisir des techniques d'échantillonnage et des sources d'information adaptées. On utilisera avantageusement une méthode de tirage au hasard pour constituer l'échantillon, plutôt qu'un mode empirique.

On notera qu'un sondage par questionnaire de vastes secteurs professionnels peut permettre d'obtenir une vue globale de l'exposition au bruit avant d'entreprendre, dans des secteurs d'étendue limitée, des sondages visant à quantifier avec précision l'exposition et nécessitant des mesures en entreprises.

Dans ce cas, l'enquête sera considérablement enrichie si le recueil des données in situ comprend, outre les effectifs de travailleurs et leurs niveaux d'exposition sonore, des informations sur les sources de bruit, les parcs machines. Car cet ensemble de résultats paraît particulièrement pertinent pour orienter les choix de travaux d'insonorisation.

<table>
<thead>
<tr>
<th>Auteurs de l'étude</th>
<th>NIOSH</th>
<th>BBN 1</th>
<th>TNO</th>
<th>BBN 2</th>
<th>MECV</th>
<th>INSEE</th>
<th>CRAM/IURS</th>
<th>NIOSH (B.I.)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pays</td>
<td>U.S.A.</td>
<td>U.S.A.</td>
<td>Pays-Bas</td>
<td>U.S.A.</td>
<td>France</td>
<td>France</td>
<td>France</td>
<td>U.S.A.</td>
</tr>
<tr>
<td>Objectif</td>
<td>Exposition au bruit</td>
<td>Coût social du bruit</td>
<td>Exposition au bruit</td>
<td>Coût social du bruit</td>
<td>Coût social du bruit</td>
<td>Conditions de travail</td>
<td>Méthodologie</td>
<td>Bruits impulsionnels</td>
</tr>
<tr>
<td>Champ de l'étude</td>
<td>Industrie</td>
<td>Industrie</td>
<td>Industrie</td>
<td>Industrie</td>
<td>&quot;11 entreprises bruyantes&quot;</td>
<td>Tous les travailleurs</td>
<td>Métallurgie (une seule région)</td>
<td>Industrie, Bâtiment et T.P., Mines</td>
</tr>
<tr>
<td>Pourcentage de travailleurs exposés :</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
</tr>
<tr>
<td>- Leq &gt; 85 dB(A)</td>
<td>14 %</td>
<td>26 %</td>
<td>35 %</td>
<td>19 %</td>
<td>20 %</td>
<td>/</td>
<td>13 %</td>
<td>/</td>
</tr>
<tr>
<td>- Leq &gt; 90 dB(A)</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
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<tr>
<td>Autre indicateur</td>
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<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
</tr>
<tr>
<td>Nombre d'entreprises sondées</td>
<td>341</td>
<td>/</td>
<td>/</td>
<td>68</td>
<td>11</td>
<td>/</td>
<td>23</td>
<td>31</td>
</tr>
<tr>
<td>Nombre de travailleurs concernés</td>
<td>800 000</td>
<td>/</td>
<td>95 000</td>
<td>61 700</td>
<td>900</td>
<td>18 500</td>
<td>22 000</td>
<td>42 725</td>
</tr>
<tr>
<td>Nombre total des travailleurs du champ étudié</td>
<td>17 №</td>
<td>14 №</td>
<td>1,2 №</td>
<td>13 №</td>
<td>/</td>
<td>17 №</td>
<td>70 000</td>
<td>21 №</td>
</tr>
<tr>
<td>Unités statistiques croisées</td>
<td>Entreprises</td>
<td>Branches industrielles</td>
<td>Travailleurs</td>
<td>Entreprises</td>
<td>/</td>
<td>Travailleurs</td>
<td>Entreprises</td>
<td>Entreprises</td>
</tr>
<tr>
<td>Type d'échantillonnage</td>
<td>Aléatoire</td>
<td>/</td>
<td>Empirique</td>
<td>Empirique</td>
<td>/</td>
<td>Aléatoire</td>
<td>Aléatoire</td>
<td>Empirique</td>
</tr>
<tr>
<td>Sources des informations recueillies</td>
<td>Entreprises</td>
<td>Experts</td>
<td>Experts</td>
<td>Experts</td>
<td>Entreprises</td>
<td>Experts</td>
<td>Entreprises</td>
<td>Experts</td>
</tr>
<tr>
<td>Mesures de bruit effectuées particulièrement pour l'étude</td>
<td>Non</td>
<td>Non</td>
<td>Non</td>
<td>Oui</td>
<td>Oui</td>
<td>Non</td>
<td>Oui</td>
<td>Non</td>
</tr>
<tr>
<td>Précision du sondage</td>
<td>?</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>30 %</td>
<td>/</td>
</tr>
</tbody>
</table>

* Pourcentage de travailleurs n'entendant pas si on leur parle à 2-3 m, à cause du bruit.
** N.B. : Pour l'industrie seule, ce pourcentage doit être doublé.
** L'étude citée donne le nombre de travailleurs exposés à des bruits impulsionnels : 2,7 millions. Le calcul de ce pourcentage est effectué à partir de données statistiques de 1975 ; ce pourcentage n'est qu'indicatif.
SENSIBILITE DIFFERENTIELLE AU BRUIT (GENE SUBJECTIVE) SELON LE PATTERN COMPORTEMENTAL DE TYPE A OU B

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INTRODUCTION

De nombreuses recherches ont mis l'accent sur l'importance de certaines attitudes comportementales dans l'identification des situations de stress et dans leurs effets sur l'homme. On a ainsi plus particulièrement décrit un pattern comportemental de type A (coronary prone behavior, JENKINS et coll, 1971) qui prédisposerait aux maladies coronariennes. Ce-lui-ci caractériserait les personnes ambitieuses, toujours pressées, mettant tout en œuvre pour atteindre leurs buts. À l'inverse, les individus de type B seraient ceux présentant des attitudes opposées.

BUT

Notre objectif dans cette étude a été d'examiner la sensibilité différentielle au bruit de sujets de type comportemental A ou B. À la suite de certains résultats (LOVALLO et PISHKIN 1980, PITTNER et HOUSTON 1980, GLASS 1977) ayant souligné les effets différentiels du bruit sur l'homme tant sur le plan physiologique que dans l'appréhension cognitive du stress auditif et dans l'évaluation de la gêne, nous nous attendions à ce que les individus de type A tolèrent un niveau de bruit plus élevé que ceux de type B et ceci pendant plus longtemps.

METHODOLOGIE

Echantillon

40 sujets de sexe féminin, étudiants en première année de psychologie à l'Université PARIS VIII, ont été sélectionnés à partir d'une population de 104 personnes. La classification en type A ou type B a été effectuée à l'aide de l'adaptation française de l'échelle de BORTNER (DEFOUNRY et FRANKIGNOUL 1973). Ont ainsi été retenus 20 sujets de type A et 20 sujets de type B, respectivement issus du quartile inférieur ou du quartile supérieur de la distribution des notes à l'échelle de BORTNER. Celle-ci est constituée de 14 items notés de 1 à 24 selon l'endroit auquel se situe le sujet interrogé. Une note élevée à cette échelle traduit l'appartenance au type comportemental A, une note faible, au type B.
. Matériel

Un générateur de bruit a permis de délivrer aux sujets un son de 3000 hertz dont on pouvait faire varier l'intensité de 60 DbA à 110 DbA. Les niveaux d'intensité sonore notés de 1 à 8 correspondaient à des accroissements de 6 DbA en 6 DbA. Les différents stimuli auditifs étaient présentés au casque (SONY DR 53).

. Tâche

Les sujets étaient invités à participer de façon individuelle à une expérience présentée comme portant sur la discrimination perceptive de stimuli auditifs et visuels. Après les avoir avertis que des sons pourraient leur parvenir à travers le casque, on leur demandait d'examiner une série de diapositives présentées à allure libre, représentant des œuvres de peinture abstraite. On leur disait qu'ils seraient ultérieurement interrogés sur les couleurs des diapositives. Après une période de familiarisation avec le matériel et les modalités expérimentales, le son augmentait toutes les 20 secondes à partir du seuil de 68 DbA jusqu'à ce que le sujet signale qu'il était gêné. Celui-ci devait donc indiquer une première fois quand il commençait à être gêné (niveau de tolérance) et une seconde fois quand il voulait que l'on arrête le son (durée de tolérance à un seuil donné).

. Résultats

1) Niveau de tolérance au bruit :

Nous avons comparé les distributions des réponses des sujets selon qu'ils appartiennent au type A ou B. Nous avons obtenu les résultats suivants :

![Diagramme des résultats](image)

Tableau I : Distribution des notes de niveau de tolérance au bruit des sujets, selon le type A ou B
Nous voyons (tableau I) que les distributions des notes des deux groupes de sujets diffèrent fortement. Les sujets de type A supportent des niveaux de bruit plus élevés que ceux de type B, certains ayant même accepté des seuils maxima (note 8) sans jamais exprimer la moindre gêne. Le test non paramétrique U de Mann-Whitney appliqué aux données confirme les résultats observés sur les distributions (U : 3,23 p < .01).

2) Durée de tolérance au bruit :

Tableau II : Durée de tolérance au bruit selon le type comportemental A ou B

Les résultats (tableau II) diffèrent significativement selon le type comportemental A ou B (χ² = 4,94 p < .05). 65% des sujets de type A (13/20) contre 25% des sujets de type B (5/20) supportent le niveau sonore correspondant à leur seuil de tolérance, pendant une durée au moins égale à 60s (temps maximum accordé aux sujets dans l'expérience).

DISCUSSION

Dans cette expérience, tandis qu'ils sont censés effectuer une tâche de reconnaissance visuelle, les sujets de type A acceptent des niveaux d'intensité plus élevés que ceux de type B et corrélativement supportent plus longtemps les sons présentés. On peut émettre l'hypothèse selon laquelle l'engagement dans la tâche, le désir de réussir peuvent expliquer le peu d'attention qu'ils semblent prêter à leur environnement sonore. Peut-être cependant repugnent-ils plus que les autres à reconnaître qu'ils sont gênés...

En effet, à la fin de l'expérience on a présenté à tous les sujets, dans le même ordre, trois sons (faible, fort et moyen) que ceux-ci devaient évaluer sur une échelle en sept points. Les sons ont été bien différenciés entre eux que soit le groupe : F(2,36) = 267,49 p < .001. Mais on n'observe pas de différence selon le type comportemental F(1,38)= 0,54 n.s. L'appartenance à l'un ou l'autre des patterns comportementaux
n'a aucune influence sur l'évaluation. L'interaction évaluation sonore/type comportemental A ou B est non significative $F(2,76) = 0,14$ n.s.

On peut s'interroger sur les risques encourus par les individus de type A qui dans certaines conditions tentent de s'adapter de façon parfois excessive à leur environnement et n'en reconnaissent pas le caractère désagréable.

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EVALUATING THE EVIDENCE ON NOISE AND HEALTH

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Introduction

Numerous problems beset any investigation of the health effects of low-level environmental toxic agents such as noise. Three of these are fundamental: the multiple causation of most diseases which have been linked to noise exposure; the long latency period between noise exposure and the diagnosis of disease; and the ambiguity of the role of noise as a causal agent especially with respect to non-auditory health outcomes. Each of these problems seriously complicates precise determination of the causal effect of noise on human health.

Given that noise effects on health are not amenable to easy detection, very carefully designed studies are required to provide reliable evidence on the existence or non-existence of such effects. To assess the validity of statements linking noise and health it is necessary to evaluate the adequacy of the studies from which supporting evidence is drawn. In the remaining sections of this paper we outline criteria which can be employed to assess the strength of evidence linking noise and human health, and we report the results of an application of these criteria to studies which investigate the effects of environmental noise on a range of human health outcomes.

Criteria for evaluating studies linking noise and health

Two sets of criteria are proposed for assessing the research evidence for a causal link between noise and human health. Application of the first set logically precedes use of the second set. The first set comprises basic criteria for judging the methodological adequacy of a study. Their purpose is to identify studies which are fundamentally sound in design, analysis and reporting and which therefore are worth examining in terms of the strength of evidence they provide for a causal association between noise and health. The second set of criteria deals explicitly with the diagnosis of causation. A brief outline of both sets of criteria follows.

Ten criteria make up the first set:

1. Health outcomes: does the outcome measure deal with ill health (e.g. cardiovascular disease) or with physiological changes of unknown consequences for well-being?
2. Problem statement: is the research problem clearly stated and are the hypotheses explicit?
3. Research setting: was the study conducted in a laboratory or field setting?

4. Sample design: is the sample design explicitly stated? Was the health status of subjects recorded/reported prior to noise exposure? Is the sample size adequate to avoid sampling bias? Are response rates and other possible sources of bias reported? Was a control group used?

5. Noise measure: are full details given about the frequency, intensity and duration characteristics of the exposure?

6. Compliance: where appropriate, is degree of compliance with intervention strategies (e.g. wearing hearing protection) reported?

7. Confounding factors: are factors which might confound outcome measures acknowledged (e.g. stress-induced physiological changes in a laboratory experiment).

8. Outcome measurement: are the outcome measures appropriate, objective, sensitive, valid, reliable, blind, rigorous?

9. Analysis: are the analytical procedures replicable and valid?

10. Interpretation: are the study conclusions justified in light of the research methodology and analytical results?

Studies judged to be methodological sound in terms of these criteria can be subsequently assessed using the diagnostic tests of causation described by Sackett, (1978). There are nine tests which are described here in descending order of importance for diagnosing causation.

1. Study design: randomized control trials involving human subjects provide the strongest evidence for causation. Successively weaker evidence is provided by the following study designs: cohort studies, case controls, field surveys, laboratory studies.

2. Strength of association: normally based on a statistical comparison of differences in outcomes between the experimental and control groups.

3. Consistency: occurs where several independent studies produce the same conclusions about the causal association.

4. Temporality: established by ensuring that the health outcome postdates exposure to the hypothesized causal agent.

5. Gradient: shown where there is a direct relationship between the degree of exposure and the severity of the outcome.

6. Epidemiologic sense: satisfied when the hypothesized cause-effect link makes sense in terms of the known distribution of the health outcome in the general population.

7. Biologic sense: satisfied when the hypothesized cause-effect link makes sense in terms of biologic mechanisms observed in non-human studies.

8. Specificity: applies only in those few cases where the outcome is due to a single cause (e.g. asbestosis caused by asbestos exposure and ingestion).

9. Analogy: hypothesized cause and effect link is analogous to a causal association already confirmed by research evidence.

Notice that the diagnostic tests are not mutually exclusive. In particular the study design dictates to a considerable extent the strength of evidence determined by the other tests.

Assessment of the literature on noise and health

These two sets of criteria were used to assess the evidence for a causal link between noise and a range of auditory and non-auditory health outcomes (Taylor et al., 1980). Studies dealing with noise and health
were identified by a comprehensive search of the research literature using computerized bibliographic data bases. The time period covered by the search was 1966 to 1980. Approximately 3000 titles were initially identified. This number was reduced to 1038 by a screening of abstracts which revealed that many of the papers did not actually examine the effects of noise on health. The literature assessment based on the 1038 titles was divided into three stages: classification, evaluation and diagnosis of causation.

The classification stage included a preliminary assessment of the potential importance of the contribution of each study to knowledge about the effects of noise on health. Three criteria were used in combination to make the importance rating: study design; level of knowledge about the health outcome examined; and degree of recognition accorded to the study in the current literature (particularly in review papers). On this basis 287 papers were rated of primary importance and these were carried forward to the evaluation phase.

The set of 287 papers was subsequently reduced to 146 for full evaluation. The remaining 141 were not evaluated for one of three reasons: 60 were not available in either French or English translation (four papers in Russian and one in German judged to be of especial importance were translated for this project); 28 papers were not available as full texts; and 53 were review papers not involving original research and therefore not applicable to the evaluation.

The evaluation stage revealed methodological problems in many of the studies. The final criterion, 'interpretation', provides a summary indicator of methodological adequacy. 40 papers were judged to state conclusions which were fully justified in light of the study design and analytical results. In a further seven papers the conclusions were rated as possibly justified. Lack of justification stemmed from failing to acknowledge limitations falling under one or more of the criteria. Of the 47 papers fully or possibly justified, only 20 dealt with auditory outcomes and ten with non-auditory health affects, including hypertension, mental health and birth defects. In short, the application of a reasonable, if stringent, set of methodological criteria reduced a large literature to a small number of studies providing the strongest basis for causal inference. The 47 "justified" papers were carried forward to the final stage of the assessment, the diagnosis of causation.

The results from the final stage of the assessment are most usefully described by reference to each of the major types of health outcome dealt with in the "justified" set of papers: noise-induced hearing loss; hypertension; mental health; birth defects and self-reported health problems.

Ten papers dealt with permanent hearing loss due to noise and these included studies with the strongest designs. The net conclusion is that there is strong evidence for concluding a causal link between noise exposure and hearing loss.

Evidence on the effect of noise on hypertension is far less clear which is partly due to the absence of strong study designs. The overall conclusion is that there is a lack of strong evidence for either the presence or absence of a causal link between noise and hypertension.

There was also an absence of strong study designs in the papers dealing with the link between noise and mental health outcomes. Researchers acknowledge that the relationship, if present at all, is complex and that existing evidence is not strong enough to either confirm or deny the
hypothesis that noise affects mental health.

Only one paper dealing with birth defects was included in the 47 "justified" papers. Here again however the quantity and strength of the evidence are insufficient to support valid causal inference.

The final "justified" paper dealing with non-auditory health outcomes focussed on self-reported health problems and showed an association between fatigue and headaches and level and duration of industrial noise exposure. No data were obtained to determine the relationship between self-reported symptoms and objective outcome measures.

By restricting attention to only those studies judged methodologically sound, the full range of health outcomes which have been hypothetically linked to noise is not represented in this brief summary on the diagnosis of causation. Our approach to assessment makes methodological adequacy a sine qua non as far as diagnosing causation is concerned. It follows therefore that the evidence for a causal link (or lack thereof) for outcomes not mentioned is weaker than for those for which there were methodologically sound studies.

Conclusion

We have proposed criteria for a systematic assessment of the research evidence on the causal link between noise and health. Our use of these criteria revealed that only a small percentage of the studies examined were methodologically adequate and therefore useful for diagnosing causation. The lack of strong evidence cautions against wholesale acceptance of statements implicating noise as a causal factor in a range of disease outcomes. Careful assessment of the research evidence on which such statements are based reveals that the conclusions cannot be supported as definitively as the statements sometimes suggest. This is not to argue that noise is necessarily benign. With respect to hearing loss we can confidently conclude that it is not. For non-auditory outcomes the evidence is not strong enough to come to definite conclusion which leaves us currently with the dilemma of deciding in the absence of the "burden of proof" what policy statements are justified on the basis of the "burden of prudence".

References


STRESSFUL NOISE AND VIGILANCE

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Introduction

Widespread use of the term stress is largely due to Hans Selye, who has defined it as "... the nonspecific response of the body to any demand". (Selye, 1976). The response is physiological, and is characterised initially by hyperactivity of the pituitary adrenal-cortical system. Mason (1974) has provided evidence against the non-specificity of the stress response. His data suggest that the adrenal cortical response can be prevented if certain conditions of novelty, psychosocial stimulation and discomfort are removed. What remains are physiological responses specific to the physical stressor. Mason's later discussion (Mason, 1975) suggests that the two types of response may reasonably be called 'psychological' and 'physiological' stress respectively.

This position still defines psychological stress in terms of a physiological response. If psychological stress is to be a useful concept which can be related to noise exposure then both its behavioural characteristics and the environmental conditions giving rise to it must be identifiable.

A substantial step in this direction was made by Glass and Singer (1972) when they showed the importance of predictability and control in determining post-noise behavioural after-effects. These after-effects included impaired performance on tasks involving response conflict (the Stroop test) and frustration tolerance (the Feather test). These impairments could reasonably be used as measures of psychological stress due to noise and other stressors.

The present study attempted to carry this approach further by having subjects carry out a vigilance task in quiet or intermittent noise of varying predictability. Correlations were sought between noise 'treatment' and various characteristics of task performance, physiological response, subjective ratings of the task and noise, and behavioural after-effects as measured by the Stroop test.
Method

72 male subjects carried out a 55 minute vigilance task somewhat similar to that used by Broadbent and Gregory (1963, 1965). Three lights flashed simultaneously for 0.3 second every five seconds. On about half of these trials one light flashed more brightly than the other two. This was the 'signal'. This signal occurred on any of the three lights at random. The subject was required to press one of five buttons after each flash to indicate certain signal, uncertain signal, uncertain either way (or don't know), uncertain no-signal, or certain no-signal. Failure to press any button was also recorded. These failures to respond were designated 'gaps'.

Subjects were randomly allocated to one of four 'treatment' groups. Group 1 carried out the task in quiet, group 2 in regular bursts of noise (one per minute), group 3 with variable intervals between noise bursts, and group 4 a noise schedule combining variable interval and variable burst durations. All noise schedules had the same acoustic energy summed over the 55 minutes. Noise bursts were all 92 dB, one-third octave bands with centre frequency 4.0 kHz.

Results

The results presented here are limited to the GAP scores, defined as the ratio of failures to respond to the total number of signal and no-signal light flashes. Figure 1 gives the mean GAP score by treatment (noise/quiet) condition and block. Each block is one (13' 45") quarter of the 55 minute task period.

Figure 1 shows an apparent increase in failures to respond as instructed as time on watch proceeds. This increase is more pronounced in those subjects exposed to noise, particularly those given the most irregular noise schedule (treatment 4). The GAP distributions are highly skewed however and the median GAP scores do not show the same trends. This skewness, largely due to the large number of zero GAP scores, also precludes the use of parametric statistics. The data were therefore categorised into GAP scores equal to or less than 0.02, and greater than 0.02, for each subject and block. Frequencies of GAP 1 (GAP ≤ 0.02) and GAP 2 (GAP > 0.02) were then calculated for each treatment and block. Frequencies of subjects obtaining GAP 2 scores on any of the four blocks were also calculated for each treatment group.

χ² tests of these frequencies showed significant noise/quiet differences only in the fourth block and over all blocks, although there is a progressive reduction in the values of p from block 1 to block 3. Similar results are found whether the noise treatment groups are treated separately or are pooled. There appear to be no systematic differences in frequency of GAP 2 scores according to type of noise schedule. When compared with Figure 1 this suggests that irregular noise increases the magnitude of the effect in already susceptible individuals, rather than increases the number of individuals showing the effect. For reasons of space only the GAP frequencies pooled across the three noise conditions are given here, in Table 1.
Discussion

The $\chi^2$ analysis described above gives no additional weighting to the very high GAP scores which appeared late in the session under the highly irregular noise condition. Many previous studies of vigilance have used longer task durations than the 55 minutes used in the present study. Again, the level of the noise bursts (92 dB) is less than those used in many previous studies of noise effects on vigilance (e.g. Broadbent and Gregory, 1963; 1965). Under these more severe conditions more subjects may have shown GAP scores greater than 0.02 and large GAP scores. The possible effect of noise schedule may then have become more pronounced.

One interpretation of a high GAP score is that these subjects tended to 'give up' the task under noise conditions. This interpretation suggests that the GAP score is a within task period measure of the same process as that measured by Glass and Singer (1972) after the task period has ended. There are other possible interpretations however, and as well, a variety of theories have been proposed to account for post-stress after-effects (Cohen, 1980).

Attempts were made to link GAP score with Stroop scores and with other measures referred to above. The results of these analyses are given elsewhere.

References


Table 1. $\chi^2$ tests of the significance of differences in frequency of GAP score categories, by noise/quiet treatment, block and all blocks. The $\chi^2$ incorporates Yates' correction for continuity. df=1.

<table>
<thead>
<tr>
<th>Block No.</th>
<th>GAP</th>
<th>Treatment</th>
<th>$\chi^2$</th>
<th>P</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Quiet</td>
<td>Noise</td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>≤ 0.02</td>
<td>17</td>
<td>54</td>
<td>0.0</td>
</tr>
<tr>
<td></td>
<td>&gt; 0.02</td>
<td>0</td>
<td>1</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>≤ 0.02</td>
<td>17</td>
<td>48</td>
<td>1.166</td>
</tr>
<tr>
<td></td>
<td>&gt; 0.02</td>
<td>0</td>
<td>7</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>≤ 0.02</td>
<td>17</td>
<td>45</td>
<td>2.23</td>
</tr>
<tr>
<td></td>
<td>&gt; 0.02</td>
<td>0</td>
<td>10</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>≤ 0.02</td>
<td>17</td>
<td>40</td>
<td>4.320</td>
</tr>
<tr>
<td></td>
<td>&gt; 0.02</td>
<td>0</td>
<td>15</td>
<td></td>
</tr>
<tr>
<td>All Blocks</td>
<td>≤ 0.02</td>
<td>17</td>
<td>37</td>
<td>5.775</td>
</tr>
<tr>
<td></td>
<td>&gt; 0.02</td>
<td>0</td>
<td>18</td>
<td></td>
</tr>
</tbody>
</table>
ON THE ANNOYANCE OF THE COMPOSITE NOISE FROM VARIOUS SOUND SOURCES

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I. Introduction

It have been agreed by many investigators that the equivalent continuous sound pressure level, Leq is the suitable quantity for rating of the annoyance due to noise.

This is the report about the experimental results of subjective rating on the annoyance due to various kinds of noises and these composite noise. The relationship between the equivalent continuous sound pressure level averaged in five minutes (written as $\text{L}_{\text{Aeq5}}$) and the annoyance of these noises are describe.

II. Experimental procedure

II-1 Sound stimuli: The sound stimuli used for experiments are the noises of road traffic, Shinkansen (rapid trains), tram-cars and jet aircrafts. Each sound was tape recorded during five minutes on each channel of four channels tape recorder. The intermittent noises such as aircraft noise were recorded five times repeatedly during five minutes. One or more of these noise samples were reprinted into one channel. Figure 1 shows the "A" weighted level record of all the noise samples.

![Fig. 1 "A" weighted level recording of noise stimuli](image-url)
II-2 Subjects and test room: The subjects are fourteen male students and two female ones. Three experiments were performed. The third experiment was especially performed for examining of reappearance of noise rating results. The test room is illustrated in figure 2. The left loudspeaker was used for presenting of sound stimuli of noise, and the right one was used for presenting of the Japanese sallables for articulation test.

II-3 Presenting of the noise stimuli, and the rating of annoyance:

The sound levels were determined in L_{Aeq} at the point of the height of the ear shown by mark A in figure 2.

The articulation test of Japanese sallables were performed as the duty work of subjects under the noise exposure. The subjects were requested to estimate the degree of disturbance due to annoyance at the end of every session.

In second experiment, the subjects were requested to estimate the annoyance without regarding to the articulation test. Table 1 shows the examples of sound stimuli and these L_{Aeq} values.

<table>
<thead>
<tr>
<th>No.</th>
<th>L_{Aeq}</th>
<th>combination of sound stimuli</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>69.0</td>
<td>aircraft</td>
</tr>
<tr>
<td>4</td>
<td>65.2</td>
<td>train</td>
</tr>
<tr>
<td>5</td>
<td>60.8</td>
<td>traffic</td>
</tr>
<tr>
<td>1</td>
<td>67.3</td>
<td>traffic+train</td>
</tr>
<tr>
<td>10</td>
<td>74.4</td>
<td>traffic+aircraft</td>
</tr>
<tr>
<td>14</td>
<td>71.0</td>
<td>train+aircraft+tram</td>
</tr>
<tr>
<td>24</td>
<td>68.2</td>
<td>traffic+aircraft+train+tram</td>
</tr>
<tr>
<td>13</td>
<td>69.3</td>
<td>train+aircraft</td>
</tr>
</tbody>
</table>

III. Experimental results and discussion

Figure 3 shows the relationship between annoyance and the equivalent continuous sound pressure level, L_{Aeq} . The plots encircled by marks ··· show the results by the intermittent noises from Shinkansen trains, jet aircrafts and tram-cars. It is observed that the intermittent noises are estimated
less annoyed than continuous noises at equal sound level in the range from 65 dBA to 70 dBA. It is supposed that the silent interval gave the negative influence to rating of annoyance.

If the levels of these sound were reduced by 5 dBA, the correlation coefficient is shifted from 0.79 to 0.92.

Figure 4 shows the relationship between annoyance and $L_{10}$. The correlation coefficient is 0.84. This value is slightly better than the one of $L_{Aeq}$. Both of $L_{Aeq}$ and $L_{10}$ are suitable for estimating of annoyance. However, $L_{Aeq}$ is more excellent rating scale, because of the usefulness for noise prediction.

IV. Conclusion

The following conclusions were obtained.

a. The annoyance of the noise from a kind of source or these composite noise can be uniformly estimated by equivalent continuous sound pressure level, $L_{Aeq}$.

b. The intermittent noise are estimated less annoyed by 5 dBA than continuous noise in the range from 65 dBA to 70 dBA.

![Fig. 3 Relationship between annoyance and equivalent continuous sound pressure level, $L_{Aeq}$. Marks show the results of intermittent noises.](image-url)
Fig. 4 Relationship between annoyance and upper limit of 80% range, $L_{10}$

Mark○ show the results of intermittent noise.

Reference

1) T. Yamamoto; "Noise rating from the point of view about $L_{eq}$" Journal of acoust. soc. Japan, 38(5), pp. 293-300 (1982), (and Japanese), and etc.

INFLUENCE DU BRUIT DES AVIONS SUR LE NIVEAU D'ANXIÉTÉ

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Introduction

Ce travail prétend évaluer la possible influence que le niveau du bruit supporté par des écoliers d'une zone proche à l'aéroport de Madrid, puisse avoir sur le niveau d'anxiété de ces écoliers lorsqu'ils réalisent une tâche intellectuelle.

Pour faire cet étude on a disposé d'un échantillon de 334 sujets appartenant au 8ème cours de l'Enseignement Général Elementaire, distribués en cinq écoles, trois d'elles de la zone bruyante proche aux pistes de l'aéroport et dont les sujets ont constitué le groupe expérimental, et les autres deux écoles, dont les caractéristiques culturelles, socio-économiques, familiales, etc. étaient semblables à celles du groupe expérimental, en exceptant ce qui se rapporte au bruit, ont formé le groupe contrôle.

Comme instrument pour la mesure de l'anxiété on a utilisé le STAI (State Tract Anxiety Inventory). La mesure du bruit a été réalisée avec un Noise Level Analyser, B & K, type 4426.

Pour la mesure du rendement, on a employé une preuve de raisonnement numérique, qui se compose d'une série d'éléments de facile réalisation, mais qui exigeait pour son développement un certain degré d'attention et de concentration.

Procédé

Les preuves furent appliquées de façon collective dans chacune des écoles sélectionnées. Le groupe contrôle a réalisé ces preuves dans des conditions sonores normales et le groupe expérimental sous l'estimulation du bruit d'avions propre de la zone.

L'expérience pourrait se schématiser comme suit:

1. Mesure de l'anxiété
2. Execution de la tâche (duration 30')
3. Mesure de l'anxiété

4. Execution de la tâche (duration 30')

Dans les analyses du rendement dans la tâche, on a pris en considération les indices suivants: succès, erreurs et rendement effectif (succès moins erreurs).

En ce qui se rapporte à la mesure de l'anxiété on a obtenu cinq mesures, deux de trait d'anxiété et deux d'état d'anxiété, par moyen de l'application du STAI complet en deux moments du procès expérimental.

A partir des donnés obtenus dans les 334 individus de l'échantillon on a réalisé des plusieurs analyses de la variance.

Résultats

Première application (Trait d'anxiété)

Le trait d'anxiété fait référence aux caractéristiques de tendance à l'anxiété, relativement stables pour chaque individu. Ces différences individuelles peuvent se déduire de la fréquence et l'intensité des réactions de l'état d'anxiété à travers le temps.

Dans ce point on analyse le comportement différent des sujets dans le trait d'anxiété en fonction du milieu bruyant, en partant de l'hypothèse de que le stress psychologique produit par le bruit augmentera le niveau d'anxiété.

En étudiant les données obtenus de la première application du STAI trait, on peut tirer les conclusions suivantes:

- Le niveau du bruit n'a pas une influence significative sur le trait d'anxiété des personnes soumises au bruit, ne confirmant pas pourtant l'hypothèse théorique dont on avait parti.
- On a trouvé un unique résultat significatif, correspondant au facteur sexe. Les femmes ont ponctué plus en trait d'anxiété que les hommes.

Deuxième application (Trait d'anxiété)

D'une façon globale on montre des différences pas significatives dans les deux groupes (avec du bruit ou sans bruit) entre la première et la deuxième application, en appuyant de cette façon la supposée stabilité des mesures du trait d'anxiété.

Le résultat de la présente analyse montre que les fonctions en trait d'anxiété ne sont pas influencées par la situation du stress associé au niveau du bruit, ce qui est d'accord avec d'autres études ou a été démontré que les ponctuations en trait d'anxiété appréciées par le STAI ne changent pas en réponse du stress situationnel.
Lopez, I. - Influence du bruit des avions sur le niveau d'anxiété.

État d'anxiété

L'état d'anxiété est conçu comme un état émotionnel transitoire qui varie en intensité et fluctue dans le temps en fonction de la quantité de stress incidant sur l'individu. Le niveau de l'état d'anxiété sera haut dans des circonstances qui sont perçues par le sujet comme menaçantes. L'intensité de l'état d'anxiété sera baissée dans des situations sans stress, ou dans des circonstances ou le danger objectif n'est pas perçu comme menaçant.

Dans la plupart des études réalisées, l'anxiété tend à corrélérer négativement avec le rendement. L'anxiété ne se relationne pas avec l'exécution. Dans la preuve ou tâche on considère comme de peu d'importance, mais lorsque la tâche est d'une grande difficulté, l'intensité détrône l'exécution de l'individu.

Par le moyen de cet étude nous avons pretendu connaître l'influence de l'ambient bruyant sur l'état d'anxiété des sujets, lorsqu'ils réalisent une tâche difficile.

Notre hypothèse initial fut que la réalisation d'une tâche de difficulté élevée dans un milieu bruyant produira des niveaux plus élevés sur l'état d'anxiété des écologistes, que si cette même tâche s'aurait réalisée dans des conditions du bruit ambient normales.

Pour corroborer cette hypothèse on a mesuré différentes fois l'état d'anxiété pendant le temps de durée de la tâche, à l'objet de connaître les possibles variations du niveau d'anxiété tout au long de la situation expérimentale.

Pour chaque sujet on a obtenu deux mesures de l'état d'anxiété par l'application de le STAI complet en deux moments du procès expérimental.

Les résultats obtenus dans les analyses de variance, montrent une similitude des résultats dans les trois mesures de l'état d'anxiété dans les deux groupes, avec du bruit et sans bruit. C'est à dire les points d'influence de l'anxiété se sont manifesté d'une façon pratiquement la même dans les deux groupes.

Dans tous les deux groupes on observe une certaine tendance à augmenter progressivement son niveau d'anxiété pendant la réalisation de la preuve, étant cette tendance légèrement supérieure dans le groupe des sujets d'ambient bruyant, cependant, le niveau d'anxiété n'a pas été affecté différentiellement par le bruit, par ce qui n'a pas resté confirmé l'hypothèse initiale.
ASPECTS CENTRAUX DE LA FATIGUE AUDITIVE

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La fatigue auditive (FA) obtenue après exposition à un bruit intense et/ou prolongé se manifeste par des modifications temporaire de l'audition. De nombreux travaux se sont attachés à décrire les conséquences anatomiques, physiologiques et psychoacoustiques de l'exposition à un bruit fatigant (ELDRIDGE et al 1973, BOTTE et GROCHOLLE 1979).

Le but de notre étude est de préciser comment ces effets du bruit affectent la perception et la compréhension de parole et, à travers les résultats obtenus, de discuter des hypothèses sur l'existence d'une composante très centrale, voire non-auditive, de la FA.

Une étude antérieure (THOUIN, SORIN, BOTTE 1981) a montré qu'une FA légère (TTS ≤ 15 dB entre 1500 et 3000 Hz) entraîne une diminution nette du taux d'intelligibilité.

Une autre expérience, associant à une tâche de décision Mot/Non-mot une tâche de répétition du stimulus, a confirmé la dégradation de l'intelligibilité en présence de FA même légère et a mis aussi en évidence une modification des mécanismes d'accès au lexique (SORIN et THOUIN 1982). Toutefois les temps de réponse Mot/non-Mot n'étaient pas significativement modifiés par la FA.

Pour préciser nos hypothèses sur les effets plus centraux de la FA sur la compréhension de parole, nous avons ensuite réalisé une expérience visant à observer les modifications de la capacité de traitement cognitif complexe après exposition à un bruit fatigant. Des travaux récents (MAC KOWN et RATCLIFF 1980, LE NY 1981) ont permis d'élaborer un indice de cette capacité de traitement. Cet indice peut être mesuré dans une tâche où le sujet doit se prononcer sur l'existence ou non d'une relation sémantique entre un mot sonore et une phrase présentée antérieurement. L'indice correspond alors au temps de réponse.

DESCRIPTION DE L'EXPERIENCE

1/ Sujets et fatigue:

13 sujets jeunes, audiométriquement normaux, ont participé à l'expérience. Le bruit fatigant était un bruit blanc filtré dans la bande 800-2500 Hz, choisi de manière à ce que la zone de TTS se situe dans la plage de fréquence la plus critique pour l'intelligibilité de la parole (1500 à 3000 Hz). Ce bruit était présenté de façon monaurale sur l'oreille droite des sujets. Le niveau du bruit ainsi que la durée d'exposition ont
été ajustés pour chaque sujet de manière à obtenir un même TTS de 15 dB pour tous les sujets, 2 min après la fin de l'exposition au bruit.

2/ Elaboration du corpus:

Le matériel comportait 80 phrases de 20 mots et 80 mots-sonde correspondants. Ces phrases ont été réparties en 2 listes de 40 phrases comportant chacune:
- 25 phrases-cible de type 1 où la réponse attendue était "oui".
- 9 phrases-cible de type 2 où la réponse attendue était "non".
- 6 phrases-cible de type 3 relevant d'une autre problématique pour éviter que ne se créent des stratégies d'anticipation.

Exemples:
- Phrase de type 1: "Sur la scène, l'homme s'écroula brutalement et les spectateurs applaudirent; l'enfant ne semblait pas impressionné par la violence de la scène." / Mot-sonde "Acteur" / Réponse attendue "Oui"
- Phrase de type 2: "Dans la forêt blanche, sous un ciel jaune et lourd de nuages, les corbeaux battaient des ailes en criant la faim" / Mot-sonde "Pendule"/ Réponse attendue "Non".
- Phrase de type 3: "À la maison, dans la frénésie de préparatifs, personne n'avait regardé l'heure; la pâtisserie embaumait l'air et les crêpes s'amonticiaient" / Mot-sonde "Fête"/ Réponse attendue "Oui".

Toutes ces phrases ont été conçues pour que la relation mot-sonde/phrase (quand elle existe) soit évidente, afin d'obtenir un nombre important de bonnes réponses. En effet l'indice dont on voulait étudier la variation sous l'effet de la FA correspondait au temps de réponse sur les seules bonnes réponses.

3/ Déroulement de l'expérience:

Les phrases étaient présentées sur l'oreille droite du sujet, à un niveau de 45 dBA. Une seconde après la fin de chaque phrase, le mot-sonde apparaissait sur une réglette de visualisation. Le sujet avait alors pour tâche d'indiquer le plus rapidement possible s'il existait ou non une relation sémantique entre le mot-sonde et la phrase.

L'expérience s'est déroulée en deux séances. Dans l'une des séances, le sujet entendaient l'une des deux listes de phrases en l'absence de toute exposition au bruit; dans l'autre séance, le sujet entendaient la deuxième liste de phrases, 2 min après cessation du bruit fatigant. L'ordre des listes ainsi que celui des séances avec ou sans FA était contrebalancé.

4/ Résultats principaux:

A l'issue de cette expérience on observe:

- une augmentation significative du nombre des erreurs après exposition au bruit fatigant:

<table>
<thead>
<tr>
<th></th>
<th>Sans FA</th>
<th>Avec FA</th>
<th>Test t</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pourcentage d'erreurs</td>
<td>5,3 %</td>
<td>8,8%</td>
<td>t(12)= 2,6</td>
</tr>
</tbody>
</table>
THOUIN et SORIN - Aspects centraux de la fatigue auditive

- une augmentation significative du temps de traitement pour les bonnes réponses:

<table>
<thead>
<tr>
<th></th>
<th>Sans FA</th>
<th>Avec FA</th>
<th>Test t</th>
</tr>
</thead>
<tbody>
<tr>
<td>Temps moyen de réponse</td>
<td>842 ms</td>
<td>1013 ms</td>
<td>t(12)=3,09 p&lt;0,01</td>
</tr>
</tbody>
</table>

Si l'augmentation significative du nombre d'erreurs en présence de FA peut être éventuellement attribuée à une diminution de l'intelligibilité des phrases et donc à des dégradations du système auditif lui-même, par contre, l'allongement des temps de réponse suggère nettement l'existence d'une composante centrale non-auditive dans les effets postérieurs de l'exposition au bruit.

En effet, pour fournir un avis sur l'existence ou non d'une relation sémantique entre le mot-sonde et la phrase, le sujet doit effectuer une comparaison entre la signification du mot-sonde d'une part et la trace laissée en mémoire à court terme de la signification de la phrase d'autre part. L'allongement de ces temps de comparaison peut être interprété comme une diminution de l'accessibilité des représentations sémantiques engendrées par la phrase. Un tel phénomène ne peut s'expliquer par les simples modifications introduites dans le système auditif périphérique par l'exposition au bruit.

DISCUSSION:

L'existence d'une fatigue auditive centrale après exposition à un bruit a été discutée chez les psychoacousticiens et chez les psychologues. Les premiers ont tenté, par des mesures controlatérales essentiellement (pour une revue voir GRAUER et DUNN 1980), de déceler la présence d'altérations dans le fonctionnement des centres auditifs après exposition au bruit. Bien que certains résultats expérimentaux divergent, il semble actuellement justifié, en accord avec certaines mesures physiologiques (KLEIN et MILLS 1981 par exemple), d'admettre que la cochlée n'est pas le seul lieu où la FA s'exerce. Ceci a peut avoir des conséquences importante sur le dépistage de la FA, la présence d'une élévation temporaire du seuil d'audition (TTS) n'étant pas nécessairement le meilleur indice de dépistage de cette FA. On a en effet observé que certains mécanismes comme la sélectivité en fréquence pouvaient être dégradés après exposition au bruit même dans le cas où l'on ne décelait aucun TTS significatif (PETH et al 1979).

Les psychologues, quant à eux, ont montré que diverses tâches pouvaient voir leurs scores de réalisation décroître après exposition au bruit (GLASS et SINGER 1973 par exemple). Ils montrent que ces dégradations dépendent non seulement des caractéristiques physiques du bruit fatigant (et donc des dysfonctionnements purement auditifs qu'il peut provoquer) mais aussi et parfois dans des proportions équivalentes, de facteurs tels que la prédicibilité du bruit et la possibilité d'exercer un contrôle sur ce bruit.
Dans la série d'expériences évoquées ici, on a tenté de lier les deux approches psychoacoustiques et cognitives pour faire la part, dans les effets postérieurs de l'exposition à un bruit, des phénomènes auditifs (centraux et périphériques) et des conséquences extra-auditives (dégénération de performances). Si des diminutions d'intelligibilité peuvent être dues à des modifications de certaines propriétés auditives (effet de masque et selectivité en fréquence par exemple), la dégradation de la capacité de traitement cognitif complexe, par contre, traite manifestement un effet central non-auditif postérieur à l'exposition au bruit.

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INFLUENCE OF DIFFERENT PRESENTATION PATTERNS OF A GIVEN NOISE DOSE ON HEARING IN GUINEA - PIG

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Most of present standards designed to prevent hearing impairment are based on the equal energy principle. This means, that two noises, regardless of their presentation mode, will injure hearing in the same way, if they contain the same energy. Some of them however apply to impulse noise an energy majoration of 5 to 10dB, which implies, that impulse noise is more hazardous than an equal dose of continuous noise.

This standards are supported by publications of NILSSON et al. and of DAMONGEOT. Our own seemed at first to point in the same direction: e.g. a 127dB SPL band-pass noise presented for 9 seconds did not at all affect the AP-threshold of guinea-pigs, whereas the same noise, cut in 900 10ms-bursts and presented with a repetition-rate of 1 per second, induced about 43dB of TTS. but the hypotheses, according to which this effect is due to the impulse character of the second noise became doubtful when we observed during further measurements, that an isoenergetic continuous noise presented for the same time (900s; 107dB), induces as much TTS as the impulse noise.

We proceeded then to a more detailed study in order to get more data about:

a) the influence of the repetition rate of noise bursts,
b) the influence of the presentation time of isoenergetic continuous noise, on TTS.

Fig. 1 Stimuli used in the first experimental series
For this measurements we chose the following stimuli (figure 1):
- Impulse noise: 900 trapezoid shaped noise bursts with an effective duration of 10ms, cut out of a 127dB SPL noise; the repetition rate ranged from .25 to 32 per second.
- Continuous noise: Noise with a duration between 9 and 3600 seconds, and a level which was adapted to get isoenergetic stimulation (L=101dB SPL to 127dB SPL).

All this stimulations were isoenergetic and had the same spectrum.

![Figure 2](image)

**Figure 2** shows the TTS as a function of the audiometric frequency (a: continuous noise; b: impulse noise). We can see, that the frequency which shows maximum TTS does not coincide with the frequency-range which contains the maximum of energy, but is located about one octave higher. CODY and JOHNSTONE that the frequency of maximum TTS depends essentially on the amplitude of the stimulus. This could explain, that the maximum TTS for impulse noise (the level of all stimulations was the same; 127dB) is always at the same frequency, whereas for continuous noise, this maximum is less sharp and located at lower frequencies. We can also see, that the maximum TTS ranges in the two cases, for different presentation patterns, between 5dB and almost 50dB.

In figure 3 TTS is plotted as a function of exposure time (a: continuous noise, b: impulse noise; parameter: audiometric frequency). For continuous noise and exposure time between 9 and 29 seconds we cannot observe any TTS for any frequency.

For longer exposures, TTS increases quite fast (except for 3kHz), and reaches, for the most hurted parts of the cochlea (7, 8.5 and 10kHz), about about 40dB for 113s (L=116dB SPL). Between 113 and 1800 seconds the TTS grows with a slope of approximately 3dB per octave, as reported by WARD or KRITTER.

For durations beyond 1800 seconds TTS decreases from about 50dB at 1800 seconds to about 20dB at 3600 seconds.
The absence of TTS between 9 and 57 seconds points to the acoustic reflex, which could become less efficient after having been triggered continuously for one minute. This hypotheses seems not realistic, because of:

a) the spectral distribution of the noise, which has been chosen in a frequency range which is almost not affected by the acoustic reflex,

b) additional experiments, during which we suppressed the acoustic reflex by curarizing the animals. The circled symbols in figure 4a show, that this results are not different to those recorded without curare.

This first series of experiments showed, that there was no difference between the maximum TTS which can be evoked by isoenergetic continuous noise or impulse noise (noise bursts) if they have the same spectrum, but that the different presentation patterns could modulate TTS between 0 and 50dB.

In a second experiment we tried to get data concerning the influence of the crest-factor of a "steady state" noise. To do this, we submitted animals to two different stimulations, which had the same spectrum but very different time-functions. This stimulations were (fig. 4):

a) short insulated pre-sure-impulses, with a duration of approximately 200μs. The repetition rate was 100 per second. This impulses were presented. They were presented for 20 minutes at RMS-levels of 103, 106 and 109 dB SPL. (Following the standards, this signals have to be
treated like continuous noise.)

b) a continuous noise which was presented for 20 minutes at RMS-levels of 106, 109 or 112 dB SPL.

**Figure 5** TTS as a function of frequency for different stimulation levels

Figure 5 shows the TTS as a function of audiometric frequency (a: pressure impulses; b: continuous noise; dashed line: spectrum of the stimulation). Once more we can realize that the most affected areas correspond to a frequency which is about one octave higher than the stimulation. Like in the first series, the continuous noise seems to affect a larger part of the cochlea than the pressure impulses.

The maximum TTS as a function of stimulation level is plotted in figure 6 (a pressure impulses; b continuous noise). This figure shows that the continuous noise must have a 3 dB higher level than the pressure impulses, to produce the same effect.

In the present state of knowledge, isoeenergy can be quite a good indicator for the maximum effect of noise. To get better and more complete indicators, a simple distinction between continuous and impulse noise does not seem very useful, because there are other parameters, like the couples level - duration, or repetition rate - peak level (for impulse noise), which seem to be necessary to describe the physiological effects.
IMPULSE NOISE EXPOSURE CRITERIA AND THEIR APPLICATION

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Introduction

Serious progress towards damage-risk criteria (DRC) for weapon noise exposure began with the pioneering studies of Murray and Reid (1946). With the advent of transducers, etc., capable of providing better impulse noise measurements it became possible to look for relations between temporary threshold shifts (TTS), measured with an audiometer, and impulse noise exposure parameters. TTS studies were conducted in many laboratories on the assumption that there must be at least an ordinal relation between temporary and permanent hearing losses. (Today, in 1983, that issue is still open for discussion!)

At least four DRC for the auditory effects of impulse noise exposure have been published so far: Coles, Garinther, Hodge and Rice (1968); CHABA (1968); Pfander (1975); and Snoorenburg (1980). All of these DRC have two limitations: (1) they are based on noise parameters that were easy to measure in the 1960's, rather than on the characteristics of the auditory system; and (2) they are based almost exclusively on exposures to small arms' noise. Despite these shortcomings, the DRC have been used in weapon design standards, or evaluation criteria, covering the gamut from small arms to artillery, and are being used by the military forces in many countries. Minor differences among the basic criteria are sometimes magnified by differences in the degree of protection which is assumed to be provided by earplugs, earmuffs or other types of hearing protective devices.

This paper summarizes the main features of the several impulse noise DRC, and compares the predictions that result from application of the DRC to real weapon noise data. The implications of various assumptions about hearing protection are explored. Comments are also included about topics that should be addressed in developing better impulse noise DRC in the future.

Current Impulse-Noise Damage-Risk Criteria

A. Coles, Garinther, Hodge & Rice (1968). This DRC is based on the same assumptions about the orderliness of TTS as were assumed in the CHABA (1965) DRC for standy and intermittent noise. The CHABA limits on TTS measured 2 minutes after exposure (10 dB at or below 1 kHz; 15 dB at 2 kHz; 20 dB at or above 3 kHz) were applied to develop a limit curve for
exposure to 100 impulses per day that would protect 75% of exposed ears. The important impulse parameters were peak pressure level and B-duration, an envelope duration measurement based on pressure fluctuations that are within 20 dB of the peak pressure level.

B. CHABA (1968). The CHABA Committee essentially adopted the Coles, et al., formulation (see above) but lowered the limit curve by 10 dB: 5 dB was subtracted to make the curve apply to normal incidence exposures, and another 5 dB to increase the level of protection to 95%. The limit curve was flattened at 200 msec to account for the action of the acoustic reflex. CHABA also proposed a correction for number of impulses per day based on a 5 dB change in peak level for a 10-fold change in number. The CHABA DRC was subsequently incorporated into U.S. Army MIL-STD-1474 (1979), in which the basic limit curve was increased by 29 dB to account for the protection afforded by ear plugs.

C. Pfander (1975). Pfander's approach differs from the two DRC discussed above. His DRC is based on TTS measured 24 hours after exposure; it assumes that any exposure from which there is 95% recovery in 24 hours is acceptable. TTS need not be measured immediately after exposure, and no allowable hearing losses are identified with specific frequencies. The impulse noise parameters of importance are peak pressure level and C-duration. (C-duration is the sum of all the positive and negative spikes that are within 10 dB of peak level; it is not an envelope duration.) The abscissa of his limit diagram is labeled "effective duration" which is obtained by multiplying the C-duration of a single impulse by the number of impulses in a day's exposure. As applied to protected exposure situations, Pfander assumes 30 dB of attenuation for good ear plugs (e.g., Comfit).

D. Smoorenburg (1980). In the Netherlands, Smoorenburg's approach combines some of the features of both the CHABA and Pfander criteria. The allowable TTS is 15 dB averaged over the frequencies of 1, 2 and 3 kHz, in 10% of the population. The limit diagram is based on peak level, and on D-duration, an envelope-type duration measurement (similar to CHABA), but based on the 10 dB down points (similar to Pfander). The Dutch are more conservative in their assumptions about earplug attenuations, allowing 15 dB for small arms and 14 dB for large-caliber weapons (also 23 dB for earmuffs, and 30 dB for double protection).

Published Comparisons of the Noise Criteria

Pfander, et al. (1980) reported on tests in which detailed examination was made of the TTS recovery of nearly 500 soldiers. The test conditions included the noise of small arms, rocket launchers, and mortars, as well as both protected and unprotected exposure. The analysis of the impulse noise parameters included peak level and both B- and C-durations. They compared the predictions of the Pfander and CHABA DRC, both in terms of TTS measured immediately after exposure and that measured 24 hours later. In addition they contrasted the relative hazard predictions of the two DRC based on use of a 30-dB hearing protector. The overall conclusion was that the CHABA DRC is too conservative, as it would prohibit noise exposures which the German data indicated did not cause excessive TTS.
In Smoorenburg's (1980) review, he transformed all the noise measurements to peak level and D-duration, and all the TTS's to his acceptable maximum of 15 dB averaged over the frequencies of 1, 2 and 3 kHz. He then used this database to formulate a DRC for unprotected exposure and concluded with a comparison of the CHABA and Pfander DRC. He notes that his DRC curve (actually a straight line) is parallel to but 1.4 dB lower than that of Pfander. The CHABA DRC, by contrast, has an entirely different slope and the lines for different numbers of impulses per day do not coincide. The Dutch DRC is more conservative than CHABA for a very short total D-duration, but less conservative than CHABA at the longer exposure durations per day.

Comparative Assessment of Impulse Noise Data

With the possible exception of a few small arms, all weapons' noise characteristics require at least single hearing protection for safe use. So comparison of the predictions resulting from applying the DRC to real data will be based on protected exposure. For this purpose it should be noted that MIL-STD-1474 assumes 29 dB of protection for ear plugs, Pfander 30 dB, and Smoorenburg 15 dB. The following table presents the allowable number of exposures per day for five generic types of weapons.

<table>
<thead>
<tr>
<th>WEAPON</th>
<th>PEAK (dB)</th>
<th>DURATION (msec)</th>
<th>ALLOWABLE DAILY IMPULSES</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>B</td>
<td>C</td>
<td>D</td>
</tr>
<tr>
<td>Small Arm</td>
<td>155</td>
<td>5.4</td>
<td>1.0</td>
</tr>
<tr>
<td>Rocket</td>
<td>179</td>
<td>13.2</td>
<td>0.6</td>
</tr>
<tr>
<td>105 How</td>
<td>174</td>
<td>18.5</td>
<td>2.5</td>
</tr>
<tr>
<td>155 How</td>
<td>180</td>
<td>34.8</td>
<td>5.6</td>
</tr>
<tr>
<td>Recoiless</td>
<td>186</td>
<td>17.8</td>
<td>0.8</td>
</tr>
</tbody>
</table>

These data confirm Pfander's contention that MIL-STD-1474 (i.e., CHABA) is more conservative than his DRC, as in all but one case his DRC allows a larger number of daily exposures. By contrast, Smoorenburg's DRC allows no exposure at all to large caliber weapons with earplugs alone. To determine the effect of earmuffs, for which Smoorenburg assumes 23 dB of protection, the last column of the table was recomputed with the following results for earmuffs: Rocket 3; 105 How 2; 155 How 0; and Recoiless 0. So to get a realistic number of allowable exposures per day with the Smoorenburg DRC requires double hearing protection, for which 30 dB of protection is assumed for large caliber weapons only.

Limitations of Current DRC

The foregoing discussion suggests that we have not yet arrived at the perfect DRC for impulse noise exposure. There is growing evidence that the spectrum of impulses and, possibly, a critical level as well, are important factors in assessing the risk of hearing loss. It appears that large-caliber weapons (long durations but low in frequency) may be less hazardous than small arms (short durations but high in frequency) despite the fact that all current DRC predict exactly the opposite! G.R. Price (1983) argues this point in another paper at this Congress.
None of the existing DRC provide a means of combining noise exposures. Impulse noise frequently occurs in a background of potentially hazardous steady or intermittent noise; no existing DRC can predict the effects of combined steady and impulse noises. Likewise, no existing DRC can combine impulses having different levels and/or durations into a single risk prediction. This is a particularly important problem for artillery crews, as a day’s firing nearly always involves a mix of charges, each with its own unique noise characteristics. The attenuation of hearing protectors is a key issue, especially the standardization of measurement techniques for determining the impulse noise attenuation values. In Europe there is a great deal of interest in using tiny microphones to measure impulse characteristics inside the ear canal; this technology does not seem to have spread to the U.S. so far as impulse noise is concerned. The selection of ear protectors for military field applications is also of great concern; for one thing the new foam type of earplug has a number of significant drawbacks, despite its apparently superior attenuating characteristics. And the effect of interval between impulses needs to be more completely addressed in future DRC formulations.

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A DAMAGE-RISK CRITERION FOR IMPULSE NOISE BASED ON A SPECTRALLY DEPENDENT CRITICAL LEVEL

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Introduction

Damage-risk criteria (DRCs) used for rating hazard from intense impulsive sounds (CHABA, 1968; Coles, Garinther, Hodge & Rice, 1968; Pfander, 1975; and Smoorenburg, 1980) differ in their details; but are consistent in several respects. They use a measure of peak pressure combined with a measure of duration to rate effect. Consequently, they generally agree that for any given pressure, long duration impulses, such as those produced by cannons, are more hazardous than shorter impulses such as those produced by rifles. The DRCs also share a common bond in that they have been derived empirically rather than theoretically.

An analysis of the ear's response to intense sounds has focussed attention on two salient points (Price, 1981). First, where susceptibility to damage from intense sounds is concerned, the ear is spectrally tuned, showing greatest susceptibility in the mid-range. Secondly, at high levels, there is a critical level (CL) at which the damage mechanism becomes mechanical stress within the organ of Corti. Taken together, a spectrally dependent CL threshold has been calculated and is reproduced in Figure 1.

![Figure 1. The Relative sound pressure level reaching the CL for human ear (from Price, 1981).](image-url)
It can be seen from Fig. 1 that the susceptibility of the ear is relatively sharply tuned with its peak of sensitivity at about 3.0 kHz. A low frequency slope of about 6 dB/oct and a high frequency slope of 18 dB/oct describe the CL curve relatively well.

Many weapons produce impulses with Friedlander-like waveforms (pistols, rifles, mortars, cannons, etc.) and as a consequence, their spectra all look similar, differing primarily in the location of the spectral peak. Given the spectral similarities and the CL curve, two interesting predictions follow for an ear exposed to an impulse of this type (Price, 1982). First, because the susceptibility curve is more sharply tuned than the spectrum of the impulse, loss is predicted to occur first in the mid-range, regardless of the location of the spectral peak. Second, relative susceptibility is predicted to vary as a function of peak pressure and spectral peak or A-duration (duration of first positive pressure excursion) as shown in Fig. 2.

![Figure 2. The relative hazard from weapons impulses with differing spectral peaks/A-durations. Typical rifle and cannon data are shown as are the ratings by current DRCs.](image)

Fig. 2 shows that impulses with their spectral peaks in the mid-range should be most hazardous and as the spectral peak gets lower, the impulse should be progressively less hazardous. This latter prediction is especially significant because it runs contrary to the predictions from all existing criteria (the shaded area in Fig. 2) and forms the basis for a crucial experiment.

**Experiments**

An experiment based on the foregoing prediction was conducted using a single exposure to 60 rounds of either cannon or rifle fire, 38 cats (76 ears) as the experimental animals, and electrophysiological measures of
loss (Price, 1983). In order to increase the number of ears exposed to
cannon fire (105mm Howitzer), increase the range of sound pressures, and
get histological measures of loss, an additional experiment has been run
with 28 additional animals (56 ears).

Result and Discussion

Figures 3 and 4 show the losses measured the day of exposure (within
1 to 5 hours after exposure) for both the cannon and rifle impulses.
The pattern for both is similar, and for each weapon the data match expec-
tations, i.e., loss increases with increasing intensity and the range
of loss grows with higher losses. The animals were allowed to recover
for 2 or more months and sensitivity was redetermined. After recovery,
both groups of animals again looked similar, recovery being about the
same for both groups. Histological results paralleled the electrophysio-
logical data.

Figure 3. Threshold shift following 60 rifle rounds.

Figure 4. Threshold shift following 60 rounds from a cannon.
From a theoretical standpoint, the really important question was that of the relative hazardousness of the two types of impulse. The least-squares regression lines in Figs. 3 and 4 indicate that ears began to lose sensitivity at about 137 dB for the rifle and 146 dB for the cannon, a 9 dB difference. The regression lines fitting the data after recovery also support this relationship. Given that the A-durations and spectral peaks were about 3 octaves apart, the prediction from the diagram in Fig. 2 is met.

One attractive possibility for rating hazard might be that an A-weighted energy measure would provide the appropriate spectral weighting and be both accurate and easily done. Unfortunately, such measures, when calculated for the rifle and cannon impulses, were in error by about 15 dB.

The major finding, of course, was that the cannon impulse was less hazardous than the rifle impulse, exactly the reverse of the prediction of present DRCs. The prediction of lessened hazard from low spectral frequency impulses has been supported by data from a number of different investigators and for a variety of species, including the human (Dancer, Franke, Lombardo, and Pujol, 1980; Patterson, 1982; Price, 1978). Given that a solid data base continues to grow and mathematical modeling of the loss process at high intensities improves, it is reasonable to suppose that a new, theoretically based, DRC for impulse noise can be developed in the next several years.

References


THE ONSET OF HANDICAP DUE TO NOISE-INDUCED HEARING LOSS

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To set legally enforceable limits for occupational noise exposure, a balance has to be struck between the effects on those exposed and the costs and organization needed to reduce their noise exposure below the limiting value. Here we review relevant aspects of the relation between noise and noise-induced hearing loss (NIHL). The review is prepared in the context of an on-going study of handicap due to NIHL, and the paper also provides an outline of the experimental approach that is being employed.

TERMINOLOGY Various interpretations of the terms impairment, disability and handicap are evident in the literature. Here we adopt the definitions of the World Health Organization (1) as indicated in the top row of the diagram. Noise can give rise to a chain of effects, but at each stage the linkage will be of variable strength - or even absent - as indicated by the diverging arrows. Although exposed to similar noise some individuals, being less susceptible, suffer less impairment than others; people with essentially the same hearing impairment may suffer different disabilities due to their varying capacities to comprehend auditory information; and those with the same disability may suffer very different handicaps because of their different personalities and life styles. The three aspects of NIHL are measured in different ways (row 2 of the diagram), showing a general trend from the biological manifestations through the psychological to the sociological domain.

PREVIOUS RESEARCH There are extensive data relating noise exposure to impairment of auditory function as measured by pure-tone audiometry. Studies have also shown that damage to the cochlea reduces the ear's ability to discriminate one sound in the presence of others (frequency selectivity) and to respond to rapidly changing sounds (temporal resolution) but definitive data on these aspects are lacking. However, it is likely that continuing research will uncover reliable measures of these and other impairments, as there already is for loss of sensitivity to faint sounds.

The test of performance most often used in the laboratory and the clinic is speech audiometry, employing lists of words or sentences presented over headphones in quiet conditions. Sometimes competing noise is added to 'sensitize' the test, and occasionally free-field listening, reverberation or signal shaping is used, either to provide greater realism or just to make the test more difficult. There are clearly other aspects of disability such
Terminology & Causality

Method of observation

Example

Noxious agent

Physical measurement

Noise (and/or age)

Impairment: loss or abnormality of structure or function of an organ

Biological or physiological examination; Subjective tests

Cochlear degeneration

Threshold elevation

Loss of frequency and/or temporal resolution

Loudness recruitment

......

Disability: inability to perform normal human functions

Tests of task performance

Inability to:

- hear speech
- hear other sounds
- localize sources
- resolve competing messages

......

Handicap: limitation of an individual's rôle fulfilment

Questionnaires;

Monitor behaviour in social environment

Inadequacy

Withdrawal

Alienation

Change of life style

......

......

Syllogisms:

Idealistic

Cause

Permitted exposure

Effects ..............

......

......

Criterion selections

Administrative target

Cause

Permitted exposure

Effects ......

......

......

Criterion selections

Practical realization

Noise immersion

Ageing

Population distribution of threshold shifts

Threshold shift averaged over frequency

Speech discrimination loss

Equivalent audiometric criterion e.g. av. HL at 1,2,3 kHz < 30 dB

Criterion: retention of conversational listening ability (at given age)

Permitted noise immersion

Criterion: % persons not covered (minus % due to age alone = 'risk')
as perception of non-speech sounds, resolving competing messages and localizing sound sources which have rarely been measured in this context.

Numerous attempts have been made to determine an exact relation between the predominant measures of impairment and disability, i.e. the pure-tone audiogram and speech discrimination score. Noble (2) reviewed 23 studies of this type and indicated that the results did not consistently establish an association between the two measures, and that in general the correlations were relatively weak. Given that speech perception involves auditory and cognitive processes not involved in the detection of pure tones and that there are many aspects of impairment besides loss of sensitivity at threshold, this observation is not particularly surprising.

There have been relatively few large scale studies of hearing handicap. Various questionnaires have been developed and indices of handicap proposed (3 - 9) but up to now these have not been administered uniformly to sufficient numbers of subjects to lead to a standardized technique (10). Moreover the questionnaire approach has not been systematically validated with studies of observed behaviour, and comparisons with measures of disability and impairment are sketchy. If the object of noise regulation and hearing conservation is to procure the avoidance of hearing handicap - as has often been stated - there is pressing need to determine these relationships as precisely and realistically as possible.

SPECIFYING PERMISSIBLE EXPOSURE The lower half of the diagram illustrates three syllogisms for setting noise exposure limits in the context of various effects of NIHL. Ideally the limit would be based on a specific degree of handicap in a certain proportion of the population at risk - a criterion of this kind being unavoidable since no system of prevention can cover 100% of cases - and this requires a specification of how handicap is to be measured, and of what is an acceptable 'size' of the change from normal. This approach has never been implemented because of difficulties in linking the noise exposure to any observable changes and the mediating role of idiosyncratic factors.

A more readily achieved objective would be to set the limit on the basis of an appropriate set of task performance tests. It could then reasonably be assumed that the larger the measured disability, the greater on average the resulting handicap.

In practice the approach follows the tortuous path shown at the bottom of the diagram. It relies on an index derived from the pure-tone audiogram, but sets a limit by reference to the weakly correlated measures of speech discrimination in quiet, and more recently in noise. A criterion of maintaining listening ability for conversational purposes, taking age into account, can be applied in a general manner, but an exact specification for this is elusive because of the wide variety of possible conditions. The current position is that the link between noise exposure and pure-tone audiogram is standardized (11) but this has been achieved only by rather crude averaging of disparate results from various studies. The link from audiogram to speech test performance, measured for example by the discrimination score for monosyllabic words, cannot be agreed internationally. Even if these links were perfected, the present approach would fall far short of covering the whole problem of preventing disability and handicap, as will be evident from the diagram.
RESEARCH PROGRAMME. The specific focus of our study is the onset of handicap rather than its later development or more severe manifestations, and accordingly the experiments are designed to elucidate the differences between normals and those with a mild degree of NIHL.

Impairment is being assessed by a battery of audiological tests including pure-tone audiometry, frequency selectivity and temporal resolution.

Disability is being determined in simulations of everyday listening situations. The essential acoustical and visual elements of listening to speech at a social gathering, over a public address system, in a motor vehicle and at a public meeting are re-created in a laboratory setting. Performance at these tests is measured both in the conventional manner (as a percentage of the material correctly reproduced) and in terms of the subjects' ability to answer questions relating to the messages conveyed. In addition, conventional speech audiology in quiet is also administered.

Handicap is assessed by questionnaires. One section obtains a general self-assessment through a series of questions similar to those used in previous studies. A second section elicits attitudes to nine kinds of communication situations (domestic, social and public) in some detail; these include familiarity with the situations described, self-assessed ability to cope, any particular difficulties encountered, and the relative importance of such difficulties in the subjects' everyday lives. Some of these situations correspond to the laboratory simulations described above. In these cases, a third section of the questionnaire assesses subjects' reactions to the simulations with the object of uncovering any changes in attitude to the general situation given in answer to the earlier questions, as a direct consequence of actually experiencing a particular situation of the same kind.

This three-pronged study should provide the following information:

(a) the starting point of disability by self-assessed and measured performance in a range of listening situations of common occurrence

(b) the onset of self-assessed handicap and its relationship to disability as determined by the two approaches in (a) above

(c) measures of impairment of hearing which individually or in combination act as predictors of the onset of handicap.

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DIFFERENCES IN ITS AFTER EXPOSURES TO EXPECTED AND UNEXPECTED NOISE

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INTRODUCTION

As is well known there is a surprisingly large individual variation in sensitivity to noise. The variation can obviously depend on several factors such as transmission characteristics of the middle ear, different susceptibility of the sensorineural elements in the cochlea, differences in the cochlear circulation etc (Ward, 1967, 1976; Jansen, 1970; Sandén & Axelsson, 1981). One possible factor which might contribute to such individual differences in development of permanent threshold shift could be variations in predictability of harmful noise. There are some studies that indicates a relation between the predictability of the noise and the effects on hearing. It has been shown that the acoustic reflex (AR) could be elicited before the onset of an impulsive noise and that elicitation of the AR could be recorded even when the impulse-sound-creating weapon mis-fired (Brasheer et al., 1969). Similarly, Marshall et al. (1975) found an anticipatory effect of the AR to toys when these were presented visually or before the onset of the acoustical stimulation. Both these investigations suggest a possible protective effect of the AR, elicited before the onset of the impulsive sound. In the reviewed experiments no attempts were made to establish temporary shifts in hearing thresholds in order to objectively demonstrate a possible protective effect of the predictability.

The aim of the present study was to determine individual differences in ITS after multiple exposures to predictable and unpredictable noise.

MATERIAL AND METHOD

The experimental subjects were eight voluntary males and two females with a mean age of 20.7 years (range = 17 - 37 years). Each individual showed normal hearing on manual pure tone audiometry in the frequency range 250 - 8000 Hz. In addition, middle ear pressures were within 0 - 0.25 kPa for all subjects confirmed by tympanometry.

The noise exposure consisted of 100 one third octave band filtered noise bursts with 4000 Hz center frequency. The duration of each burst was 0.5
sec. In order to obtain two "identical" exposures with different degrees of predictability, all subjects were exposed as "producers" at the first session. The subjects were instructed to press a button when a red lamp went from flashing to steady light. When the button was pressed the noise burst was immediately delivered to the subject's left ear. The time between the onset of flashing as well as the number of flashes prior to steady light was randomized within each "producer"-session and controlled by a computer which also supervised the whole session and checked that the button was properly handled. Each session lasted 10 min. The noise bursts were presented at individually determined levels in the range 100 to 113 dB SPL. The individual time sequence of the "producer"-sessions was stored in the computer and used to produce a subsequent "consumer"-session where the subjects were exposed to an unpredicted but identical noise dose from the computer-stored "producer"-data. Each subject participated on subsequent days in five producer- and five consumer-sessions.

At all sessions each subject's pre- and post-exposure hearing thresholds on the left ear were established with a computerized sweep-frequency audiometer (type Békésy) in the frequency range 4000 - 8000 Hz (Ivarsson et al, 1980). In order to minimize variations caused by misplaced earphones the test tone was delivered to the subject over a test-fixture consisting of a TDH-39 ear-phone attached to an ear speculum (Rölandsson et al, 1980). Pre- and post-exposure hearing thresholds were calculated by a computer from the sweep recordings at test frequencies 4000, 5000, 6000, 7000, and 8000 Hz.

The post-exposure pure tone threshold determination started at 4000 Hz 2 minutes after cessation of the exposure (TTSp). All hearing tests were carried out in a sound-proof booth and conducted by the same operator at all sessions.

RESULTS

Evaluation of collected TTS-data from the 100 sessions (10 subjects exposed five times as "producers" and five times as "consumers") revealed that 6000 Hz was the most TTS-sensitive frequency for both test-conditions. The mean TTSs for all "producer"- and all "consumer"-sessions did not show any difference (Fig. 1). However, inter-individual differences in TTS were pronounced. Individual analysis demonstrated almost uniform shapes and equal amounts of TTS at the two test-conditions and the mean intra-individual standard deviations over sessions were below 5 dB at all test frequencies.

In order to determine if there were systematical differences of individual TTSs at the two test-conditions and taking into consideration the fact that each subject here became his own control, we calculated the total mean paired session-differences. At each test-frequency each individual's "producer"-TTS was subtracted from his "consumer"-TTS. If the intra-individual variation in TTS between the two stimuli was systematically shifted, this measurement should deviate from zero. However, the mean paired session-differences were not at any test-frequency significantly shifted from zero.
Lindgren and Axelsson, TTS after expected and unexpected noise

**DISCUSSION**

Hypothetically, individual variation in susceptibility to noise could be influenced by the predictability of the noise. This could explain differences in susceptibility frequently encountered in workers exposed to equal noise-doses. However, the present study does not support the notion that variation in hearing is due to variation in predictability of noise.

The AR is one uncontrollable cause of variation and therefore we used a stimulus with a center frequency of 4000 Hz in order to exclude the influence of AR-activity. It has been shown that the attenuating effect of the AR is mainly located at frequencies below 2000 Hz and that the attenuating effect at and above 4000 Hz could be considered insignificant (Borg, 1968). We may then conclude that when the exposure stimulus does not interact with the AR, TTS is not likely to be influenced of the predictability of the noise.

From a technical point of view the present reactions to predicted and unpredicted noise could have been different if the actual noise-bursts had been louder and particularly if the sound had been of impulsive type (e.g., gunfire). However, regarding the amount of TTS that the subjects showed with the actually adopted noise, it would be unethical to increase the sound levels of these noise-bursts.

It has been shown in many psychological studies that anticipation effects influences many reactions in the body. Many stressful events could be counteracted and diminished by anticipatory effects. From this point of view we had expected a difference in TTS depending on whether the noise
was predicted or not. The negative findings suggest that whether the noise is predicted or unpredicted there may be no difference in development of TTS or some experimental defects may have elapsed our notion.

In conclusion, it appears that an unpredicted sound and a predicted, self-elicited sound, result in the same amount of TTS under experimental conditions. However, in a real life work-situation there are numerous factors including work-load, stress, motivation and AR-activity that could interact with predictability and influence the hearing damaging effect of noise.

ACKNOWLEDGEMENTS

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REFERENCES


EFFECT OF NOISE ON EKG WITH COMPUTER ANALYSIS

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Since 1979, we investigated thousands of workers exposed to occupational noise (≥90 dBA) for more than ten years, and found that most of them complained neurological symptoms. These results are compatible with that of other authors and suggest that the noise is able to influence the function of the brain and heart. But the problem is not concluded, for the reason of that a scientific conclusion must be verified by objective evidences (such as the physiological indexes), and the results obtained by us (and the other authors also) are singly the subjective complaints. We concerned that the computer analysis of EKG and EKG would be a hopeful way to discovered some objective indexes for evidencing the effect of noise on the function of the brain and heart, and to make the problem concluded.

According to this concept, we undertaken a series of studies on the effect of noise upon EKG with computer analysis undertaken by us in recent two years.

Although the history of EKG computer analysis used to diagnosis is more than twenty years, the advances are not quiet ideal clinically. We think that the reason of thus would be attributed to the method of EKG computer analysis in previous being restrained by the concept of routine analysis method of EKG. If we forsake the concept of routine method, regard the heart as a specific machine, and analyse the EKG with information processing technique according to the concept of bio-cybernetics, we should discover some available indexes of heart dysfunctions, which would be useful for approaching the influence of noise on the heart.

Method

Subjects
The subjects divided into four groups:
1. Control group 37 normals.
2. Noise group 74 workers exposed to occupational noise (65–100 dBA) for more than ten years.
3. Myocardial occlusion group (with normal EKG*) 15 patients suffered myocardial occlusion half year ago, now their EKGs are normal with routine method.
4. Myocardial occlusion group (with abnormal EKG*) 20 patients whose condition is like to group (3), but their EKGs are abnormal*.

All the subjects mentioned are with normal eardrums, no history of events affected to the hearing ability and heart function other than the occupational noise exposure and the myocardial occlusion in patients.

EKG recording method

The EKG were amplified and recorded by a four channals electrocardiograph and a TEAC- R210 magnetic recorder, once each subject, in a magnetic proof hut.

EKG computer analysis method

The EKG recorded by the magnetic recorder were inputed to a TO8 medical computer through which the auto spectrum, transfer function, impulse response, coherence and histogram of amplitude were calculated and the results were recorded by a X-Y oscillograph.

The parameters of computing are as follows: the sampling time is 10 ms, but that of the histogram is 100 μs; each result was calculated by 15 computing sections; the gain of computer equals to 1, but that of the histogram is four. In calculating, the EKG of V5 was used as input and that of standard II was used as output.

Data analysis formula

1. Auto power spectrum \[ G(f) \]
   \[ G(f) = [G_{xx}(f)] = S_x(f) \cdot S_y(f) \]
   \[ S_x(f) \] is the Fourier's transform of the function of EKG, \( S_x(f) \) is its conjugate.

2. Transfer function \[ H_{xy}(f); \phi_{xy} \]
   \[ |H_{xy}(f)| = \left| \frac{G_{xy}(f)}{G_{xx}(f)} \right| \]
   \[ \phi_{xy}(f) = \tan^{-1} \frac{H_{xy}(f) (\text{Imag})}{H_{xy}(f) (\text{Real})} \]
   where \( H_{xy}(f) \) is the amplitude ratio of transfer function, \( \phi_{xy} \) is the phase angle of transfer function, \( G_{xy}(f) \) is the cross power spectrum, which represents:
   \[ G_{xy}(f) = S_y(f) \cdot S_x(f) \]

3. Coherence function \[ \gamma^2(f) \]
   \[ \gamma^2(f) = \frac{|G_{xy}(f)|^2}{G_{xx}(f) \cdot G_{yy}(f)} \]
   \( 0 < \gamma^2(f) < 1 \)

*By using routine method.
4. Impulse response \[ IH_x(t) \]

\[ IH_x(t) = F^{-1}(H_y(f)) = F^{-1}(G_{xy}(f)) \]

5. Histogram of amplitude

That is the histogram of the amplitude of EKG.

Result

The main results obtained were shown in table 1. From the data of table 1 we could find that the frequency of indexes (mentioned in table 1) in the controls is lowest (0%-10.6%), the noise group second (6.4%-41.5%), and the patients highest (33.3%-90.0%), especially the patients with abnormal EKG (by routine analysis method). The result suggests that the indexes developed by us in this paper could be used to diagnose the heart dysfunction and to reflect the effect of noise on the function of the heart.

Discussion and conclusion

1. The effect of noise on the heart has been an awkward problem in the noise research field. Even if many investigations have been run and evidenced the complaint of heart dysfunction being one of the effects of noise exposure, the conclusion has not been settled, because it has not been verified by objective physiological indexes. Many authors have carried out series of studies on this topic, but the advance is quiet not ideal, until now.

2. Table 1 suggest that the indexes developed by us were able to reflect dysfunction of the heart and the effect of noise on the heart, thus it would be applied to the clinical practice and the research work of environmental medicine. The significant difference between control group and noise group, as shown in table 1, suggested that the noise is able to effect on the heart function.

3. Application of computer technique to medicine is one approach of the clinical practice, and also it is a good way for approaching the effects of environmental factors on man. EKG computer analysis would be the better way for developing EKG research and practice. One of its benefits is "objective", that means the result of computer analysis could be quantified, without subjective effects by the doctors. EKG computer analysis (if used appropriately) would increase the correct diagnosing of the heart dysfunction.

4. The results obtained have been examined with a series of studies conducted by us, which may be published in nearly future.

5. Our conclusion supports the theory, that the heart could be analysed by computer according to the method of cybernetics. From this theory we had developed some indexes for diagnosing suggested a new way for applying computer technique to the EKG analysis, and bio-cybernetics to the environmental medicine as well as medical practice.
Table 1  Frequencies of the indexes of EKG computer analysis in the experimental groups

<table>
<thead>
<tr>
<th>Frequency Index (%)</th>
<th>Abnormality of auto power spectrum</th>
<th>Abnormality of histogram of Amplitude</th>
<th>Abnormality of phase angle of transfer function</th>
<th>Abnormality of impulse response</th>
<th>Abnormality of coherence function (1)**</th>
<th>Abnormality of coherence function (2)**</th>
</tr>
</thead>
<tbody>
<tr>
<td>Control</td>
<td>2.7</td>
<td>10.8</td>
<td>5.4</td>
<td>10.8</td>
<td>5.4</td>
<td>0</td>
</tr>
<tr>
<td>Noise</td>
<td>17.1</td>
<td>41.5</td>
<td>12.8</td>
<td>14.9</td>
<td>6.4</td>
<td>12.8</td>
</tr>
<tr>
<td>Myocardial occlusion (with normal EKG*)</td>
<td>53.3</td>
<td>86.7</td>
<td>80.0</td>
<td>86.7</td>
<td>53.4</td>
<td>33.3</td>
</tr>
<tr>
<td>Myocardial occlusion (with abnormal EKG*)</td>
<td>35.0</td>
<td>85.7</td>
<td>85.0</td>
<td>90.0</td>
<td>45.4</td>
<td>40.0</td>
</tr>
</tbody>
</table>

*By using routine EKG method.

**"Coherence function (1)" is the coherence of the frequency at which the amplitude ratio of transfer function is highest.

***"Coherence function (2)" is the coherence of the frequency at which the first peak of amplitude ratio of transfer function exists.

Note: Statistical significant differences were obtained between experimental groups, by x² test (p < .05).
EFFECT OF NOISE ON EEG WITH COMPUTER ANALYSIS

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The effect of noise on the brain function has been suggested by a lot of studies. The main evidences of such studies were that most workers under long term of high level noise complained a series of nervous symptoms, i.e. headache, dizziness, dysmnesia, and insomnia. The results are not concluded, because the view mentioned are basis upon subjective complaints of workers, and have not been verified by objective physiological indexes. It is evident that developing a series of objective physiological indexes for brain dysfunction, which is available to reflect the effect of noise on the brain, is a most important thing for making this problem concluded. We concerned that EEG computer analysis could be a hopeful way for discovering such indexes, if we used it appropriately.

The series of experiments were run in past three years.

Series I

We recorded the spontaneous potentials (EEG) and visually evoked potentials of 39 workers (noise group) who were exposed to 75, 85, 90, 95 dBA occupational noise exposure respectively for more than ten years. The EEG echoes were amplified in the first instance with an electro-encephalograph and recorded through a magnetic tape recorder, then analysed by a medical computer (7206) to obtain auto power spectrums, transfer functions, coherence functions and impulse responses. Two recording electrodes were arranged at the occipital lobes bilaterally (O1, O2). A white flicker (1Hz) was used as the stimulus of evoked EEG.

We used EEG functional index as a physiological index to reflect the effect of noise on the brain function. The definition of EEG function. The definition of EEG functional index (EFI) is:

\[ EFI = \frac{g_{xy}(f) \times \log |\phi_{xy}(f)| (\text{absolute}) \times \gamma(f)}{G_{xy}} \]

Where \( H_{xy}(f) \) is the amplitude ratio of transfer function, \( \phi_{xy} \) is the phase angle of transfer function, \( \gamma(f) \) is the coherence function, \( G_{xy} \) is the EEG power spectrum. All values in the formula referred to the main frequency of the left occipital lobe.
We also used the abnormality of impulse response as another index of brain dysfunction.

The main results obtained were shown as in table 1-2. From the data of table 1, we could see that there are a linear (inverse) relation between the noise level and EFI. That of the spontaneous EEG could be shown as follows:

\[ Y = 6.65 - 0.06 L_a \]

where \( Y \) is the EFI, \( L_a \) is the level of noise exposure (dBA).

Table 2 shows that the frequency of abnormality of impulse response in the controls are zero, and that the frequency in noise groups increases with the increase of noise level, when the noise level is higher than 65 dBA.

In order to test whether the EEG functional index is correlated to information process of the brain, we investigated into the nervous symptoms complained by the subjects, and found that 5.8 was the average number of nervous symptoms complained by the subjects whose spontaneous EEG functional index \( \leq 0.8 \), and that of the subjects whose EEG functional index \( > 0.8 \), and was only 3.4, the difference was statistically significant (by F test, \( P < 0.05 \)).

The results obtained not only suggested that the adverse effect of occupational noise exposure on the brain function was precisely reflected and suggested that EFI was significantly correlated to the functional state of the brain, the lower the EFI, the worse the functional state of the brain, with the more the number of nervous symptoms complained.

**Series II**

**Method**

We recorded spontaneous and visually evoked EEG of 37 normals (control group), 17 patients (7 automatic epileptics and 10 sequelae of cerebral concussion) and 62 workers (noise group who exposed to 65-100 dBA occupational noise exposure for more than ten years). The EEG were amplified and recorded by a Galileo ESA electro-encephalograph, and a TRAC magnetic data recorder. The recording electrodes were placed on bilateral Broca's canters. A white flicker (1 Hz) was used as the visual stimulus of evoked EEG.

The EEG recorded by the magnetic recorder were imputed to a 7TOS medical computer through which the auto power spectrum, transfer function, impulse response, coherence function and histogram of amplitude were calculated, and the results were recorded by a X-Y oscillograph.

The parameter of computing are as follows: the sampling time is 10ms, but that of the histogram is 100 μs; each computing section consists of 1024 sampling points; each result is calculated by 15
computing section; the gain of computer equals to 4, but that of the histogram is 52. In calculating, the EEG of left Broca's Center is used as input and the right Broca's center as output.

**Result**

The main result obtained is shown as in table 3. Table 3 suggests that the frequency of EEG computer analysis indexes in the controls is lowest (0-76%), in the patients is highest (17.6-100%), and the frequency of the noise group (8.0-73.%) is just between the 2 groups. From the fact mentioned, we could conclude that the EEG computer analysis indexes developed by this paper are able to reflect the brain dysfunction, by using these indexes the adverse effect of noise on the brain function is evidenced.

**Discussion and conclusion**

1. Through the computer analysis of EEG, we developed a series of indexes, which are able to reflect the brain dysfunction and the effect of noise on the brain, it was shown as in table 1-3. By using these indexes, the adverse effect of occupational noise exposure on the brain is evidenced precisely.

2. The indexes developed by this paper could be used to clinical diagnosis. It would be helpful for doctors to diagnosing brain dysfunctions more easily and more early than the routine analysis method of EEG. This conclusion has been verified by us in some other studies, which will be published in nearly future.

**Table 1. The relation between the EEG functional index (EFI) and the level of occupational noise exposure**

<table>
<thead>
<tr>
<th>Noise level dB(A)</th>
<th>Average value of EFI (spontaneous EEG)</th>
<th>Average value of EFI (visually evoked EEG)</th>
</tr>
</thead>
<tbody>
<tr>
<td>75</td>
<td>2.206</td>
<td>0.926</td>
</tr>
<tr>
<td>85</td>
<td>1.559</td>
<td>0.510</td>
</tr>
<tr>
<td>90</td>
<td>1.341</td>
<td>0.423</td>
</tr>
<tr>
<td>95</td>
<td>1.071</td>
<td>0.365</td>
</tr>
</tbody>
</table>

**Table 2. Influence of the level of occupational noise exposure on the frequency of the abnormality of impulse response**

<table>
<thead>
<tr>
<th>Groups</th>
<th>Frequency of abnormality of impulse response (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Controls</td>
<td>0</td>
</tr>
<tr>
<td>75 dBA</td>
<td>0</td>
</tr>
<tr>
<td>85 dBA</td>
<td>3.7</td>
</tr>
<tr>
<td>90 dBA</td>
<td>5.7</td>
</tr>
<tr>
<td>95 dBA</td>
<td>14.5</td>
</tr>
</tbody>
</table>

Note: Statistical significant difference between the experimental groups were obtained by $\chi^2$ test, (P 0.05).
Table 3. The frequency of EEG computer analysis indexes in the experimental groups

<table>
<thead>
<tr>
<th>Frequency (%)</th>
<th>Index</th>
<th>Abnormality of histogram of amplitude</th>
<th>Abnormality of auto power spectrum spontaneous EEG</th>
<th>Visually evoked EEG</th>
<th>Abnormality of transfer function and coherence function</th>
<th>Abnormality of impulse response spontaneous EEG</th>
<th>Visually evoked EEG</th>
</tr>
</thead>
<tbody>
<tr>
<td>Group</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Controls</td>
<td>27.6</td>
<td>0</td>
<td>5.4</td>
<td>16.0</td>
<td>24.1</td>
<td>27.0</td>
<td></td>
</tr>
<tr>
<td>Noise (55-95 dBA)</td>
<td>22.4</td>
<td>8.0</td>
<td>6.6</td>
<td>20.5</td>
<td>33.3</td>
<td>37.5</td>
<td></td>
</tr>
<tr>
<td>Noise (100 dBA)</td>
<td>54.5</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Patients (with normal EEG*)</td>
<td>43.8</td>
<td>33.3</td>
<td>17.6</td>
<td>44.0</td>
<td>73.3</td>
<td>47.4</td>
<td></td>
</tr>
<tr>
<td>Patients (with abnormal EEG*)</td>
<td>100.0</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>-</td>
</tr>
</tbody>
</table>

*by using routine analysis method.

Note: Statistical significant difference between the experimental groups was obtained by $x^2$ test ($P < 0.05$) in all indexes.
A HEARING PROTECTOR FOR IMPROVED COMMUNICATIONS IN INTERMITTENT NOISE

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Introduction
One of the most common reasons given for the low or intermittent usage of personal hearing protection by persons subjected to high noise exposures is their suspected interference with both verbal and non-verbal communications. In particular the perception of warning and indicator sounds is often thought to be degraded\(^1\) when wearing conventional passive hearing protectors, with the result that they are either rejected completely or only worn during exposure to very high noise levels.

Previous research\(^{2,3}\) has indicated that in high noise levels the intelligibility of recorded speech and the perception of distinctive intentional warning sounds are not in general impaired when wearing hearing protectors. In fact positive improvements in the perception of such signals have often been shown to exist for persons with normal hearing, when compared to the open ear condition. However it has also been shown that when worn in low to moderate levels of noise, verbal communications and the perception of indistinct warning sounds are significantly degraded for both normal hearing and hearing-impaired persons.

In general it may be concluded that the wearing of ear-muffs and plugs in continuous high levels of noise does not have a widespread adverse effect on auditory communication for those with normal hearing. Unfortunately a feature of many industrial noise environments, particularly in the metals manufacturing and mining industries, is the irregular variation in noise levels throughout the working day. This often results in protectors being removed during the 'quieter' periods and not being replaced until after the resumption of the noisy processes, which necessitate the wearing of protection and contribute most to the daily noise exposure of workers employed adjacent to such processes. The consequent reduction in the effective attenuation of protectors when not worn for only a small proportion of the total noise duration can be dramatic, as was first highlighted by Else\(^4\).

To overcome the above problems and to avoid the potentially serious consequences of the interference of protectors with communications at low to moderate noise levels a prototype non-linear earmuff has been developed with the financial aid of the European Coal and Steel Community. This
features variable attenuation characteristics such that during low noise levels sounds are transmitted, via a microphone and earphone located in each shell, at their natural intensity but are progressively attenuated as their intensity increases thereby protecting the wearer from an otherwise hazardous noise exposure. The device is currently being evaluated using subjective experimentation, the results of which will be presented at the conference. It is anticipated that the device will prove capable of maintaining normal communications in quiet conditions similar to those which are possible when no protection is worn, whilst at the same time providing significant improvements in communication during moderate to high levels of noise exposure when compared with other electronic communication devices currently available.

Non-Linear Earmuffs

The attenuation of conventional passive hearing protectors is generally assumed to be linear; that is the attenuation at any given frequency is constant and independent of the incident sound pressure level. Measurements of hearing protector attenuation on cadavers have shown it to be essentially constant for steady state incident sound levels over the range 75-125 dB. Similarly the attenuation was found to be approximately constant for impulsive sounds with peak sound pressure levels over the range 135 - 175 dB, thus vindicating the assumption of linear attenuation for conventional hearing protectors.

To overcome the communication problems discussed previously at low ambient noise levels, however, would require that the protector provides no attenuation at low to moderate incident sound levels but maximum attenuation at high levels.

Devices are commercially available in the U.K. which consist of a microphone amplifier and earphone built into each shell of a passive earmuff. The amplifiers are designed to transmit incident sound levels at their natural level below a pre-set value, but to peak clip or saturate for incident sound levels above this. A typical transfer function claimed for such a device is illustrated in Fig. (1).

The use of peak clipping, however, introduces harmonic and inter-modulation distortion products which may affect communications at high input levels as well as being subjectively unpleasant. Furthermore, no discrimination of changes in intensity levels is apparently possible for external sound pressure levels between about 85 and 125 dB, a range not uncommon to many industries. It might therefore be expected that communications would be degraded, particularly for the perception of warning sounds characterised by a varying intensity over this range.
In order to avoid this loss of intensity discrimination, whilst maintaining normal transmission of low level sounds, single channel A.C.C. amplifiers have been developed elsewhere (6), although such devices have never proved to be completely successful possibly due to the poor quality of the sound reproduction.

A prototype hearing protector has been constructed consisting of a microphone and earphone built into both shells of a high quality earmuff together with a novel electronic processing unit. This system has been designed to transmit sounds at their normal intensity levels in low levels of background noise; at higher noise levels the attenuation of the electronically transmitted sounds increases progressively, thereby reducing the wearer's exposure to high and potentially hazardous levels of noise.

The dB(A) attenuation that this earmuff provides has been measured using a miniature microphone located at the wearer's ear. The difference in sound pressure levels, with and without earmuffs, is shown in figure (2) when the device was worn in different levels of broadband noise.

![Graph showing attenuation characteristics](image)

fig.(2) Attenuation Characteristics of I.S.V.R. Non-Linear Earmuff.

Preliminary investigations into the effects of this earmuff on the perception of recorded speech in the quiet have shown an insignificant loss of intelligibility compared to the open ear condition. The wearing of standard earmuffs, however, showed a large and significant reduction in speech intelligibility.

Conclusions

An earmuff has been developed with variable attenuation characteristics so that during the quieter periods of any fluctuating noise environment normal communications should be permitted. The results of speech intelligibility and warning sound perception tests used to assess the performance of this non-linear earmuff, when worn in different background noises, will be presented at the conference.
References


FIELD TEST OF HEARING PROTECTORS

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Introduction

Field performance of hearing protectors is a major matter of concern for anyone involved in hearing conservation. The two properties to be considered are acoustical attenuation and comfort (and acceptability) of users. The attenuation has been measured by several researchers, following procedures similar to those outlined in the ANSI S3.19-1974 Standard. Other attempts included masking noise on the non-protected ear. However, no systematic efforts have been reported on assessing users comfort.

In our study we assessed the overall performance of protectors as worn in one nuclear and one fossil-fueled generating station. The attenuation was measured following the ANSI Standard, while the comfort was assessed by using a questionnaire and by measuring the head-band force of muffs.

Instrumentation, Subjects and Protectors

A MAICO Mod MA27 audiometer was modified for our purpose by introducing a pink noise generator. Its output, filtered in 1/3 octave bands centered at 250, 500, 1000, 2000, 4000 and 8000 Hz was amplified by a low noise, 5W amplifier and fed into a two way speaker system. The audiometer's attenuator was also modified by adding a 2.5 dB step. The whole system was tested to comply with the ANSI requirements.

Tests were conducted in audiometric rooms located in each station. In this way subjects found themselves in a familiar environment where they routinely underwent audiometric testing.
A. BEHAR - Field Test of Hearing Protectors

The head-band force apparatus used to test cap-mounted muffs is the one quoted in the ANSI standard.

Subjects were selected on a voluntary basis, the goal being 20 subjects per protector at each station. Management announced the purpose of the test and helped the scheduling of subjects, so they could show up at a steady rate of 2 per hour. Each subject came with his own protector. Only the disposable ear plugs were supplied at the testing site.

The table below shows details of protectors and subjects.

<table>
<thead>
<tr>
<th>Manufacturer</th>
<th>Model</th>
<th>Type</th>
<th>Subjects</th>
</tr>
</thead>
<tbody>
<tr>
<td>Marion</td>
<td>Decidamp</td>
<td>Plug</td>
<td>24</td>
</tr>
<tr>
<td>Peltor</td>
<td>H9P3E</td>
<td>Cap-mounted</td>
<td>36</td>
</tr>
<tr>
<td>EAR</td>
<td>EAR</td>
<td>Plug</td>
<td>18</td>
</tr>
<tr>
<td>Bilsom</td>
<td>Prop-O-Plast</td>
<td>Plug</td>
<td>23</td>
</tr>
<tr>
<td>Willson</td>
<td>Sound-Ban</td>
<td>Semi-insert</td>
<td>22</td>
</tr>
<tr>
<td>AOCO</td>
<td>1776K</td>
<td>Cap-mounted</td>
<td>16</td>
</tr>
<tr>
<td></td>
<td></td>
<td>muff</td>
<td></td>
</tr>
</tbody>
</table>

Questionnaire

A multiple choice questionnaire was prepared to assess the reasons why a particular protector was chosen by the individual. (Workers can select from several brands available at each station.) Another matter of concern was if the user was instructed on the proper wear and care of his protector.

Procedure

Each subject was tested only once. Tests started with an explanation of their purpose and procedure. The subject was then introduced into the audimetric booth. After a one minute rest, the open ear threshold was determined. Then the subject was invited to put on his protector in his normal manner, without any instruction on our part. The protected ear threshold was then determined. Next the subject was invited to fill in the comfort questionnaire. If his hearing protector was a cap-mounted ear muff, then the head-band force was measured at this time. The session ended by an explanation to the
subject in how protectors have to be fitted. It was our intent not only to measure the existing attenuation but also to improve it by instructing protector users.

Results and Discussion

The table below shows the results from the attenuation measurements.

<table>
<thead>
<tr>
<th>Protector</th>
<th>Attenuation and Standard Deviation</th>
<th>NRR</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>250</td>
<td>500</td>
</tr>
<tr>
<td>Decidamp</td>
<td>16.9</td>
<td>20.0</td>
</tr>
<tr>
<td></td>
<td>6.0</td>
<td>9.1</td>
</tr>
<tr>
<td>H9P3E</td>
<td>6.9</td>
<td>19.2</td>
</tr>
<tr>
<td></td>
<td>4.3</td>
<td>5.5</td>
</tr>
<tr>
<td>EAR</td>
<td>17.7</td>
<td>16.5</td>
</tr>
<tr>
<td></td>
<td>7.2</td>
<td>7.6</td>
</tr>
<tr>
<td>Prop-O-Plast</td>
<td>9.6</td>
<td>13.1</td>
</tr>
<tr>
<td></td>
<td>4.7</td>
<td>5.1</td>
</tr>
<tr>
<td>Sound-Ban</td>
<td>12.2</td>
<td>10.9</td>
</tr>
<tr>
<td></td>
<td>8.0</td>
<td>7.0</td>
</tr>
<tr>
<td>1776K</td>
<td>11.4</td>
<td>16.7</td>
</tr>
<tr>
<td></td>
<td>4.6</td>
<td>7.3</td>
</tr>
</tbody>
</table>

Measured attenuations were smaller, and standard deviations were larger than those reported by manufacturers, resulting in much smaller NRR. Results are in line with these reported elsewhere. Poor fitting, already observed during the test, is obviously the main reason for these findings.

The main results from the questionnaire were:

a) By providing choice of protectors, workers ended up wearing brands they liked and felt comfortable with.

b) Most of the workers were not formally instructed on fitting and care of protectors.

Conclusions

Field measurements are a useful tool in a hearing conservation program by assessing the effective attenuation and acceptance of protectors.
ATTENUATION OF HEARING PROTECTORS AT THE FREQUENCY EXTREMES

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When the attenuation of hearing protection devices (HPDs) is evaluated according to the standardized real-ear attenuation at threshold (REAT) methods (ASA STD 1; ISO 4869), the measurements are normally limited to the frequency range of 125 Hz - 8 kHz. Some laboratories routinely test to frequencies as low as 63 or 75 Hz (Brinkmann, 1977; Camp, 1979; Martin, 1977), but few data are available at frequencies above 8 kHz (Townsend and Bess, 1973). The purpose of this report is to present extended frequency attenuation data for a representative sample of devices so that users who require such data have it available, and also to provide an indication of whether or not such information should be more routinely reported in the future.

METHOD

The attenuation of the HPDs was estimated via an REAT method in conformance with the requirements of ASA STD 1 (and therefore also in substantial conformance with ISO 4869). The test procedures have been described elsewhere (Berger, 1980). The equipment was modified for the current tests by the addition of six Motorola Piezo Ceramic tweeters (model KSN 1005A) to the existing speaker systems, and a switching network to energize them for presentation of the test stimuli at the two highest 1/3 octave bands.

Two separate complete attenuation tests (10 subjects by 3 replications) were conducted on each HPD. One test covered the standard 1/3 octave test bands centered at the frequencies .125, .250, .500, 1.0, 2.0, 3.15, 4.0, 6.3, and 8.0 kHz and the other test (referred to as the extended frequency test) included the bands centered at .080, .125, 1.0, 8.0, 12.5, and 16.0 kHz. Separate tests were conducted, since at the time of this experiment all of the devices had been previously tested at the standard test frequencies. Three frequencies were common to both tests to assure similarity between the current and previous results, so that the extended frequency data could be appended to the standard frequency data.

The hearing threshold levels of all subjects were within the specifications of ASA STD 1 for the standard test frequencies. At
16 kHz the average open ear 1/3 octave band diffuse field hearing threshold level was 30 dB SPL re 20 μPa, with the values ranging from 20 - 46 dB SPL across subjects.

The HPDs that were evaluated were selected to represent the spectrum of the currently commercially available products. These included a fiberglass plug (Bilsom Soft), a pre-molded PVC plug (Mediprint V-51R), three different insertions of a foam earplug (E-A-R™ Plug), a semi-aural device (Caboflex™ 600), a small volume earmuff (Bilsom UL-1), two medium volume earmuffs (Silenta Super and Willson 358A), and a large volume earmuff (David Clark 19A). The three insertions for the foam earplug were: partial, as an estimate of attainable real world usage (E-A-R:PI, 15 - 20% of the plug in the canal); laboratory standard (E-A-R:SI, 50 - 60% of the plug in the canal); and deep, which was the maximum depth of insertion that a subject could achieve before he experienced significant discomfort (E-A-R:DI, 80 - 100% of the plug in the canal). In addition, two plug-plus-muff combinations were evaluated, one of which (E-A-R:DI + 1650 g/cur lead earmuffs) had been previously shown to provide a good estimate of the bone conduction limits to HPD attenuation with deeply occluded ear canals (Berger, 1982).

The method of fitting the HPDs could be characterized as experimenter supervised fit except for the foam earplug, which was tested with three distinctly different experimenter fitted insertions. Subjects read the manufacturers' instructions and then fitted the device while the experimenter watched to make sure that there were no gross errors in the fitting technique. If the subject experienced problems or had questions, he was free to ask for assistance. A 65 - 70 dBA fitting noise was presented in the room prior to each test. The V-51R was initially sized by the experimenter working in conjunction with the subject.

RESULTS

The results are depicted in Figures 1 - 3. The standard and extended frequency tests were combined, with the data at the three overlapping frequencies averaged together since the differences in mean attenuation for each device averaged no more than 1.3 dB at any of those frequencies. On all three graphs, the bold black line at the bottom represents an estimate of the bone conduction limits as previously discussed.

The earplug data (Fig. 1) demonstrate that the 80 Hz attenuation never differs from the values at 125 Hz by more than 2.8 dB. Additionally, for three of the earplugs, the 12.5 and 16 kHz data are not appreciably different from the 8 kHz results (within 2.6 dB). However, for the E-A-R:SI and DI, the highest frequency data are lower by 8 - 9 dB. This is probably because the attenuation was great enough at those frequencies to be controlled to some extent by the bone conduction limits which also exhibit the same trend.

The semi-aural and earmuff data are shown in Figure 2. As with the plugs, the 80 and 125 Hz data are substantially the same (within 2.0 dB, except within 4.1 dB for the UL-1). For the cases in which the 80 Hz
Fig. 1 - Real-ear attenuation for 5 earplugs

Fig. 2 - Real attenuation for 4 earmuffs and 1 semi-aural device

Fig. 3 - Real-attenuation for a plug and muff, alone and in combination
data are higher, such as the UL-1, the results are suggestive of data that are spuriously elevated due to physiological noise masking of the occluded ear thresholds (Berger and Kerivan, 1982). At the high frequencies the attenuation is essentially uniform (within 4.1 dB) from 6 - 16 kHz with the exception of the Caboflex, where the 16 kHz data are 7.6 dB less than at 6.3 kHz.

In Figure 3, the attenuations of a muff plus plug combination is compared to that of the individual devices and to the bone conduction limits. The combination of the two devices yields attenuation that is equivalent to the bone conduction limits from 2 - 16 kHz, and offers at least 30 dB of attenuation at the lower test frequencies.

CONCLUSIONS

Examination of the attenuation of five earplugs, one semi-aural device, four earmuffs, and one plug + muff combination suggests that the attenuation of HPDs at 80 Hz, and at 12.5 and 16 kHz can be approximated by assuming it is equivalent to the adjacent upper and lower 1/3 octave band attenuation, respectively. Thus, there appears to be little need to increase the range of test frequencies that are normally evaluated by the standardized test procedures.

REFERENCES

LA PROBLEMATIQUE DU RISQUE QU’ENCOURTENT LES FEMMES ENCEINTES POUR L’OUÏE
DE LEUR FOETUS LORSQU’EXPOSÉES QUOTIDIENNEMENT À DES BRUITS INDUSTRIELS
INTENSES

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Problématique

Un nombre important de travailleuses sont exposées quotidiennement à des niveaux sonores élevés de bruit industriel, en particulier dans les industries du papier, du tabac, du vêtement, du textile et de l’alimentation. De ce nombre, plusieurs poursuivent leur travail lorsqu’elles sont enceintes, du moins durant les premiers mois de leur grossesse. Les risques qu’elles encourrent alors pour l’organe de l’ouïe de leur foetus ne sont pas vraiment connus.

En effet, chez les humains, une seule étude a porté sur les effets nocifs du bruit sur la fonction auditive d’enfants dont les mères avaient été exposées quotidiennement au bruit d’une usine de tissage pendant leur grossesse (Laciak et Majcherska-Matuchniak, 1970). Cette étude a révélé que des 75 enfants sélectionnés, 35 présentaient une perte auditive permanente unilatérale ou bilatérale. Cette perte variait entre 20 et 55 dB pour les fréquences de 4000 et de 8000 Hz. Puisque les autres facteurs de risque d’une surdité neurosensorielle avaient été contrôlés, cette perte pouvait en conséquence être imputée à l’exposition in utero au bruit industriel.

L’étude de Laciak et al. (1970) suggère donc que les enfants de mères exposées pendant toute leur grossesse à un bruit industriel intense (100 dB), ont une chance sur deux de subir une détérioration irréversible de leur audition pour certains sons. Toutefois, cette étude ne nous permet pas de répondre aux questions suivantes:

- Quelle dose quotidienne de bruit et quel niveau sonore maximal sont inoffensifs pour l’organe de l’ouïe du foetus?
- Certains types de bruit sont-ils plus nuisibles que d’autres? Par exemple, à égale énergie sonore, un bruit riche en sons graves est-il plus nocif pour l’ouïe du foetus qu’un bruit avec prédominance de sons aigus? Ou encore, à égale énergie sonore, une certaine répartition temporelle du bruit est-elle plus dommageable qu’une autre (bruit intermittent ou bruit impulsive versus bruit continu)?
- L’exposition à des bruits industriels intenses est-elle plus dangereuse pour l’ouïe du foetus lorsqu’elle prévaut durant les premiers mois de la grossesse (période au cours de laquelle l’organe de l’ouïe se forme, en particulier l’oreille interne) que lors des derniers mois de grossesse?
(période où s'effectue entre autres la maturation des récepteurs cochléaires)?

L'exposition combinée à un bruit industriel intense et à un autre agent agresseur tel des vibrations, a-t-elle un effet plus marqué sur la détérioration de l'ouïe du foetus, que la seule exposition au bruit? Dans l'affirmative, cet effet est-il additif ou s'agit-il plutôt d'un effet de potentiation?

Les quelques études menées chez les animaux, ne permettent pas davantage de répondre à ces questions (Daniel, 1976; Szmeja et al., 1979). Néanmoins, les résultats divergents de ces études suggèrent que tout travail éventuel portant sur l'évaluation de la nocivité pour l'ouïe, d'une exposition au bruit pendant la vie foetale, devrait prendre en compte non seulement le niveau sonore et la durée d'exposition, mais aussi les caractéristiques spectrale et temporelle du bruit. Daniel (1976) a en effet rapporté que sur 15 cobayes exposés pendant toute leur vie foetale au bruit d'une usine de tissage (103-105 dBA), la plupart d'entre eux avaient une détérioration de leurs seuils d'audition à l'âge de 60 jours. Par contre, Szmeja et al. (1979) n'ont trouvé ni détérioration des seuils d'audition, ni changement histologique ou histochemique au niveau de la cochlée, chez dix cobayes exposés pendant toute leur vie foetale à un bruit continu d'usine dont le niveau sonore était de 95 à 100 dBA. Il est possible que les résultats contradictoires de ces études soient dus uniquement à la différence de niveau sonore. Toutefois, pour l'affirmer, il faudrait s'assurer que la composition spectrale et temporelle du bruit était la même.

En l'absence de données précises concernant les limites sécuritaires pour protéger l'ouïe du foetus, certaines réglementations, notamment celle du Québec (Anon., 1981), se réfèrent aux limites d'exposition jugées admissibles pour prévenir la surdité professionnelle. Or il existe de bonnes raisons de croire que ces limites ne sont ni sécuritaires ni applicables pour le foetus.

Premièrement, on ignore si la relation établie pour l'adulte entre les effets du niveau sonore et de la durée d'exposition sur la détérioration temporaire ou permanente de l'acuité auditive (Melnick, 1978), vaut pour le foetus. En conséquence, on ne sait pas si la dose quotidienne jugée admissible pour l'adulte, par exemple 90 dBA/8 heures, l'est également pour le foetus. Il en est de même du niveau maximal admissible, par exemple 115 dBA/15 minutes.

Deuxièmement, que penser des milieux de travail pour lesquels les niveaux de bruit deviennent acceptables au sens de la loi par l'usage de moyens individuels de protection.

Troisièmement, les réglementations actuelles sur le bruit recommandent que le calcul de la dose quotidienne de bruit soit fait à partir de mesures obtenues en utilisant une pondération A au sonomètre. De cette façon, on accorde relativement peu d'importance aux sons de basses fréquences. Par contre, comme l'illustre la Figure 1, les études menées tant chez les humains que chez les animaux ont montré que l'atténuation offerte pendant la grossesse par les parois abdominale et utérine de la mère de même que par le liquide amniotique, est nettement plus faible pour une bande de fréquences centrée sur une fréquence inférieure à 500 Hz. Alors qu'elle se situe entre 16 et 39 dBA à 500 Hz, elle n'est plus que de quelques décibels pour les fréquences de 300 Hz et moins (1 à 20 dBA suivant
LALANDE Nicole - HETU Raymond, Le bruit industriel: facteur de risque pour le foetus?

Bench (1968), H: 1 sujet
Grimwade et al. (1970), H: \( \bar{x}, N = 9 \) △
Walker et al. (1971), H: \( \bar{x}, N = 14 \) ○
Szmeja et al. (1979), H: \( \bar{x}, N = 19 \) △
H: \( \bar{x}, N = 11 \) ▼
Armitage et al. (1980), A: 2 sujets ■

Figure 1. Selon diverses études, valeurs moyennes (\( \bar{x} \)) ou valeurs individuelles de l’atténuation offerte par les parois abdominale et utérine de la mère de même qu'occasionnellement le liquide amniotique (Szmeja et al., 1979; Armitage et al., 1980), en fonction de la fréquence. Les mesures ont été obtenues chez l'humain (H) et chez l'animal (A).

Les auteurs. En conséquence, lorsqu'il s'agit d'évaluer la nocivité du bruit pour le foetus, la mesure des niveaux de bruit obtenue en pondération A n'a plus de sens. Ceci est d'autant plus vrai lorsqu'on est en présence d'un bruit riche en sons graves.

Finalement, l'évaluation de la nocivité d'une ambiance sonore en pondération A s'est avérée utile parce qu'elle simulait bien la sensibilité de l'oreille à des sons transmis par voie aérienne. Mais dans le cas du foetus baignant dans le liquide amniotique, on ne peut certainement plus évaluer l'effet des sons d'après les conditions de transmission aérienne.
Conséquences pratiques

Compte tenu de l'analyse qui précède, les conséquences d'une exposition quotidienne au bruit industriel pendant la grossesse, doivent faire l'objet d'études expérimentales et épidémiologiques. Ces études permettront d'une part, de connaître la réponse de l'oreille en milieu liquide et d'autre part, de préciser les limites de bruit qui ne seraient pas nuisibles pour l'ouïe du foetus.

Dans cette perspective, nous prévoyons débuter sous peu une étude rétrospective sur les enfants de mères ayant été exposées quotidiennement à un bruit industriel voisin d'un L_{Aeq-8h} de 90 dBA pendant leur grossesse. Les seuils d'audition des enfants seront étudiés en fonction a) de la dose quotidienne et totale d'exposition au bruit, b) de la période et du nombre de mois auxquels ils auront été exposés au bruit pendant leur vie foetale, c) des caractéristiques physiques du bruit et d) de la présence ou non d'un deuxième facteur environnemental susceptible d'avoir affecté l'organe de l'ouïe lors de sa formation. Cette étude permettra ainsi a) d'obtenir des données sur l'influence d'une exposition au bruit pendant la grossesse sur l'audition des enfants de mères-exposées, cette exposition comportant un niveau de bruit équivalent plus faible que ceux étudiés jusqu'ici, b) de vérifier la validité de la réglementation implantée au Québec par la Commission de la santé et de la sécurité du travail en ce qui a trait au retrait préventif de la femme enceinte travaillant en milieu industriel bruyant et c) de contribuer à établir éventuellement une norme sécuritaire pour cette condition.

Références


3.5

Acoustique animale.
Génération et détection
par des systèmes biologiques

Animal acoustics.
Generation and detection
by biological systems

Tierakustik.
Schallerzeugung- und Wahrnehmung
durch biologische Systeme
SENSITIVITY OF MONKEYS (M. nemestrina) TO INTERAURAL TIME DISPARITIES IN NOISE BANDS.

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Attempts at explaining the ability of mammalian auditory systems to localize sound in space have led to several possible cues. The first of these cues is a transient onset difference at the two ears, which occurs when a sound is produced by a source located away from the midline. The other cues provide ongoing information, that is, the cues are available for the duration of the signal to be localized. For pure tones, the so-called "duplex theory" parcels out the auditory spectrum into high frequencies which are localized on the basis of interaural level differences, and low frequencies which are localized on the basis of cycle by cycle interaural time differences. This last cue, the ongoing time difference cue, has been investigated in lateralization experiments and is generally considered to be effective for man up to approximately 1300 Hz (Zwislocki & Feldman, 1956). For monkeys, experiments in our laboratory using pure tones have shown the time difference cue to be effective for frequencies up to 2,000 Hz (Houben & Gourevitch, 1979). The interaural time difference (ITD) function for the monkey stops fairly abruptly at 2 kHz, i.e., none of the three monkeys in the study was able to lateralize pure tones at 2.25 kHz, regardless of time difference.

Monkeys can localize noise signals often more accurately than pure tones (Brown et al. 1978;1980). Ongoing time difference in complex signals may be a major contributor to such localization since this cue appears to be ubiquitous, i.e., it exists at low and high segments of the spectrum. Since monkeys are able to detect interaural time differences for pure tones up to 2 kHz, it is expected that they should respond to time differences in low frequency noise bands.

Recently Henning (1974) and McFadden and Passanen (1976) have pointed out that discrimination of time differences at high frequencies where ongoing pure tone time disparities are not perceived, can also be made, if the signal is a complex one. Klumpp & Eady (1956) showed that bands of high
frequency noise could be lateralized on the basis of time differences only. The work of McFadden and Pasanen (1976), Henning (1974a,b; 1980) and others suggests that in such cases lateralization is achieved by means of time differences in the envelopes of the complex signal, rather than in the fine structure time differences which are too rapid for man to detect at these frequencies.

A similar lateralization mechanism would be very advantageous for primates smaller than man. Since their heads are smaller, the effectiveness of interaural intensity difference, which depends on wave length relative to head size, does not emerge at middle frequencies as in man, but at higher frequencies (above \( \sim 6 \) kHz). The monkey's extended range for detecting cycle by cycle time differences up to 2 kHz only partially compensates for this head size disadvantage and leaves a gap of frequencies over which neither of the cues of the duplex theory is effective.

The present experiment was undertaken to determine the sensitivity of the binaural system of monkeys to ongoing time differences in noise bands centered at various segments of the audible spectrum.

Three pig-tailed monkeys (\( \text{Macaca nemestrina} \)) served as subjects. They were trained psychophysical observers; one has served over the course of three or more years as subject in lateralization experiments involving interaural time and intensity differences in pure tone stimuli.

The animals were restrained in a standard primate chair where they were faced with three manipulanda and a 'time-out' light. Pressing the center key resulted in the presentation of the stimulus complex, and pulling either of the two side levers constituted the monkey's response choice on that trial. Correct responding was rewarded with applesauce, incorrect responding was followed by a six second time-out from the experimental procedure.

The stimulus complex consisted of two 250 ms duration noise bursts presented via TDH 39 headphones. The first noise burst was presented diotically (no interaural disparities), and served as a reference image. The second noise burst, which followed after a 250 ms silent interval was presented dichotically and contained an interaural delay. Both bursts were gated simultaneously in the two ears with 10 msec rise/decay times. Bands of noise with extremely sharp skirts (96 dB/octave) were generated by multiplying low-pass filtered noise with a sine wave. The resulting noise band could be centered at any frequency set on the oscillator (Palin and Gourevitch, 1970). A delayed version of this wave form was obtained by use of a 'bucket-brigade' type analog delay module with 32 taps and adjustable clock frequency. At the output of this kind of charge-coupled device (TAD 32, Reticon) all frequency components were delayed by exactly the same amount. From this point to the headphones, the signals proceeded via two totally independent channels. All stimuli and contingencies were controlled by a
PDP 12 computer.

The subjects were tested at six center frequencies ranging from 500 Hz to 4 kHz. Different noise bandwidths were presented at each center frequency. The overall level of the signals was held constant at 78 dB SPL as the bandwidth was varied. Interaural delays were presented according to the method of constant stimuli (five levels). Left ear leading and right ear leading was randomized, and a correct response was defined as a lever pull on the leading side.

Thresholds were calculated from psychometric functions by interpolating the delay at which 75% correct responding would occur. These psychometric functions were obtained for each animal under each experimental condition and were based on a minimum of 400 trials.

As expected, results obtained with noise bands centered at low frequencies (500 Hz to 2 kHz) show that monkeys can discriminate ongoing time differences in this portion of the spectrum. The interaural time difference thresholds are considerably lower than those obtained for pure tones, even at the narrowest noise bands used (50 Hz); they are as low as about 23 usec, whereas time difference thresholds for pure tones are no lower than about 44 usec.

It also appears that a bandwidth effect exists, i.e., time difference thresholds are lower at wider bands (400 Hz and 800 Hz) than at narrower bands. However, this effect is not equally strong at all frequencies. A similar finding was reported in the localization of noise bands centered at low frequencies (Brown et al., 1980).

Preliminary results for ongoing time disparities for noise bands centered at high frequencies (above 2 kHz) indicate that monkeys are able to detect time differences in this portion of the spectrum. It seems, however, that these thresholds are considerably higher than those at low frequencies and are above the lowest thresholds for pure tones.

ACKNOWLEDGEMENTS

This research was supported by National Science Foundation grant BNS-7915834.

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GOUREVITCH, G. SENSITIVITY OF MONKEYS TO ITD


IMPAIRED FREQUENCY SELECTIVITY IN MONKEYS AFTER
NOISE-INDUCED TEMPORARY HEARING LOSS

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Tuning curves, both physiological and psychophysical, provide measures of the frequency selectivity of the auditory system; that is, of its ability to analyze a complex signal into its components. (Pickles, 1980; Weber et al., 1980; Serafin et al., 1982) The majority of studies on tuning curves from impaired ears have demonstrated that the sharply tuned tips of the curves, both physiological and psychophysical, are the most susceptible to cochlear insult. Even with relatively small hearing losses the tips show elevated thresholds and/or greater width, indicating loss of selectivity. (McFadden and Pasanen, 1980; Feth et al., 1980; Florentine et al., 1980; Wightman et al., 1977) Occasionally the tips are lost completely, in which case the tuning curves resemble the mechanical excitation patterns of the basilar membrane described by Bekesy—a fact which has led Evans and others to suggest a vulnerable second filter mechanism which is responsible for the sharply tuned tip. Although the need for the second filter construct has recently been questioned by Khanna among others, the vulnerability of sharp tuning to cochlear insult is readily apparent from these recent data.

There are, however, a small number of animal studies which have demonstrated retention of the sharply tuned tip even though absolute threshold is elevated by as much as 40–50 dB and the curves are otherwise abnormal. These studies, most notably those of Dallos and associates (Dallos et al., 1977; Ryan et al., 1979), have been carried out in kanamycin-treated chinchillas in which the treatment has resulted in areas of the membrane which are devoid of outer hair cells but which have a full complement of normal-appearing inner hair cells. There is general agreement that, at frequencies with greater than about 50 dB of shift, corresponding to cochlear loci at which there is inner-hair-cell loss, all measures of selectivity show some degradation.

The present investigation was concerned with the measurement of psychophysical tuning curves (PTCs) with a forward masking paradigm following repeated TTS in monkeys. Previous studies have been carried out with mild TTS in humans (Feth et al., 1980) and with PTS in animals (Kiang, Schmiedt and others). These studies have generally shown a reduction in selectivity. The magnitude of the threshold shift employed in this study was 50 dB or less, within the range in which Dallos and
colleagues have shown retention of the sharply tuned tip. In the monkey, repeated shifts of this magnitude are possible without incurring permanent hearing loss. Recovery occurs over the course of several days so that changes in PTCs during the process of recovery can be followed.

Three male pigtai11e macaques (Macaca nemestrina) served as subjects; all had several years experience in tuning curve experiments. The masking stimulus consisted of 130-msec pure-tone pulses presented at a rate of 3.8 pulses per second. Masker frequencies were chosen in the vicinity of the test tone, and levels were determined by a modified method of tracking. If the subject responded correctly on a given trial, the level of the masker was increased by 10 dB on the next trial; failure to respond within the 2.5-second trial resulted in a decrease in masker level. The test tone pulses were 25 msec in duration and there was a 2-msec interval between the time the masker was completely off and the time the test tone began to turn on. Test-tone level was always 10 dB SL based on absolute thresholds for pulsed pure tones determined immediately prior to each tuning curve session. Test tone frequencies were 500 Hz and 1, 2, and 4 kHz.

The exposure stimulus used in this experiment was a 2-kHz, 100-dBA octave band of noise. Exposure duration was 1 hour, and exposures took place immediately prior to testing sessions. Exposures were carried out only when recovery was complete from previous exposures and only one frequency was tested following each exposure. Thus, a full set of data from each exposure consisted of a pre-exposure baseline tuning curve, followed by tuning curves obtained immediately after the noise exposure, and on successive days until baseline was recovered. There was essentially no threshold shift at 500 Hz and 1 kHz and a 30-50 dB loss at 2 and 4 kHz. The amount of the loss was fairly uniform across animals.

A set of tuning curves obtained at 2 kHz are shown in the center of Figure 1. It is clear from these data that significant changes resulted from the exposure. There is still a slight tip remaining on the post-exposure function, but it is considerably truncated and probably does not represent useful frequency selectivity; in fact, it is not possible to compute a Q₁₀ from these data because the low-frequency limb of the function which is unusually flat, never reaches 10 dB above the tip. In these data, considerable selectivity has returned by 1 day post-exposure. In fact, Q₁₀ is within the normal range. On the second post exposure day, the tip of the tuning curve is somewhat sharper than that of the baseline function, but this finding should be considered an exception rather than the rule.

A set of tuning curves obtained at 4 kHz is shown on the right of the figure. Once again, the tip of the immediate post-exposure curve is truncated. Again, computation of Q₁₀ is not possible from this function. Recovery proceeds as in the previous functions—a well-defined tip reappears on the first day of recovery, but is considerably elevated relative to baseline. Recovery towards baseline continues on the second day, but both recovery days show decreased Q₁₀'s relative to baseline. It should also be noted that there was usually a third recovery day before baseline levels were completely reestablished, but these have been omitted.
from the functions for clarity. These last two functions present typical findings for all subjects for changes in tuning curves at 2 and 4 kHz following exposure.

At these frequencies below the 2-kHz exposure band, where there was little or no threshold shift, there was sometimes a slight reduction in selectivity observed following exposure but these reductions were usually quite small. An extreme example of reduced selectivity with normal thresholds is seen on the left of the figure, which presents data obtained at 500 Hz. Here, the tuning curve obtained immediately after exposure is considerably broader, although there is only a very slight elevation in the tip. Recovery functions, however, are somewhat more sharply tuned than the baseline function.

In summary, the most frequent finding from this experiment was a reduction in frequency selectivity following exposure to moderately intense noise. This reduction was most strongly manifested for threshold shifts of 40 dB and greater, but was occasionally seen with little or no shift. Even when the tip was completely missing immediately following exposure, it returned within 24 hours, but was usually broader until recovery was complete—usually by 4 days after exposure. Recovery of normal selectivity usually, but not always, occurred simultaneously with recovery of absolute threshold. It is clear that the physiological changes which lead to the changes in tuning curves are fully reversible, but we cannot yet state what those changes might be. The occasional presence of impaired tuning with near-normal thresholds and of normal selectivity with elevated thresholds suggests that tuning curves may reflect mechanisms which are to some extent independent of those manifested by elevated thresholds.


BAUDOIN, VALENTIN - Audition et émissions acoustiques du lérot.

EVOLUTION DES CAPACITÉS D'AUDITION ET DES ÉMISSIONS ACOUSTIQUES EN FONCTION DE L'ÂGE CHEZ LE LÉROT

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Dans le cadre d'une étude étho-écologique des communications chez le lérot, seul rongeur européen présentant des bulles tympaniques hypertrophiées, nous avons étudié les relations entre le développement du système acoustique de communication et celui des capacités d'audition. Cette étude permet de mieux comprendre comment des contraintes physiologiques peuvent intervenir dans le réglage des relations interindividuelles spécifiques.

I. Les capacités d'audition

I.1. Apparition des réactions comportementales en réponse à des sons de fréquences pures:
Le développement de l'audition a été étudié au moyen de critères comportementaux. L'apparition des réactions comportementales des jeunes à des stimulations de fréquences pures a été testée quotidiennement chez 11 jeunes âgés de 0 à 30 jours. Le stimulus sonore a une durée de 1 seconde et une fréquence de 1600 Hz. Le jeune est placé dans une enceinte isolée acoustiquement. Les réactions comportementales du jeune aux stimulations acoustiques sont des sursauts et des réactions

Fig.1: évolution en fonction de l'âge:
- du % de jeunes aux conduits auditifs ext. ouverts (→), n=II.
- du % de réponses positives aux sons de:
  1600 Hz (→); 16000 Hz (→), n=II.
- du nombre moyen de sifflements émis par un jeune (→-→) en 3 minutes, n=34.
d'immobilisation. La réaction est dite positive lorsque l'un et/ou l'autre de ces comportements est enregistré. La figure I montre que l'âge moyen d'apparition de la réaction positive est de 19,27 ± 1 jours pour I son de 1600 Hz et de 20,45 ± 0,93 jours pour un son de 16000 Hz. L'ouverture des conduits auditifs externes a lieu entre 17 et 19 jours (18,09 ± 0,5 jours).

1.2. Évolution de l'audition à l'âge adulte:
16 adultes (7 O° 0° 9 O Q) âgés de 5 à 60 mois ont été étudiés par la technique des potentiels évoqués auditifs (P.E.A.) enregistrés au niveau colliculaire. Le stimulus acoustique est un train d'ondes sinusoidales de 5 ms de durée avec I pente de 1ms. Le rythme de stimulation est de I par seconde. Les fréquences choisies varient de 0,125 à 96 kHz. La stimulation est réalisée en champ libre. Après amplification, les P.E.A. sont sommés par un moyenneur qui permet d'extraire du bruit de fond cérébral un P.E.A. de 2 mV considéré comme seuil objectif d'audition du lérot. La courbe des seuils d'audition (fig. 2) a une allure bimodale. Le pic de meilleure sensibilité se situe entre 1 et 2 kHz et le second entre 16 et 28 kHz. Les 2 pics sont d'égale amplitude lorsque la température corporelle des animaux est maintenue stable à 38°C.

![Fig. 2: courbe des seuils moyens d'audition du lérot](image)

Les résultats montrent également que les seuils d'audition sont d'autant plus élevés que les animaux sont âgés. En particulier, au-delà de 3 ans il y a une élévation des seuils pour toutes les fréquences et la gamme des fréquences perçues est restreinte par modification simultanée des limites inférieure et supérieure d'audition.

2. Les émissions acoustiques
2.1. Les cris des jeunes:
Les enregistrements ont été réalisés à l'intérieur du nid où se trouvent la mère et les jeunes et dans des conditions de tests standardisés de la naissance au 47e jour de vie, c'est-à-dire au-delà du sevrage. Des observations complémentaires ont été faites sur les réactions des adultes aux émissions
acoustiques des jeunes dans des conditions de vie en groupe en enclos. Les cris des jeunes sont de 4 types : 1- les transitoires, et en particulier les clics émis lors des changements de position de la mère dans le nid et dans les situations expérimentales de manipulation. Ils disparaissent au-delà du sevrage, vers 35 jours. 2- les sifflements, caractérisés par la fondamentale à 12 kHz et H₂ à 24 kHz. Cette harmonique disparaît vers l'âge de 21 jours. La durée de cette émission est alors voisine de 0,10 s et les syllabes sont émises en série, espacées de 0,10 s environ. Ce cri est émis par les jeunes isolés ou tombant du nid et provoque le comportement de "retrieving" de la part de la mère. En situation expérimentale d'isolement à 22°C le nombre des sifflements émis pendant le test de 3 minutes augmente progressivement jusqu'au 15e jour puis décroît progressivement jusqu'au 30e jour (fig. 1). 3- les cris pulsés de basses fréquences (0-4 kHz) de durée variable (0,01s-0,20s) sont émis par les jeunes au-delà du sevrage dans une situation de forte perturbation (contention, situation agonistique). 4- les cris pulsés à large bande de fréquences, de brève durée (< 0,05 s), émis en série caractérisées par un rythme de plus en plus net du 10e jour au sevrage. Ils sont émis dans une situation de forte perturbation du jeune.

2.2. Les cris des adultes :
5 groupes de lérot réunissant une trentaine d'adultes ont été étudiés tout au long du cycle annuel d'activité dans des conditions de semi-liberté. Les 4 catégories de cris décrites chez les jeunes sont également observées chez les adultes. On distingue : 1- les cris brefs de basses fréquences avec un maximum d'énergie situé entre 1,7 et 2 kHz. Ces cris émis à un faible niveau sonore accompagnent les déplacements et l'exploration de l'environnement. Le rythme d'émission de ces cris varie avec le statut social de l'émetteur. Ils accompagnent également les états dits "d'excitation" et traduisent un faible déséquilibre de l'émetteur par rapport à son environnement. Ce type de cri pourrait constituer le support d'une identification de l'émetteur par les autres membres du groupe et permettre sa localisation spatiale. 2- les sifflements, situés dans une bande de fréquences comprise entre 10 et 13 kHz et dont la durée varie de 0,10s à 1s. Ce sont surtout des cris d'appel sexuel des femelles qui émettent 10 à 12 syllabes pendant les comportements agonistiques et tendent à un niveau élevé d'excitation. 3- les cris pulsés à large bande de fréquences sont émis selon un rythme de 3 à 6 syllabes /s pendant les comportements agonistiques et traduisent un niveau élevé d'excitation. 4- les cris pulsés à large bande de fréquences et comprenant un maximum d'énergie à l'attaque du signal et pour les fréquences allant de 5 à 8 kHz. Très intenses, ce cri est souvent porté de 100 mètres environ dans la nature et constitue un signal d'alarme en relation avec les comportements territoriaux.

3- Relations entre l'audition et les cris.
Une concordance précise est observée entre la date d'apparition des réponses comportementales des jeunes aux sons de fréquences pures (à partir du 15e jour) et l'évolution des émissions acoustiques comme les sifflements dont le nombre émis au cours
d'une situation test décroît à partir du 12e jour de vie (fig.1).
Par ailleurs, une bonne relation est observée entre les caractéristiques des cris spécifiques et les capacités d'audition. La plupart des cris aussi bien chez les jeunes que chez les adultes, sont situés dans une bande de fréquences s'étalant de 0,5 à 15 kHz. La quasi-totalité des cris de faible niveau de pression sonore ont au moins une partie de leur spectre de fréquences qui correspond à la zone du 1er pic de meilleure sensibilité auditive (1-2 kHz). Ce sont les cris d'isolement des jeunes qui coïncident avec la zone du 2e pic de meilleure sensibilité auditive (16-28 kHz). Enfin, les cris les plus intenses (cris agonistiques, cris d'alarme) sont situés dans la zone intermédiaire (2-16 kHz). La diminution de sensibilité auditive surtout nette à partir de 3 ans correspond d'une part à un appauvrissement du contingent cellulaire du ganglion spirale et d'autre part à la faible proportion de lérot de cet âge dans les populations naturelles.

Références
HEART RATE AND THE BEHAVIORAL ORIENTING RESPONSE IN RATS EXPOSED TO A
TIME VARYING NOISE SIGNAL.

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N-7034 Trondheim-NTH.
Norway

Summary

The behavioral orienting response (OR) and heart rate (HR) were simultaneously recorded in rats in response to 85 dB auditory stimuli. The behavioral OR habituated after 50 trials whereas (HR) responses failed to habituate. There was no differential cardiac responding with or without the behavioral OR and the preponderance of all HR changes were accelerations, followed by decelerations and bimodal responses. It is suggested that HR changes are concomitant responses rather than specific components of orienting, and are determined by the somatic and autonomic requirements of the response made by the organism.

Method

Rats exposed to noise were monitored with regard to orienting response, OR, whereas the heart rate was telemetered and recorded directly by a computer. The electrode was surgically secured to one costa and connected via a plug fixed to the skull to a radio transmitter strapped to the animal's back. The animal was kept in a sound insulated box with controlled temperature and humidity, and observed via CCTV.

The stimulus was modulated white noise recorded on tape. The background noise was 75 dBA with variable fluctuations. The stimuli were identical noise bursts with a level of 85 dBA. Interstimulus intervals varied between 30 and 80 sec, see figure 1.

![Stimulation signal](image)

**Figure 1.**
Time plot of a section of the stimulation signal.

Experimental procedure

Twelve naïve male Möll Wistar rats were used in the final experiment. They were approximately 120 days old and were housed individually with free access to food and water.
The animals were habituated to the background noise for 30 minutes per day for three consecutive days prior to the test. On the test day each animal was conditioned to the background noise for 30 minutes and then presented with 50 stimulations.

An orienting response was obtained by quantifying the change in post-stimulus as compared to pre-stimulus behavior.

The heart analysis was restricted to 50 pre-stimulus and 100 post-stimulus beats. A 10-beat change in either direction was considered a cardiac response. Both acceleration, deceleration and bimodal response was observed.

Figure 2 represents the combined data for one single animal. The pre-stimulus HR is given in the top line. The black dots indicate the occurrence of a stimulus. The second line is the HR response and the third line the behavioral response.

We used two different background noise signals, but as the modulation depth seemed to be of no importance, all the results are combined in the following.

Figure 3 shows the combined data for orienting response. The results are combined for blocks of 5 stimuli. Each column represents 5 x 12 stimuli. The diagram shows percent of subjects with OR. The subjects showed distinct responses to the initial stimulations and habituated gradually upon repeated stimulations.
Figure 4 shows the combined data for cardiac response. The results are combined for blocks of 5 stimuli. The responses were predominantly accelerations (63% of all) followed by decelerations (24%) and bimodal response (13%).

![Figure 4](image)

The HR response do not show a clear habituation. 84% of the animals showed some form of cardiac response on trials 1-10 and 60% continued to show a response on trials 41-50. Likewise, there was no relationship between the direction of HR change and the number of stimulus presentations.

No consistent relationship was found between the occurrence of the behavioral OR and the direction of HR change. However, some form of cardiac response was exhibited more often when an OR was present than when it was not, see figure 5.

![Figure 5](image)

Discussion

Our behavioral criteria successfully delineated an orienting response which habituates. The accompanying HR response, however, does not relate unequivocally to the OR. There is thus no support for any "cardiac OR" in the rat in our data. It was expected that the heart response accompanying the behavioral OR would show more accelerative activity due to the somatic requirements inherent in any behavioral response. However, there was a predominance of accelerative activity that was unrelated to the occurrence of the behavioral OR. The heart response during orienting appears to be a concomitant response that is neither dependent on nor specific to the behavioral OR. Rather, the heart responds synchronously with all other physiological systems. For example, arrest is followed by a heart rate decrease, while searching behavior entails muscular contractions with a corresponding heart rate acceleration.
EFFECT OF CHRONIC EXPOSURE OF LOW FREQUENCY ULTRASOUND ON SOME PHYSIOLOGICAL PARAMETERS OF MATURE RATS

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Abstract

For the present investigation a colony of 15 rats, male and female were exposed to ultrasound (18 kHz, 7 W) for a period of 40-50 days. The rats were caged individually in a room where the ultrasound source was hanging from a wall support. The ultrasound source having the following specifications, was kept on for 24 hours each day.

Output power 7 W
Ultrasound Propagation Angle 70°
Sound Pressure 96dB

The animal cages were kept in the direction of maximum intensity within a distance of 3-4 meters.

These data (experimental) were compared with those for control group of animals of the same age and sex, kept under identical environmental conditions. Haematological parameters considered for comparison included total erythrocyte counts (TEC), total leucocyte counts (TLC), differential leucocyte counts (DLC), haemoglobin concentration (HB) and erythrocyte sedimentation rate (ESR). In addition to these parameters, normalized weights (weight of the organ) of different organs (Heart, Kidneys, Ovaries/ Testes, Stomach, Small Intestine, Large Intestine, Liver, Skull and Brain) were also compared.

It was found (Table I) that Hb concentration, ESR and TEC were higher in experimental group as compared to the control ones, DLC did not show any difference within measurable limits while TLC showed an decline.

In case of normalised organ weights (Table II) exposed animals were found having higher weights for heart, kidneys and ovaries/tests, while no such difference was found in stomach, small intestine and large intestine. A decline in weights were found for liver, skull and brain in experimental group of animals as compared to control.
In all these experiments there was found to be no difference in measured parameters among male and female animals of the same age.

It is inferred that ultrasound at this frequency has an effect on the physiological state of the animals.

**Table I**

<table>
<thead>
<tr>
<th>Sr. No.</th>
<th>Parameters</th>
<th>Exp.</th>
<th>Control</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.</td>
<td>Hb (gam/100 ml)</td>
<td>17.32</td>
<td>15.10</td>
</tr>
<tr>
<td>2.</td>
<td>TEC ($10^6$/cumm)</td>
<td>7.35</td>
<td>6.23</td>
</tr>
<tr>
<td>3.</td>
<td>ESR (mm/hr)</td>
<td>2.4</td>
<td>2.0</td>
</tr>
<tr>
<td>4.</td>
<td>DLC (P%)</td>
<td>30.4</td>
<td>29.8</td>
</tr>
<tr>
<td></td>
<td>(L%)</td>
<td>69.6</td>
<td>70.2</td>
</tr>
<tr>
<td>5.</td>
<td>TLC (/cumm)</td>
<td>10,550</td>
<td>11,980</td>
</tr>
</tbody>
</table>

P = Polymorphs
L = Lymphocytes

**Table II**

<table>
<thead>
<tr>
<th>Sr. No.</th>
<th>Organ</th>
<th>Exp.</th>
<th>Control</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.</td>
<td>Heart</td>
<td>0.00320</td>
<td>0.00286</td>
</tr>
<tr>
<td>2.</td>
<td>Kidneys</td>
<td>0.00678</td>
<td>0.00645</td>
</tr>
<tr>
<td>3.</td>
<td>Ovaries/Testes</td>
<td>0.0231</td>
<td>0.0153</td>
</tr>
<tr>
<td>4.</td>
<td>Stomach</td>
<td>0.0137</td>
<td>0.0132</td>
</tr>
<tr>
<td>5.</td>
<td>Small Int.</td>
<td>0.0477</td>
<td>0.0476</td>
</tr>
<tr>
<td>6.</td>
<td>Large Int.</td>
<td>0.0308</td>
<td>0.0309</td>
</tr>
<tr>
<td>7.</td>
<td>Liver</td>
<td>0.0424</td>
<td>0.0451</td>
</tr>
<tr>
<td>8.</td>
<td>Skull</td>
<td>0.0601</td>
<td>0.0762</td>
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<tr>
<td>9.</td>
<td>Brain</td>
<td>0.00464</td>
<td>0.00673</td>
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</table>
3.6

Applications médicales
Medical applications
Medizinische Anwendungen
Summary
The paper describes a μP-controlled Békésy-Audiometer which adapts automatically to the reactions of the patient by varying the speed of frequency and level shiftings. The evaluation of the registered frequency and level data in the reversing points of the curve yields a smoothed audiogram with a frequency resolution of 1/6 octave. The data can be stored on floppy-discs for further investigations and comparisons.

Resumé
Il s'agit dans ce compte-rendu d'un audiomètre Békésy qui peut s'adapter de façon automatique dans ses paramètres aux patients par variation de la progression de la fréquence et du niveau d'intensité. L'évaluation des données aux points d'inversion de la courbe donne un audiogramme égalisé de résolution de fréquence d'1/6 octave. Les résultats peuvent être stockés en vue d'une utilisation ultérieure ou de comparaisons.

Zusammenfassung
1. Einleitung

Békésy-Audiometer werden in der ohrenärztlichen Praxis viel verwendet, weil die mit ihnen gewonnenen Ergebnisse zumindest bei kooperationswilligen Patienten eine detaillierte Übersicht über die vorhandene Hörfähigkeit liefern.

Die Vermessung der Hörschwellen bzw. Mithörschwellen erfolgt beim B.-A. durch pendelndes Angleichen, d. h. der Patient reagiert auf die Grenze zwischen Hören und Nichthören eines Testtones, ggf. in Anwesenheit eines Störtones oder Maskierers. Der pendelnde Pegel des Testtones wird in einem Frequenz-Pegel-Diagramm in Form einer Zuckenkurve registriert. Die Auswertung erfolgt i. a. manuell, indem nach Augenmaß eine Mittelwertkurve in das Diagramm eingezeichnet wird. (Fig. 1) Um ein gut auswertbares Audiogramm zu erhalten, muß vor der eigentlichen Aufnahme des Audiogramms in einem Probelauf eine Anpassung des Frequenz- und Pegelvorschubs an die Reaktionsfähigkeit des Patienten erfolgen. Diese Aufgabe wird vom Praxisteam erledigt.

2. Entwicklung eines µP-gesteuerten Békésy-Audiometers

Bei der Entwicklung eines neuen B.-A. im Institut für Kommunikationswissenschaft der TU Berlin wurde die Frage gestellt, ob das B.-A. sich für die Automatisierung eignet. Folgende Gesichtspunkte sollten dabei berücksichtigt werden:
- automatische Anpassung des Frequenz- und Pegelvorschubs an die Reaktionsfähigkeit des Patienten
- sichere automatische Verfolgung der Hörschwellen auch bei stellflankigen Verläufen der Hörschwelle
- automatische Auswertung der Zuckenkurve und Ausgabe eines gedruckten Protokolls
- programmgesteuerte Vertäubung bzw. Maskierung
- automatische Kalibrierung der Kopfhörer mit Hilfe eines Ohrkupplers
- Verwendbarkeit als Festfrequenz-Audiometer
- Erstellung einer Hörschwellendatei auf Disketten für spätere Vergleiche und wissenschaftliche Untersuchungen

Die Aufgabe sollte mit Hilfe eines Minirechners und dessen üblicher Peripherie - Bildschirmterminal, Drucker und Disketten Speicher - gelöst werden.

Die Lösung führte zu dem in Fig. 2 dargestellten Blockschaltbild. Als Minirechner dient ein Commodore CBM 3032. Die Kopplung des Rechners an die Audiometer-Komponenten erfolgt über den 8 bit breiten "user port". Ein Interface übernimmt die Verteilung der vom Rechner gelieferten Informationen zur Einstellung der Frequenz und des Pegels für den Testton, das Vertäubungsgerät bzw. den Maskierer und die Zuordnung des zu prüfenden Ohres. Die Reaktion des Patienten auf die dargebotenen Signale erfolgt in der üblichen Weise durch Drücken bzw. Loosen einer Taste, deren Schaltzustandsänderungen über das Interface den
Rechner veranlassen, die Einstellungen gemäß dem Programm zu variieren.

3. Komponenten des Audiometers

- Testsignalgenerator (SG)
  Als Sinustongenerator wird ein spannungsgesteuerter Oszillator verwendet /1/. Der Frequenzbereich von 19.5 Hz bis 20 kHz (± 10 Oktaven) wird mit einer Steuerspannung von 0 bis 10 V (± 1 V/Oktave) überstrichen. Die digitale Auflösung von 8 bit (± 256 Stufen) ergibt Frequenzstufen von jeweils 2,75 %.

- Pegelabschwächer
  Das Ausgangssignal des Testsignalgenerators wird mit einem multiplizierenden D/A-Wandler (MDAC) mit 8 bit Auflösung abgeschwächt. Die Dämpfung erfolgt in 240 Stufen zu je 0,375 dB und reicht somit von 0 bis 90 dB.

- Vertäubung
  Das Signal zur Vertäubung des nicht geprüften Ohres wird im einem Rauschgenerator (RG) erzeugt. Es kann mit einem zweiten multiplizierenden D/A-Wandler in gleicher Weise wie das Testsignal abgeschwächt werden. Das Rauschen kann mittels eines digital einstellbaren Terz/Oktavfilter (FI) in seiner Frequenzlage und seiner Bandbreite beeinflußt werden.

- Maskierung
  Anstelle des Rauschens kann ein beliebiges externes Signal in den Vertäubungssignalweg gegeben werden, das dann in der gleichen Weise von einem Programm in Pegel und Frequenzlage eingestellt werden kann.

- Kommutator
  Mit Hilfe eines programmgesteuerten Umschalters kann die Zuordnung der Signale zu den beiden Ohren beliebig erfolgen.

- Automatische Kalibrierung (AUTOCAL)
  Mit Hilfe eines Ohrkupplers (AR), dessen Ausgang einem Triggerverstärker zugeführt wird, kann der Frequenzgang der ver-
wendeten Kopfhörer automatisch ermittelt werden. Als Prüfsignal wird das terzgefilterte Rauschen aus dem Rauschgenerator (RG) benutzt. Die Frequenzgangdaten werden registriert und bei der Auswertung der Audiogramme automatisch verrechnet.

4. Funktion des Audiometers

- Anpassung an die Reaktionsgeschwindigkeit des Patienten
  In einem Probelauf, der normalerweise zwischen 734 und 1332 Hz stattfindet, wird fortlaufend der mittlere Pegelhub errechnet. Der Pegel- und Frequenzvorschub wird dabei stufenweise so lange geändert, bis der Pegelhub 10 dB nicht mehr übersteigt.

- Anpassung des Pegelvorschubs an steile Hörschwellenverläufe während der Audiogrammaufnahme kann es vorkommen, dass wegen eines steilen Schwellenverlaufs die Vorschubgeschwindigkeit von Frequenz und Pegel zu gering ist (Fig. 3). Tritt $\Delta T < 300$ ms nach der letzten Reaktion des Patienten keine neue Reaktion auf, wird die momentane Vorschubgeschwindigkeit verdoppelt. Sie wird wieder reduziert, wenn die Reaktion innerhalb des halben Zeitintervalls $\Delta T (= 150$ ms) erfolgt.

- Registrierung, Auswertung und Ausgabe der Audiogramme
  Während der Audiogrammaufnahme werden die Frequenz- und Pegelwerte der Umkehrpunkte der Zackenkurve gezeichnet. Bei der Auswertung wird zunächst eine Treppenkurve bestimmt, indem für jede Frequenzstufe im Intervall einer Zackenflanke der Pegelmittelwert zwischen den Umkehrpunkten eingesetzt wird (Fig. 4). Zur Speicherung und für den Druck werden jeweils Werte für eine halbe bzw. ganze Terz zusammengefasst und gemittelt.

/1/ Wolfgang Schulz: "Untersuchungen zur Frequenzgenauigkeit von spannungsgesteuerten Funktionsgeneratoren für elektrotonische Musikinstrumente" Studienarbeit am Institut für Kommunikationswissenschaft der TU Berlin

Die Arbeit wurde als Studienarbeit von cand. ing. Wolfhard Homma am Institut f. Kommunikationswiss. der TU Berlin durchgeführt
SIMULATION OF A LOSS OF FREQUENCY RESOLUTION IN THE EAR.

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Introduction
Circuitry has been developed which aims to simulate the loss of frequency resolution associated with sensorineural hearing loss. Each frequency component in the unprocessed input signal is "blurred" into a band of noise in the processed output. Fine detail in the spectrum is thus lost and the listener experiences an impairment of perception analogous to that of an observer looking through an out-of-focus optical system. Fig.1 shows the simplest form of the speech processing circuitry. The input signal (speech) and a band of low-pass-filtered noise are applied to an analogue multiplier. (The noise has in fact a second-order fall-off, rather than the sharp cut-off shown). The output consists of sum and difference frequencies between the frequency components in the input signal and the noise components.

Fig.1. Block diagram of the basic speech processing system.
In a more advanced three-channel version, the audio frequency range is divided into bands (0-2 kHz, 2-4 kHz, 4 kHz upwards), each of which can be processed separately to allow different degrees of "blurring" in different parts of the spectrum. (The cut-off between the bands has a second-order fall-off at 12 dB/octave, allowing some overlap at the transition points).

Testing

Processed speech signals were played to normally-hearing young adult subjects at a nominal peak level of 65 dB(A) via a pair of loudspeakers in a soundproof testing room. The speech was additionally passed through an A-weighting network and presented in conjunction with a white noise background at 45 dB(A). This gives a "normal" threshold shape which can be modified with extra filtering if it is desired to simulate a loss of sensitivity in particular frequency regions.

Each subject was played a recorded sequence of four different lists, each composed of ten words from a male speaker. These were taken from phonetically balanced lists compiled by the Medical Research Council (U.K.). Different listening conditions were used for each of the four lists. Subjects were asked to identify each word and their responses were scored on the basis of the percentage of words totally correct. Table 1 shows results for the twelve different listening conditions used in all. Each score is the average over twenty subjects (i.e. 200 words).

Table 1. Test results.

<table>
<thead>
<tr>
<th>Condition</th>
<th>0-2kHz</th>
<th>Δf(Hz) 2-4kHz</th>
<th>4kHz upwards</th>
<th>Score (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>95</td>
</tr>
<tr>
<td>B</td>
<td>50</td>
<td>50</td>
<td>50</td>
<td>79</td>
</tr>
<tr>
<td>C</td>
<td>50</td>
<td>100</td>
<td>200</td>
<td>83</td>
</tr>
<tr>
<td>D</td>
<td>50</td>
<td>200</td>
<td>200</td>
<td>76</td>
</tr>
<tr>
<td>E</td>
<td>50</td>
<td>200</td>
<td>400</td>
<td>74</td>
</tr>
<tr>
<td>F</td>
<td>50</td>
<td>400</td>
<td>400</td>
<td>70</td>
</tr>
<tr>
<td>G</td>
<td>50</td>
<td>50</td>
<td>50</td>
<td>70</td>
</tr>
<tr>
<td>H</td>
<td>100</td>
<td>100</td>
<td>100</td>
<td>64</td>
</tr>
<tr>
<td>I</td>
<td>100</td>
<td>100</td>
<td>100</td>
<td>60</td>
</tr>
<tr>
<td>J</td>
<td>200</td>
<td>100</td>
<td>50</td>
<td>58</td>
</tr>
<tr>
<td>K</td>
<td>200</td>
<td>200</td>
<td>200</td>
<td>41</td>
</tr>
<tr>
<td>L</td>
<td>200</td>
<td>200</td>
<td>200</td>
<td>33</td>
</tr>
</tbody>
</table>

The standard error on the scores is typically five percentage points or less.

Condition A has no "blurring".

* Conditions G, I and L were obtained using the single-channel system. All other conditions were obtained with the three-channel system. The lack of correlation between the noise sources in the three channels means that conditions such as H and I are not exactly equivalent.
Discussion

For the range of $\Delta f$ used in these experiments, the observed impairment of speech perception is determined very largely by the amount of "blurring" in the 0-2 kHz frequency range. This can be clearly seen in fig. 2, where subjects' scores are plotted against the $\Delta f$ value used in processing the 0-2 kHz band.

Fig. 2. Test Results

"Blurring" of the higher frequencies (2kHz upwards) with values of $\Delta f$ up to 400 Hz (i.e. noise bands at the output of up to 800 Hz width) produced little or no significant effect on subjects' scores.

These results are perhaps not surprising, since the 0-2kHz region contains much of the formant structure which is much more susceptible to
the "blurring" process than the broad-band noise from fricatives, etc., found in the higher frequency bands.

Future plans

It is hoped to develop this system into a device which could form part of a realistic simulation of sensorineural hearing loss. As well as giving insight into the problems of partially-hearing patients, a realistic simulation of a particular loss would assist in the design of a compensatory hearing aid, which could incorporate the "inverse transform" of features in the loss simulation.

References


EINIGE EIGENSCHAFTEN DES AU迪MOTORISCHEN REFLEXES

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Um die Untersuchungen des audiomotorischen Reflexes zu ermöglichen wurde eine Aparatur nach folgender Blockscheme zusammengestellt (Bild 1). Der Akustische Stimulator gibt uns folgende diskontinuierliche Tonfrequenzen: 500, 1000, 2000 und 3000 Hz. Das ausgewählte Tonsignal wird dem Tonimpulsgenerator zugeführt. Der Tonimpulsgenerator enthält einen regulierbaren elektronischen Schalter mit dem Tonsignale von einer Zeitauf 1 msec bis 1 sec durchlässt. Dieser elektronische Schalter durchlässt auf äussere Anregung von nur einmaliges Tonsignal und gibt den Startimpulse für eine einmalige Strahlbelenkung des Zweistrahlenosilloskopa,
1. Akustischer Stimulator 1a. Tonfrequenzgenerator 1b. Tonimpulsgenerator 1c. Lautstärkeregler und Programmschalter
I = Startimpuls für die Zeitbasis.


Nachdem die Parameter der Tonstimulation festgelegt wurden, erfolgte die Hervorjung, Registrierung und Analyse des audiomotorischen Reflex. Das Reflexpotential von m. orbicularis oculi ist am Bild 2 ersichtlich. Die Parameter des Reflexpotentials sind folgende:

1. $T_L$ = Latenzzeit (Zeitabschnitt zwischen dem Stimmulations-
anfang und dem Anfang des Reflexpotentials.

2. $T_R$ = Dauer der Reflexkontraktion (Dauer des Reflexpotentials)

3. $A_R$ = Amplitudenmaximum des Reflexpotentials.

Im Bild 2 ist dieser Wert mit dem Potential zwischen den Scheitelwerten der maximal positiven und negativen Halbwelle bezeichnet.

Eine andere Art von Messungen wurde ebenfalls mit dem Ton von 1000 Hz bei konstanten Schalldruckpegel von 110 dB durchgeführt wobei die Impulsduer geändert wurde. Man prüfte mit Impulsen von 10, 30, 100 und 300 msec um den Einfluss von dessen Dauer auf die Latenz T₁, Zeitdauer T₂ und die maximale Amplitude des Reflexpotentials A₁ festzustellen. Die Ergebnisse dieser Prüfungen sind auf Bild 4 dargestellt.


Literatur:
HI-FI-HEARING AID

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One of the common characteristics of all commercial hearing aids is poor fidelity. During the past 50 years hearing aids have been miniaturized, but acoustic quality has not improved to any great extent. Improving the electro-acoustical quality enhances the fidelity of hearing aids. Not only the intelligibility of speech, but the enjoyment of theatre and music is aimed at. Electronical means nowadays enable the realization of just such high-fidelity hearing aids.

1. TARGETS

The target of the development of high-fidelity hearing aids is to develop a hearing aid which is free of all whistling due to feedback and avoids noises of poor microphones in addition to showing good electro-acoustical characteristics (like frequency response etc.).

For the customer the main target is to have an invisible hearing aid which means to him not to be discovered as a handicapped man. With respect to this psychologically very important target care has to be taken so as to obtain an acceptable hearing aid.

2. The way of achieving high-fidelity in hearing aids necessitates a large number of regulation units which require a lot of space. The individual fitting of a hearing aid to the customer is a condition sine qua non to ensure high-fidelity.

3. ANALOGOUS SOLUTION

The solution to the problems involved in the previously mentioned targets of a high-fidelity hearing aid lies in separating the one-piece hearing aid into a split-unit made up of a small receiver in the ear (without any means of regulation) and a spacious transmitter in the pocket, in which are incorporated a hi-fi capacity microphone and all the necessary regulation units. This pocket transmitter transmits a high frequency FM or PCM signal to the receiver in the ear.
The receiver in the ear with a volumina of less than 1 cm³ is not visible and it is the spacious transmitter that incorporates all the merits modern electronics can offer.

4. ELECTRONIC CIRCUITS

An electronic diagram for such hearing aids can be seen in figures 1 – 4. Simple circuits in which the frequency response and the compression in every frequency band can be regulated are shown in figure 1, while a more complex system, transponding frequencies from one range to the other, is shown in figure 2. It is even possible to leave the receiver in the ear without a battery when high power transmitting is involved (figure 3).

5. DIGITAL EQUIPMENT

The incorporation of digital electronics into hearing aids has been tested during the past few years. The main problem of making digital filters with low battery consumption is as yet not solved so that a hybrid solution represents the present state of the art. Future development of digital hearing aids will be accelerated by the VLSI-development which will enable a low battery consumption miniaturized hearing aid. In this way it will probably be possible to design all digitized hearing aids as a split unit or later as one-piece unit in the ear, an aim which will be the focus of the industry in the nineties.
DIGITAL HEARING AID WITH AN EMPHASIZED CONSONANT

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The hearing disability of patients with sensorineural hearing loss involves threshold elevations and is often accompanied by a decreasing ability in frequency and time analysis and in slower transient responses. From psychoacoustical experiments using siwa waves, clicks and monosyllables, it was revealed that when the amplitude or duration of a speech signal decreases, discrimination becomes more difficult for hearing loss patients as well as for persons with normal hearing. It would seem that when the amplitude or duration of the speech is increased, the hearing ability of a subject could improve. This theory holds true for persons with normal hearing and persons with conductive hearing loss. However, on increasing the duration of voiceless plosives, speech discrimination becomes heavily impaired. Furthermore, the duration of vowels is considerably longer than that of consonants. Increasing the duration time of the entire syllable would lengthen the speech so as to make regular conversation tediously slow. On the other hand, conversation would be better facilitated if the duration time of consonants could be increased while decreasing that of vowels. There has been no report to date concerning this method of improving the speech discrimination of hearing impaired patients. Although increased amplitude with conventional hearing aids will help increase the hearing ability of sensorineural hearing impaired patients, there is a limit to the improvement of the speech discrimination score. It has been the dream of those patients with poor speech intelligibility to be able to distinguish speech (at least) in quiet places.

This report is concerned with psychoacoustic experiments and their application to the needs of sensorineural hearing impaired patients.

On discriminating a monosyllable consisting of a consonant and a vowel, the patient is liable to miss the consonant part and catch only the following vowel, or confuse one consonant with another. Voiceless plosives such as /p/, /t/ or /k/ which have the short consonantal duration and small energy are the most difficult to discriminate. The nasal consonants /m/ and /n/ are also often confused with each other. The following 3 methods were chosen to emphasize a consonant part of a monosyllable: 1) the amplification of the consonant part, 2) the extension of the duration of a consonant part and 3) the insertion of a silent pause between the consonant and following vowel.

Fig. 1 shows the 3 methods of emphasizing the consonant part (a) and the presentative case of /ka/ (b) and /na/ (c).
From the psychoacoustic experiments performed on voiceless plosives /p/, /t/, /k/, it was found that the most effective amplification of the consonant part was 10 - 20 dB and the length of the silent pause between the consonant and vowel was less than 100 msec, and for nasal consonants /m/, /n/, the amplification was also 10 - 20 dB and the length of duration time was found most effective at less than twice the original. By combining these methods, it was not only possible to limit the amplitude and duration time to a lower level, but definite improvement was seen in patients who had in the previous experiment shown no effect.
The results of improvement are shown in fig. 2. In the case of /pa/, /ta/, /ka/, with the increased amplitude of consonant and insertion of silent pause, recognition improved by 46.7%. Note the remarkable improvement in /ka/ where synthesized sound was recognized by all the subjects. A considerable improvement was seen also in /ma/ and /na/ whose recognition improved to 70% from the original 22.5%.

These experiments proved that emphasizing the consonant part could aid sensorineural hearing loss patients in their speech discriminating ability. With assurance of improved hearing for these patients, we developed the computer hearing aid. The speech recognition of synthesis system of the computer automatically recognizes the monosyllable and separates the consonant from the vowel. After placing appropriate emphasis depending on the results yielded from tests on the consonant part of each patient, resynthesizes the monosyllable and the sound is transferred on line to the earphone. This system was named the speech processing hearing aid.

The hardware equipment used to design the system consists of a microprocessor (Z-80), a wave memory to store speech wave, a 15-channel filter bank which allows monosyllable recognition in real time. A hardware board for monosyllable recognition and an A/D, D/A converter.

The software equipment stores the speech wave in wave memory. Simultaneously the speech wave is put through the filter bank and the spectral envelope is obtained. Monosyllable recognition and discriminating division between consonant and vowel is then automatically achieved. Depending on the monosyllable and the patients ability to discriminate, the system will determine the emphasis to be placed on the consonant. After these process the monosyllable is resynthesized. This process takes approximately 200 msec.

**Fig. 3.** The hardware and the software of the computer hearing aid.
The method used for emphasis in the case of /p/, /t/, /k/ involves the amplification and/or the insertion of the silent pause. For /m/, /n/, the method includes both amplification and extension of the duration time. In the case of other monosyllables, the system is programmed to simply amplify the consonant by 10 dB. If the silent pause is inserted or the consonant is extended, the same time is subtracted from the middle section of the following vowel.

The experimental advantage of this system is that once the test results of the patients are stored in the computer, no pretesting or training is necessary.

We are at present in the process of designing a wearable hardware system to better serve the needs of our patients.

Fig. 4. The outlook of the computer hearing aid.

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Introduction

La communication présente les travaux d'une équipe interdisciplinaire dans le domaine des prothèses auditives de substitution. L'objectif de la recherche est de fournir aux enfants sourds sévères et profonds, déjà en bas âge, des moyens de communication substituts du sens auditif, leur donnant d'abord une certaine ouverture sur le monde sonore et leur facilitant ensuite l'apprentissage de la lecture labiale.

Le principe d'une prothèse de substitution est de transmettre par un autre canal sensoriel l'information acoustique perçue par les entendants : elle fournit donc un équivalent, visuel ou tactile par exemple, de l'environnement sonore sous formes d'associations entre les éléments perçus par le sens de substitution et les événements extérieurs.

Le sens de substitution retenu est le tactile, et dans certains cas le somesthésique. La vue n'a pas été envisagée, car elle est déjà très sollicitée en situation de communication normale. Dans une première étape, les travaux ont porté sur l'étude des capacités informationnelles du toucher, les techniques de réduction de l'information acoustique et l'analyse des composantes de motivation à l'emploi par des enfants sourds d'une prothèse de substitution par voie tactile. Une seconde étape voit la réalisation et l'utilisation en milieu clinique de différents dispositifs acousto-tactiles : jouets, indicateurs et alarmes, prototypes d'une prothèse d'aide à la lecture labiale.

Collaborent au projet : des ingénieurs acousticiens et microtechniciens (École Polytechnique Fédérale de Lausanne), des physiologistes, un psychologue, des médecins et des orthophonistes (Faculté de médecine de l'Université de Lausanne).

Etude du sens tactile

La possibilité d'utiliser le sens tactile comme substitut à l'audition se heurte à de nombreux problèmes aussi bien technologiques que psychophysiologiques. Notre étude a permis de classer les sensations tactiles induites par les vibrations complexes d'une matrice de pointes vibrantes en trois types : vibration, mouvement apparent et texture.
La sensation de vibration apparaît lorsqu'une ou plusieurs pointes vibrent simultanément et en phase entre 10 et 1000 Hz. La sensibilité maximum est aux environs de 300 Hz. La discrimination de deux fréquences est relativement grossière, mais constante ($\Delta f/f \sim 0,5$).

La sensation de mouvement apparent peut être induite lorsque les pointes d'une matrice sont activées en succession. Nous avons développé un bracelet muni de huit microvibrateurs (il s'agit en fait de micromoteurs alimentés par un courant alternatif de 20 à 200 Hz et exécutant de petits mouvements angulaires). Les limites de perception de mouvement, avec ce bracelet, sont de 7,5 à 70 cm s$^{-1}$. Lorsque le signal est modulé (présentation successive et répétitive de deux vitesses différentes), la discrimination de vitesse perçue $\Delta v/v$, relativement constante, est de l'ordre de 0,2.

En situation naturelle, la sensation de texture apparaît le plus souvent lors de la palpation active de surfaces. Nous avons tenté de créer des textures artificielles, perceptibles en mode passif, en utilisant une matrice de 16 pointes vibrantes sur une surface de 0,8 mm$^2$. Cette matrice est composée de 4x4 éléments piézoélectriques dont la fréquence de vibration peut être réglée de 40 à 500 Hz; la surface de contact de chaque pointe avec la peau est d'environ 0,1 mm$^2$. Un microprocesseur commande la présentation des patterns vibratoires. Nous avons testé la sensation de texture provoquée par la vibration simultanée de toutes les pointes à quatre fréquences différentes (deux pointes adjacentes ne vibrent jamais à la même fréquence) avec l'espoir que la discrimination de tels patterns soit meilleure que celle de vibrations simples. Or nous n'avons pu confirmer cette hypothèse.

Face aux difficultés de créer des stimuli d'un seul type induisant des sensations facilement discriminables à l'intérieur de l'un de ces trois modes de perception, nous avons testé la discrimination de cinq stimuli discrets appartenant à ces trois modes. Les résultats préliminaires indiquent que dans ces conditions l'on peut atteindre un débit d'information de 10 bits s$^{-1}$ qui serait probablement suffisant pour une aide à la lecture labiale.

Informations acoustiques

La capacité informationnelle relativement faible du sens tactile oblige à réduire drastiquement le débit d'information acoustique en n'en conservant que les éléments essentiels porteurs de signification, dépendant du type de prothèse considéré.

Dans le cas d'une aide à la lecture labiale, ces éléments sont ceux complémentaires à cette dernière, permettant de lever l'ambiguïté entre sosies labiaux, définis au niveau phonémique. Les démarches suivies sont : 1) définition de ces informations à partir des traits distinctifs articulatoires (un système simplifié a été élaboré pour le français); 2) établissement des relations entre traits articulatoires des sosies labiaux et paramètres observables du signal acoustique; 3) élaboration des algorithmes d'extraction de ces paramètres; 4) implantation dans une technologie appropriée.

Le premier prototype d'aide à la lecture labiale valorise le fait que certains paramètres sont relativement aisément décelables dans le signal.
acoustique : à une valeur d'un paramètre articulatoire correspondent des valeurs données des paramètres observables du signal de parole. Dans ce cas, les algorithmes sont encore assez simples, le débit d'information résultant en sortie compatible avec le sens tactile et l'implantation en temps réel possible au moyen d'un système multiprocesseur. Ont été d'abord retenus : les traits de sources glottique (valeurs : muette, voisée) et articulatoire (valeurs : muette, plosive, constrictive) et les éléments prosodiques de tonie et d'intensité acoustique. Sont prévus dans l'étape suivante (après évaluation clinique de la première aide) : les traits du lieu d'articulation, d'aperture et de nasalisation.

Afin d'avoir un maximum de souplesse et de performances, et une utilisation clinique aisée, un système multiprocesseur modulaire a été spécialement développé. Il met en oeuvre des processeurs de signal rapides NEC μPD7720 associés en un réseau, géré par un séquenceur spécialisé, et des interfaces d'entrée et de sortie ad hoc. Chaque processeur se charge d'une ou plusieurs tâches (filtrage, calcul de valeur efficace, etc..) programmables en fonction des algorithmes choisis.

Les prothèses de substitution apportant une information sur des événements sonores particuliers (appels, alarmes, détections, etc.) ont été abordées par une autre approche. Les principes appliqués sont d'une part un traitement approprié du son (analogique ou numérique) pour en extraire les éléments sémantiques concernés, d'autre part une stimulation tactile ou somesthésique. Pour la seconde nous avons développé un module stimulateur électrique de faibles dimensions et consommation, très efficace et bien supporté, car se basant non pas sur la perception électrocutanée mais sur celle des mouvements musculaires induits par l'excitation électrique.

L'enfant sourd face à une information tactile

On connaît mal les réactions et l'intérêt des enfants sourds à une présentation d'informations par le canal tactile. Afin d'étudier cette question nous avons développé deux jouets recourant à une stimulation tactile.

Le but du premier jouet est d'identifier et de localiser, dans un ensemble de dix objets, deux types d'objets. L'information pertinente permettant de discriminer les deux types est donnée par un pattern vibrotactile que l'on obtient que par une recherche active. Cette recherche est contrôlée par une pression fine du pouce alors qu'un feedback visuel permet d'identifier l'objet dont provient le signal vibrotactile perçu. La maîtrise du jeu implique une coordination efficace entre la pression du pouce, les indices visuels de localisation et l'information vibrotactile. Les 17 enfants sourds testés (7 âgés de 5 à 6 ans et 10 âgés de 7 à 12 ans) apprirent la tâche en moins de 10 minutes et comprirent les règles du jeu. Néanmoins la mémorisation de la position de chacun des deux types d'objets s'est avérée impossible, à moins que l'enfant n'ait recours à une stratégie de verbalisation. Cette observation montre toute l'importance du langage comme élément de structuration des tâches complexes.

Le deuxième jouet a pour but d'identifier quatre sources sonores différentes. Le signal acoustique est traité par un filtre passe-bande de largeur variable et dont la fréquence centrale est commandée par le sujet. La sortie de ce filtre est ensuite divisée pour obtenir un signal de fréquence
perceptible tactilemente. Le spectre de la source sonore peut ainsi être exploré activement en contrôlant la position de la fréquence centrale du filtre au moyen d'une pression du pouce. Un feedback visuel peut être utilisé pour contrôler la position du filtre dans le spectre de la source, lequel est obtenu par un banc de filtres passe-bande incorporé, puis affiché par des LEDS (photo en fin de texte). Tous les enfants testés avaient préalablement utilisé le premier jouet décrit et étaient donc familiarisés avec ce genre d'investigations. Après quelques essais, ils étaient tous capables d'identifier les sources sonores utilisées.

Ces expériences, aussi simples soient elles, montrent que des enfants sourds peuvent apprendre à utiliser activement l'information tactile dans le cadre d'une activité ludique. On peut donc espérer que l'acquisition du langage chez l'enfant sourd puisse bénéficier de la présentation d'une partie de l'information acoustique par le canal tactile.

Deuxième jouet tactile, avec de gauche à droite : le casque d'écoute du moniteur (contrôle); la "tête", tenue en main par l'enfant malentendant et comprenant un microphone, un palpeur de commande (pouce) et un vibreur (paume); le boîtier d'électronique et le bloc d'alimentation secteur. La fenêtre noire correspond à l'affichage pouvant être mis en ou hors fonction.
A NEW HEARING AID AND ITS FITTING PROCEDURE BY MASTER HEARING AID

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INTRODUCTION
There are some complaints from users of hearing aids. The majority of them are as follows, (1) annoying noises, (2) hard to understand speeches, (3) getting tired with hearing. In order to solve these problems altogether, a new type hearing aid and new fitting procedure called here "fitting circle" has been developed. One of the main points of this procedure is for measuring the auditory sense by flat response earphone which is identical earphone used for the new type hearing aid. This point intend to reduce the fitting error of sound pressure on actual hearing. Another point is for adjusting the acoustic characteristic of the new type hearing aid so as to the dynamic range characteristic of individual hearing impairments.
This study is related to some researches on wearable Master Hearing Aid (MHA).(1)

FITTING CIRCLE(2)
Figure 1 shows the new procedure for fitting of hearing aid and auditory training repeatedly.

![Diagram of fitting circle]

NOUVEL APPAREILS AUDITIFS ET SON SYSTEME D'ADAPTATION AVEC AIDE AUDITIVE PRINCIPALE.
Upper flow is for measuring the auditory sense of hearing impairment by MHA. Lower flow is for adjusting the new hearing aid and for training auditory sense including in daily life.

**SPL AUDIOMETRY**

As a preliminary research, SPL audiograms of 100 hearing impairments who intended to use hearing aids have been measured at random. Relationships between characteristics of UCL and MAL are shown in Table 1.

- ○ type UCL holds 35%. It is hard to fit by conventional hearing aids.
- ○ type UCL holds 23%. It is hard to fit if dynamic range is narrow.
- ○ type UCL is only 3%. This type is easy to fit by conventional hearing aid.
- ○ type MAL holds 51%.
- ○ ○ type UCL, MAL combination holds 27%.

Data to decide the specifications of MHA and the new type hearing aid were obtained from the analized results of these audiograms.

<table>
<thead>
<tr>
<th>Type of UCL</th>
<th>Type of MAL</th>
<th>Examples</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Total</td>
</tr>
<tr>
<td>⌒</td>
<td></td>
<td>7</td>
</tr>
<tr>
<td>⌒</td>
<td>⌒</td>
<td>8</td>
</tr>
<tr>
<td>⌒</td>
<td>⌒</td>
<td>51</td>
</tr>
<tr>
<td>⌒</td>
<td>⌒</td>
<td>2</td>
</tr>
<tr>
<td>⌒</td>
<td>⌒</td>
<td>11</td>
</tr>
<tr>
<td>⌒</td>
<td>⌒</td>
<td>18</td>
</tr>
</tbody>
</table>

Table 1 Matrix of MAL, UCL classified by type. Rejected 3 examples are impossible to measure.

Tableau 1 Matrice de MAL, UCL classifiée par type. Les 3 exemples rejetés sont impossibles à mesurer.
FITTING CONDITION

MHA: This instrument is made up of 5 channel AGC amplifiers.
Centre frequency: 250, 500, 1k, 2k, 4kHz
Maximum SPL: 90 - 130 dB SPL
Maximum acoustic gain: 70 dB

New type hearing aid: This is made up of 3 channel AGC amplifiers, other characteristics are similar to MHA.

Subjects: M.I., M.S., H.O., are aged impairments, whose ages are 57 - 78.
C.K., M.K., are sensorineural impairments, whose ages are 19, 21.

Procedure: According to the fitting circle, measuring and fitting by MHA, and auditory training by the new type hearing aid were repeated during several weeks.

RESULTS AND DISCUSSION

Typical data by this fitting circle is shown in Table 2.

<table>
<thead>
<tr>
<th>Item</th>
<th>Subject</th>
<th>M.I.</th>
<th>M.S.</th>
<th>H.O.</th>
<th>C.K.</th>
<th>M.K.</th>
</tr>
</thead>
<tbody>
<tr>
<td>MAL</td>
<td>500 Hz</td>
<td>95</td>
<td>95</td>
<td>75</td>
<td>80</td>
<td>75</td>
</tr>
<tr>
<td>dB SPL</td>
<td>1000 Hz</td>
<td>80</td>
<td>90</td>
<td>95</td>
<td>90</td>
<td>95</td>
</tr>
<tr>
<td></td>
<td>2000 Hz</td>
<td>75</td>
<td>95</td>
<td>120</td>
<td>95</td>
<td>130</td>
</tr>
<tr>
<td>UCL</td>
<td>500 Hz</td>
<td>125</td>
<td>125</td>
<td>130</td>
<td>125</td>
<td>105</td>
</tr>
<tr>
<td>dB SPL</td>
<td>1000 Hz</td>
<td>120</td>
<td>120</td>
<td>125</td>
<td>125</td>
<td>115</td>
</tr>
<tr>
<td></td>
<td>2000 Hz</td>
<td>120</td>
<td>125</td>
<td>125</td>
<td>120</td>
<td>130</td>
</tr>
<tr>
<td>Dynamic range</td>
<td>500 Hz</td>
<td>30</td>
<td>30</td>
<td>55</td>
<td>45</td>
<td>30</td>
</tr>
<tr>
<td>dB</td>
<td>1000 Hz</td>
<td>40</td>
<td>30</td>
<td>30</td>
<td>35</td>
<td>20</td>
</tr>
<tr>
<td></td>
<td>2000 Hz</td>
<td>45</td>
<td>30</td>
<td>5</td>
<td>25</td>
<td></td>
</tr>
<tr>
<td>Discrimination (syllable)</td>
<td>own HA</td>
<td>80</td>
<td>54</td>
<td>25</td>
<td>26</td>
<td>22</td>
</tr>
<tr>
<td>%</td>
<td>new HA (start)</td>
<td>65</td>
<td>58</td>
<td>34</td>
<td>24</td>
<td>22</td>
</tr>
<tr>
<td></td>
<td>new HA (finish)</td>
<td>96</td>
<td>78</td>
<td>44</td>
<td>37</td>
<td>35</td>
</tr>
<tr>
<td>Discrimination (vowel)</td>
<td>own HA</td>
<td>100</td>
<td>94</td>
<td>74</td>
<td>88</td>
<td>78</td>
</tr>
<tr>
<td>%</td>
<td>new HA (start)</td>
<td>100</td>
<td>99</td>
<td>78</td>
<td>71</td>
<td>72</td>
</tr>
<tr>
<td></td>
<td>new HA (finish)</td>
<td>100</td>
<td>100</td>
<td>88</td>
<td>96</td>
<td>86</td>
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<tr>
<td>Subjective evaluation</td>
<td>noisy or not</td>
<td>O</td>
<td>O</td>
<td>O</td>
<td>O</td>
<td>O</td>
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<tr>
<td></td>
<td>articulation</td>
<td>O</td>
<td>O</td>
<td>Δ</td>
<td>Δ</td>
<td>Δ</td>
</tr>
<tr>
<td></td>
<td>fatigu or not</td>
<td>O</td>
<td>O</td>
<td>O</td>
<td>O</td>
<td>O</td>
</tr>
</tbody>
</table>

Table 2 Results of auditory improvement.
Tableau 2 Résultats de l'amélioration auditive.
M.I.'s discrimination score decreased from 80% to 65% at the beginning of this experiment. Then it increased to 96% at the end of this experimental term. This result is considered that his auditory sense has been trained by his own hearing aid for a long time and new training by new type hearing aid brought about final improved score of discriminations.

M.I. and M.S. had relatively wide and flat dynamic range characteristics, and they obtained high discrimination scores.

H.O. and M.K. had narrow dynamic range characteristics in high frequency range and they could not obtain high discrimination scores, even by applying the fitting circle.

C.K. had relatively wide and flat dynamic range characteristic but she should not obtain high discrimination score, even by applying the fitting circle. She was handicapped for speaking naturally.

Scores in discrimination tests has been improved at the rate of 20 - 76% by applying the fitting circle. Subjective evaluation has been improved comparing with own conventional hearing aid.

CONCLUSION
In order to solve the major complaints for using the conventional hearing aid, a new type hearing aid and fitting circle has been developed.

As a preliminary research, 100 SPL audiograms of hearing impairments were measured for designing the new type hearing aid and MHA. Results of this preliminary measurements showed that UCL and dynamic range (UCL-MAL) characteristics were more important than MAL for exact fitting, and also showed that dynamic range of some patient were narrow especially in mid and high frequency ranges. For compensating some differences of sound pressure generated by earphones between the hearing aid and audiometer, flat response earphone was used for the new type hearing aid, and the identical earphone was used for the MHA. Comfortable fitting under the condition of all kinds of noise and conversation was set with the fitting circle. After several weeks, fitting circle was finished and the scores of discrimination were improved distinguishly.

REFERENCES
(3) M. Uemura, K. Miura, T. Yoshiba. The hearing inspection from the view point UCL and MAL. J.A.S.J. 1982. 3.
ETUDE EXPERIMENTALE DE L'OTOTOXICITE COMPAREE DE QUATRE
AMINOSIDES PENDANT LA PERIODE CRITIQUE DU DEVELOPPEMENT COCH-
LEAIRE

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Introduction

Des études électrophysiologiques récentes, réalisées chez le rat, ont montré
l'existence d'une période critique de sensibilité aux antibiotiques ototoxiques
pendant le développement (Osako et col., 1979 ; Carlier et Pujol, 1980 ; Marot
et col., 1980). Cette période correspond à la phase d'entrée en fonction des
récepteurs cochléaires (2ème semaine postnatale chez le rat). Au cours de
cette période, l'antibiotique a un effet ototoxique à des doses 10 à 15 fois
plus faibles que chez l'animal adulte.

Ces résultats suggèrent le risque d'utilisation des aminosides chez le nou-
veau-né humain. De ce fait il nous a paru utile d'étudier sur l'animal, pendant
la période critique du développement, l'ototoxicité comparée des quatre amino-
sides les plus couramment utilisés en clinique humaine : dibékacine, tobramy-
cine, gentamicine et amikacine. Cette étude physiologique et histologique
a été faite sur le rat blanc.

Matériel et méthode

- L'étude physiologique a été réalisée sur 36 rats nouveau-nés. Les animaux,
  répartis en 4 groupes, ont été traités par une injection intra-musculaire quoti-
dienne d'antibiotique, pendant 8 jours consécutifs, du 9ème au 16ème jour
  postnatal. Les doses d'antibiotiques choisies correspondaient à 20 fois la dose
  thérapeutique humaine (D.T.H.) pour la dibékacine (soit 60 mg/kg/j) et 15
  fois la D.T.H. pour les trois autres antibiotiques (soit 225 mg/kg/j pour l'
amikacine, et 45 mg/kg/j pour la gentamicine et la tobramycine).

La fonction auditive des animaux a été testée par électrocochléographie
un mois après la fin du traitement. Le seuil des potentiels cochléaires (potentiel
d'action et potentiel microphonique) a été déterminé avec des tone-bursts
de différentes fréquences : 0.5, 1, 2, 4, 8 et 16 kHz.

- L'étude histologique a été réalisée à la fin des enregistrements électro-
  cochléographiques sur 16 animaux, soit 4 animaux par antibiotique testé. Dans
  chaque groupe, 2 rats ont été utilisés pour des comptages cellulaires de la
cochlée à partir de la technique de préparation de surface. Les cochlées des 2 autres rats ont été observées au microscope à balayage.

Résultats

- Physiologiques :
Les traitements par dibékacine et tobramycine n’ont pas entraîné d’altération des potentiels cochléaires. La gentamicine a provoqué une élévation discrète des seuils des potentiels cochléaires dans les fréquences aiguës. Au contraire l’amikacine a déterminé une perte auditive sévère sur l’ensemble des fréquences testées.

- Histologiques :
Aucune perte cellulaire et aucune anomalie des cils des cellules ciliées n’ont été observées après traitement à la dibékacine.

Pour les animaux traités à la tobramycine on a observé au niveau du tour basal de la cochlée, la perte de quelques ciliées externes et des signes de dégénérescence ciliaire.

La gentamicine a provoqué des dommages plus importants des cellules ciliées externes au niveau du tour basal: perte des cellules, fusion des cils et formation de cils géants. Certaines de ces anomalies ciliaires ont également été retrouvées à l’Apex.

Le traitement à l’amikacine a provoqué une disparition massive des cellules ciliées externes sur l’ensemble de la cochlée, et des cellules ciliées internes du tour basal.

Conclusions

L’ensemble des résultats physiologiques et histologiques s’accordent pour établir que la dibékacine est l’aminoside le moins toxique des quatre antibiotiques étudiés. La tobramycine a une faible toxicité. La gentamicine et surtout l’amikacine présentent des risques ototoxiques plus évidents, en particulier dans la période du développement cochléaire. Cette étude constitue une base expérimentale à l’utilisation clinique des aminosides en Néonatalogie.

Références

