A Representation of the Ear for Parameter Identification by External Kemp-Echo Measurements*

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ABSTRACT A digital filter representation of the ear is given which yields Kemp-echos as its unit pulse response, and is, hence, suitable for ear parameter identification via external Kemp-echo measurements. This provides analytic simplifications to the more physical previous models which allow for noninvasive comparison of healthy versus damaged ears.

INTRODUCTION

Because almost every human will experience some hearing loss during aging, it would be useful to have means of determining the precise nature and extent of damage through external measurements. Toward this there appears to be much progress in the convergence of two technologies, one being the experimentally determined Kemp-echos and the other the more theoretical digital systems models of the ear.

The remarkable Kemp-echos give an external means of measurement that does differentiate between healthy and damaged ears [1]. However, it remains to be able to use these echos in a meanginful way in terms of damage assessment. In regard to this latter it does appear that digital signal processing has a lot to contribute in that the reflected signal recovered in the Kemp-echo experiments can be related to parameters of the ear via the digital filter models.

BACKGROUND

Figure 1 shows in outline form the cross-sectional structure of the ear as it is of interest here and with the cochlea unfolded [2, pp. 23-25]. Sound is channeled by the outer ear, a, to vibrate the ear drum, b, which via the bony structure of the middle ear, e, acting as a mechanical lever arm (transformer), excites the choclea at the oval window, c. The choclea is a rather complex transmission system but for our purposes it can be considered as a partition, h (the basilar membrane), separating two channels, called the scala vestibuli, f, and the scala tympani, g. The excitation at the oval window sets up a pressure difference across the basilar membrane which travels up and down the membrane. The fluid in the two scala is coupled via the helicotrema, i, at the far end ($x=\ell$); it also retransmits signal to the eardrum via the oval window and the round window, d (which seals the scala tympani at the excitation end of the basilar membrane). These retransmitted signals appear in the outer ear as reflected signals picked up (as Kemp-echos) after a short sound pulse excites the outer ear. Further, the vibration of the basilar membrane appears to excite neural transmitters which send auditory signals to the brain (and, very interestingly, vice versa, neural feedback seems to excite motion of the basilar membrane making it look "active").

The Kemp-echos appear after initial reflections (due essentially to the eardrum and middle ear), following about 5 ms the excitation and continuing for some time [1, p. 1387]. They seem to be primarily due to mechanical vibrations in the cochlea and, hence, models depicting the pressure transmission and reflections in the cochlea are pertinent. A number of circuit-like models of the cochlea do exist [3][4] but these are not within a scattering basis, as is most pertinent to the auditory system, nor within the digital systems domain. However, recently such a scattering parameter digital filter model based upon mechanical properties of the cochlea has been developed as a cascade of 2-port sections corresponding to lengths Δx of the choclea [5][6][7]; each section is as in Fig. 2 with the full cascade as in Fig. 3. Because the scala and the basilar membrane are both tapered, the characteristics of the kth 2-port section depend upon the position, $x=k\Delta$, along the choclea to which it corresponds; thus, the parameters are space dependent, a property which complicates the situation.

To the cochlea models one should insert models for the outer and middle ear and for the neural interconnections.

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Although much more complicated models do exist for the outer and middle ears [2, pp. 41-45], for Kemp-echo purposes we consider that the outer ear can be considered as a simple delay and the middle ear (that is eardrum and bony structure) as simply scaling (via the transformer action). Here we ignore the effects of the neural connections to the basilar membrane though we do point out below how this can be inserted into the model. We also ignore various nonlinearities in this first order treatment while again pointing out below how some of these can be inserted in the model.

In short, and in concluding this section, we observe that for the purposes of Kemp-echos the ear can be considered as a cascade of sections of the form of Fig. 2 with the incident input wave being a scaled and delayed Kemp-echo excitation pulse and the associated reflected wave being similarly scaled and delayed Kemp-echo echos. And since these scalings and delays are uniformally the same for the excitation and response we ignore them.

SECTION DESCRIPTION

Our goal is to obtain the parameters of a cascade of M sections of the form of Fig. 2 from the knowledge of the Kemp-echos, these latter being considered as the reflected signal G_0 of the system subject to a unit pulse input and a termination of the Mth section that represents the helicotrema. For this $\Delta x = \ell/M$, where ℓ is the length of the cochlea, roughly 3.5 cm [3 ,p. 370] and M is the number of sections used in the model (about which we say more later). Considering the nature of the helicotrema, we will assume that the termination of the Mth section has a reflection coefficient that is +1. Our problem is then one of system identification of a cascade of scattering parameter described digital filters of similar structure but parameters which vary with the section number and z-transform variable (of delay 1/z). That is we wish to determine $\gamma_k(z)$, $\rho_k(z)$, and $\tau_k(z)$. We note that both γ_k and ρ_k are linearly proportional to Δx and that [7]

$$\tau_k = 1 - \rho_k \tag{1a}$$

$$\frac{\rho_k}{\tau_k} = \frac{(n_{k0}+n_{k1}z^{-1}+n_{k2}z^{-2})\Delta x}{d_{k0}+d_{k1}z^{-1}+d_{k2}z^{-2}} = \frac{N_k(z)}{D_k(z)} \tag{1b}$$

which give

$$\rho_k = N_k/(N_k+D_k), \quad \tau_k = D_k/(N_k+D_k) \tag{1c,d}$$

Here n_{k0}, n_{k1}, n_{k2}, d_{k0}, d_{k1}, and d_{k3} are to be determined as are the parameters in $\gamma_k(z)$. Previously we have given a set of these which are directly interpretable in terms of the physical constants of the cochlea evaluated for some $x=x_k$ within the kth section by formulas presented in [7, eqs. (33)-(36)] where a rather complicated dependence of the space damping constant γ_k on z is derived on physical grounds [7, eq.(20)]. With these results we see that the kth section can be described by the transfer scattering matrix Θ_k via [8]

$$\begin{bmatrix} F \\ G \end{bmatrix}_{k-1} = \Theta_k \begin{bmatrix} F \\ G \end{bmatrix}_k \tag{2}$$

where

$$\Theta_k = \left. \begin{bmatrix} e^{+\gamma} & 0 \\ 0 & e^{-\gamma} \end{bmatrix} \begin{bmatrix} 1 & -\rho \\ -\rho & 1 \end{bmatrix} \Big/ (1+\rho) \right|_k \tag{3}$$

The inverse of Θ is readily found and allows us to determine the variables at the kth step in terms of those at the (k-1)st.

$$\Theta_k^{-1} = \left. \frac{1}{1-\rho} \begin{bmatrix} 1 & \rho \\ \rho & 1 \end{bmatrix} \begin{bmatrix} e^{-\gamma} & 0 \\ 0 & e^{+\gamma} \end{bmatrix} \right|_k \tag{4}$$

Corresponding to the three factors in this form for Θ_k^{-1} we introduce new variables for use in the identification. Thus, let

$$F_k' = [\exp(-\gamma_k)] \cdot F_{k-1} \tag{5a}$$
$$G_k' = [\exp(+\gamma_k)] \cdot G_{k-1} \tag{5b}$$
$$F_k'' = F_k' + \rho_k \cdot G_k' \tag{5c}$$
$$G_k'' = \rho_k \cdot F_k' + G_k' \tag{5d}$$

In which case (4) yields

$$F_k = F_k''/(1-\rho_k) \tag{5e}$$
$$G_k = G_k''/(1-\rho_k) \tag{5f}$$

Equations (5) are convenient ones for the estimation of parameters.

PARAMETER ESTIMATION TECHNIQUE

Given an ear, the outer ear excitation, and the resulting Kemp-echo, we can obtain the cochlea excitation and response by time shifting (for outer ear delay) and combining them (for middle ear transformer action, see (12) below). Thus, we can assume that $F_0(z)$ and $G_0(z)$, the z-transformed cochlea input port incident and reflected signals are known. These serve to start an iteration since, if we know the variables F & G at the (k-1)st iteration (port), eqs. (5) show how we can obtain them at the kth (port) if we know the two parameters γ_k and ρ_k. So the real problem is to determine γ_k & ρ_k, which are both functions of k with ρ of degree two in z. In an iteration on k the input energy into the kth section is known (since F_{k-1} & G_{k-1} are). Therefore, taking our clue from speech estimation theory [9], to determine γ & ρ we minimize the output energy with respect to these parameters. Switching to the (discrete) time domain, for each k we then wish to minimize the two energy functions

$$E_k' = \sum_{n=0}^{P} \{f_k'(n) - g_k'(n)\}^2 \tag{6a}$$

$$= \sum_{n=0}^{P} \{[\exp(-\gamma_k)]*f_{k-1}(n) - [\exp(+\gamma_k)]*g_{k-1}(n)\}^2 \tag{6b}$$

$$E_k'' = \sum_{n=0}^{P} \{f_k''(n) - g_k''(n)\}^2 \qquad (6c)$$

$$= \sum_{n=0}^{P} \{\tau_k * [f_k' - g_k'](n)\}^2 \qquad (6d)$$

where P is the number of samples of the signal being used and * is discrete time convolution, with the resulting time being denoted by n. The reason for our introduction of the primed variables is now clear in that a separation of parameters has taken place so that (6b) is to be minimized over the parameters of γ_k and (6d) over the parameters of τ_k.

To proceed one takes the partial derivatives of the above two energy functions with respect to the parameters of the section under consideration. In order to do this one needs to know the form of γ to be used. Since the ear is a physical device one can proceed in terms of physically meaningful parameters, as in [7], or in terms of mathematically more tractable parameters. Since the physically meaningful parameters lead to each section being of degree 34, we present here a mathematically tractable method which leads to much smaller degree sections. We note that γ_k is a space delay which has a frequency dependence which we put in a function h(z), writing, with α_k a constant,

$$\gamma_k = \alpha_k + \ln[h(z)] \qquad (7)$$

The simplest choice is h(z)=1 but more physically meaningful is

$$h(z) = z \qquad (8)$$

which puts in a time delay (resulting from signal travel in space). The remaining frequency dependence of a section is then placed into N_k & D_k. In this paper we will use (8) which allows us to proceed toward a minimization to obtain degree four sections. We have

$$E_k' = \sum_{n=0}^{P} \{[\exp(-\alpha_k)]f_{k-1}(n-1) - [\exp(+\alpha_k)]g_{k-1}(n+1)\}^2 \qquad (9)$$

For which our choice of γ_k from (8) using $\partial E_k'/\partial \alpha_k = 0$ yields

$$\alpha_k = \tfrac{1}{2}\ln\{-[\sum_{n=0}^{P} f_{k-1}(n-1)]/[\sum_{n=0}^{P} g_{k-1}(n+1)]\} \qquad (10)$$

Here the right hand side contains only quantities known from the previous step, while if it turns out to be complex then it is replaced by 0 since no attenuation is needed for that section (and the phase portions, which a complex α_k represents, will be achieved via N_k & D_k). We note that one could use more complicated rational h(z) terms at (8) to get rational operators for (9) in which case the same method applies. Likewise even irrational h(z) can be used and approximations, as in [7], used to bring the overall signal-flow graph transmittances to be rational.

For (6d) we next desire a minimization over the parameters n_{ki} & d_{ki} in τ_k. Thus, using (1d) and considering $N_k(z)$ and $D_k(z)$ as time domain operators, we determine $\partial E_k''/\partial n_{ki}=0$, and $\partial E_k''/\partial d_{kj}=0$, for i,j=0,1,2. These give

$$0 = \sum_{n=0}^{P} \{\tau_k * [f_k' - g_k'](n)\} \cdot \{(D_k/(N_k+D_k)^2) * [f_k' - g_k'](n-i)\} \qquad (11a)$$

$$0 = \sum_{n=0}^{P} \{\tau_k)) * [f_k' - g_k'](n)\} \cdot \{(N_k/(N_k+D_k)^2) * [f_k' - g_k'](n-j)\} \qquad (11b)$$

Assuming, as we may, $d_{k0}=1$, we have five (i=0,1,2; j=1,2) nonlinear equations to be solved for the five n_{ki}, d_{kj}; the nonlinearity coming in two forms, one from the operator denominators and the other from the products (resulting from the square in the energy functions). One can proceed in various ways to solve these. Perhaps the most preferred and convenient way being via iteration on the parameters.

Noting that $\rho_k = N_k/(N_k+D_k)]$ we obtain F_k and G_k by (5c-f). The iteration on k then can continue until one obtains F_k equal to G_k, representing the Helicotrema, to within a prescribed error. Since the model appears to be a very reasonable one for the Cochlea, the convergence of $F_k - G_k$ to zero should result, but this remains to be proven.

Because the external behaviors should be identical for the physically meaningful digital filter and the mathematically simplified digital filter, one should be able to obtain the parameters of one from those of the other. This means that eventually, using the mathematically simplified version, one should be able to characterize the physical parameters within the ear via the mathematically simplified digital filter, thus, leading to a simplified means of characterizing the ear via Kemp-echos.

DISCUSSION

Here we have presented a means of obtaining a digital filter that has the Kemp-echos as its unit pulse response, and, therefore, essentially identifies the cochlea structure from external Kemp-echos. From the identification equations it is seen that one is led to nonlinear estimation equations even when the structure assumed is linear, as it was here. Thus, the incorporation of the cochlea system nonlinearities, which seem important to the Kemp-echos, should impose no great changes in the techniques presented, especially since they can be conveniently incorporated in cascade with the sections of Fig. 2. Further research is, therefore, needed on the nonlinear theory as well as on properties of the identification parameters along with more efficient means of obtaining them.

The effect of the middle ear needs also to be taken more into account. Because of its transformer action, the middle ear can be described by the transfer scattering matrix of a transformer of turns ratio t, this being found as [10, p.51]

$$\theta_{11}=\theta_{22}=(1+t^2)/2t; \quad \theta_{12}=\theta_{21}=(-1+t^2)/2t \quad (12)$$

Thus, before the identifications made in this paper are carried out the transformations of (12) need to be made on F_0 and G_0 along with a determination of t.

References

[1]. D. T. Kemp, "Stimulated Acoustic Emissions from within the Human Auditory System," The Journal of the Acoustical Society of America, Vol. 64, No. 5, November 1978, pp. 1386-1391.

[2]. J. Zwislocki, "Analysis of Some Auditory Characteistics," Chapter 15 of Handbook of Mathematical Psychology, Edited by R. D. Luce, R. R. Bush, and E. Galanter, John Wiley and Sons, NY, 1965, Vol. 3, pp. 1-97.

[3]. L. C. Peterson and B. P. Boyert, "A Dynamical Theory of the Cochlea," The Journal of the Acoustical Society of America, Vol. 22, No. 3, May 1950, pp. 369-381.

[4]. J. B. Allen, "Cochlear Modeling," IEEE ASSP Magazine, Vol. 2, No. 1, January 1985, pp. 3-29.

[5]. P. Gómez, "Modelo Incremental Recursivo para el Analisis de la Mecanica del Oido Medio e Interno," Doctoral Thesis, Universidad Politecnica de Madrid, December 1982.

[6]. V. Rodellar, "Metodos no Invasivos para la Medida de las Caracteristicas de Transmision del Aparato Auditivo," Doctoral Thesis, Universidad Politecnica de Madrid, November 1983.

[7]. P. Gómez, V. Rodellar, and R. W. Newcomb, "A Digital Lattice for Cochlear Parameter Identification," Proceedings of MECOMBE'86, Seville, Spain, September 1986 (to appear).

[8]. P. Dewilde, A. C. Vieira, and T. Kailath, "On a Generalized Szego - Levinson Realization Algorithm for Optimal Linear Predictors Based on a Network Synthesis Approach," IEFE Transactions on Circuits and Systems, Vol. CAS-25, No. 9, September 1978, pp. 663-675.

[9]. J. D. Markel and A. H. Gray, Jr., "Linear Prediction of Speech," Springer-Verlag, NY 1976.

[10]. R. W. Newcomb, "Linear Multiport Synthesis," McGraw-Hill, NY, 1966.

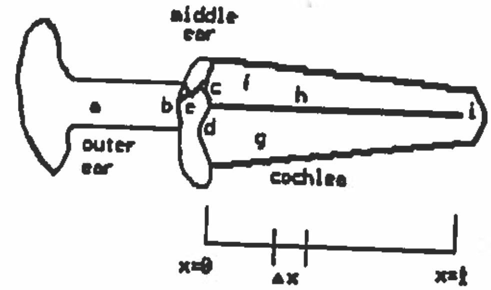

Figure 1
Ear Cross Section-Unfolded Cochlea

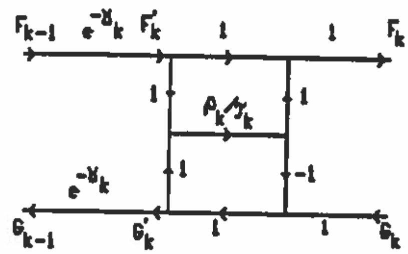

Figure 2
kth Cochlea Digital Section

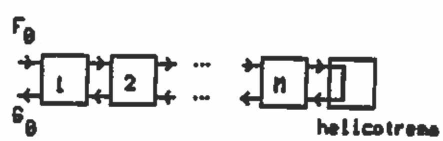

Figure 3
Digital System Structure

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