MEDDIS IHC BASED MULTI-CHANNEL COCHLEAR MODEL SETUP AND INVERSION USING THREE TYPES OF AUDITORY FILTER BANKS

Houshang Habibzadeh-Vaneghi and Shigeyohi Kitazawa
Graduate School of Science and Engineering, University of Shizuoka, Japan

ABSTRACT

This paper deals with the forward and reverse multi-channel implementations of the Meddis's inner hair cell (IHC) based auditory models. During the forward model implementation, we have tested three different auditory models and used a suitable filter bank as a pre-processor to Meddis IHC model and calculated the auditory neural firing data for each channel.

In the reverse model implementation and test phase, different sections of the model like Meddis IHC, half wave rectifier (HWR) and filter bank have been inverted. Using the outputs of the forward model as an input to the reverse model and regenerating the estimate of the original signal, the performance of the reverse model has been evaluated.

1. INTRODUCTION

Previously two attempts were made to reverse process the auditory system to re-generate the input acoustic pattern. In the first cochlear model inversion study, Daniel Naar [2] and Malcolm Slaney [3] reverse processed the Lyon's cochlear model, to resynthesis the original signal from it's Correlogram. Their primary target was to separate a sound from noisy background. Second study was our previous work[7], which was based on Meddis's inner hair cell (MIHC) model [1].

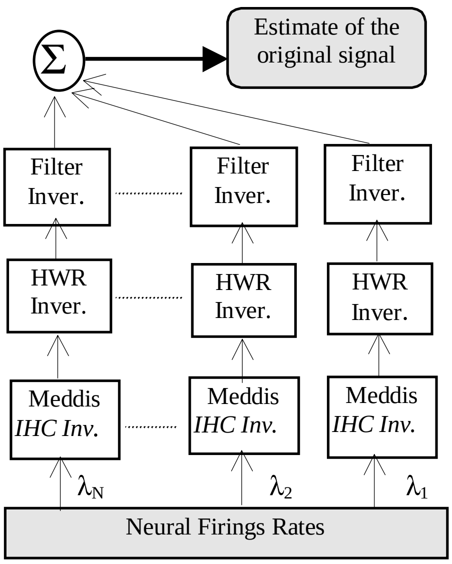


Figure 1. MIHC based cochlear reverse model.

The later method, affords us an straight forward way of regenerating an acoustic signal from its auditory neural firings. Our main goal was to construct an efficient and relatively simple model for cochlea and then use its outputs as a test base for our cochlear model inversion algorithm. Previously, we successfully inverted a single channel Meddis IHC model and regenerated the input signal to the model[7,9].

Our current study deals with the full multi-channel implementation of the cochlear model in forward and reverse modes for Meddis IHC, half wave rectifier and filter bank sections. For the best simulation of the model, we have tested and used three different auditory filter banks as a front end to the Meddis IHC model. According to our test results, Gamma-tone filter bank had the best match with the Meddis IHC model, however at the inversion phase, Ohdaira's FIR filter bank showed better results.

2. METHOD

After a complete multi-channel forward cochlear model setup using Ohdaira's 28 channels filter bank and following Delugutte's auditory system, we used one Meddis IHC model for each channel, and captured the necessary data (neural firing rates) for all channels from the outputs of cochlear model and then used it in inversion process. During the inversion phase, data for each channel was passed through a Meddis model inversion section, a half wave rectifier inversion section and finally filter bank inversion section. At the end, outputs of the all channels were summed to make the estimated version of the original input signal to the cochlear model.

To have some ideas about earlier study, first, we shall give a brief explanation of Lyon's forward and reverse model. Then Meddis based auditory model's implementation in forward and reverse modes will be discussed in detail.

2.1. Lyon's forward Auditory model

The basis of the model is a *bank of filters*, implemented as a cascade of second-order *low-pass* filters, that splits the input signal into broad spectral bands. This model uses a HWR as a detector and four stages of a multiplicative AGC to model adaptation. This cochlear forward model designed by F. Lyon and implemented by Slaney[3].

We have tested the AGC section of the Lyon's model with different input levels and filter time constants. Regarding to the results, when input level is nine times or more of the target level, and there is no limiter, AGC oscillates or blows up and fails to reach to the desired target value. The reason is that, without limiter, gain goes zero or even negative. However using limiter eliminate this problem but, because of high compression rate, it amplifies the noise and makes the inversion process more difficult and inaccurate. In contrast, Meddis model is stable even with 60dB dynamic range and is suitable for use with arbitrary stimuli such as speech and noise signals [1].

2.2. Auditory reverse model

In order to estimate the original time sequence of the signal that enters to the Lyon's cochlear model, IIR Filter bank, HWR and four stage AGC sections must be inverted[2].

3. MEDDIS IHC BASED AUDITORY MODELS

The following sections explains the necessary inversion operations to estimate the input acoustic stimulus from auditory synaptic cleft contents data . For full multi-channel operation, we have to reverse process the outputs of the all channels of the Meddis IHC, HWR and filter bank section.

The construction of the forward model has been fully explained in our previous study[9].

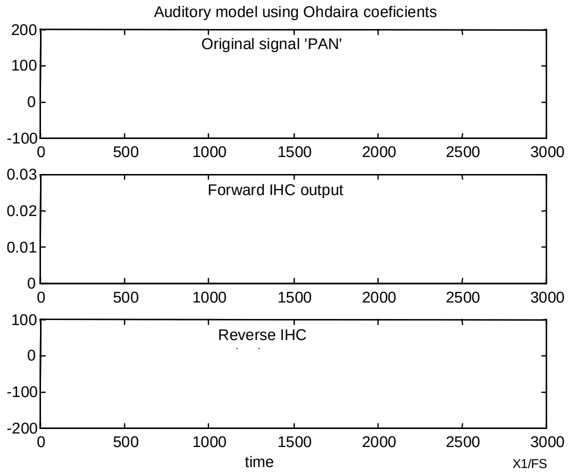


Fig.3. MIHC based Cochlear Model's forward and reverse outputs using Ohdaira's FIR filter coefficients

3.1. Meddis IHC Inversion

To regenerate the original time sequence of the filtered signal which entered to the IHC section, we have to get the auditory neural firings data as input and reverse process the IHC section. Estimated signal would be half wave rectified version of the original signal. Detailed Meddis IHC inversion algorithm has been presented in our previous works[7,9].

3.2. HWR Inversion

At this stage reconstructed *half wave rectified* signal from IHC inversion stage of each channel has been fed into the HWR inversion algorithm . Then with suitable number of iterations, the negative part has been recovered.

For HWR inversion we have used the same convex projection algorithm which Slaney and Naar have used in their 'Correlogram Inversion' work [2,3] but with different filtering and parameters. Due to usage of fast and efficient frequency domain, overlap and add FFT filtering, which works only for FIR filters.

Number of acceptable iteration have been reduced to less than 10. while in previous study it was at least 15 iterations[2].

To do the HWR inversion the following steps have been carried out :

a) Get the outputs of the Meddis IHC inversion module as a 28x(data-length) matrix(X_{hwr}).

b) Prepare the coefficients of the band pass FIR filter for 28 channels .

c) Then use the following method to recover the negative part [2,3]:

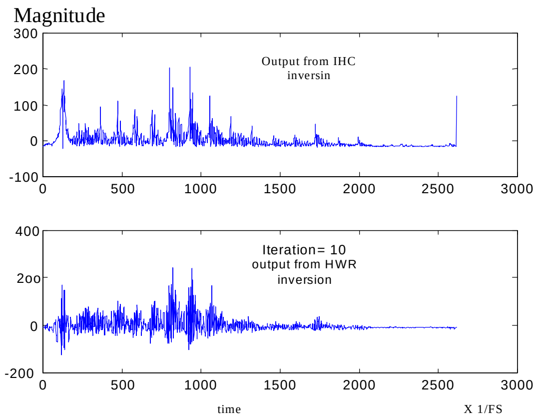
1. Estimated signal (X_{est})$\leftarrow$Half wave rectified signal (X_{hwr}).
2. Do frequency domain Band pass filtering on estimated signal.
3. Modify the estimated signal :

$$X_{est} = \begin{cases} X_{hwr} & \leftarrow X_{hwr} > 0 \\ X_{est} & \leftarrow X_{hwr} = 0 \; or \; X_{est} < 0 \\ 0 & \leftarrow otherwise \end{cases}$$

4. Go to step 2 (stop after N iterations).

Here is a test run of the program under Matlab with 10 iterations. N is defined regarding to a threshold which specifies difference between original and recovered signal.

Fig. 4 .(Upper) Input from IHC inversion section shown in Fig.3. (lower) Recovered signal, which is out put of HWR inversion



module. Both signals are sum up of 28 channels.

3.3 Filter bank inversion

Motivated by the work of Yang et al. (1992) and Daniel Naar[2], we employ the following technique with different filtering method which simplifies the inversion process .

The inversion of a linear filter bank must produce an estimate of the original input signal, x(t), given the output of each channel, y(λ,t). An estimate of x(t) can be realized by convolving the signal y(λ,t) by using the matched filter h(λ,-t). The estimate of x(t), $X_{est}(t)$ is given by the sum of convolutions:

$$X_{est}(t) = \sum_{\lambda} y(\lambda,t) * h(\lambda,-t)$$

Each term in the summation is the output of a channel convolved with the time-reversed impulse response of that channel.

Since in the forward model we have used the Ohdaira's coefficients for FIR filter bank, in the reverse mode also I have used the impulse response of the same filter for inversion. For the inversion , the following steps have been done:

1. Extract and arrange Ohdaira's FIR filter coefficients in an 28x255 matrix(28 channels with filter order of 255).
2. Filter the 28 channels input signal using the mentioned coefficients and save the result as one channel per row. Use the Matlab zero phase *FILTFILT* function for filtering. This function performs digital filtering by processing signal in both the forward and reverse directions.
3. Calculate the impulse response of the each filter channel and save the results as *H(t).* Use the Matlab *IMPZ* function for calculation.
4. Then use the following relation to calculate filter reversed signal for each channel:

   $$Fin(t) = Fout(t)*H(-t)$$

   Using the results of the step 2 and3.
5. Sum the calculated *Fin* for each channel and do phase and gain correction if necessary. Use the Matlab *SUMmation* function to sum the channels inputs which have saved in the output matrix at the previous step.

It is worthy to mention that, however using Forward / Reverse filtering removes phase problem from filtered signal it slightly increases the calculation time. Figure 5. Shows the regenerated signal which resulted from IHC, HWR and filter bank inversions successively. Regenerated signal is the estimated version of the original signal shown in Figure 6.

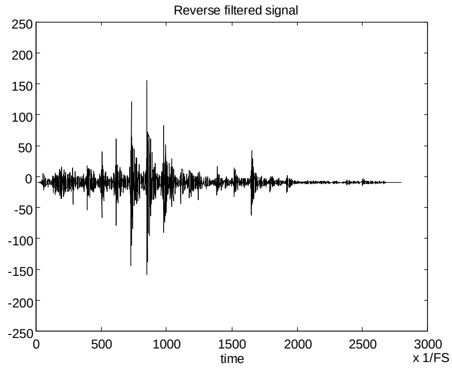


Fig. 5. Regenerated signal by reverse filtering of signal shown in Fig. 4. [SOUND 0566_INV.WAV].

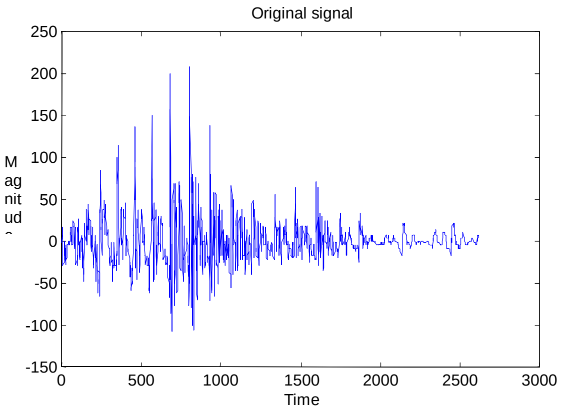


Fig. 6. Original input signal to the forward model [SOUND 0566_PAN.WAV].

4. FILTER BANKS

Here are the results of the implementation and test over three auditory filter banks which we have used them as a preprocessor to Meddis IHC model to calculate synaptic cleft contents data. Our goal was to find a suitable filter bank that has the following two properties:

- good match with Meddis IHC model.
- easily invertable.

According to our test results, filter bank based on Ohdaira's coefficients was sufficient enough to be used as a preprocessor for Meddis IHC forward model and easily invertable in reverse mode. However Gamma-tone and Seneff's filter banks had better frequency response, they were not easily invertable. Figures 8,10 and 11 show the frequency response of these three filters.

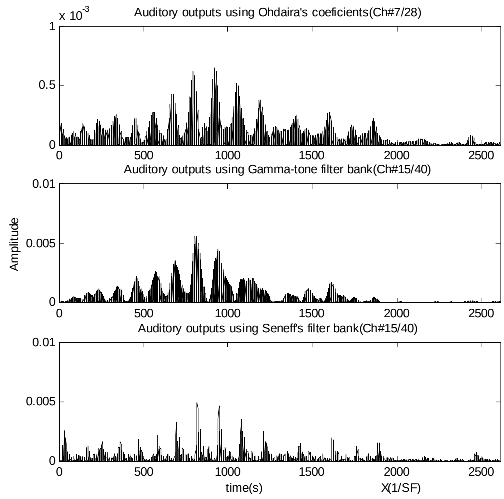


Fig. 7. MIHC forward model outputs using signal in figure 6 and three different types of auditory filter banks.

4.1. Ohdaira's Cochlear Filter Coefficients

We have tested and used a 28-channel FIR cochlear filter coefficients, which was designed by Ohdaira [6]. The frequency response of this 28-channel filter bank has been calculated and shown in the Figure 8 (Only 6 selected channels are shown).

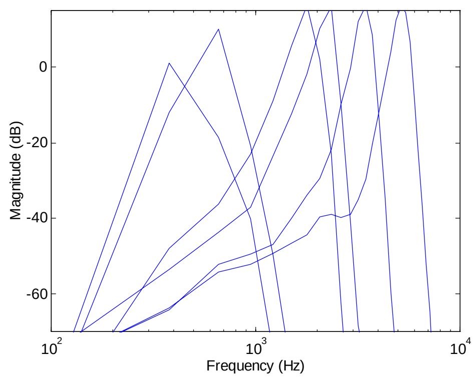


Fig .8. Frequency response of the Ohdaira's FIR filter channels 5,7,15,17,21,25.

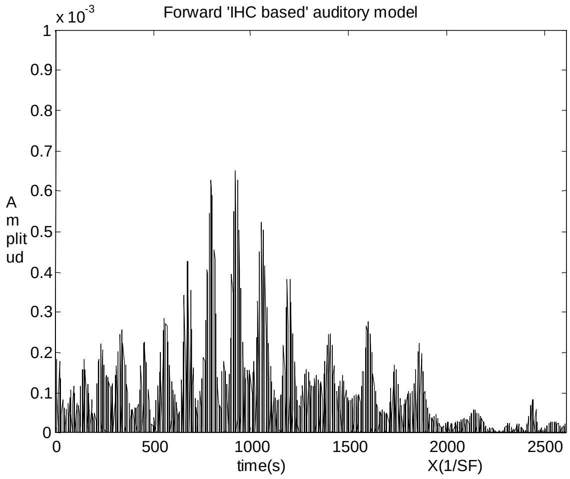


Fig. 9. MIHC based forward model output using Ohdaira's filter coefficients.

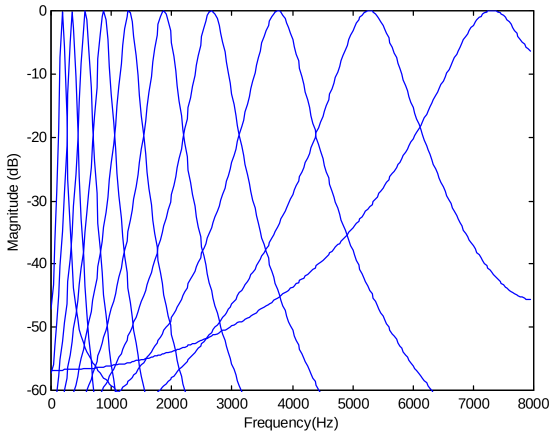


Fig. 11. Frequency response of the gamma-tone filter bank for 10 channels .

4.2. Seneff's critical band filter bank.

This filter bank was designed and described by Seneff in the 1988. She provided the coefficients required to produce the 40 channel IIR filters, which collectively covers the range from 50 to 7000 Hz. Fig. 5 shows the frequency response of the Seneff's Filter bank.

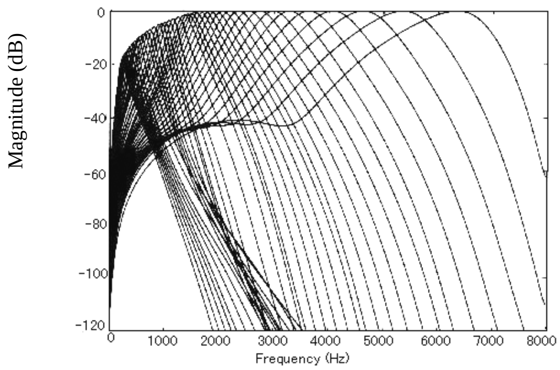


Fig 10. Frequency response of a 40 channel Seneff's filter bank (after Seneff 1988).

4.3. Gamma-tone cochlear filter bank

A Gamma-tone auditory filter bank is defined by its impulse response in time domain as follow:

$$gt(t) = at^{n-1}e^{-2\pi bt}\cos(2\pi f_c t + \phi) \quad \bullet \quad t > 0$$

R.D. Patterson et al. [8] have given some reasons for choosing the gamma-tone function as an auditory filter bank. For example they show that: it has good impulse response and its magnitude characteristic is very similar to the representation of the human auditory system. Frequency response of the gamma-tone filter bank for 10 channels is shown on figure 11.

5. CONCLUSIONS

Since Meddis IHC model inversion simplifies the four-stage AGC inversion of the earlier work to a single task , it is faster and also is less sensitive to the input signal gain. HWR and filter bank inversion sections of the earlier techniques have been used with some modifications. Due to usage of Ohdaira's FIR filter coefficients and a special zero phase forward/reverse FIR filtering , there is no phase distortion in forward mode and phase correction almost is unnecessary at inversion stage. This makes the filtering inversion section easier with less errors compare to earlier work.

REFERENCES

[1] Rey Meddis et. al.," Implementation details of a computational model of the IHC/AN synapse" *J. Acoust. Soc. Am.* 87(4), April 1990.

[2] Daniel. Naar "Sound resythesis from a correlogram" *Apple technical Rep.* No.3 May 1993.

[3] Slaney et al." Sound Analysis and Resynthesis using Correlogram " *Apple computer Inc*. Dec. 1995.

[4] Seneff , Stephanie "A joint synchrony/mean-rate model of auditory processing" *Journal of phonetics*, V.16, 1988

[5] Benjamin D. Bryant ,John N. Gowdy "Simulation of stages I and II of Seneff's auditory model (SAM) using Matlab" Clemson University.

[6] S. Ohdaira(1991) "Simulation of speech processing in auditory nerves.", *Master Thesis(In Japanese).* Shizuoka University, Japan

[7] Habibzadeh V. Houshang, Kitazawa Shigeyoshi "Stimulus signal estimation from auditory neural transduction inverse processing" *Proceeding of the ESCA. Eurospeech97*. ISSN-1018-4074, V.5 ,P2795, 1997

[8] R.D.Patterson , J.Holdthworth "A functional model of neural activity patterns and auditory images" *Speech, hearing and language processing* vol.3, JAI press, London (1992).

[9] Habibzadeh V. Houshang, S. Kitazawa "Reverse processing of the auditory-neural transduction for auditory model inversion ", *Proceeding of ASJ, 2-8-16 pp-377*, March 1997.

[10] B. Delgutte "Representation of speech like sounds in the discharge patterns of auditory-nerve fibers", *JASA, 68(3),* (1980.9)