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Ear-type analog and digital systems

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Abstract – This paper reviews the research work done at the Microsystems Laboratory of the University of Maryland in the area of analog and digital ear-type circuits and systems, which includes modeling, analysis, synthesis, simulation, and design.

1 Introduction

Following the philosophy which has proven successful in the field of neural networks, we wish to mimic characteristics of the biological ear to create systems which have ear type of behavior. In our research we focus on ear-type circuits and systems of the type shown in Figure 1, both analog and digital; this includes design, analysis, and simulation. In particular we base our systems on fluidics and the mechanical structure of the cochlea and processing of signals through its interactions by hair cells with neurons.

Ear-type systems are by definition ones that are based upon the structure of the ear abstracted in block form by Figure 1, with ear-like characteristics for the sub-blocks. In particular electrical (nonbiological) structures which mimic the mechanical, hair cell, and neural behavior of the ear are considered. In this structure the upper layer consists of lattice sections described in terms of incident and reflected pressure variables, P^i and P^r, mimicking the mechanical fluidic properties of the cochlea. The middle layer consists of transducer-like circuits coupling signals in hair-cell-like fashion from the lattices to the lower layer. The latter is an artificial neural network which is a primitive mimic of the brain.

Since the cochlea has two channels and the characteristics we work with are best explained through the fluid waves in these two channels, we wish to work with multi-channel systems described on a scattering basis. Considering the cochlea as a pipeline of sections we work with a cascade of sections each of which contains nonlinearities due to the limited motion available of the membrane separating the channels as well as the limited and nonlinear motion of the hair cells. When motion of the separating membrane is sensed the hair cells send signals to the brain cascade and conversely signals from the brain through the hair cells can cause motion of the separating membrane.

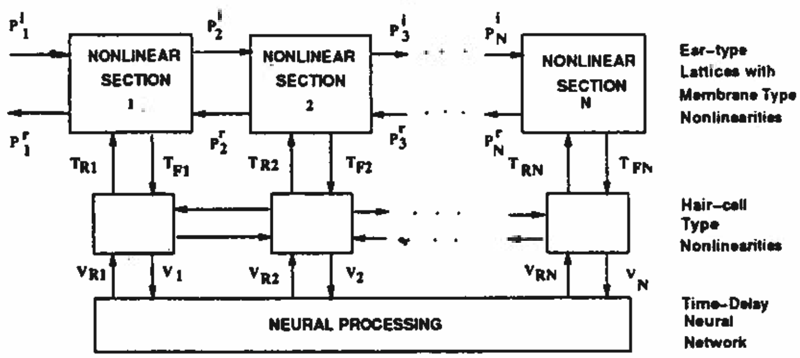

Figure 1: Ear-type signal processing system.

In our research we mimic most of these behaviors in system theoretic form, via nonlinear digital and analog filter-like structures through scattering and semistate theory. Toward this we build upon the background obtained by our researches over the last ten years on studies of stimulated emissions, known as Kemp Echoes, from the ear. This research includes the following:

- Analog and digital scattering models of the cochlea.
- System identification techniques for cochlea transfer function estimation.
- Degree-two two-port and degree-one four-port lossless and lossy lattice synthesis methods.
- Estimation techniques for parameter determination for design from experimental data.
- Hair cell modeling and incorporation of hair cell type of nonlinearities in the digital scattering model of the cochlea.
- Semistate representation of hair cell circuits and their use for VLSI design.

Possible applications could be made to sound localization, improved hearing aids, noninvasive detection of damage to animal and human ears, multi-aural sound systems, chaos cancelling, speech recognition, and sensitive digital filters.

In the following we concentrate on reviewing research carried out at the Microsustems Laboratory of the University of Maryland, often in conjunction with researchers, particularly Dr. Pedro Gomez and Dr. Victoria Rodellar, of the Laboratorio de Communicacion Oral of Universidad Politecnica de Madrid. Since our research stems from our previous research on Kemp echoes, we first briefly describe the nature of Kemp echoes and models for their generation.

2 Kemp Echoes and Cochlea Properties

In 1978 Kemp [1] reported the existence of stimulated acoustic emissions within the human auditory system. These emissions, also known as Kemp echoes, have been shown to possess nonlinear characteristics and are modified by by damage to the auditory system [1, 7]. Although of small magnitude, Kemp echoes can be isolated with good filtering techniques. Because there are significant differences in the Kemp echoes for normal versus certain types of damaged ears, it is felt that the Kemp echoes can provide a noninvasive way to quickly and easily characterize some types of damages to the inner ear, which should be especially useful for diagnosis of babies, the elderly, as well as some animals. Furthermore, these emissions can be a reliable technique for demonstrating objectively the presence of normal activity in the cochlea, detecting changes in its functioning, as well as detecting hearing loss of noncochlear origin [2].

2.1 Kemp Echoes

The peripheral (internal) auditory system has the ability to generate audio-frequency sounds in the external auditory ear canal which can be detected by external measurements. Kemp was able to record echoes from human ears by sealing a miniature sound source and a microphone into the ear canal. Since then, other researchers [3, 4, 5, 6] have confirmed Kemp's discovery and investigated the influence of different stimuli on the characteristics of these acoustic emissions.

The sound generating property of the auditory system manifests itself in various forms: spontaneous emissions are detectable from some healthy human ears and consist of narrow-band signals, of one or more fixed frequency tones, often audible to the subject, that can be measured in the absence of acoustic stimulus. In contrast, Kemp echoes, also referred to as stimulated acoustic emissions, or transiently evoked oto-acoustic emissions (TEOAEs), in the literature are exhibited by the majority of normal human ears and half of abnormal ears [4] in response to an incident acoustic pulse stimulus. They occur several milliseconds after the stimulus is applied and persist for some tens of milliseconds. The response latencies can be divided roughly into two intervals: 0-5 ms post-stimulus time, in which one observes the impulse response of outer and middle ears, and a later part at greater than 5 ms, in which emissions from the inner ear appear [4].

In Figure 2 we show one sample of the Kemp echoes recorded by H. P. Wit and P. Van Dijk from the Institute of Audiology, the Netherlands. This signal was obtained by averaging 3000 echoes from the same ear in order to get rid of microphone noise. The horizontal axis is

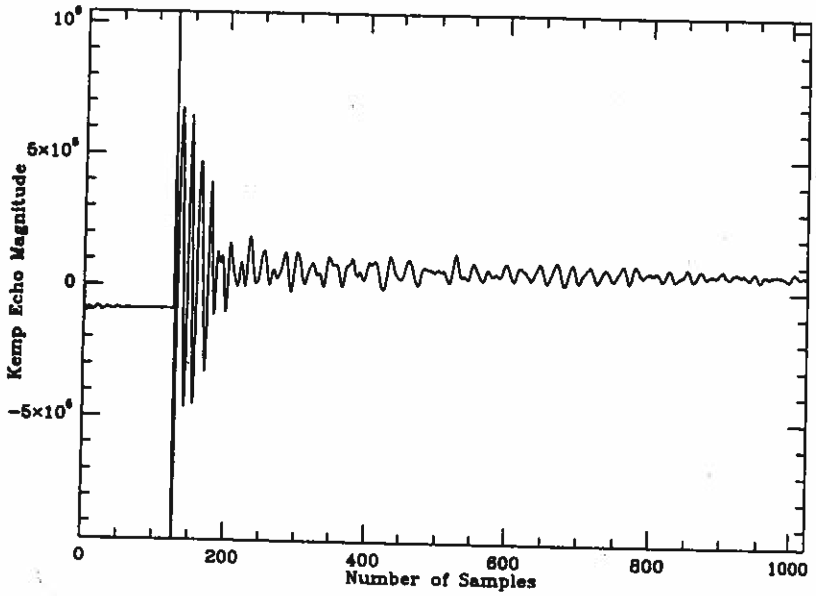

Figure 2: Kemp echo signal, sample 1. Data provided by Dr H. P Wit and Dr P. Van Dijk, Institute of Audiology, the Netherlands.

scaled to sampling units (about 100 samples per ms) and the vertical axis represents pressure. The first peak occurs at 125 samples following the initial excitation. This peak indicates reflections from the middle ear, which extend to sample number 200, while the remainder of the response characterizes the inner ear.

2.2 Background Description of the Ear

Before we get to the digital scattering cochlea model, the digital lattice filter synthesis techniques, and other Kemp echo related topics, we give a brief description of the cochlea and some of its properties.

The human ear is divided into three parts: the external ear, the middle ear, and the inner ear including the hair cells and their connections to the brain (Figure 3 after [8]).

External Ear: It consists of the auricle and the auditory canal. The auricle, which is the visible part of the ear, is the main receiver of sound stimuli. Its principal function is to conduct the sound energy into the auditory canal. The auricle serves, however, some important additional functions, of interest to this research, such as sound localization [9]. The auditory canal is a tapering canal of about 27 mm of length and 7 mm of diameter which leads the sound to the eardrum or tympanic membrane, a conical membrane of about 0.1

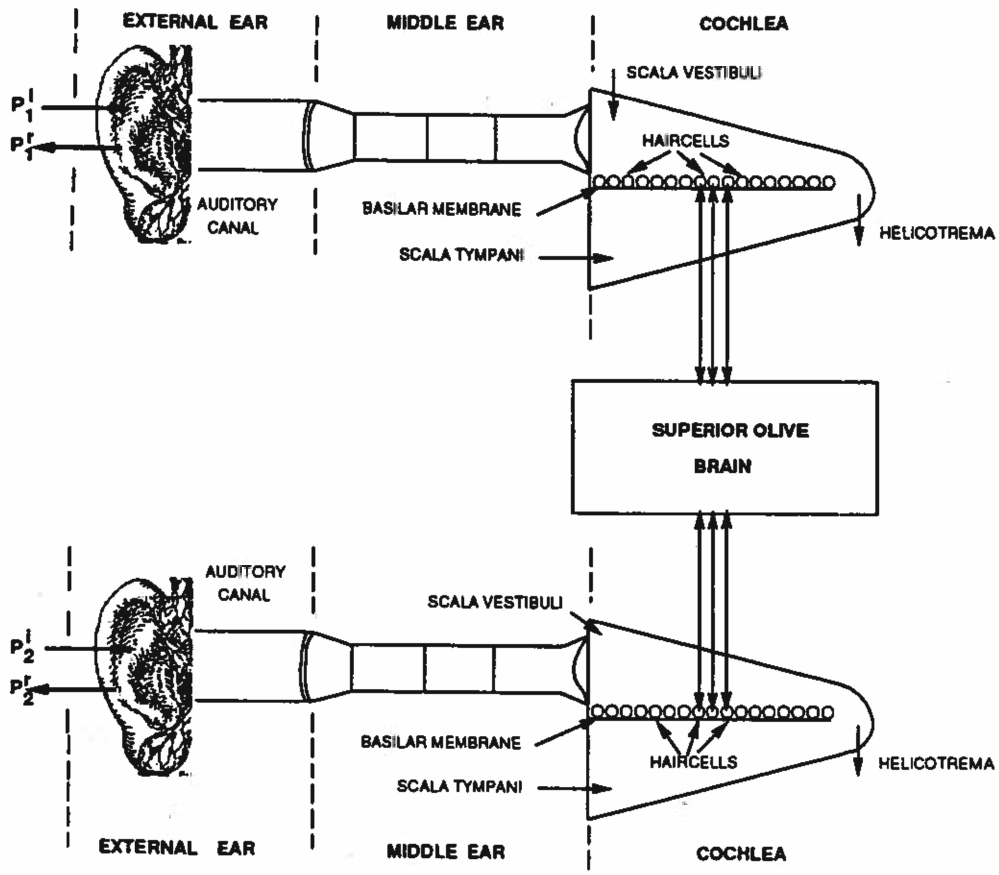

Figure 3: Schematic representation of two ears and interactions of their haircells with the brain. P^i and P^r are the incident and reflected pressure waves respectively.

mm of thickness [10].

Middle Ear: The sound pressure at the eardrum is the input of the middle ear system. The middle ear is an air-filled cavity separated from the external ear by the eardrum and from the inner ear by two windows: the oval window and the round window. The essential acoustic function of the middle ear appears to be to adapt the high fluid-borne input impedance of the cochlea to the low airborne impedance of the outer ear in order to improve the transfer of acoustic energy [11].

Inner Ear: It is composed of the vestibular organs, the cochlea, and the haircells. The cochlea is a liquid-filled spirally coiled and tapered tube of 30-35 mm of length and with an average cross-sectional area of 2 mm^2 in man. In a cross-section of the tube, three chambers can be discerned: scala vestibuli, scala media or cochlear duct, and scala tympani. The cochlear duct is relatively small in cross-sectional area, compared to the scala vestibuli and the scala tympani, which are larger and roughly equal in area, and does not communicate with the other chambers. The other two chambers are indicated in Figure 3. The scala media is separated from the scala vestibuli by a very thin and flexible membrane called Reissner's membrane, and from the scala tympani by the basilar membrane. The cochlear duct begins at the base of the cochlea, i.e., near the windows, and ends just before the apex, thus leaving a small passageway, the helicotrema, through which the fluid in the other two scalae communicate. The basilar membrane supports the organ of Corti which extends from the base to the apex of the cochlea. The organ of Corti consists of haircells arranged in four rows: three rows of outer hair cells and one row of inner hair cells. The hair cells are the receptor cells that transduce the mechanical vibrations of the basilar membrane into nerve impulses transmitted to the brain by the auditory nerve fibers and vice versa [12].

Superior Olive - Brain: From the cochlea about 28,000 nerve fibers in each of two auditory nerves carry auditory impulses which take various pathways before reaching the auditory cortex where neural processing is undertaken. In particular, signals from both ears converge on the superior olive where signals are sent on primary fibers to the opposite side of the brain as well as locally returned to hair cells of the opposite ear [13].

3 A Cascade Digital Scattering Model of the Cochlea

With the discovery of Kemp echoes and their experimentally well established properties, the development of a new category of models for cochlea assessment (cochleography) and hearing loss correction, through the design of new and better hearing aids, become possibilities. The new models that we have been working on are hoped to lead to the achievement of these objectives.

Cochlea modeling, primarily in the frequency domain, has been the subject of many studies for decades and several models exist in the literature, each one describing one or more particular functional aspects of the cochlea [14, 11, 15, 16, 17]. However, most of these studies were focused on the mechanical frequency response of the basilar membrane to an acoustical sinusoidal stimulation, in an attempt to understand and reproduce the observations and the frequency response measurements of Bekesy [10], Rhode and Robles [18], and Wilson and Johnstone [5]. These studies led to the development of one-dimensional transmission line models first, then to more complete two and three-dimensional models. However, all of these primarily linear models are not directly appropriate for Kemp echo phenomenon since the latter is based on the non-linear behavior of incident and reflected pressure waves of fluid flow in the cochlea [1, 6].

The work that we have done recently differs in three essential ways from past research. First, the model itself is different, in that it is of a scattering nature, i.e., based on incident and reflected pressure waves. Second, the model is converted to a cascade of digital scattering lattice filters with the linear part of each lattice described by a transfer scattering matrix containing the characteristic parameters of that section of the cochlea. The third difference lies in the purpose of this work, which is the development of digital signal processing systems, based on these models, that can possibly be used to design more sensitive signal processing systems which also should allow us to simulate Kemp echoes in their impulse response and from which a characterization of ear-type behavior can be made.

A cascade digital scattering linear model of the cóchlea, suitable for Kemp echo cochlea characterization, was developed among our researchers [19, 20, 21, 22]. The analog model is based on a unidimensional fluid flow model into which nonuniform, nonlinear, and loss properties can be incorporated, and whose equations are derived from three considerations: conservation of mass, Newton's law for force on a fluid, and basilar membrane motion. The linear digital model is of a scattering nature and its lattice structure is obtained by rephrasing the analog model equations, given in (1) and (2), in terms of incident and reflected scattering waves, linearizing and digitizing the resulting equations in both space and time.

$$\nabla p(s,x) = -P(s,x)u_V(s,x) \tag{1}$$

$$\nabla u_V(s,x) = -\left[1 + \frac{8}{D^4(x)}\left(\frac{p(s,x)}{sQ(s,x)}\right)^2\right]\frac{p(s,x)}{Q(s,x)} \tag{2}$$

where s is the derivative operator, x the distance down the cochlea, $p(s,x)$ the pressure difference between the fluids in the two channels of the system, $u_V(s,x)$ the fluid velocity, and $D(x)$, $P(s,x)$, and $Q(s,x)$ are functions of the characteristic parameters of the system. The resulting structure is a cascade of digital lattice filters where each lattice corresponds to one section of the cochlea, as shown in Figure 4, where p_k^i and p_k^r are the incident and reflected pressure waves, respectively, of pressure difference between the top and bottom of the basilar membrane, ρ_k a reflection coefficient, and γ_k a propagation function yielding a delay through the k^{th} section. These functions are reasonably approximated in the z-transform domain by

$$\rho_k(z) = \frac{A_{k2}z^2 + A_{k1}z + A_{k0}}{B_{k2}z^2 + B_{k1}z + B_{k0}} \tag{3}$$

$$\gamma_k(z) = \sqrt{\frac{C_{k2}z^2 + C_{k1}z + C_{k0}}{D_{k2}z^2 + D_{k1}z + D_{k0}}} \tag{4}$$

with the A_k, B_k, C_k, and D_k coefficients being functions of the geometrical, fluid, and me-

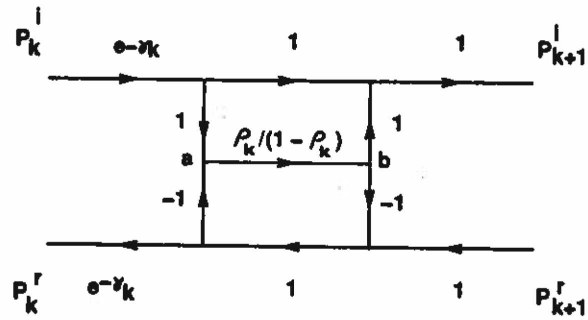

Figure 4: Signal-flow graph of a cochlea section.

chanical properties of the cochlea. The lattice filters are described by transfer scattering matrices whose entries are functions of these properties, thus allowing a systematic extraction of these characteristics [23]. For the k_{th} section, the expression of the transfer scattering matrix is

$$\theta_k(z) = \frac{1}{1+\rho_k(z)} \begin{bmatrix} e^{\gamma_k(z)}\rho_k(z) & e^{\gamma_k(z)} \\ e^{-\gamma_k(z)} & e^{-\gamma_k(z)}\rho_k(z) \end{bmatrix} \quad (5)$$

4 System Identification

To be able to use the digital model for cochlea characterization, its transfer function, assumed stable, minimum phase, and of unknown order, was estimated [23]. Kemp echo signals recorded from human ears and provided to us by Dr. H. P. Wit and Dr. P. Van Dijk from the Institute of Audiology, the Netherlands, and considered as the impulse response of the cochlea, were employed as the output signal in a new ARMA system identification technique developed by Pillai, Shim, and Youla [24, 25]. This technique is deterministic and utilizes the theory of positive real and bounded real functions [26], Richard's theorem, the concept of degree reduction, and determines both the order and the predictor coefficients of a stable digital ARMA(n,m) filter with a minimum phase rational transfer function (here n and m are the degrees of the numerator and the denominator of the transfer function respectively). The results of the estimation showed that the linear behavior of the cochlea is closely approximated as an ARMA(16,32). Below we give the expression of the numerator $N(z)$ and denominator $D(z)$ of the corresponding transfer function for the Kemp echo of

Figure 2.

$$
\begin{aligned}
N(z) \;=\;& 0.575130 - 1.182959z + 0.327421z^2 + 0.430116z^3 + 0.738268z^4 - \\
& 1.354292z^5 - 1.809802z^6 + 0.970980z^7 + 0.923793z^8 + \\
& 1.418271z^9 + 0.098415z^{10} - 0.295472z^{11} + 0.312645z^{12} - \\
& 0.630343z^{13} - 1.440973z^{14} - 1.869601z^{15} + 4.462128z^{16} \qquad (6)
\end{aligned}
$$

$$
\begin{aligned}
D(z) \;=\;& 180(0.139683 - 0.286067z + 0.174111z^2 - 0.135610z^3 + \\
& 0.010562z^4 + 0.163818z^5 - 0.150244z^6 + 0.088931z^7 + \\
& 0.031265z^8 - 0.050757z^9 - 0.155118z^{10} - 0.123422z^{11} - \\
& 0.063228z^{12} + 0.097258z^{13} + 0.216830z^{14} - 1.137432z^{15} + \\
& 2.769446z^{16} - 1.603863z^{17} - 0.583684z^{18} - 1.661542z^{19} + \\
& 2.207658z^{20} + 1.978895z^{21} - 2.653648z^{22} + 0.392582z^{23} - \\
& 1.000299z^{24} + 1.166679z^{25} + 0.328678z^{26} - 1.447006z^{27} + \\
& 0.440116z^{28} + 1.684127z^{29} + 3.255450z^{30} - 9.090247z^{31} + \\
& 5.005533z^{32}) \qquad (7)
\end{aligned}
$$

5 Digital Lattice Filter Synthesis

Our research realizes the linearized cochlea model as a pipeline of cascaded lattice filters through ARMA lattice filter synthesis techniques. We chose such a representation because of the interesting features that lattice filters have. Three synthesis techniques have been developed: degree-two real lossless two-port lattice synthesis, degree-one four-port lossless lattice synthesis, and degree-two real lossy two-port lattice synthesis.

5.1 Lossless Synthesis

There has been considerable success in modeling AR and MA processes in the past decade and various realizations for ARMA digital filters have been proposed. Among these, the digital lattice realization has received an increasing amount of interest [27, 28, 29] because of its superiority in round off performance [30] and its ease of VLSI implementation [31]. The lattice ARMA filter structure has advantages especially when applied to speech and signal processing [32]. Besides the practical usefulness, the lattice form has a number of interesting features from a theoretical point of view, such as its connections to the theory of orthogonal polynominals [33], network synthesis [34], and linear prediction [35, 36].

a. Degree-Two Real Two-Port Lattices

In their paper [27], Deprettere and Dewilde dealt with the realization of a cascade of orthogonal multiport digital filters from the overall transfer scattering matrix of the system. Vaidyanathan and Mitra introduced a discrete version of Richard's function and used it to extract degree-one real sections each of which realizes a real zero of transmission [28]. Following up on that, in [29] we have extended the ideas presented in [28] to complex zeros. But because the zeros of transmission are complex, the degree-one sections extracted are also complex and the cascade of two degree-one complex sections does not necessarily yield a degree-two real section.

To remedy the shortcomings of previous work, we proposed a new technique to synthesize a real, stable, single-input, single-output, digital ARMA(n,m) filter in the form of a cascade of degree-one or degree-two real lossless two-port lattice filters of the two types shown in (Figure 5), from the input reflection coefficient, S_I, or the transfer scattering matrix, and using complex or real zeros of transmission [37]. The technique relies on a four-step complex Richard's function extraction to calculate the transfer scattering matrices that characterize the lattices. Two of the steps are used to reduce the degree of the transfer function or the reflection coefficient of the filter through the extraction of the zeros of transmission and the other two steps are used to obtain a real realization. The synthesis is minimum, that is, with the number of delays equal to the degree of the transfer function. Compared to [28, 29], the proposed technique offers the advantage of realizing real degree-two sections from complex degree-one lattices.

This technique was used to synthesize the linear digital scattering model of the cochlea as a cascade of 16 degree-two real lattice filters. We proceed from the transfer function, considered here as the input reflection coefficient $S_I(z)$, and extract degree-two real lattice filters which reduce the degree of the reflection coefficient by two at each extraction. Below

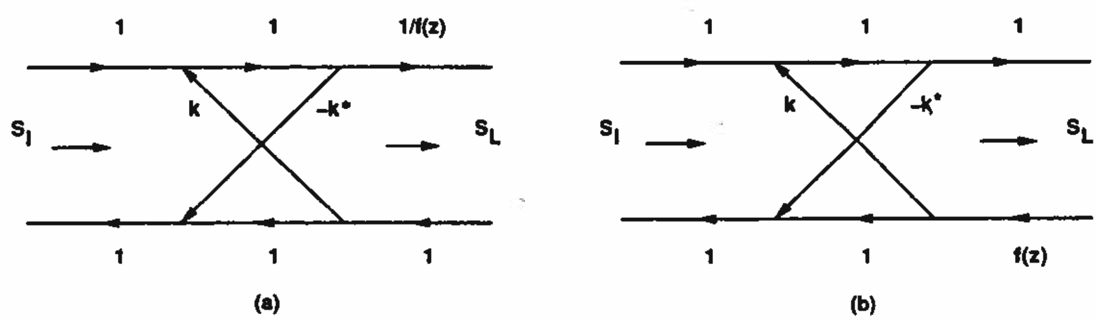

Figure 5: Two-port lossless lattice filters for transmission zeros inside (a) and outside (b) the unit circle.

we give the transfer scattering matrix of the eighth section and the parameters of the corresponding degree-two lattice for the cochlea model giving the Kemp echo of Figure 2.

$$\begin{bmatrix} \frac{0.665988+0.154261z+1.020000z^2}{0.645988+0.143261z+z^2} & \frac{0.000078-0.000284z+0.000206z^2}{0.645988+0.143261z+z^2} \\ \frac{0.000106-0.000285z+0.000278z^2}{0.645988+0.143261z+z^2} & \frac{0.645988+0.148261z+1.013000z^2}{0.645988+0.143261z+z^2} \end{bmatrix} \tag{8}$$

$$a_8 = -0.906311 + 0.40187j, \quad a_8^* = -0.906311 - 0.40187j \tag{9}$$

$$k_8^1 = 0.000086 - 0.00043j, \quad k_8^2 = 0.002578 + 0.0042j, \quad f(z) = \frac{1 - a_8^* z^{-1}}{z - a_8} \tag{10}$$

where the pairs (a_8, a_8^*), (k_8^1, k_8^2), and $f(z)$ are the zeros of transmission, cross arm coefficients, and Richard's function of section 8.

The last section has a constant output reflection coefficient, for Figure 2 this being $S_{16} = .98$, an indication that no more extractions are possible. This output reflection coefficient is the input of the terminating section which corresponds to the helicotrema and which we represent by a one-port as shown in Figure 6. From S_{16} we calculate a terminating "resistance" R_{term} as follows

The particular lattice structure developed leads to a systematic characterization of the parameters of the ear indicated by equations (3) and (4). The realization generated by this technique is lossless, but because of the type of lattice structure used, the realization can be converted to a lossy one (refer to section 5.2). The technique can be generalized to nonlinear realizations, by suitably inserting nonlinear factors between lattices, as well as to multiport

$$R_{term} = \frac{S_{16} + 1}{1 - S_{16}} = 117.474407 \tag{11}$$

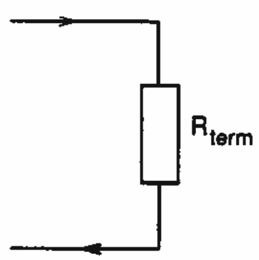

Figure 6: Terminating helicotrema-type section of the cochlea model.

systems. In particular, it can be used to synthesize the ear-type nonlinear lattices for one or more coupled ears.

b. Degree-One Four-Port Lattices

One of the two-channel lattice filter models of interest because it allows us to mimic of the two-channel structure of the cochlea, is that of Miyanaga, Nigai, and Miki. In their paper [38], they proposed estimation algorithms according to a given power spectrum, but they did not investigate the minimal degree realization or relationship to the structure of the ear. To improve on this technique, we developed new algorithms suitable for the ear structure for a minimal degree four-port realization with no delay free loops within the filter [39] which consists of two coupled channels corresponding to the channels above and below the basilar

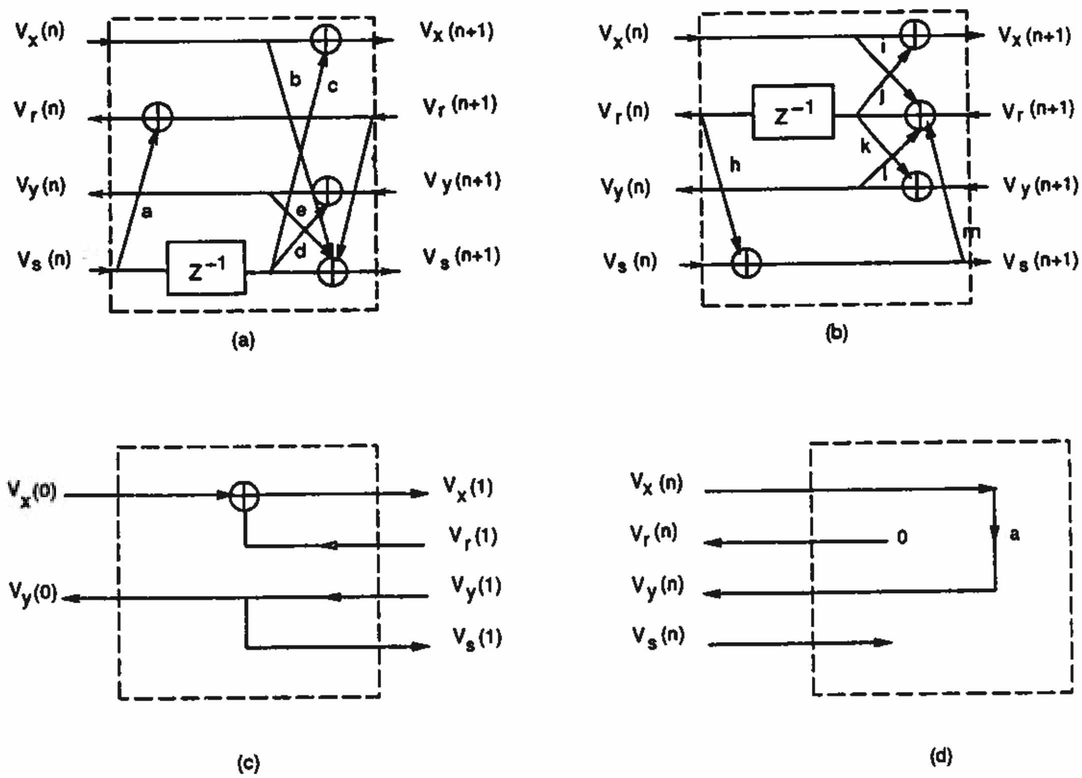

Figure 7: (a) Four-port lossless AR-type degree-one lattice filter. (b) Four-port lossless MA-type degree-one lattice filter. (d) Input section. (d) Load section.

membrane (Figure 3) with incident and reflected inputs and outputs. The algorithms lead to a systematic filter parameter identification. The structure itself is a pipeline of AR and MA filter sections (Figure 7) cascaded in a way to achieve a minimum degree realization.

This technique was used to realize the ear-type filter as a cascade of 7 degree-one four-port lattices. To indicate their nature, below we give the expressions of the transfer scattering matrices of the input, load, 4^{th} and 5^{th} lattice sections.

$$T_{in} = \begin{bmatrix} 1 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0 \\ 0 & 0 & 1 & 0 \\ 0.0015 & 0 & 0 & 1 \end{bmatrix}, \quad T_{load} = \begin{bmatrix} 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 \\ 0 & 0 & 1 & 0 \\ -0.001 & -1 & 0 & 0 \end{bmatrix} \qquad (12)$$

$$T_4 = \begin{bmatrix} 1 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0 \\ 0 & 0 & 1 & 0 \\ 0 & 0 & 0 & z^{-1} \end{bmatrix}, \quad T_5 = \begin{bmatrix} 1 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0.15z^{-1} \\ 0 & 0 & 1 & 0.15z^{-1} \\ 0 & 0 & 0 & z^{-1} \end{bmatrix} \qquad (13)$$

5.2 Lossy Synthesis

Since the cochlea actually has a small amount of distributed loss, a new method for converting a lossless cascade lattice realization of an ARMA(n,m) filter with a lossy terminating section to a lossy realization was proposed [23]. The conversion process was carried out through the factorization of the terminating lattice and the distribution of the loss term, contained in the terminating section, among the various lossless lattice sections (Figure 8). This involved finding the transfer scattering matrix of the terminating section from its input reflection coefficient, factoring it into a cascade of lossy constant sections of the same structure as shown in Figure 8a, translating these load sections from the end of the cascade to the appropriate location in the cascade, and finally, calculating the new lattices. This technique changes the nature of each section from lossless to lossy but does not alter the realness and the passivity characters of the cascaded structure. This is because the transfer scattering matrix of the load section is real and passive and its factors are all real and passive. The technique can be applied to degree-two as well as to degree-one sections. This technique is best suited for the lossless synthesis described in [37], but can be modified to suit other types of ARMA synthesis such as the one developed in [29] or that of Figure 7.

6 Characterization from Data

The characterization from data of the cochlea was accomplished via the scattering method. The idea was to embed the characteristic parameters of each cochlea section in the entries of

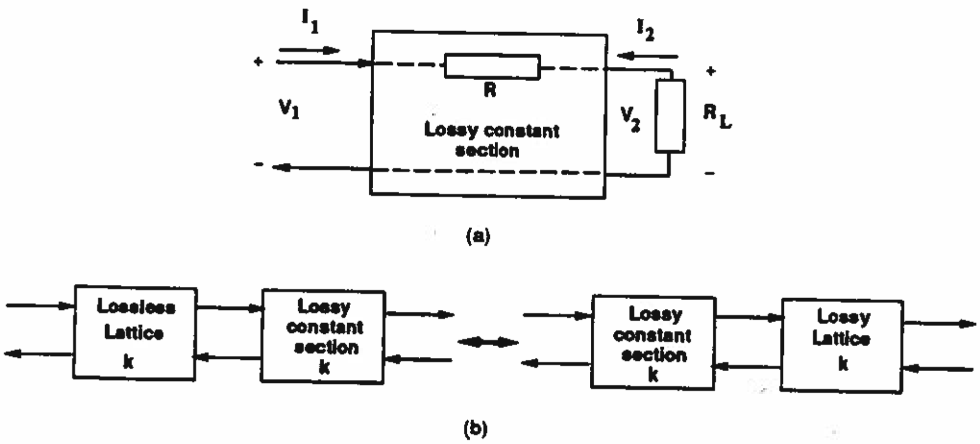

Figure 8: (a) A lossy terminating section, (b) Transformation of section k from lossless to lossy.

the corresponding transfer scattering matrix (see equations (3), (4), and (5)). The degree-two two-port synthesis method described earlier extracts cochlea lattice filters with transfer scattering matrices that satisfy this description. From the entries of the transfer scattering matrix of each section, we determine the values of the coefficients A_k and B_k for each of the 16 sections of the cochlea. From these coefficients, the width $D(x)$, damping $\sigma(x)$ and stiffness $\phi(x)$ of the basilar membrane are evaluated for each of the 16 sections. The values of $D(x)$, $\sigma(x)$, and $\phi(x)$ are found to vary from section to section, being smallest at the first section and increasing linearly for $D(x)$, and largest at the first section and decreasing exponentially for $\sigma(x)$ and $\phi(x)$. Their functional dependency on the displacement x along the basilar membrane is obtained by curve fitting the estimated values for each one, using MATHEMATICA. The resulting expressions are given explicitly by

$$D(x) = 0.002911 + 0.000248x \quad cm \tag{14}$$
$$\sigma(x) = 650e^{-1.7x} \quad dyne.sec.cm^{-3} \tag{15}$$
$$\phi(x) = 1.1\,10^9 e^{-3.4x} \quad dyne.cm^{-3} \tag{16}$$

7 Hair Cell type Circuits

The presence of nonlinearities and active behavior in the cochlea is supported by experimental data that show discrepancies between basilar membrane and neural tuning. Two major categories of sources of cochlear nonlinearity have been suggested in the literature [18, 40, 41, 42]. The first category finds its origins in the basilar membrane and the cochlear fluids, and the second one resides in the mechano-electrical transduction process that occurs in the hair cells. The discovery of Kemp echoes in 1978 [1] raised the question whether these phenomena could be explained by transmission line models. Furst and Lapid developed a nonlinear transmission line model but were unable to predict evoked emissions [43].

Experiments conducted by Crawford and Fettiplace [44] and Howard and Hudspeth [45], have shown that hair cells are not only capable of transducing mechanical vibrations of the basilar membrane and the tectorial membrane into electrical signals but also electrical signals to mechanical signals that drive their hair bundles. These experiments have also shown that this bidirectional transduction process, besides being an active one, is nonlinear.

New models, called bidirectional, of hair cells have been proposed by Weiss and Leong [46], Weiss [47], and Secker, Searle, and Wilson [48] to explain the sharp frequency selectivity exhibited by cochlear hair cells and their associated neurons in the case of lizards, red-eared turtles, and bullfrogs. They attributed this selectivity to two different physical mechanisms that occur in the hair cells: a mechanical resonance of the stereociliary-tectorial structure and an electrical resonance of the hair cell membrane. Experimental evidence shows that these mechanisms are nonlinearly coupled because the current in the hair cells depends on the angular displacement of the hair bundle and the electrical system of the hair cells feeds back a torque, that depends on the membrane potential, to drive the hair bundle [47, 48, 45]. We believe that the bidirectional process of the hair cells could be the physical basis for a variety of phenomena observed in mammals, including the generation of spontaneous acoustic emissions.

The cochlea lattice model developed in [23] and summarized above is passive and linear. But since experimental evidence shows that Kemp echo phenomena are nonlinear, it is important to incorporate activity and nonlinearities into the full ear-type system. We assumed that these nonlinearities and activity were due to the hair cells and, therefore, proposed a nonlinear bidirectional model of a hair cell, following from that of Weiss and Leong [47, 46] and Secker, Searle, and Wilson [48]. We then estimated its parameters from experimental data, obtained by Crawford and Fettiplace [44], Howard and Hudspeth [45], and Secker, Searle, and Wilson [48], analyzed its behavior, and compared the simulation results with the mentioned experimental findings of actual ears.

The proposed active and nonlinear hair-cell model (Figure 9) is based on the mechano-electrical behavior of the hair cells and embodies two parts: an electrical part and a mechanical part. The electrical part models the electrical activities of the hair cell, namely the opening and closing of Potassium and Sodium channels, and special types of channels, called transduction channels, specific to the hair cells, that regulate the fluctuations of the membrane voltage, through the control of ion transport into the cell, and via which neural signals, encoding acoustical signals, are transmitted to the brain. These channels were modeled by nonlinear conductances converted to current and voltage controlled nonlinear current

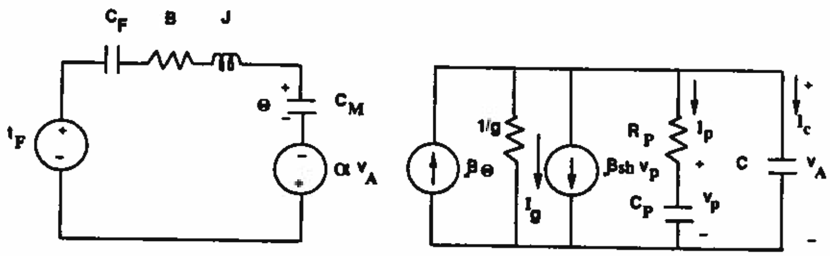

Figure 9: The electrical equivalent nonlinear model of a hair cell.

sources ($\beta\theta$ and $\beta_{sh}v_p$). The slow process of the opening and closing of potassium channels is symbolically represented by the series path R_pC_p. The hair cell membrane is represented by the right hand capacitor C.

The mechanical part of the hair cell concerns mainly the movement of the bundle and its bidirectional coupling with the body of the cell. The hair bundle is modeled by a second order mechanical system with an inertia J, a viscous drag B, and a compliance C_M, and the electro-mechanical feedback by a torque controlled by the membrane potential. The mechanical system was then converted to an electrical equivalent circuit consisting of an RLC circuit and a nonlinear voltage controlled voltage source (αv_A). In their experiment, Howard and Hudspeth [45] used a thin flexible glass fiber to drive the hair bundle. This fiber is represented by a capacitor C_F, to account for its compliance, and the torque stimulus by a voltage source t_F.

Considering Figure 1, in the ear-type lattices we have inserted nonlinearities which correspond to those of the basilar membrane and subtectorial coupling. This was done using a procedure similar to the one used to insert fluid-type nonlinearities into our mechanical operator theory models [23] as summarized in equations (1) and (2). For the subtectorial coupling, which represents the effect of the fluid movement in that area on hair cell membrane potential and vice versa, we model the angular velocity, $\frac{d\Theta}{dt}$, of the hair bundle as a saturating nonlinear function $T(.)$ of the basilar membrane velocity, $\frac{d\xi}{dt}$, from which the resulting driving torque t_F can is determined as follows

$$\frac{d\Theta}{dt} = T\left(\frac{d\xi}{dt}\right) \tag{17}$$

$$t_F = J\frac{d^2\Theta}{d^2t} \tag{18}$$

The circuit components were estimated using experimental data and the circuit model was simulated with Pspice for different levels of input stimulus. The responses obtained from the circuit simulation reproduce qualitatively and quantitatively the responses obtained from real hair cells for a certain range of input stimulus (see Figure 10).

8 Semi-State and Circuits for Hair-Cell Model

The canonical semi-state representation takes the form of the equations

$$E\frac{dx}{dt} = A(x,t) + Bu \quad (19)$$

$$y = Cx \quad (20)$$

where u, x, y are the input, semistate, and output vectors, B, C, E constant matrices (with E possibly singular), and $A(.,.)$ a possibly nonlinear function, and are very useful for VLSI design using switched current mode circuits. Consequently, we obtain such a representation

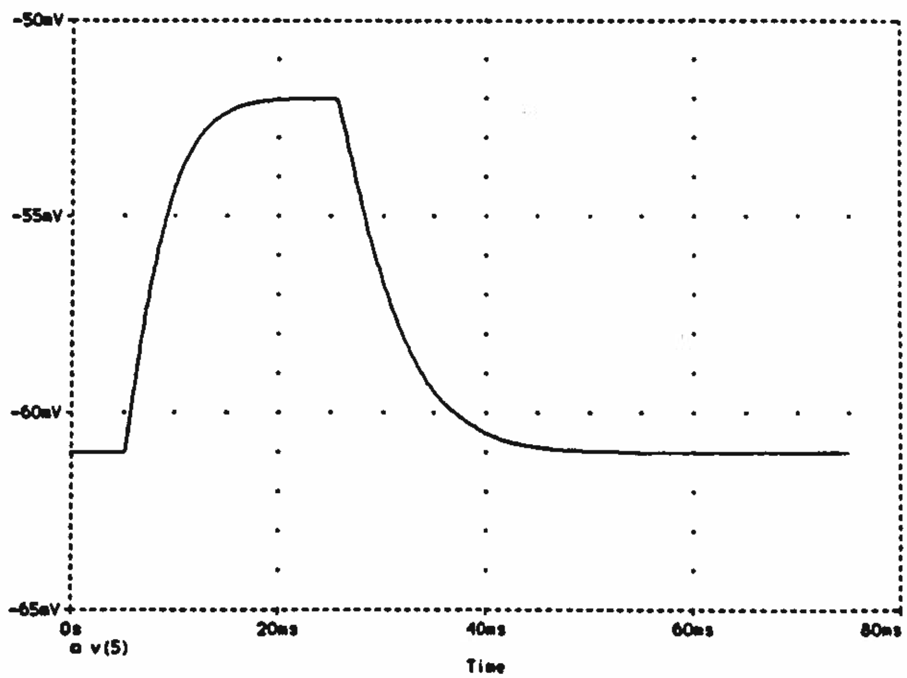

Figure 10: Membrane potential V_A.

for the nonlinear hair cell-type structures considered here and represented by Figure 9 with refinements to include ion channels as mentioned below. Setting up these equations generally is a straightforward matter of applying component and connection laws. In our case, however, we would like to put the canonical equations into a form that specifically displays the delay-differential structure of the ear-type systems under study. Once that is achieved then we will be able to make a correspondence between the equations and switched current mode VLSI circuits incorporating integration and delay, for which the switched current mode circuits appear to be ideal.

In the hair cell model the slow process of the opening and closing of Potassium and Sodium channels is modeled by allowing the effective conductances of these channels to vary according to laws determined by Weiss and Lelong [47, 46] and given below as equations (21) and (22)

$$G_i = \frac{G_M}{1+\exp[-(V_m(t)-V_{oi})/\mu_i]} \qquad i = Na, K \tag{21}$$

where V_m is the membrane potential, and G_M, V_{oi} and μ_i are constants associated with the basic sodium, $i = Na$, or potassium, $i = K$, channels. Physically, G_M is the maximum conductance, reached when all channels are open, V_{oi} a half-activation voltage and μ_i a characteristic voltage for a channel. In addition to the characterization in terms of voltage dependent conductances, there is the slow opening and closing of these channels which are modeled by the following differential equation

$$\tau_{Ki}\frac{dG_{Ki}}{dt} = \frac{G_{MKi}}{1+\exp[-(V_m(t)-V_{oi})/\mu_i]} - G_{Ki} \qquad i = 1, 2 \tag{22}$$

where τ_{K1} and τ_{K2} are time constants for these two channels.

To be able to develop hardware implementations for the hair cell model, we interpret these laws in terms of a basic circuit (Figure 11) for which canonical semi-state equations were developed following a treatment in [49, 50]. Although a large number of other sets can be used, the ones developed are of special interest since they should lead to transistor circuit realizations of the hair cell circuits. However, they do present a number of difficulties for realization in terms of electronic hardware largely due to the fact that the equations include differential equations for conductance rather than for voltage and current. Consequently, a transfer of the equations into voltage and current ones was needed and that was carried out by numerically converting the conductances into voltages and then realizing the resulting currents via voltage controlled current sources (VCCS). It is in this direction that our equations have evolved. The best way to make a VCCS is usually to use a differential amplifier with a current mirror [51]. When BJTs are used this leads to an hyperbolic tangent gain function whereas when MOS transistors are used the result is a square root function, both of which can closely approximate the nonlinearities of (21) and (22).

State variable types of equations work with inputs and outputs while the equations of hair cells are somewhat like resistors in that sometimes voltage is an input and sometimes current is an input, a property which one might call "non-orientedness." To reflect this non-orientedness of the hair cell we developed two sets of canonical semi-state equations which will be useful for ear-type systems incorporating hair-cell types of behaviors.

9 Applications

Applications where our research can be of possible usefulness include:

1. Real time voice processing,
2. Chaotic behavior of Kemp echoes and ear ringing,
3. Cochlea assessment and detection of damage,
4. Hearing aids,
5. Signal localization,
6. Vibration fault detection (abnormal sounds, cracks, etc ...),
7. Sensitive filters.

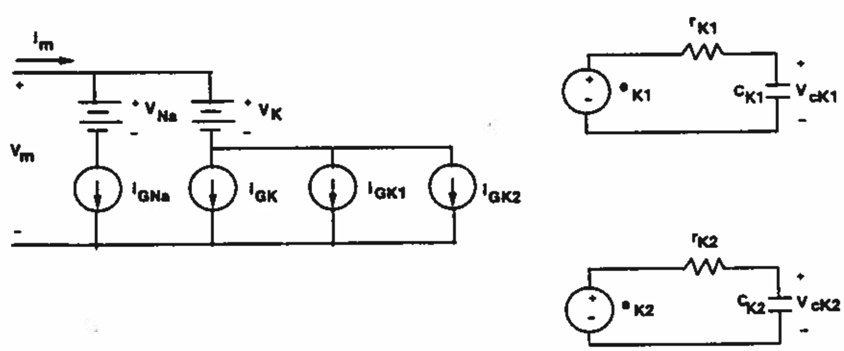

Figure 11: Equivalent circuit for the hair cell Potassium and Sodium channels semistate equations

These applications fall into four major categories, these being speech processing (real time voice processing), chaos (chaotic behavior of Kemp echoes and ear ringing), noninvasive detection (signal localization, vibration fault detection, and ear damage determination) and noninvasive correction (hearing aids and sensitive filters). Because of a lack of space we are elaborating on only the last two categories.

a. Noninvasive detection of internal damage: In those cases where the ear-type configuration is applicable, analysis of well operating systems could give a set of parameters, possibly within numerical intervals, which define the normal situation. Then by taking input-output measurements on an operating system, its parameters can be determined and compared with the normal. If they are outside of the acceptable interval, then one could claim that internal damage has occurred and by a good design the place where the damage occurs could be located. Some examples where this may be useful are the human inner ear via use of the Kemp echoes, internal cracks in drive shafts via acoustics of vibrations, animal lungs via breathing patterns, hip and knee bone damage via sonics, and internal analysis of concrete as in bridge structures via acoustical vibrations.

b. Noninvasive correction of internal damage: In the case where internal damage has been discovered via noninvasive detection and the structure is amenable to analysis or design via an ear-type configuration, it may be possible to make noninvasive corrections. One technique is to use the input output signals that allow determination of the internal damage and from that construct an ear-type filter that yields those input-output signals. Then a filter inverse to it, called the inverse-actual, can be cascaded with the actual system to yield an "identity" system which is in turn cascaded with an ear-type filter with the normal characteristics. The actual "aid" system will then be the cascade of the normal and the inverse-actual ear-type filters. An example would be a new type of hearing-aid which is the cascade of these two ear-type filters. Since this kind of hearing aid would correct the specific damage, it should be much more effective than just increasing the volume in specified frequency bands.

10 Discussion

In this paper, we have summarized the developments and results of researches associated with the Microsystems Laboratory on Kemp echo cochlea modeling and characterization, and digital lattice synthesis. From the analog cochlea model, which is based on a unidimensional fluid flow, we derived a digital scattering model in the form of a cascaded pipeline. This particular structure is crucial to the characterization from data. The idea was to embed the characteristic properties of the cochlea into the entries of the transfer scattering matrices of the cochlea sections so as to allow for their systematic extraction. With this model we are able to reproduce stimulated acoustic emissions comparable to the ones obtained via experiments. Also, with the system identification and the lattice synthesis techniques we are able to estimate the transfer function and realize it as a cascade of 16 degree-two real lossy lattice filters from which the parameters of the cochlea are estimated via our estimation method.

Our structures for modeling Kemp echoes are in a very appropriate form for VLSI construction, essentially being degree one or two cascades of real sections. By obtaining the semistate equations for each section and then realizing the semistate equations in hardware with one or two delays or integrators, a pipelined form of VLSI layout can be obtained. Similarly for the two channel ear-type systems where, however, different structures must be investigated for the different crossovers between channels. In all of these circuits, normalizations are needed to bring the parameters into range of the VLSI components available. This is rather straightforward but the hardware realization of the hair cell components on both the linear and nonlinear portions will be brought into proper range and functional form.

As seen in Figure 3, the hair cells send signals to the brain for processing in a real biological system. Consequently, one can do the similar processing for ear-type systems in an artificial neural network with time-delay as introduced in [52, 8]. This because there is a delay in processing the signals in the brain before the signals are fed back to the ear and prior to perception by a mammal. The feedback through the hair cells can cause oscillations, that is, ringing in the ear and because of the nature of the system various chaotic types of responses. The particular mathematical characterization of our ear-type systems should allow for investigation of sets of parameters which lead to ringing and chaos, as well as to excitations which will damp these effects.

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