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Speech Processors for Auditory Prostheses

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submitted by

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1.0 Introduction

Work performed with the support of this contract is directed at the design, development, and evaluation of speech processors for use with auditory prostheses implanted in deaf humans. Major research efforts are proceeding in four areas: (1) developing and maintaining a laboratory based, software controlled, real time, speech processing facility where processor/stimulator algorithms for monaural and binaural eight-channel implants can be implemented/tested and a wide range of psychophysical measurements can be made, (2) using the laboratory facility to refine the sound processing algorithms used in the current commercial and laboratory processors, (3) using the laboratory facility to explore new sound processing algorithms for implanted subjects, and (4) designing and fabricating programmable, wearable speech processers/stimulators and using these systems to: (a) field test processor algorithms developed and tested in the laboratory, (b) evaluate the effects of learning using longitudinal evaluations of speech reception, and (c) compare asymptotic performance of different speech processors across subjects.

The material of this report relates to two of the research areas mentioned above: (1) longitudinal testing using a wearable sound processor developed together with colleagues at the University of Geneva[1], the Geneva Engineering School and the Research Triangle Institute (RTI) and (2) the refinement of sound processing algorithms for intracochlear stimulation.

2.0 Geneva, MIT and RTI Implementations of CIS Sound Processors

A number of investigators have studied the speech reception of Ineraid implantees as they move from their commercial Compressed Analog (CA) processor to a version of the Continuous Interleaved Sampling (CIS) processor in the laboratory and in the field [1-4]. Wilson has noted that virtually every Ineraid subject fit using their laboratory CIS processor demonstrated significant improvement in speech reception during the several-day fitting session [5].

The Geneva Group also reports that most Ineraid users experience a significant improvement in speech reception at the initial fitting of a CIS processor [6]. Figure 1, for instance, shows CA and CIS scores for 22 subjects obtained with a 14-item consonant test. The CA processor was the commercial Ineraid system that each subject had been wearing for at least six months and the CA score for each subject represents the mean of the last five scores obtained with that system. The CIS processor score also represents the mean of five tests, conducted on different days. Except for the testing and the short conversations related to adjusting the CIS system, the subjects did not have an opportunity to practice and learn with this system. While at least five subjects do not show significant improvement with the CIS processor, the majority probably demonstrate functional improvements.

Our experience with Ineraid subjects moving from their CA system to a CIS

[1] Partial support for this work contributed by the Wilsdorf Foundation of Geneva.

sound processor is summarized in Figure 2 and does not show a majority of subjects demonstrating an immediate improvement. The top panel of Figure 2 plots the scores achieved on the Iowa 16-Consonant test [7] by 18 subjects on the day they were fit with a wearable CIS sound processing system. The CA scores were obtained using the subjects' Ineraid processors (at least six months experience). The CIS processor was implemented using the Geneva Wearable Processor (GWP) running software developed by us (MIT/GWP) [8]. Note that unlike the experience of Wilson and the Geneva Group, at least half of our subjects do not demonstrate an immediate improvement.

The bottom panel of Figure 2 compares the same CA scores with CIS scores obtained after each of the subjects had worn the CIS system for at least six months. Even with this added experience with the CIS system, at least seven (or 39%) of the subjects do not demonstrate marked improvements in speech reception (although virtually all express a strong preference for the CIS system and none will return to their Ineraid system).

A number of possible reasons for this apparent difference between subject performance at MIT and that reported by Geneva and RTI have been suggested (e.g., implementation, fitting methods, testing methods, and small numbers as well as different groups of subjects). One intriguing possibility is that because the developer of the Ineraid system fit the subjects in Boston, they received better optimized systems than other Ineraid subjects. Figure 3 presents mean data for Ineraid and CIS systems from Geneva (14-consonant test) and MIT (16-consonant test) that relates to this possibility. Notice first the difference between the Geneva (40.8%; stderr: 4.1) and MIT (51.2%; stderr: 3.6) mean scores for the Ineraid (CA). This difference is consistent with the hypothesis that one reason the MIT subjects do not show the improvements in speech reception reported by RTI and Geneva is that the scores used as a basis for judging improvement are, on average, higher. Note also that the mean performance (at the time of fitting) of the MIT group of subjects using the MIT/GWP CIS processor is at least as high as the Geneva subjects. Because mean scores of information transfer computed from consonant recognition results collected for 11 subjects by RTI are similar to those collected for 15 subjects in Geneva [9] and because the mean consonant scores collected with CIS processors at the time of fitting are similar to those reported by Geneva [6], it is possible that a difference in performance does not exist between any of these three versions of CIS processing.

In order to more directly address this issue, we, together with Geneva and RTI, have developed plans to test a few subjects that have been fit with at least two of these three versions of CIS processing. Because Marco Pelizzone and others from his group will be in the U.S. attending scientific meetings in August, we decided that they would come to Boston and fit two of our subjects with their version of CIS processing. The hardware platform will be the GWP that is common between the two groups. The GWP will be programmed with the Geneva software implementation and the visitors from Geneva will use their techniques to fit the two Boston subjects. The subjects will wear this Geneva/GWP system for at least one month and speech reception tests will be administered at the time of fitting and at the end of the one-month trial.

At the end of September, Colette Boëx-Spano will finish her two-year postdoc at MIT and return to Geneva. She (and possibly one senior staff person from MIT) will then fit the MIT/GWP system to at least two Geneva subjects using the fitting techniques used at MIT. These Geneva subjects will also be tested at the time of fitting and at the end of the one-month trial period.

Now that RTI has also ported their version of CIS processing to the GWP (RTI/GWP), we hope to transport the same two Boston subjects tested with the MIT/GWP and Geneva/GWP processors to RTI for CIS fitting with the RTI laboratory (RTI/lab) system and the RTI/GWP system. These subjects would then be tested with both systems at the time of fitting and with the RTI/GWP system after the end of a one-month trial.

These CIS trials with a small number of subjects should reveal large differences in performance between the three systems. No matter what the outcome, the results of these trials are likely to be interesting. If substantial differences in performance are found, it will be possible to isolate the hardware/software/fitting differences responsible. If performance does not differ within a subject across the three systems, the differences in hardware/software/fitting will be placed in a functional perspective.

In preparation for the testing described above, we thought it would be useful to summarize our understanding of the three systems we will be testing: MIT/GWP, Geneva/GWP, RTI/GWP and RTI/lab. This serves two purposes. First, describing our understanding makes it more specific and concrete in our own minds. This will be important as we both prepare for the testing and interpret the results. Second, we hope that others who read the following characterizations will identify and bring to our attention misunderstandings and areas that are not clear.

2.1 General CIS Structure and Fitting Issues

General CIS channel operations are shown in Figure 4. To produce current stimulation for six electrodes, six channels would be implemented. As shown, front-end operations like automatic gain control (AGC), and/or a frequency dependent gain (preemphasis) operate between the microphone (MIC) and the set of band-pass filters that span the frequency range to be represented. Each channel consists of: (1) a band-pass filter; (2) an operation that measures the envelope variations in each band-pass filter's output waveform; (3) low-pass filtering of the derived envelope signal; (4) a nonlinear mapping operation which reduces the relatively large range of band levels (50-60 dB) to a narrower range consistent with the dynamic range of an electrically-stimulated implanted electrode (5-25 dB); and (5) a modulation operation which imposes the mapped envelope variations upon the amplitude of the pulsatile stimulation waveform.

As shown in Figure 4, each channel's operations can be viewed as an input process (with an equivalent gain, Gin) whose output is the input signal to the nonlinear mapping operation. The output of the mapping is adjusted by another equivalent gain

(Gout) and converted to a stimulation current. Because of the nonlinear mapping, various pairs of Gin,Gout will produce a comfortable loudness sensation, but, for each of these pairs, the distribution of the band levels will be positioned differently with respect to the mapping function. One important aspect of fitting a CIS processor to a particular subject is the adjustment of the Gin,Gout pair.

Fitting procedures that specify the CIS processor parameters usually include psychophysical measures of the threshold level (THR) and the most comfortable loudness level (MCL) for each of a subject's implanted electrodes. In some cases, the manner in which the perception of loudness grows with increasing current level and statistical measures of the input speech signal are also employed. These measures are used to establish values for Gin and Gout that determine a suitable operating point.

The next three Sections describe our understanding of the implementation and fitting procedures associated with the MIT/GWP, the Geneva/GWP, and the RTI/GWP processing schemes. Table I summarizes the operations of these three systems and Figures 5, 6 and 7 show the magnitude responses of their preemphasis, band-pass and low-pass filters respectively.

2.2 The MIT/GWP Implementation

Referring to Figure 4 and Table I, there is no front-end processing in the MIT/GWP implementation. The microphone signal is applied to the band-pass filters after some amplification that is not frequency dependent. There is no automatic gain control applied before the filtering. Each of the filters is an equiripple, finite impulse response (FIR) design which spans the frequency range from 250 to 7000 Hz with log-spaced crossover frequencies at –6 dB, and –60 dB rejection in the stop bands (see Figure 6). The envelope is derived using quadrature channels to form a complex signal whose magnitude is the measured envelope. This envelope is smoothed and limited in its frequency range by an equiripple low-pass FIR filter whose –3 dB bandwidth is at 400 Hz and is –60 dB at 1000 Hz[2] (see Figure 7, top left). The output of this filter is applied to the mapping function at a level described in the fitting procedure below. The input range of the mapping is 60 dB, and is implemented with a table of 1024 values of the function: Out = A*(In**0.001) + B. The constants are used to fit the Out signal to the range from the THR current level to a maximum current (Imax) determined from the psychophysically measured MCL. This mapped quantity, (Out) is used to modulate a cathodic-first, bipolar pulse train with repetition rate of 2000 Hz and phase duration of 31.25 microseconds. Six of these envelope modulated pulsatile stimulation currents are applied to electrodes starting at the most basal, in the order 6-3-5-2-4-1. We have not observed a strong effect when using other stimulation orders (e.g. 6-5-4-3-2-1, or 1-2-3-4-5-6).

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[2] This is only true for the lower five channels of the CIS implementation. The sixth channel is smoothed by the filter shown on the top right of Figure 7. This sixth channel's low-pass filter also has a –3 dB bandwidth of approximately 400 Hz but does not attenuate at –60 dB until a frequency of approximately 1500 Hz.

As mentioned earlier, the fitting procedure serves to define the Gin and Gout parameters based upon: (1) psychophysical measures of THR and MCL made for each subject's electrodes and (2) measures of level histograms made for each CIS processing channel. We use a subset of the TIMIT database [10], sentences spoken by male and female talkers from eight dialect regions in the US, as the speech input to compute the band-level histograms. The sentence subset used consists of one male and one female talker from each of the eight dialect regions, each speaking ten sentences that are played at a comfortable listening level. Gin is set to be uniform across channels and is adjusted so that the highest 1% of the envelope levels generated by the TIMIT subset reach the clipping or saturating level at the mapping input for the highest energy channel (in our processor, channel #2) [8]. When speech is applied to the processor at this level and the output of the mapping is allowed to range from a highest stimulation current of MCL to a lowest value of THR (envelope levels below –60 dB re: max input are set to zero), the perceived loudness of the CIS processor is above the level of comfort. We maintain the minimum output level at THR, and reduce Imax from MCL to a level determined by Imax = k[MCL-THR] + THR, where k is adjusted to produce a comfortable listening level. For the twenty Ineraid subjects we have fitted, the value of k ranges from 0.4 to 0.7. The computed mapping function uses an input dynamic range of 60 dB and an output range (for each electrode) of 20*(log10(Imax/THR)). This fitting procedure was used with the twenty Ineraid implanted subjects who are currently using the MIT/GWP processor.

2.3 The Geneva/GWP Implementation [11]

Again referring to Figure 4 and Table I, the Geneva/GWP system employs both an AGC operation and frequency shaping of the microphone signal before the band-pass filters. The AGC operation is set so that conversational-level signals are not affected by the AGC amplifier, but loud signals are limited in energy so as not to cause uncontrollably loud stimulation. The output of the AGC amplifier is pre-emphasized by an analog high-pass filter whose -3 dB frequency is set to 1200 Hz and rolls off below this frequency at 6 dB/octave (as shown on the top of Figure 5). This preprocessed signal is then filtered by six FIR band-pass filters spanning the range from 150 to 6400 Hz with log-spaced crossover frequencies at -6 dB. The stop band attenuation for each filter is -60 dB. The envelope of each filter output is measured using the same quadrature operation as the MIT/GWP system. Each envelope signal is then low-pass filtered with an FIR filter whose -3 dB point is at 400 Hz and is down by -55 dB at 1000 Hz[3]. Each of the six smoothed envelope signals is multiplied by a separate gain value before the mapping operation, so that each of the envelopes reaches the peak input (saturation) mapping level 1% of the time, for conversational level speech input. Because the Gin varies across channels, the input envelope range that is mapped varies for each channel from approximately 50 to 55 dB. We will say more about this below. The mapping function is

[3] As in the MIT/GWP processing, the Geneva/GWP processing uses two low-pass filter designs. The filter shown in the middle left of Figure 7 is used for the lower three channel envelopes, and the middle right filter is used for the upper three channel envelopes. The use of two filter designs is necessitated by two different computation rates for the upper and lower filters.

implemented with a table of 512 values of the function: Out = A*(In**0.001) +B. The output of this mapping table varies over a range described in the next section. The mapped output is applied as amplitude modulation to a cathodic-first, bipolar pulse train with repetition rate of 1667 Hz and phase width of 50 microseconds. The update order of electrode stimulation is (starting at the apical end of the cochlea) 1-2-3-4-5-6.

The Geneva fitting procedure measures THR and MCL current stimulation levels, but begins by adjusting the Gin for each channel based only on: level statistics measured using four-talker speech babble. The speech babble is constructed by starting with four-talker babble and then creating three new versions by delaying the original by three seconds three times. These four babble tracks are then mixed to form a composite babble of "sixteen" talkers. This composite recording is played in a small anechoic chamber at 75 dBSPL(A) through the earhook microphone of the processor used for the fitting. Using all of the processing described in the previous paragraph, histograms of the six envelopes, before the mapping operation, are measured. For *each* channel, the peak of the mapping input is set so that 1% of the envelope values are at or above the saturation level. To determine the lowest point of the mapping input, the babble is turned off and histograms are made of the noise envelopes for each channel. For *each* channel, the band level that is higher than 99% of the noise levels is used as the lower boundary of its mapping function input range and is mapped to THR. For this procedure, the mapping input ranges from approximately 55 dB for channel #1 to approximately 50 dB for the remaining channels. The adjustment of the Gin for each channel imposes a frequency weighting (at least on a channel by channel basis) that is much like pre-emphasis.

When the input is adjusted as described, the output range of the mapping must be converted to a range of stimulation currents that provide comfortable loudness for conversational speech. The minimum output current is the measured THR. The peak stimulation current for each channel is chosen from psychophysical measurements of each electrode's loudness growth function using 500 millisecond bursts of the CIS pulsatile waveforms. A starting choice of current which produces the same loudness level at each electrode is used as the upper limit of each channel's stimulation current (Imax). This initial loudness level is chosen to be approximately half of the loudness range reported by the subject. If these Imaxs produce a processor whose loudness is either too loud or soft, a new loudness level is chosen either higher or lower as is appropriate. Then currents from this new loudness level are read from each loudness growth curve. These currents are the new Imaxs. This process is repeated for as many iterations as are required to obtain a processor which the subject finds to have comfortable loudness listening level. Once this process is finished, neither output loudness nor input sensitivity need further adjustment by the subject while the processor is in use (especially given that the input level is limited by the AGC circuit).

2.4 The RTI/GWP Implementation

Referring again to Figure 4 and Table I, the RTI/GWP uses no input AGC but does employ the same pre-emphasis processing (in this case as a software operation) as described for the Geneva system: a high-pass filter with cutoff frequency at 1200 Hz and a roll-off of 6 dB/octave below that frequency (Figure 5, bottom). One of the standard band-pass filter configurations used by RTI is realized by six, 6th order Butterworth filters which are log-spaced over the frequency range from 300 - 5500 Hz, with crossover frequencies at the -3 dB points. The envelope of each band-pass filter output is measured by computing the magnitude of the waveform samples (referred to as full-wave rectification of the band-pass waveform) and smoothing this signal with a low-pass filter. The low-pass filter is implemented with a 4th order Butterworth filter with a -3 dB point at 400 Hz (Figure 7). The smoothed envelope is the mapping function input. The mapping function uses an input range of 54 dB and implements the function: Out = A*(In**-0.001) +B, with a 512 point table. The output value is used to modulate an anodic-first pulsatile waveform with repetition rate of 2525 Hz and phase duration of 32 microsecond. The update order of electrode stimulation varies from subject to subject -- base to apex, apex to base, or alternating 6-3-5-2-4-1. The output range of the stimulation currents are based upon psychophysical measurements made during the fitting procedure.

The fitting procedure for RTI subjects is based upon psychophysical measures of MCL, and THR using a 50 millisecond burst of a CIS waveform (2525 Hz, 32 microsecond/phase). These two levels are used to compute the "A" and "B" of the mapping function and, therefore, the output currents range from MCL to THR. In contrast to the MIT and Geneva procedures, there is no attempt to control the input range of envelopes that are mapped. Instead, with the output ranges as described, the input gain from the microphone is adjusted so that conversational speech provides stimulation currents that evoke comfortable loudness levels. There may be some slight adjustment of the output stimulation, after interaction in the laboratory, such that MCL and THR currents are reduced by an overall factor, K. That is, MCL is reduced to K*MCL and THR is reduced to K*THR, where K ranges from between 0.8 to 1.0. Using this range of stimulation currents suggests that an RTI processor implementation probably operates at lower levels into the mapping than either the MIT or Geneva implementations.

FUTURE WORK

During the next two quarters we expect to fit at least two Boston subjects with Geneva/GWP processors and at least two Geneva subjects with MIT/GWP processors. We also hope to fit the same two Boston subjects with RTI/GWP processors. The speech reception results from these experiments should help to determine whether any of the differences between these CIS processing schemes are functionally important. We also plan to record input and output waveforms that characterize each of these processors when fitted for the same subject.

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Figure Captions

Figure 1. Data from the Geneva Group comparing CA and CIS scores for 22 Ineraid-implanted subjects for a 14-item consonant test. The CA processor used was the Commercial Ineraid system.

Figure 2. Data from MIT comparing CA and CIS performance for 18 Ineraid-implanted subjects for a 16-item consonant test. The top panel presents data obtained on the day that subjects were fit with the CIS system running on the GWP. The bottom panel presents data obtained after six months of CIS use. The CA scores used were obtained with the subjects' Ineraid processors (at least six months experience).

Figure 3. Data comparing mean scores for CA and CIS processor-use at Geneva and MIT. The Geneva scores are based upon 14-item consonant tests, the MIT scores used 16-item consonant tests.

Figure 4. The signal processing used in the CIS processor. The figure emphasizes the nonlinear mapping operation and the input and output gains around it. The front end processing may include an AGC and/or a frequency equalization. For this study a CIS processor is using six channels.

Figure 5. The frequency response of the high-pass filters used in the front end processing for the Geneva and RTI CIS implementations used for the GWP. The Geneva filter is implemented with analog hardware, while the RTI filter is implemented in software. The responses are indistinguishable.

Figure 6. The frequency responses for the band-pass filters used in the MIT, Geneva, and RTI processors implemented in the GWP. The Geneva and MIT designs use finite-impulse-response designs, while RTI uses infinite-impulse-response designs. For MIT and Geneva the lower filters (1-3) are computed at a lower rate than the upper filters (4-6).

Figure 7 The frequency responses for the low-pass smoothing filters used in the three sites' CIS processors. For MIT and Geneva, two designs are required since the envelopes are computed at two rates. A single design is used in the RTI processor.

TABLE I

CIS Implementation Comparison

Operation	Geneva	MIT	RTI
AGC	peak limiting only	none	none
Equalization	HPF 1200 Hz (6dB/oct)	none	HPF 1200 Hz (6dB/oct)
Bandpass Filters	6 FIR, Hamming wind, log-spaced 150-6440 Hz - 6dB cross, -60 dB stop	6 FIR equiripple,log-spaced, 250-7000 Hz -6dB cross, -60dB stop	6 IIR, Hamming (6th), log-spaced, 300-5500Hz -3dB cross
Envelope Measure	Quadrature (complex)	Quadrature (complex)	Magnitude (FWR)
Lowpass Filters	FIR, Hamming wind, -3dB@400, -55dB@1k	FIR, equiripple, -3dB@400, -60dB@1k	IIR, 4th order Butterwth -3dB@400Hz
Map Input Level (Gin)	All channels @ 1% sat.	strongest channel@1%	set for comfort. loudness
Map Input Range	50-55dB	60dB	54dB
Mapping Function	Out= A*(In**0.001)+B	Out= A*(In**0.001)+B	Out= A*(In**-0.001)+B
Psychophysical Measures, Adjustments	Uses 500ms CIS signal, THR, Loudness growth, use 50% loudness point	Uses 50ms CIS signal, THR, MCL, Imax= k[MCL-THR] + THR	Uses 50ms CIS signal, THR, MCL. kTHR, kMCL
Stimulation Waveform	Cathodic 1st, 1667 Hz, 50 microsec./phase	Cathodic 1st, 2000 Hz, 31.25 microsec./phase	Anodic 1st, 2525 Hz, 32 microsec./phase
User Controls	none	Input Sens.	Input Sens, Output Vol.

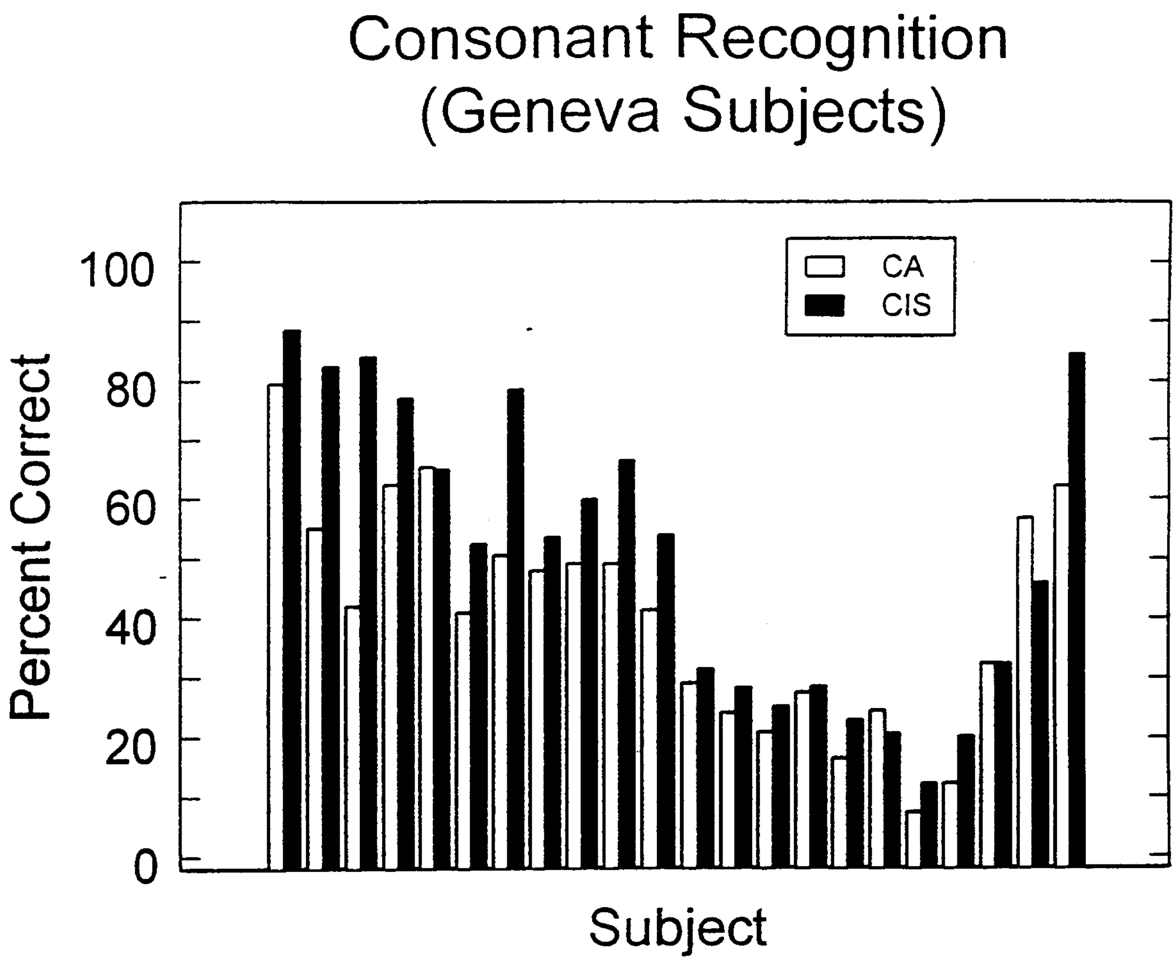
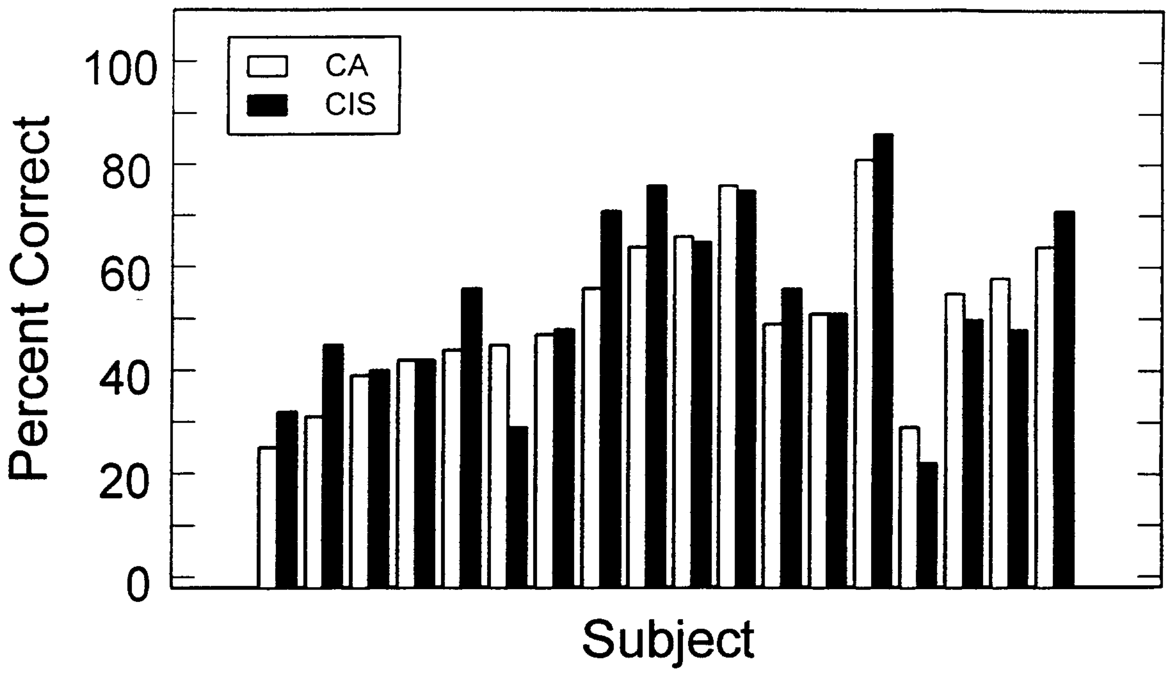

Consonant Recognition
(Geneva Subjects)
100
80
60
40
20
0
Percent Correct
CA
CIS
Subject

FIG. 1

MIT/GWP (0 months)
16 Consonant Test



MIT/GWP (6 months)
16 Consonant Test

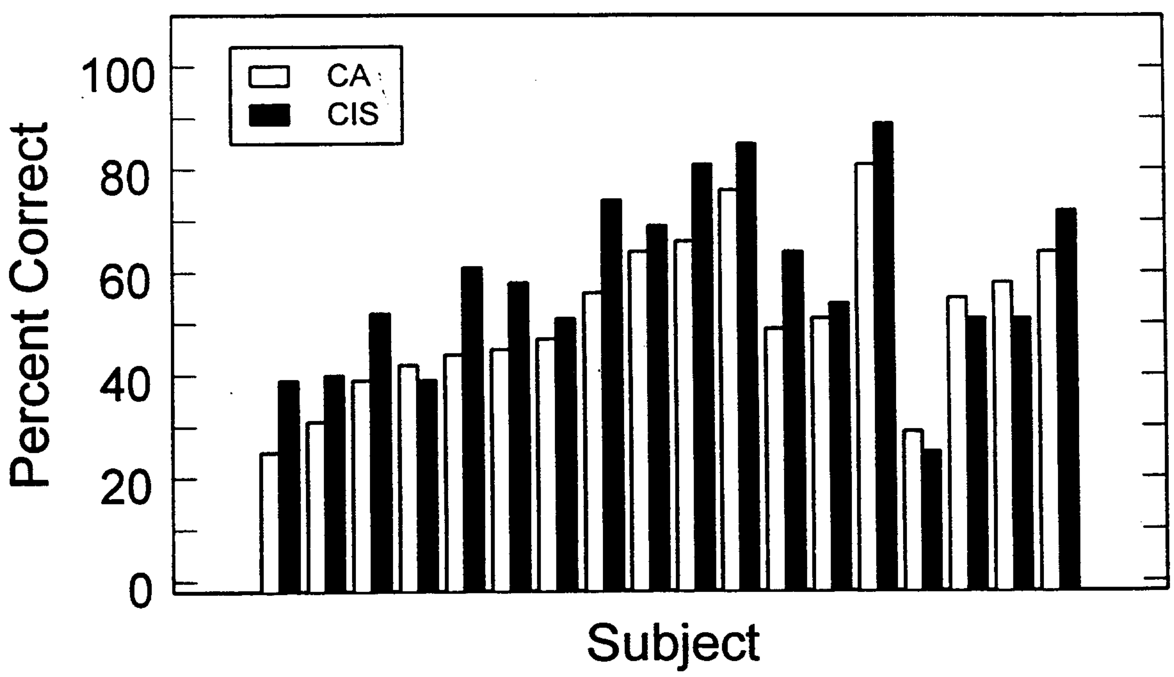


FIG. 2

13

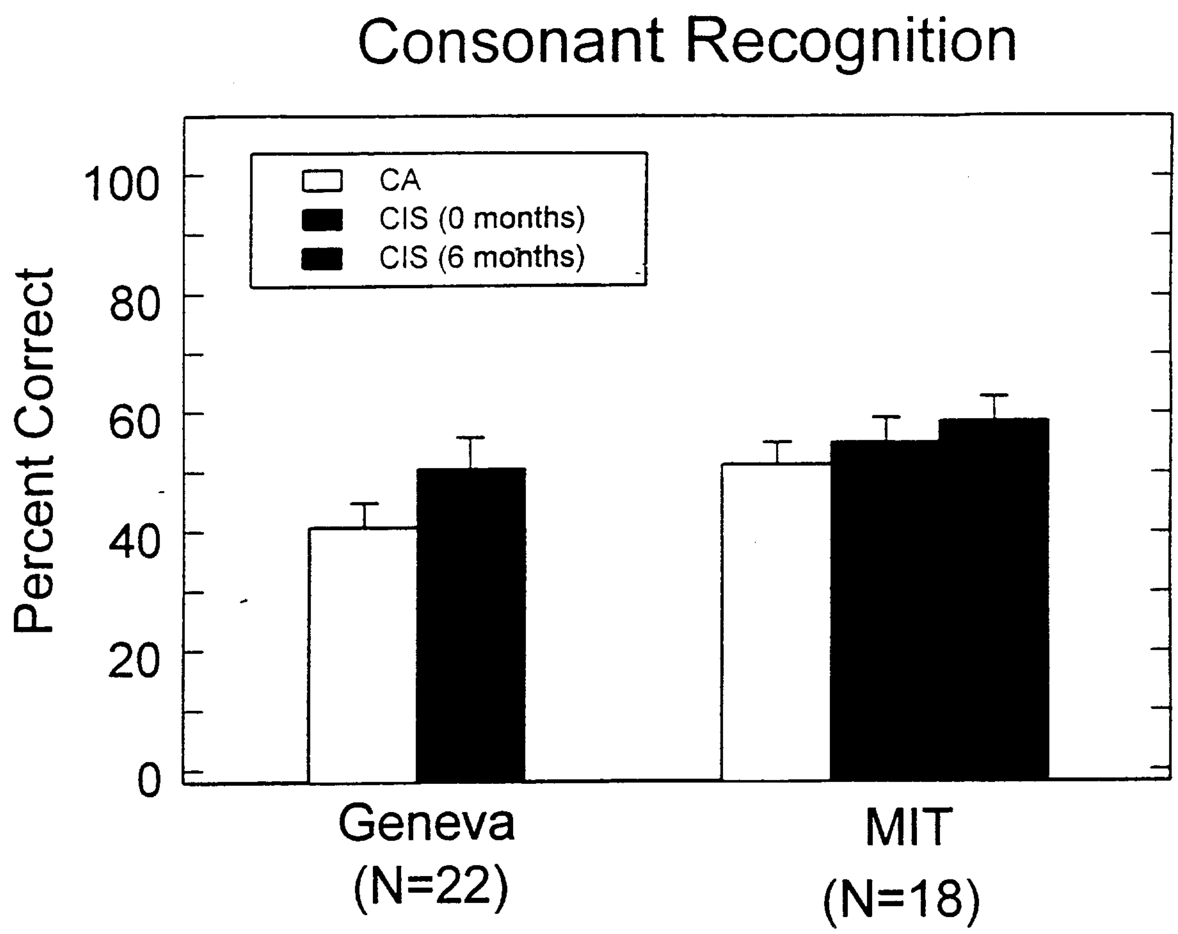

Consonant Recognition
100
80
60
40
20
0
Percent Correct
CA
CIS (0 months)
CIS (6 months)
Geneva
(N=22)
MIT
(N=18)

FIG. 3

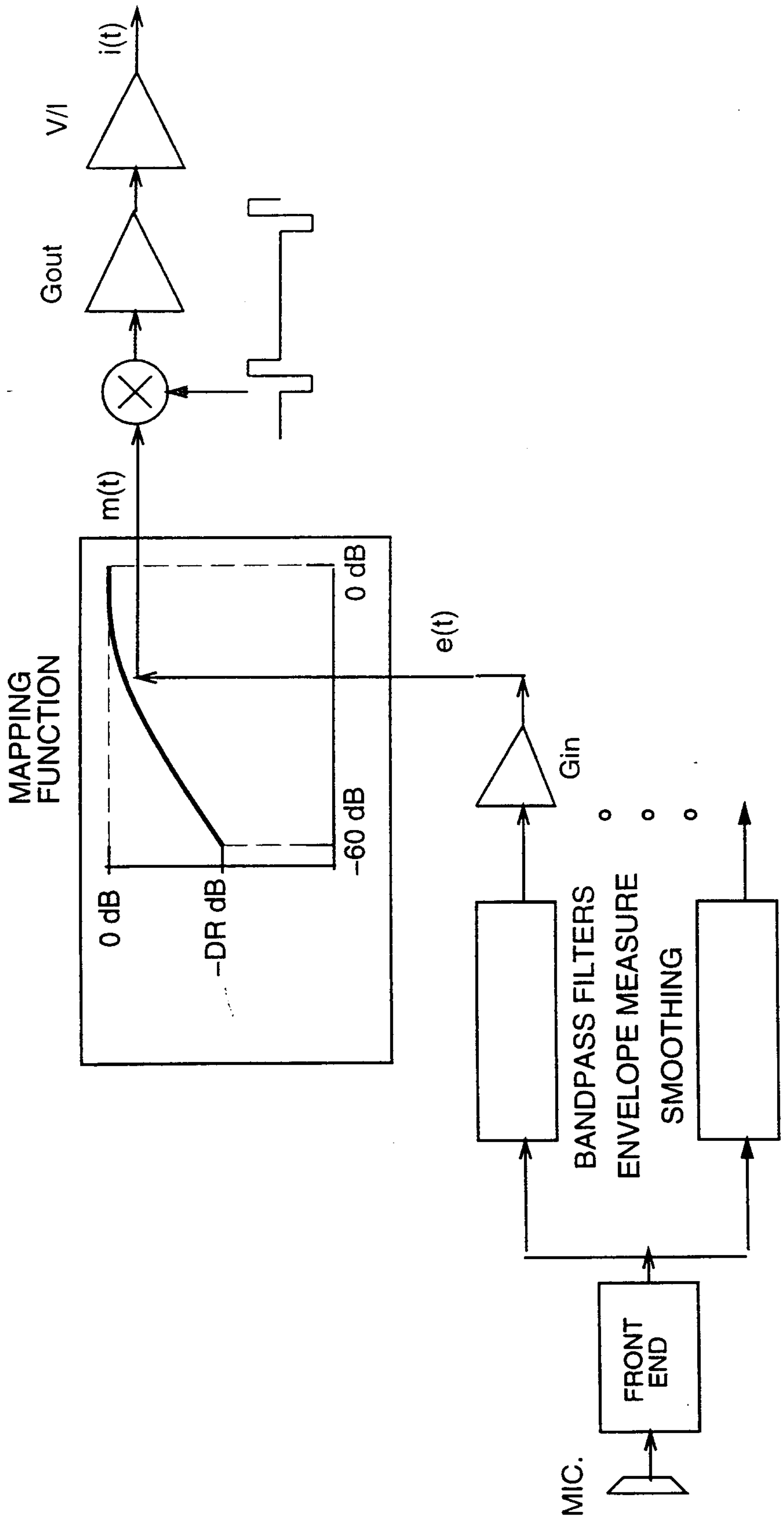

MAPPING FUNCTION
0 dB
-DR dB
-60 dB
0 dB
m(t)
e(t)
Gin
Gout
V/I
i(t)
BANDPASS FILTERS
ENVELOPE MEASURE
SMOOTHING
FRONT END
MIC.

FIG.4

FRONT END HIGH-PASS FILTER RESPONSE

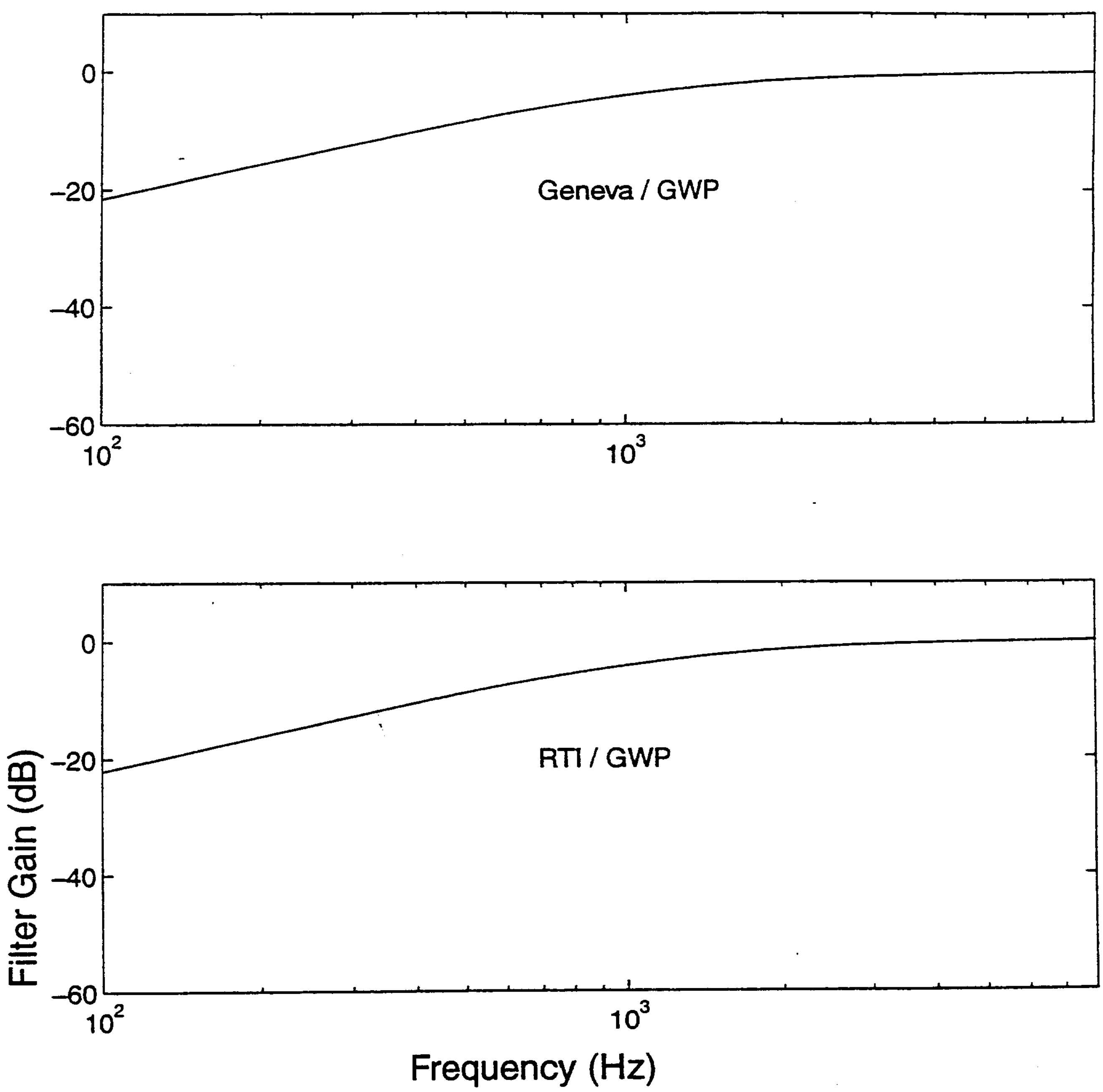


FIG. 5

16

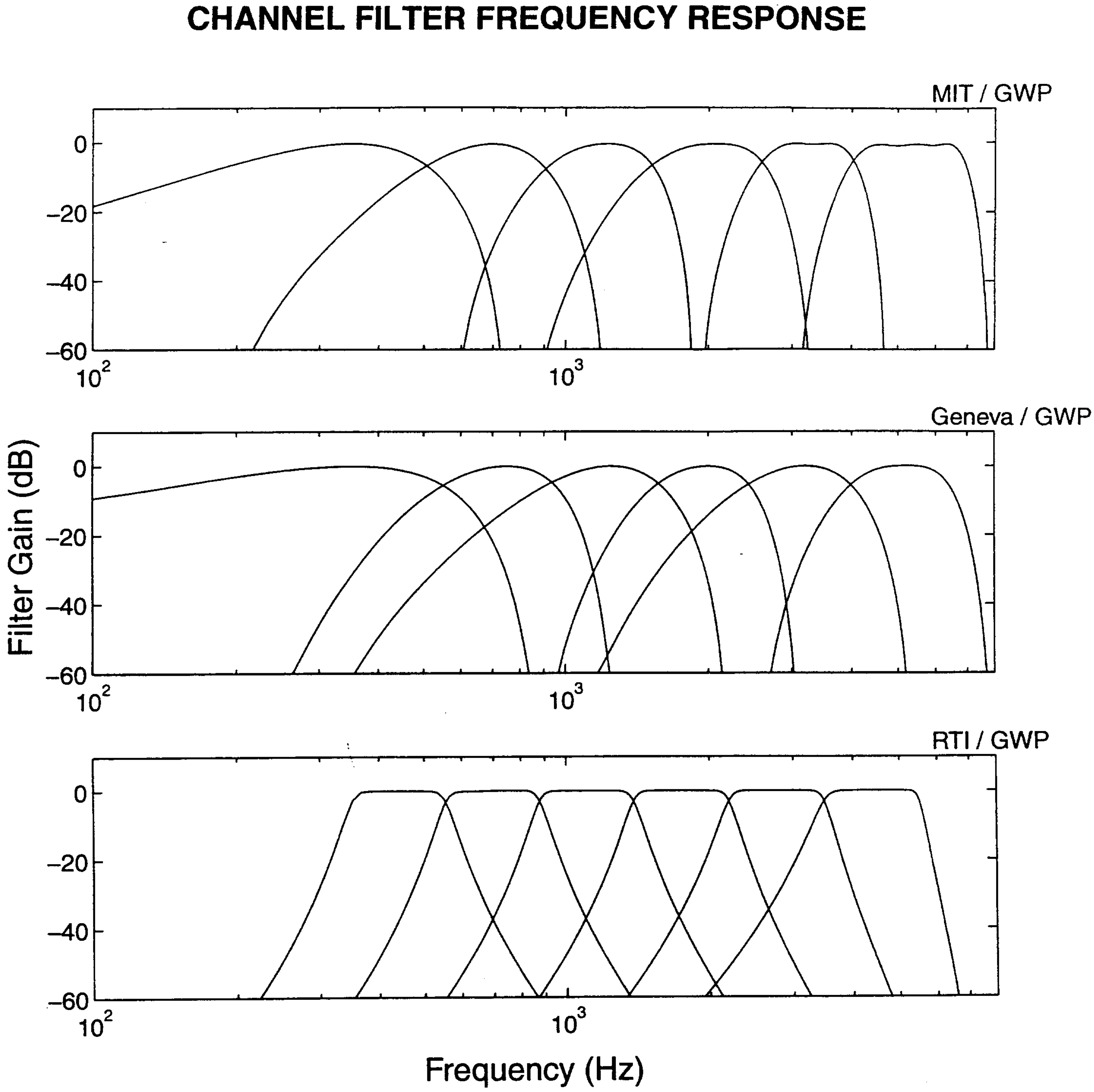

CHANNEL FILTER FREQUENCY RESPONSE
MIT / GWP
Geneva / GWP
RTI / GWP
Filter Gain (dB)
0
-20
-40
-60
10^2
10^3
Frequency (Hz)

FIG. 6

LOW-PASS FILTER FREQUENCY RESPONSE

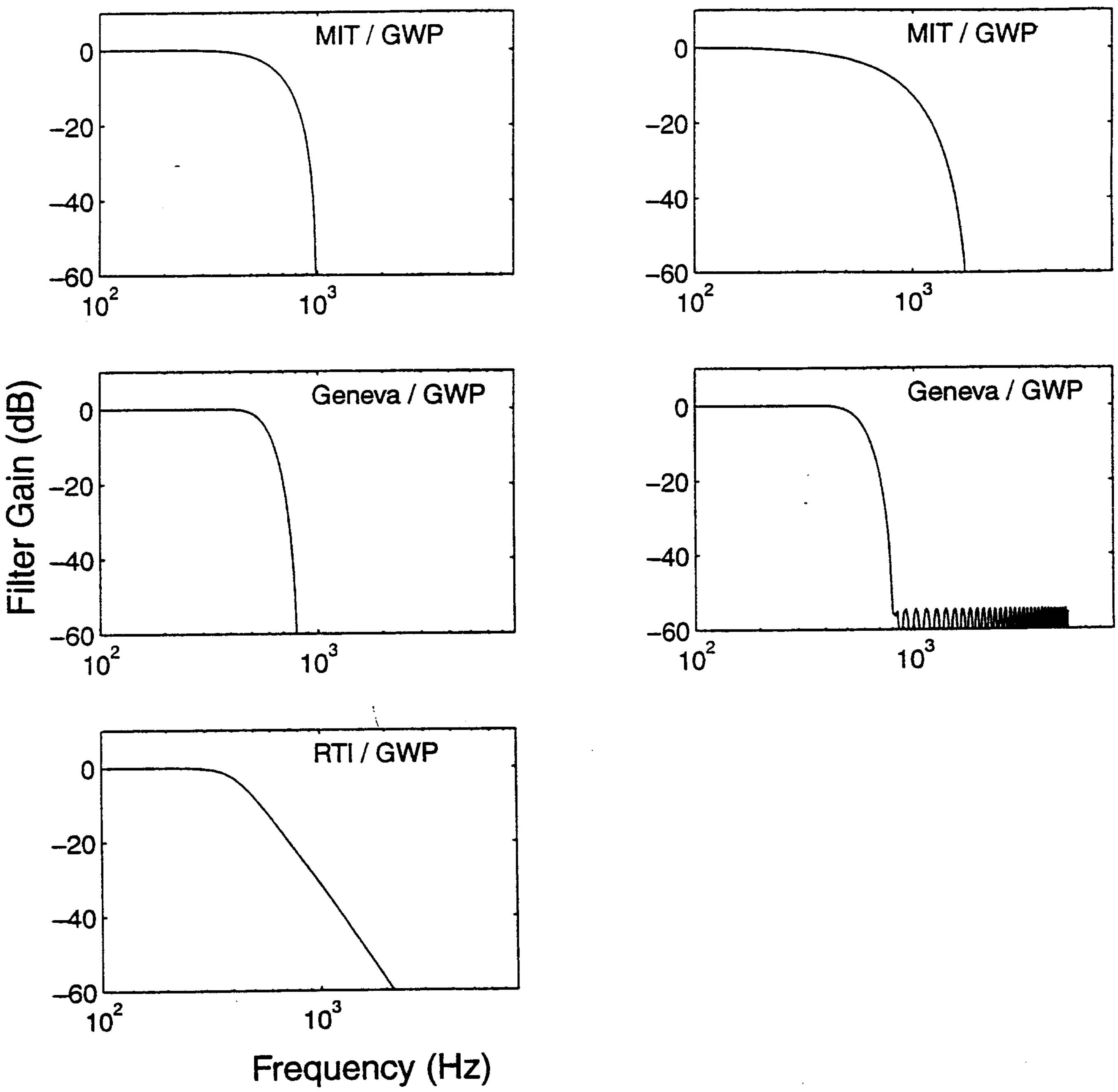


FIG. 7