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UNIVERSITY OF Miami

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23 June, 1992

Dr. Clifford G. Lau
Scientific Officer Code 1114SE
Office of Naval Research
800 North Quincy Street
Arlington, VA 22217-5000

Dear Dr. Lau:

In accordance with instructions received re my Grant N. N00014-92-J-1192 enclosed please find 3 copies of a Semiannual Performance Report. It contains mainly new ideas on which I have been working, but does not include reference to Neural Net ideas suggested to me by Professor Fred Wu here at the University of Miami, EE Dept.

I realize that as PI on this grant the sole responsibility for exercising scientific judgement regarding its execution rests on me. However, I am happy to say that I have benefitted from your counsel both at the start and when I spoke with you by phone not long ago. I would like very much to hear your response to the enclosed report.

I believe I will be ready to submit a full report and possibly also a publication by the end of September. While I realize it may be too early to broach the subject, I am currently optimistic that a workable device can be implemented and would like to inquire what level of funding might be available for continuation beyond the Feasibility Assessment stage.

Sincerely,

Jorge F. Willemsen

Rosenstiel School of Marine and Atmospheric Science
Division of Applied Marine Physics
4600 Rickenbacker Causeway
Miami, Florida 33149-1098
(305) 361-4160

Jorge F. Willemsen
ROSENSTIEL SCHOOL OF MARINE AND ATMOSPHERIC SCIENCE
UNIVERSITY OF MIAMI
4600 Rickenbacker Causeway
Miami, Florida 33149

PROGRESS REPORT
submitted to

OFFICE OF NAVAL RESEARCH

Feasability Assessment of a Transient Sound Sensor Based on the Silicon Retina Architecture

June, 1992

Statement A per telecon Clifford Lau
ONR/Code 1114
Arlington, VA 22217-5000

NWW 6/30/92

I. Review of the Project Goals

The principal purpose of the project as specified in the proposal of September 1991 is "to evaluate the conceptual design of an acoustic system which is purposefully sensitive to the identification and localization of brief transient sounds".

In the Background section of the proposal I described the basic conceptual elements which underlie the Silicon Retina (SR) and those which underlie passive mammalian hearing. My original intention was to utilize the frequency discrimination capability manifested in mammalian hearing as the mechanism on which to base an acoustic analogue to the SR concept.

Specifically, mammals have sophisticated frequency dependent sensors which transmit responses to discrete and well-resolved frequencies upward to higher processing levels. Each intermediate processor transmits the precise frequency information upstream. Thus one may think of acoustic frequencies as being analogous to visual positional information, since in the ray limit of optics there is a one to one correspondence between the relative locations of external objects and the retinal sensors which are stimulated at any given time. The SR mechanism was to be implemented by comparing individual frequencies with running averages over groups of nearby frequencies, eliciting a response when the intensities of these differed by a prescribed amount.

The next step in the program was to remain open-minded regarding the implementation of directional acoustic arrays, with the idea pf steering an array making use of the "alert" response created by the introduction of a non-background signal. In this way it was envisioned that the "cocktail party effect" would be achieved.

II. Modifications to the basic strategy

In the time intervening since the proposal was submitted and approved, I have become aware that the mammalian auditory system is far from unique in purpose and function. Evidently hearing has been "invented" somewhere near 20 different times in nature, many implementations being radically, rather than merely superficially, different from one another.

Not surprisingly the auditory system of the fishes is adapted to three dimensional perception, as is appropriate for the underwater environment. As such it is essentially different from that in mammals. There is no reason of principle to constrain the conceptual framework to one or another biological paradigm. We have the freedom to draw desired elements from any of these. In what follows I develop reasons why hearing in fishes may provide methods for implementing the desired "SR cocktail party effect solver" which go beyond what I had originally envisioned.

III. Physiological and physical aspects of fish auditory systems

A. Physiological characteristics

Mammalian hearing achieves its precise frequency discrimination through a specialized organ known as the Organ of Corti. In idealized terms, a shaped nonlinear membrane has evolved which maximally responds to different frequencies at different locations along its length. The localized vibrations of this membrane stimulate "hairs", which in turn stimulate nerve endings to transmit the signal up to higher processing levels.

Fish, it seems, have no direct analogue of the Organ of Corti, nor do they possess the same degree of frequency sensitivity and discrimination exhibited by mammals. Their needs are different. To begin with their body mass has nearly the same density as the surrounding acoustic medium, and in consequence they may be roughly described as oscillating bodily in response to the acoustic field. Consequently hearing is possible for them only through utilization an *inertial* sensory system.

I now describe this inertial system in broad outline [1]. Rather than developing a nonlinear frequency discriminator, the fishes evidently evolved a system based on the presence of a "dead-weight" bony structure which is inertially stable in the presence of the acoustic oscillation of the bulk of the fish. As in mammals, "hairs" are utilized to sense the relative motion between the fish and the bone. These hairs are anchored on one end to the bone and on the other to a membrane containing nerve endings. Thus one may think of the tugging of the hairs caused by the relative motion of the endpoints as being the primary stimulus to "hearing".

The design of the receptor is such as to enable directional hearing. Both the magnitude of the soundspeed in water (many times that of sound in air) and the comparatively small distance across the head of a typical fish render the mammalian methods of directional discrimination (time delay and refraction) ineffective. Instead, the fish (in a manner of speaking) hook the bone to hairs pointing in different directions. This wiring scheme enables an x-y-z decomposition of an incoming signal, although 180° ambiguities must be resolved through subsequent processing. (A lucid discussion of a model for how th is is accomplished is contained in Rogers et al, Ref.[1]. These authors also discuss frequency discrimination. I will not include any of these details in the present report.)

A moment's reflection reveals that the system described above must rely upon pressure gradients rather than absolute pressures to achieve its function. A further elaboration of the scheme utilizes the absolute pressure in conjunction with the three Cartesian components of the pressure gradients. While the details of the biological implementation are interesting, I will not dwell on them here. *Rather, I pass to a discussion of how oceanic noise would stimulate a system designed on the above conceptual basis, and how a desired signal could be identified.*

B. Physical considerations

In what follows I will assume that it is possible to construct acoustic transducers which can detect the absolute pressure level and the three Cartesian components of the pressure gradient as functions of time. Such transducers could either be of inertial design, mimicking that of the fish, or could utilize (to give one more technologically sophisticated example) laser interferometry. Models for how fish might process acoustic signals utilizing such "transducer" systems have been given by a number of authors. Here I draw upon the general ideas of Rogers et al [1], although I do not adhere to their all the specifics of their model, since my aim is different.

For the purposes of the present discussion I also limit the background noise field to consist of surface noise, though other types of noise will be introduced later in the project . I do not foresee that their introduction will require radical modification of the ideas to be presented here. Again, many researchers have contributed to our knowledge of surface noise, but I will restrict my attention to the formulation of Ffowcs Williams and Guo [2,3] in the present work. While there is no universal agreement regarding the mechanisms of surface noise generation, the work of these authors is based upon the Lighthill definition of sound sources:
"the source of sound (is) the difference between the exact statements of natural laws and their acoustical approximations."[4]

Based upon this very general formulation of the nature of the sources of transversely homogeneous and isotropic surface noise, it can be shown that the far-field pressure in the water $p(z,\mathbf{x},t)$, where z (<0) is the depth and $\mathbf{x}$ is a two-dimensional transverse coordinate, is of the form

$$p(z,\mathbf{x},t) = \mathcal{R}(t - |z|/c_W). \tag{1}$$

Here $\mathcal{R}$ is a function whose argument exhibits retardation of the arrival at soundspeed c_W, but which is otherwise arbitrary. In fact, due to the irregular motion of the ocean surface, it is best regarded as a random function. Proof of this assertion is deferred to the Appendix, in which a more general situation in which p depends on $\mathbf{x}$ is also discussed.

It follows that

$$\nabla_{\mathbf{x}}\, p = 0;$$

$$\nabla_z\, p = -\, c_W^{-1}\ \partial_t\, p. \tag{2}$$

The first result above simply reflects that the surface is assumed to be transversely homogeneous and isotropic, at least in a statistical sense, so there is no meaningful reference point from which to measure $\mathbf{x}$! (If, in a different context, p depends on $\mathbf{x}$, under certain conditions the second relation above can be generalized to $\nabla_{\mathbf{x}}\, p \cong \hat{\mathbf{x}}\,(-\, c_W^{-1}\ \partial_t\, p)$ in the far field, and the ensuing discussion would be suitably modified.) Thus at this level of discussion any transverse gradient whatever would be a "signal" and not surface noise. In practice it would be necessary to perform a running time average since we can expect $\mathbf{x}$ dependent fluctuations to the acoustic pressure to occur.

For convenience let us now define $\phi_1 = \nabla_z p$ and $\phi_2 = - c_w^{-1}\ p$. Construct the running time average

$$\begin{aligned} \langle \phi_1 \rangle_t &= \frac{1}{\Delta T} \int_{T_o}^{T_o + \Delta T} dt \ \ \phi_1 \\ &= \frac{1}{\Delta T} \int_{T_o}^{T_o + \Delta T} dt \ \ \partial_t \phi_2 \\ &= \frac{1}{\Delta T} \ (\phi_2(T_o + \Delta T) - \phi_2(T_o)). \end{aligned} \tag{3}$$

In words this relation states that the running time average over the z component of the gradient should equal a specifically weighted difference of the absolute pressure measured at the endpoints of the averaging interval. *This is a computation eminently suited to analog methods*. It will be especially useful if it can be used to differentiate signal from noise.

To investigate this possibility write a total pressure field (suppressing the z dependence)

$$P(t) = p(t) + S(t)\, \Theta\, (t - t_o)\, \Theta\, (t_f - t) \tag{4}$$

where p(t) is the noise, S(t) is a signal, and Θ are Heaviside functions.

Let the symbol Φ refer to the total field P, and Ψ refer to S using the same definitions introduced for ϕ referring to p. We find

$$< \Phi_1 >_t = \frac{1}{\Delta T} \left(\phi_2(T_o + \Delta T) - \phi_2(T_o) \right) + \frac{1}{\Delta T} \int_{T_o}^{T_o + \Delta T} dt \quad \Psi_1(t)\, \Theta(t - t_o)\, \Theta(t_f - t) \tag{5}$$

If S is a direct signal in the far-field which has no reverberation components from the surface, bottom, or intervening structures, it will, like ϕ, satisfy $\Psi_1 \cong \partial_t \Psi_2$. In this case the integration can be performed, but the endpoint values depend upon how t_o and t_f are related to the endpoints of the averaging interval. Alternatively, the signal may not satisfy $\Psi_1 \cong \partial_t \Psi_2$ at all, in which case the integral does not simplify.

For definiteness let T_o refer to the first averaging interval in which the signal appears, $T_o + \Delta T > t_o > T_o$. We assume further that at earlier times running averages were performed and Equation (3) was satisfied. Examining Eq.(5) we recognize that $\phi_2(T_0) = \Phi_2(T_0)$, but there is no reason for the rest of the right-hand side to equal $\Phi_2(T_0 + \Delta T)$. Thus the "test" of Equation (3) based upon the observed total field Φ fails.

Let us next suppose that the signal is so long that it completely overlaps at least one subsequent averaging period. In this case, if $\Psi_1 \cong \partial_t \Psi_2$, the right-hand side of Eq.(5) does reconstruct $\frac{1}{\Delta T} \left(\Phi_2(T_o + \Delta T) - \Phi_2(T_o) \right)$ over this subsequent interval, and the "test" is a success. On the other hand if $\Psi_1 \ncong \partial_t \Psi_2$ the test fails. Thus when the signal totally overlaps the noise there is no chance of using success or failure of the test in order to unambiguously decide whether the signal is present.

Finally, it should by now be clear that occurence of the end of the signal within a subsequent averaging interval will certainly produce a failure, followed by a success due to the ongoing noise.

To summarize:

- Monitor over a total time interval much longer than the averaging interval ΔT, during which the average gradient -to- pressure test is succesful. Provided the signal does not accidently begin precisely at the start of the averaging step, there will occur a first averaging interval in which the test fails, indicating the presence of the signal.

- If the signal lasts longer than successive averaging intervals, the test $< \Phi_1 >_t = \frac{1}{\Delta T} \left(\Phi_2(T_o + \Delta T) - \Phi_2(T_o) \right)$ will succeed over those intervals overlapped completely by the signal provided $\Psi_1 \cong \partial_t \Psi_2$, but will fail if this condition is not satisfied.

- The end of the signal sequence will be marked by a sure fail (barring happenstance) followed by a success as the noise persists.

IV. Remaining tasks

In the preceding section I have endeavored to report on new directions in which the sponsored research is moving. The full report at the termination of the project will address the specific issues raised in the project proposal, including simulations of the response of the proposed transient sound selector to signals superposed on realistic models of surface noise. In performing these simulations I will be especially focused on issues regarding optimal time-averaging intervals Δ, given the spectrum of the noise, and on sensitivity.

REFERENCES

[1] "Processing of acoustic signals in the auditory system of bony fish", P.H. Rogers, A.N. Popper, M.C. Hastings, W.M. Saidel, J. Acoust. Soc. Am. **83** (1), 338 (1988).

[2] "Sound generation in the ocean by breaking surface waves", Y.P. Guo, J. Fluid Mech. **181**, 329 (1987)

[3] "Mechanisms of sound generation at the ocean surface", J.E. Ffowcs Williams and Y.P. Guo, in *Sea Surface Sound,* pp.309-324, B.R. Kerman (ed.), (1988), Kluwer Academic Publishers

[4] See, e.g., A.P. Dowling and J.E. Ffowcs Williams, *Sound and Sources of Sound,* Ch.7,John Wiley and Sons, (1983)

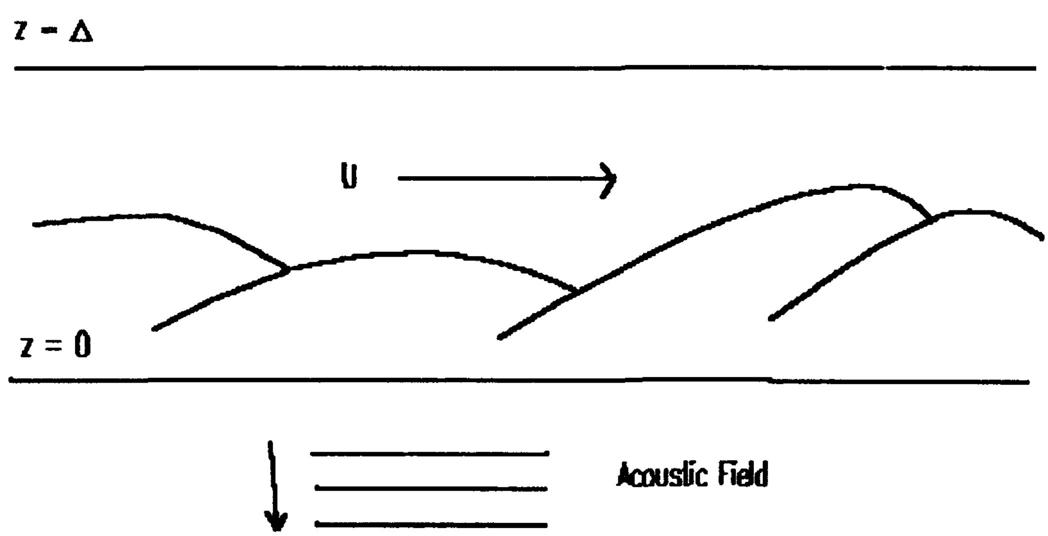
APPENDIX

The theory of Ffowcs Williams and Guo is based upon partitioning the active region of sound propagation and production at the ocean surface into domains $z > \Delta$, $\Delta > z > 0$, and $z < 0$, as indicated in the figure below. In the first and last domains there are no sound sources, so the free wave equation appropriate to each medium obtains. Let $\mathbf{p}(z, \mathbf{k};\omega)$ be the Fourier transform of the pressure field with respect to the transverse directions and with respect to time. Thus with c_i the soundspeeds

$$\partial^2{}_z \mathbf{p} + \gamma_i{}^2 \mathbf{p} = 0;$$

$$\gamma_i{}^2 = (\omega/c_i)^2 - \mathbf{k}^2.$$

The branches of the γ_i are chosen so that the pressure vanishes at spatial infinity.



Since we are dealing with fluids in motion the following is also valid:

$$\rho_w \partial_t V_z = - \partial_z p,$$

where V is the fluid velocity. Thus in Fourier space, and in particular at z=0

$$\gamma_w \mathbf{p}(z=0) = - \omega \rho_w \mathbf{V}_z(z=0).$$

Similarly

$$\gamma_a \mathbf{p}(z=\Delta) = + \omega \rho_a \mathbf{V}_z(z=\Delta).$$

Using these relations together with $\mathbf{p}\,(z<0) = \mathbf{p}\,(0)\, e^{-i\gamma_w z}$, and $\mathbf{p}\,(z>\Delta) = \mathbf{p}\,(\Delta) e^{i\gamma_a (z-\Delta)}$, one obtains eventually

$$\mathbf{p}(z) = \frac{\rho_a \rho_w e^{-i\gamma_w z}}{\rho_a \gamma_w + \rho_w \gamma_a} \left\{\frac{\gamma_a}{\rho_a} \mathbf{p}\,(0) - \omega \mathbf{V}_z\,(0)\right\}$$

By virtue of the equation before this one, an expression more symmetrical in z=0 and z = Δ may be written, but this is unnecessary for what follows.

The point of the above manipulations has been to express the acoustic field within the water in terms of pressure and velocity values at the surfaces surrounding the acoustic sources. A full solution to the problem would require examining these terms in detail. But the general character of the acoustic field is already embodied in the result above.

To see this in brief, we follow Guo and evaluate the inverse Fourier transform using the method of stationary phase, assuming only that the expressions involving $\mathbf{p}(o)$ or $\mathbf{p}(\Delta)$ are not so irregular as to spoil this estimate. The core of the computation is in the evaluation of the integral

$$I(\mathbf{x},\mathbf{y};z) = \int d\mathbf{k} \exp\{i\, \mathbf{k} \bullet (\mathbf{x} - \mathbf{y}) - z\sqrt{\nu^2 - \mathbf{k}^2}\} F(\mathbf{k},\omega)$$

Here F is a regular function of its arguments, and the structure of the phase follows from the definition of γ introduced earlier, $\nu = \omega/c_W$. The phase is stationary at

$$\mathbf{k}^* = -\nu\, (\mathbf{x} - \mathbf{y}) \,/\, R,$$

where $R = \sqrt{(\mathbf{x} - \mathbf{y})^2 + z^2}$. Introducing an angle $\tan\theta = z \,/\, |\mathbf{x} - \mathbf{y}|$, it is possible to demonstrate that the integral takes the general form

$$I \cong f(\theta,\omega) \exp(-i\,\omega R/c_W)/R^2$$

where f is polynomial in ω. Thus the inverse Fourier transform back into the time domain will introduce Dirac delta functions and derivatives of these Delta functions with argument $(t + R/c_W)$, i.e., the "sources" as described by $\mathbf{p}(0)$, etc., are retarded by the time needed to reach $\mathbf{x}$ from $\mathbf{y}$ with soundspeed c_W. Schematically,

$$p(\mathbf{x},z;t) = \int d\mathbf{y}\, \frac{g[\theta(\mathbf{x}-\mathbf{y})]}{R^2(\mathbf{x}-\mathbf{y})}\, \mathcal{P}(\mathbf{y};\, t + R(\mathbf{x} - \mathbf{y})/c_W)$$

The form of $\mathcal{P}$ indicates that we made no specific assumptions regarding the transverse ($\mathbf{y}$) dependence of the sources. There is no general reason for the spatial gradients to be simply proportional to the time derivative. It is only in the limit of homogeneous and isotropic sources that we can expect the $\mathbf{y}$ integration to correspond to a surface averaging procedure, leaving the result utilized in the main part of this report. A similar argument can be made, however, for directional noise.