An optimised and flexible Stimulation Strategy for Cochlear Prostheses Based on Digital Filter Implementation

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Abstract. *This article presents a stimulation strategy implementation for cochlear prostheses which was based on an digital filter optimisation. This strategy, implemented on the prosthesis's external part, should be flexible performing all the possibilities of programming. The purpose is to satisfy the great variety of pathological cases.*

Speech sounds will be filtered by a bench of digital filters having busy bands distributed on the audible spectre. The essential parameter of the captured signal is the energy which is going to be extracted from various wave bands, and will be then coded to the internal part in order to generate cochlea's stimuli.

We studied various synthesis methods of digital filters such as the non recursive realisation referred to as the 'FIR Finite impulse Response' filters and the recursive realisation referred to as the 'IIR infinite impulse Response' filters. DSP implementation showed that IIR filters had superiority in performance by using less coefficients and less order, but they were more delicate regarding stability compared to FIR filters. For our strategy, we offered these two types of digital filtering as well as different possibilities of adjustments for providing flexibility and handiness to clinicians. We studied henceforth the technical communication between the prosthesis's sound processor and the host computer in order to assure data transfer between them. We realized for that a graphical interface in Visual BASIC gathering various methods of the digital filters' synthesis, the various division forms of the audible spectre as well as the various filters' order.

I. Introduction

Research in the field of the deafness rehabilitations became a real necessity in our common life. Indeed, this handicap is a serious problem which does not stopping social disintegration. Hence, various laboratories notably signal processing groups were very interested in the development of more efficient stimulation techniques making able to provide more intelligibility in speech recognition when using such apparatus. Some are former as ones used in conventional hearing aids which bring a simple amplification of sounds. Others are more recent, as advanced stimulation strategies for cochlear prostheses [1, 2, 3, 4, 5, 6].

Cochlear prosthesis is intended for persons suffering from deep or total deafness where conventional prostheses were ineffective. These apparatus are based on the principle of electrical transmission of the sound information into the inner ear, the cochlea. We could distinguish main parts for the prosthesis [1, 3, 4]: A speech processing analyser (sounds' analyser) which concerns filtering and stake in suitable size for the transmission [1], a transmission module of the information and a internal stimulator (Fig. 1) [7].

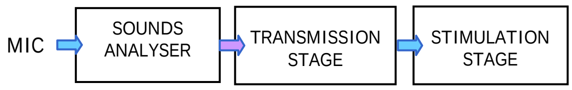

Fig. 1: Cochlear prosthesis composition

Several speech processing algorithms, often referred to as stimulation algorithms, were elaborated by research groups to extract essential parameters for the cochlea's electric stimulation [1, 3, 4]. This electric stimulation of nerve cells' leads induce a nervous message, which will be forwarded until intellectual hearing centres for the subjective interpretation [1].

In our application, dedicated to cochlear prostheses, we studied various synthesis methods of digital filters such as the non recursive realisation referred to as the 'FIR Finite impulse Response' filters and the recursive realisation referred to as the 'IIR infinite impulse Response' filters [8, 9]. This study permitted to optimise the filters' parameters and then implementation on a 'Digital Signal Processor DSP' which is the main part of the prosthesis' sounds analyser. DSP implementation showed that IIR filters had superiority in performance by using less coefficients and less order, but they were more delicate regarding stability compared to FIR filters [10]. For our strategy, we offered these two types of digital filtering as well as different possibilities of adjustments for providing flexibility and handiness to clinicians. We studied henceforth the technical communication between the prosthesis's sound processor and the host computer in order to assure data transfer between them. We realized for that a graphical interface in Visual BASIC gathering various methods of the digital filters' synthesis, the various division forms of the audible spectre as well as the various filters' order [2].

II. Cochlear Stimulation Strategy Principle

Several strategies are used nowadays and gave satisfactory results when implement on different apparatus [1]. Wilson Strategies are based on the filtering by analogical filter bench. Those belonging to the group of Melbourne University are based on formants determination of the hearing spectre [1]. These strategies present the inconvenience of the lack in flexibility and the transparency face to other types of equipments [1, 2].

Our proposed stimulation strategy which will be based on a speech processing algorithm allows the energy extraction and coding to the internal part of the prosthesis (to the implant). In this objective, we foresaw handiness and flexibility in stimulation in order to satisfy the perpetual need of the different case of patients. We took an example of a cochlear prosthesis functioning with eight electrodes of stimulation. Any time, our technique remains always valid for a possible extension in electrode number. In this example, to elaborate an adequate stimulation strategy, one should think to the exploitation of most hearing rests of the patient. Hence, the stimulation of the various sites of the cochlea should be flexible according to this conception, and this was done thanks to the various forms of hearing spectre division envisaged in this case to eight wave pass bands [1, 2].

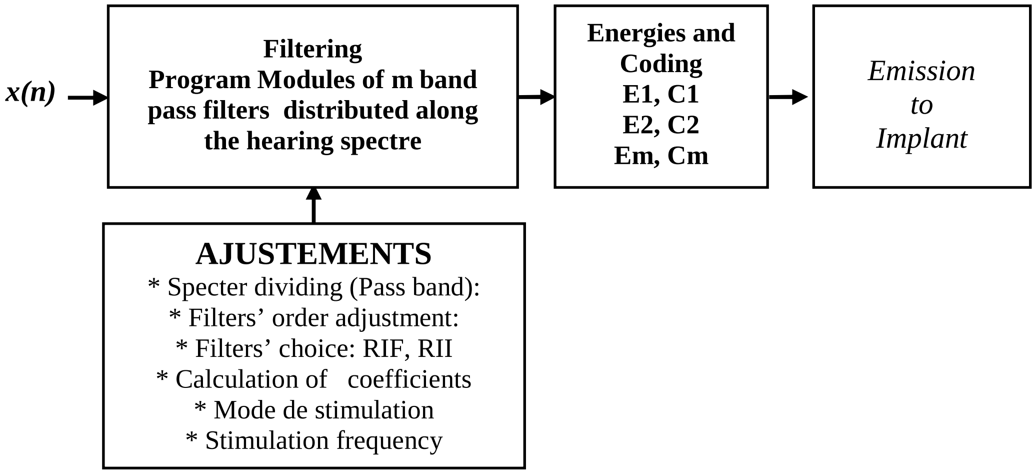

Fig. 2: Cochlear stimulation strategy based on digital filtering optimisation [1, 2]

The figure 2 illustrates the general principle of the proposed stimulation strategy based on the digital filtering approach. We envisage in this algorithm the use of various synthesis methods for determining the digital filters' coefficients, as well as different options for parameters' adaptation [1, 2].

The digital operation for signal filtering as mentioned in the first bloc would be made by all the so conceived filters at the same time, in order to have the updated outputs, i.e. energies of the processed speech signal. Parseval's relation applied in the temporal domain will allow to calculate the energies for the various selected frequency, i.e. the filters' bands. Indeed, the sum of temporal samples' squares $\sum y(n)^2$ at each filter output constitutes the energy for each considered band (Ej) of the speech signal [1, 2]. These energies would serve for the amplitude estimation of the electric stimulation that would be performed in the various sites of the cochlea. When calculating the energies Ej, the summation could be with only one term $[y(n)]^2$ either several terms $\sum y(n)^2$. The terms' number fixes the stimulation frequency since that the necessary time between two terms y(n) and y(n+n+1) is approximately equal to the period of sampling [2].

III. Main futures of the Stimulation Strategy

The principle of our proposed stimulation strategy for cochlear prosthesis consists in filtering the speech signal by using a digital and programmable filter bench offering to clinicians great flexibility for adjusting various parameters for stimulation and to choose the suitable synthesis method of filters. This flexibility is offered with a graphic interface performing all the programming possibilities [1, 2].

Hearing spectre subdivision: Filters' Distribution

Division of the considered hearing spectre for fixing band-pass filters could be made with various ways. Besides, we should decide the stimulation mode as well as the stimulation channels' number for the cochlear prosthesis. All these adjustments would take into account the pathological constraints of the patient.

The various forms of the spectre division could be as follows [1, 2]:

- Linear division of the considered hearing spectre.
- Logarithmic division concentrated towards low frequencies.
- Logarithmic division concentrated towards high frequencies.
- Division according to the evaluation of the formants positions: f1, f2, f3, f4 and f5.
- Division according to specific clinical choice.

In what follows, we are going to represent examples of division made for the case of calculated FIR filters according to the Remez method approximation. We made a linear spectre division without overlapping as well as a logarithmic concentrated division towards low frequencies. The simulation of these filters programmed under Matlab and presented satisfactory results for their following use in our stimulation strategy (Fig. 3).

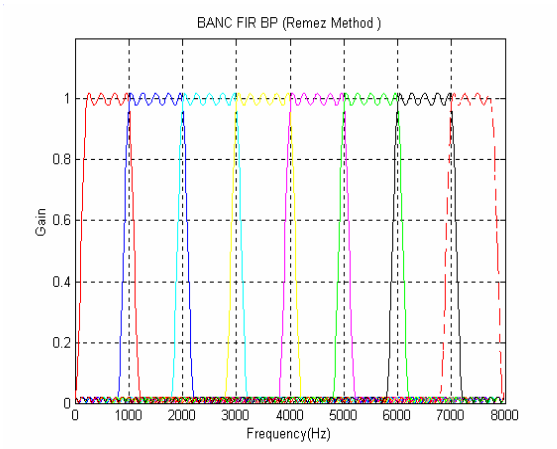

Fig. 3: Distribution of digital FIR REMEZ filters
Linear spectre subdivision

The same procedure was made for elliptic IIR filters, which we hope to implement on DSP (Fig. 4).

It was noted that for our stimulation strategy, it would possible to envisage the band filters' overlapping that could be beneficial for such pathological cases. That would take points of the spectre belonging to two consecutive bands' filters. In that case, there is a continuance between bands that could constitute sometimes an improvement in the sound perception for some case of patients.

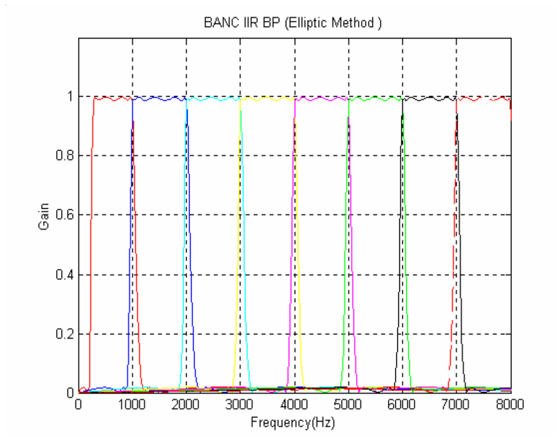

Fig. 4: Distribution of digital IIR Elliptic filters
Linear spectre subdivision

IV. DSP Implementation of the Cochlear Stimulation Strategy

Our approach in this study is going to base itself on filters cross band distributed on the considered audible spectre. The output of every filter allows the calculation of the energy which will be coded to the internal part, the cochlear implant. In this paragraph, it would be useful to present the theoretical procedures of realization of filters so used in our application.

DSP implementation was made with three stages: The main stage concerns the filtering implementation regarding FIR and IIR realisation [8, 9, 10]. The second stage concerns energy calculation by using Parseval relation, and a final stage regarding data coding to the implant [1, 2].

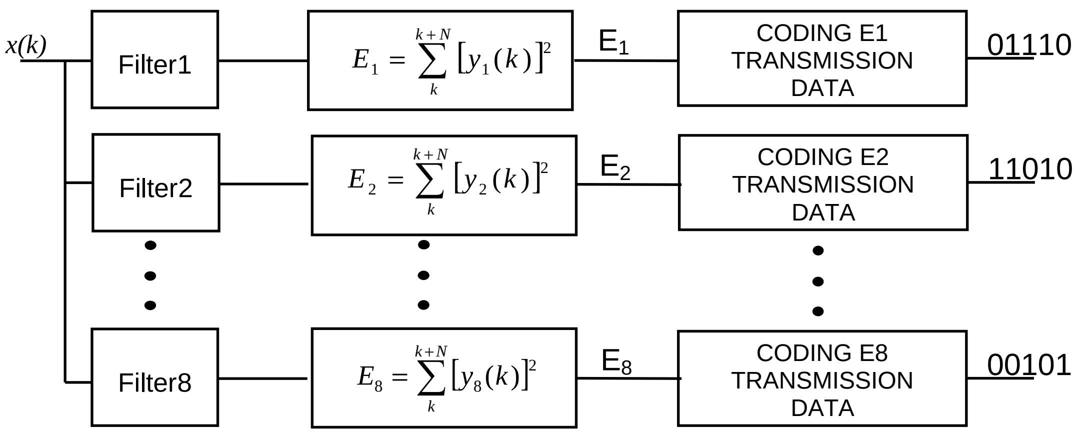

Fig. 5: Filtering, energy calculation and data transmission

The following figure shows the validation of our implementation by using harmonic sounds. In fact, we activated each filter alone by using one pure sound that had a frequency equal to the filter's central pass band frequency. In the following example, the pure sound was at 3500Hz that corresponds to the fourth filter (Fig. 6) . It was also possible, as a secondary validation, to use composite sounds in order to activate for example two filters as the first one and the fourth one.
In this illustration, the chosen sampling frequency was 16000Hz so the sounds spectre would be limited to 8000Hz. With the linear spectre dividing in order to assign filters' pass band, the fourth filter would have a band pass equal to [3000Hz-4000Hz] [2].

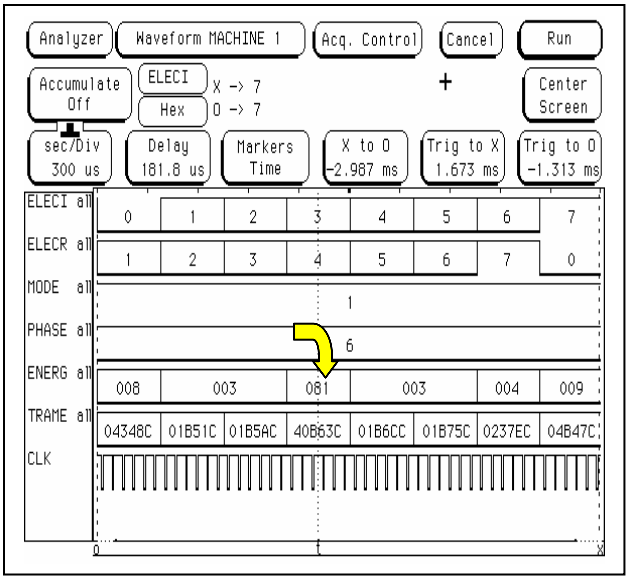

Fig. 6: Activation of filter number four [2]
Pure sound at 3500Hz

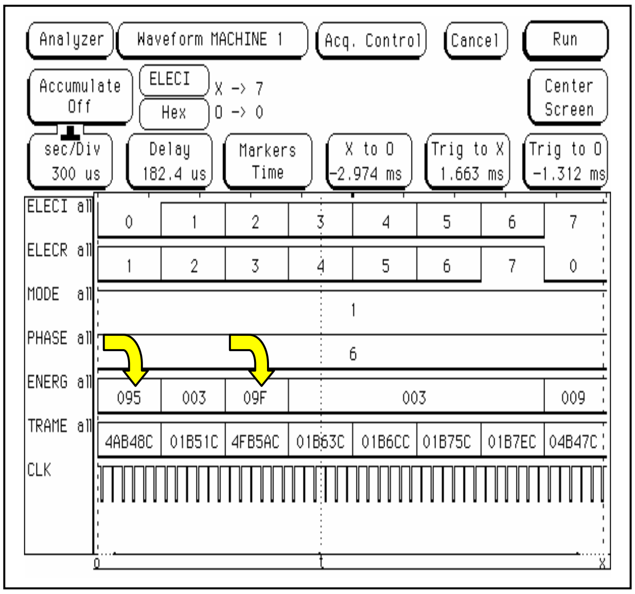

Fig. 7: Activation of filter number one and number three [2]
Composite sounds including two harmonics: 600Hz, 2500Hz

Similarly, and in order to test composite sounds, the following fig. 7 shows the validation of our implementation for two harmonics which permitted to activate two filters, number one and number three. The harmonics were at the frequencies of 600Hz and 2500Hz that corresponds to the central pass band frequency of these filters. The same procedure could be made for another composite sound including more harmonics. All filters could be activated depending on the presence of the harmonics in each band pass filter.

VI. Graphical Interface: Parameters' adjustment

We foresaw a graphic interface collecting all the parameters for algorithm adaptation. This interface was conceived in Visual BASIC and will allow communication with a host computer. All the developed theory for our temporal technique would be going to be put under software shape having as main characteristic flexibility and handiness that are essential criteria for our stimulation strategy for cochlear prostheses.

Indeed, in our completely programmable strategy, we shall propose the user interface assuring the choice of the following functions (Fig. 8):

- The sampling frequency, so the envisaged hearing spectre.
- The number of stimulation channels (or active electrodes).
- The methods of division of the hearing spectre, with the option with or without overlapping.
- The activation rhythm of each electrode (or channel stimulation).
- The digital filters FIR or IIR (method of synthesis, coefficients).
- The sizes and the orders of the filters' bench.

This interface is going to allow clinicians to experiment different possibilities during tests: various modes of filters' synthesis, spectre division, number and filers' orders… The proposed strategy would perform the most suitable stimulation for the pathological case. The stimulation parameters so found would be transferred to the specific processor ' DSP '.

VII. Conclusion

We studied a flexible stimulation strategy for cochlear prostheses which was based on digital filter optimisation. Strategy so proposed uses a temporal analysis by filtering speech by a bench of digital filters in order to extract essential parameters for the stimulation.

We have also conceived a software tool which allows strategy implementation as well as communication with the dedicated digital processor. Indeed, in our strategy, it is possible to make the synthesis of the digital filters by the IIR approach or by the FIR approach according to various synthesis methods.

It would be also necessary to specify the filter parameters according to the pathological case. We offer therefore several models of hearing spectre subdivision for exploiting the hearing capacities of these various pathological cases.

Simulations on harmonic signals as well as on speech signal were satisfactory since we note the extraction of essential parameters for stimulation. At the end, we can indeed note that our strategy reassembled not only the efficiency and flexibility criteria, which was not offered by other strategies, but also the resumption of all the performances of these existing strategies.

VIII. References

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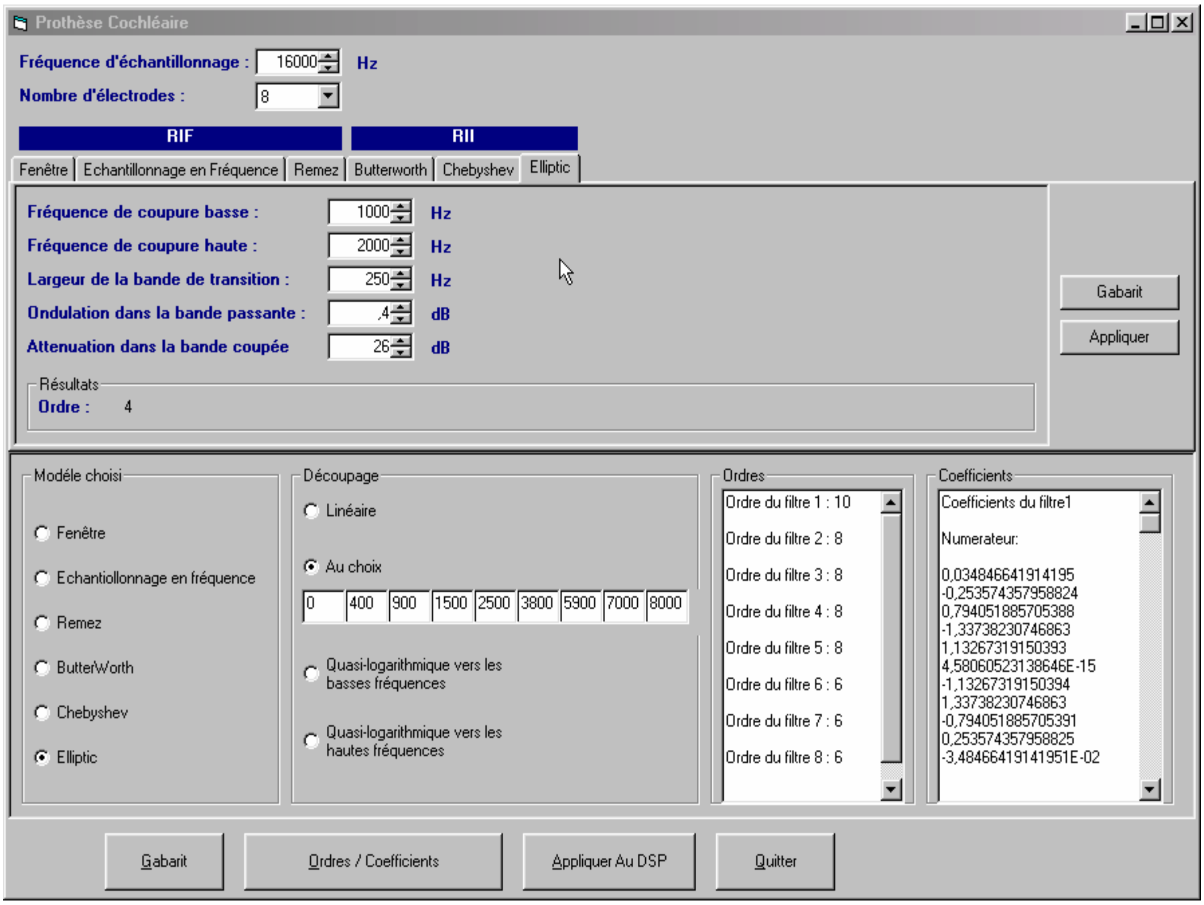

Fig. 8: Graphical Interface dedicate to stimulation parameters' adjustment [2]