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DESIGN AND IMPLEMENTATION OF AN AUDIOMETRY SYSTEM CAPABLE OF MONITORING NEURONAL ACTIVITY RELATED TO THE PATIENT'S HEARING

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ABSTRACT

The project developed a design for and use of a software controlled audiometry system. The system is able to check the neuronal changes that have relationships with the user audition. It has an audiometer to produce stimuli, an electroencephalograph to track neuronal changes, and a Labview 2012 interface that controls hardware and processes nerve signals.

This system uses an Atmel microcontroller (ATXMEGA128A4U); a device that scans the signals recorded by the electroencephalograph, synthesizes tones used by the audiometer, and is equipped with a USB port that allows for connection to a computer.

Keywords: audiometer, neuronal activity, electroencephalograph, monitoring, Labview, USB, Atmel.

1. INTRODUCTION

In today's society the generation of knowledge and technologies that contribute to the development and evolution of medical processes suggests the possibility of increasing the reliability of clinical tests such as audiometry. Audiometry is a test to evaluate the hearing of a person and involves the application of sound stimulus at different frequencies and intensities. According to Olga Gomez, speech therapist and audiologist, audiometry is the immediate objective to determine the auditory threshold [1].

In Colombia, it has developed its own technology in the design and implementation of audiometers, instrument used to perform audiometry. There is the case of the School of Engineering of Antioquia in conjunction with the University of Medellin CES virtual where an audiometer was performed in virtual instrumentation platform LabVIEW 7.1 [2]. Also, at the University of San Buenaventura who developed an audiometer screening where most of the processing is done through software [3] and Surcolombiana college in 2007 it was designed and implemented an audiometer controlled by software interface via USB [4]. A more recent work is the one from the “Universidad Autonoma del occidente” in 2011, where a type IV audiometer with PSoC technology was designed and implemented, which is different from the one used in commercial designs, says the author, the equipment is designed according to standard NTC 2884 [5].

Another type of hearing test is called auditory evoked potentials, which measures the patient’s neuro responses product to acoustic stimulus, the test is objective which reduces human errors and is very useful in the following situations: newborn screening or young children suspected of hearing loss, assessment of unconscious patients in coma, sedation or general anesthesia and when in the audiological evaluation is a pattern of asymmetrical hearing loss [6].

An audiometric examination is very susceptible to errors, because the diagnosis is made based on the information provided by the user, sometimes patients give false information to specialists to obtain favorable results, for example, will ensure continuity in the phases recruitment, procurement contracts, pensions, etc. representing a risk for companies that create jobs based on the results of such reviews.

The development of a system capable of monitoring audiometry neural activity related to the auditory organ of the patient, it is suitable for increasing the reliability of the examination making it easier to detect irregularities in them. This is achieved through the construction of audiometric competent team with latest and inexpensive technology electronic devices; and developing an electroencephalograph a channel for the neural response and using advanced filtering techniques and analysis of noise data to determine whether the auditory nerve responds to the stimulus.

2. METHODOLOGY

Figure-1 shows the modules of the deployed device.

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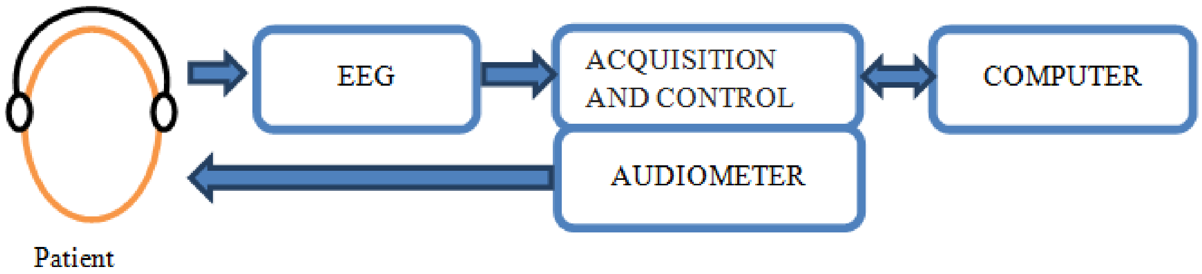

Figure-1. Block diagram of the device

Implementing an electroencephalograph that manipulates signals taken by the electrodes surface mount for subsequent digitization of data, so that an objective analysis of the brain signals is done to interpret complex related to the auditory organ of the patient.

The block diagram of the sensor system proposed is observed in Figure-2.

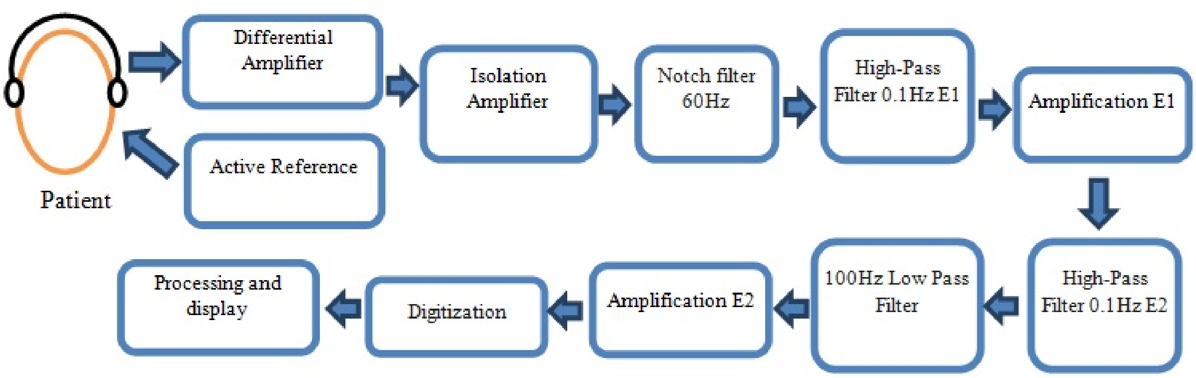

Figure-2. Block diagram of the sensor system.

To perform the acquisition of biopotential neural cup electrodes are used. The location of the electrodes will be performed according to the international standard for positioning electroencephalographic electrodes 10-20. According to the theory and various tests looking more observable results, it was decided to place the electrodes as follows: the reference electrode is placed on the nasion, while the active electrode is located in the Centrex (Cx), which corresponds the central point on the electrode system 10 to 20, finally the ground electrode is located in the front area of the skull (Fz 10-20).

2.1. Stage pre amplification and establishment of active reference

In the first stage the circuit responsible for measuring bioelectrical signals from the brain, a differential amplifier is installed in this case is the INA128 that was chosen among others for its high common mode rejection allows an obvious decrease in the noise network that is coupled to the patient, and its frequency response characteristic is suitable for the application.

The differential amplifier is configured with a gain of 50 with the aim of increasing stress levels without saturating the amplifiers noise effects. This stage is equipped with an active feedback plays an important role in reducing electrical noise of 60Hz.

2.2. Electrical insulation

In order to reduce leak currents to the patient and possible shock that this could result in an isolation amplifier that is configured with unity gain is used. There are different manufacturers of these devices, but the AD210 was chosen because of the high fidelity that exists in the output signal relative to the input, the high linearity and especially because this device provides electrical potential supply to isolated ports .

2.3. Analog filtering

The filtering system comprises:

- Butterworth low pass filter of the fourth order with a cutoff frequency of 100 Hz.
- Butterworth high-pass filter of the fourth order with a cutoff frequency of 0.1 Hz.

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- 60 Hz notch filter.

The low-pass filters and high-pass limited bandwidth of the signal is digitized and the notch filter is used to reduce electrical noise from domestic distribution network.

2.3.1. Butterworth low pass filter.

In order to eliminate the chance of acquiring noise aliasing at the time of digitization of neuronal signal, a low pass to attenuate the signal segments that have spectral components at higher frequency to 100 Hz filter is designed. This value is chosen because in considering realizing the highest density of spectral components in the range of 0 to 100 Hz was found.

2.3.2. Butterworth high-pass filter.

The system also features a filter that rejects DC signal level of neurology; it could saturate the amplifier resulting in a loss of information.

2.3.3. 60 Hz notch filter

Designing the notch filter, the IC UAF42 which is a configurable device was used. To choose the value of the resistor configuration, the manufacturer provides a manual with possible configurations depending on the type of filter to be implemented and the values of the calculated external elements compatible with DOS software that runs on Windows XP.

2.4. Amplification

Amplification was performed with an inverting amplifier mounted IC's AD 706 whose output is connected to a data acquisition board, which is responsible for performing the digitization of signals for storage, display and processing.

2.5. Audiometer

In Figure-3, a block diagram shown in the proposed hardware for implementing the audiometer.

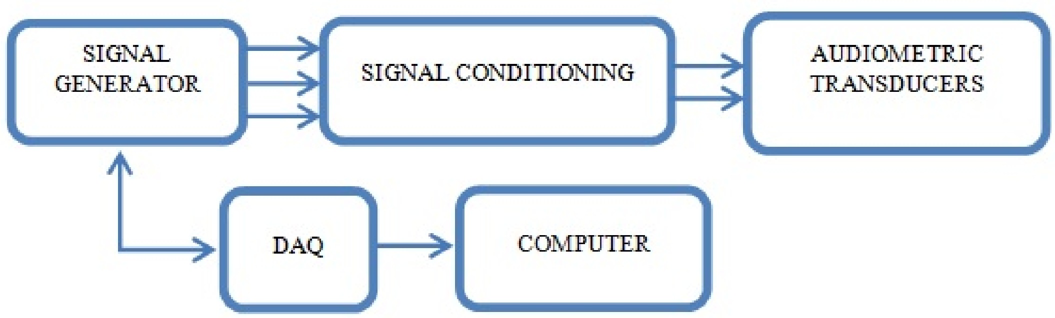

Figure-3. Block diagram of the audiometer.

2.5.1 Signal Generator

The block consists of two parts, the first is a white noise generator circuit based on avalanche effect of a zener diode, the second part is responsible for generating the tones and features a XMEGA AVR ATMEL company with reference ATXMEGA128A4U.

The circuit implemented for the white noise generator is based on the application note 3469 Building a Low-Cost White-Noise Generator Maxim Integrated; Uses the reference zener diode 1N759, resistive, capacitive elements and the integrated circuit OPA2107.

As for the tone generator, the TC DAC and AVR modules are used. The TIMER / COUNTER (TC) module is configured to generate an interrupt by overflow into a time interval equal to the sampling period of the signals; the sampling frequency is 96 KHz. Each interrupt the equivalent voltage values of the desired tone data is taken and placed in the output buffer of the DAC module to start the conversion, this interruption will run until the Stop condition is met.

The generated signals are of three types: tone pure, tone and amplitude modulated white noise.

2.5.1.1. Pure tone

Corresponds to a sinusoidal signal, the frequency of the tone to be chosen between 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 3000 Hz, 4000 Hz, 6000 Hz and 8000 Hz.

2.5.1.2. Amplitude modulated tone

The carrier frequency must be chosen between the frequencies mentioned above; the frequency of the modulating signal is between 40 and 50 Hz with a modulation equal to one.

2.5.1.3. White noise

It is characterized by having a constant power spectrum is used to perform masking.

2.5.2. Conditioning signal

The audiometer consists of two channels, right channel and left channel, the circuitry of both is quite identical. Next, the hardware for one canal. The signal conditioner comprises a step of filtering, impedance coupling stage, two stages of attenuation and multiplexing stage is described.

2.5.2.1. Passive high-pass filter of the first order

The high-pass filter is necessary to eliminate the DC component of the signal from the DAC, to this cutoff frequency is set at 6 Hz.

2.5.2.2. Passive low pass filter of the first order

Is used to eliminate harmonics of the signal generated by the sampling frequency

2.5.2.3. Coupling impedance

For this purpose a voltage follower operational with amplifiers is used using the integrated reference OPA4277P Texas Instruments, because it has four operational amplifiers circuit was implemented.

2.5.2.4. First stage of attenuation

At this point the signal conditioning, they presented values ranging from -1 to 1 Volt, being elevated to generate the intensities of the tones throughout the range required (0dBHL to 70dBHL). A first step of attenuation is necessary; it is the same integrated circuit used in the coupling impedance, giving use of the remaining two operational amplifiers, with which a circuit dimmer inverter implemented with gain factor equal to 0.1.

2.5.2.5. Multiplexer

The multiplexer selects between the passage of the signal from the microcontroller or white noise for each channel. The integrated circuit used is the Texas Instruments 74HC4053, which is digitally controlled by the microcontroller.

2.5.2.6. Second stage of attenuation

He will arrange to deliver the transducer suitable voltage levels, whose values must be calculated according to the intensity of the tone generated.

Following closely the work of Ruiz González, has arrived at the following equation for the theoretical calculations of the voltage levels for the transducers [7].

$$v = \sqrt{P_{1KHz(w)} * R_{1KHz(\Omega)}} * 10^{\frac{HTL(dB_{HTL}) + RETSPL_{(dB_{SPL})} - \Delta_{f(dB_{SPL})} - L_{P1KHz(db_{SPL})}}{20}} \qquad Ecuación 1$$

The second attenuation stage comprises an attenuator-amplifier device PGA2310 reference Texas Instruments. It is a stereo volume control with a serial three-wire control and provides a range of gain and attenuation ranging from 31.5 dB to -91.5 dB in 0.5 dB steps.

The gain for the two channels is formed by a word of 16 bits; the first 8 bits correspond to the right channel and the left channel as follows. Equation (2) shows the relationship for the gain setting is displayed, where N is the decimal equivalent of the 8 bits of the corresponding channel.

$$Gain(dB) = 31.5 - [0.5\,(255 - N)]$$

CITATION Tex01 \l 9226 |[8]} Ecuación2

2.5.3. Audiometric transducers

Circumaural earphones TDH 39 reference rate as transducers are used, they produce a sound pressure level from an electrical signal. This kind of earphones are characterized by their engagement position and completely surrounding the pinna [7] .The type headphones circumaural improve insulation of external noise, allowing the possibility of performing an audiometric test without the need for a soundproof chamber, which is why they choose to work with them.

2.6. Acquisition and control

2.6.1. Acquisition

Acquiring signals from the EEG is performed by means of the analog ports of the USB-6008 data acquisition board (DAQ). The acquisition of data is given a cup of 10000 samples per second, allowing us to acquire neuronal signal without any interference aliasing. The DAQ uses a parallel communication protocol designed for the acquisition of each of the times the examination, are synchronized with the application duly modulated tone.

2.6.2. Control device

The centerpiece of this module is ATXMEGA128A4U microcontroller, which is responsible for supervising the audiometer, synthesize auditory signals, maintain communication with LabVIEW software through the DAQ and indicate the time in which to start collecting EEG signals from the to be processed by the processing software.

The programming was done in C language under development platform integrated Atmel Studio 6. For proper performance microcontroller according to the needs, it was necessary to perform the manual configuration of each of the modules and device interrupts, since use of libraries offered by Atmel Studio difficult handling peripherals and added lines of code, making the process less efficient.

2.7. Communication

A process is vital to have a reliable communication between the control module and the software running on the computer, to solve the above we have designed a process between the microcontrollers, DAQ and user software. The process is divided into two parts, the first describes the communication between the

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DAQ and the microcontroller and the second part deals with the communication between the DAQ and software.

2.7.1. Communication between the microcontroller and the DAQ

The USB-6008 DAQ through its digital port communicates with the microcontroller. Unidirectional parallel communication with four data bits and three bits control was designed, making no errors transmitting the data to make the configuration of the device.

Because the digital ports on the DAQ are not compatible with the microcontroller, an interface is necessary to adapt the voltage levels to this buffer no inverting reference CD4050 and two transistors 2N2222 used working in the cutting region and saturation.

2.7.2. Communication between Lab VIEW and DAQ

The DAY has a USB interface, through her Lab VIEW handles the tasks to be executed by the card either acquisition or device setup. It has established five tasks which are: DATA OUT, RX, TX, START and Detain.

3. RESULTS

3.1. Printed circuit boards

Two printed circuit boards (PCB) was developed under the Multisim 11.0 and Ultiboard 11.0 National Instruments tools; the first card is for the EEG and the second corresponds to the circuitry audiometer. In Figures-4 and 5, the 3D visualization of PCB is observed, these two cards and double layer feature elements of both surface mount and insertion, all components are located on top of the cards.

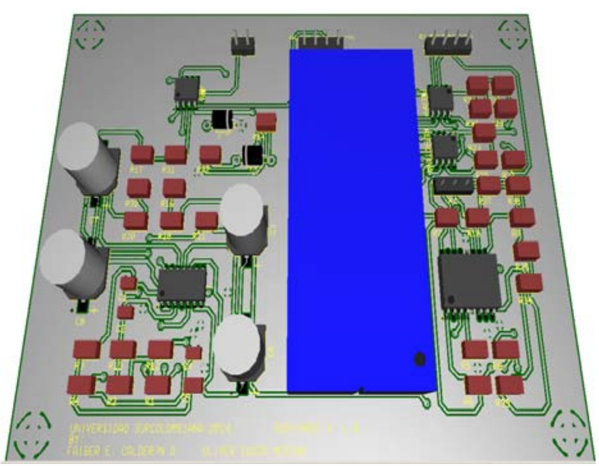

Figure-4. Top view of PCB EEG.

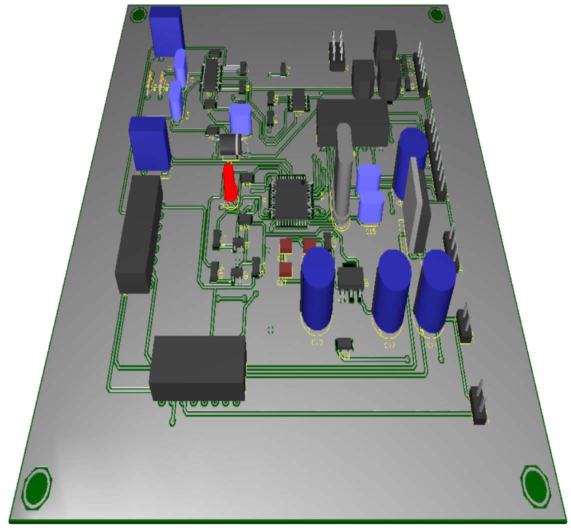

Figure-5. Top view of PCB audiometer.

3.2. Graphic interface

It is important to provide the user with a graphical interface to interact with the device in this way can make the necessary configurations, obtain the neural response and generate a report with the results. It provides the user with a control panel to manage the team and audiometer, allowing you to choose the channel to be used, the type of signal (modulated tone, pure tone and white noise) and the intensity of the tone.

Panel is also provided to control the aim of hearing (OAE) test, it is the same controls on the front panel, plus the "Stop" button to stop the test when necessary and control number to indicate the number of times.

In the results pane audiometry is allowed to enter patient data, survey and generate a document with the results. The report is made using the Toolkit: LabVIEWReport Generation for Microsoft Office, owned by National Instruments.

3.3. Audiometer

Designed and implemented equipment can generate pure tones and modulated values the following frequencies: 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 3000 Hz, 4000 Hz, 6000 Hz and 8000 Hz; the variation of intensity can be performed from 0 to 70 dB in 5 dB steps, in accordance with the requirements under the 2884 NTC [9] standard.

3.3.1. Generation of pure tones

For each of the signals generated records (Table-1) their frequency was performed using an oscilloscope GW Instek GDS-2062 and the percentage error was calculated, obtaining these values are within the range of accuracy required by the NTC 2884 (± 2% for audiometers type 3 and 4). The measurements were taken at the outlet of the first stage of attenuation.

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Table-1. Record of frequencies of sinusoidal signals.

Desired frequency of the sinusoidal signal (Hz)	**Measured frequency of the sinusoidal signal** (Hz)	**Percentage Error** (%)
250	248.5	-0,60
500	496	-0,80
1000	996	-0,40
2000	1988	-0,60
3000	2976	-0,80
4000	3975	-0,62
6000	5967	-0,55
8000	7937	-0,79

3.3.2. Generation of modulated tones

As for modulated signals similar to that indicated in the previous paragraph registration was also performed using the same equipment, the result being that the recorded values are within the range of accuracy required by the NTC 2884 standard (± 3% of the set value).

Table-2. Recording frequency modulated signals.

Desired frequency of the sinusoidal signal (Hz)	**Measured frequency of the sinusoidal signal** (Hz)	**Percentage Error** (%)
250	247.1	-1,16
500	495	-1,00
1000	1000	0,00
2000	1987	-0,65
3000	2980	-0,67
4000	3977	-0,57
6000	5963	-0,55
8000	7957	-0,62

3.4. Hearing test target

Examination of one ear at a specific frequency, lasts between 2 and 11 minutes depending on the intensity of the applied tone, and that the lower intensity, the greater must be the number of times.

The stimulus applied in testing patients, has a modulation frequency range between 35 Hz to 50 Hz and a modulation depth of 100%. The results for a modulation frequency of 50 Hz were analyzed. The procedure consists of applying 520 cycles of the signal for each time, which leads to the test has duration of 10.4 seconds per day.

The device software is responsible for performing the setup to start the test. Then he handles the registration and neural signal processing to detect patterns to infer if the patient actually experiences a sensation at the neuronal level product of the excitation of the auditory system.

The software processing of the signal is basically the coherent averaging times acquired and the development of a spectral analysis of the result. It is expected that with a stimulation frequency of 50 Hz, an overlap of auditory middle latency evoked potentials originating, generating a frequency response equal to 1/50 seconds [10].

3.4.1. ANALYSIS OF TEST RESULTS

The result of one test, in which the response was evaluated for 10 times, is observed in Figure-6.

Figure-6 shows that a large portion of the signal has a spectrum at frequencies of between 10 and 30 Hz as this range of frequencies, the neuronal events related to cognitive, motor processes and others associated with the normal operation occur brain. Neuronal activity peak at 50 Hz which shows clearly that there is a patient in response to neuronal level of the acoustic excitation product is observed.

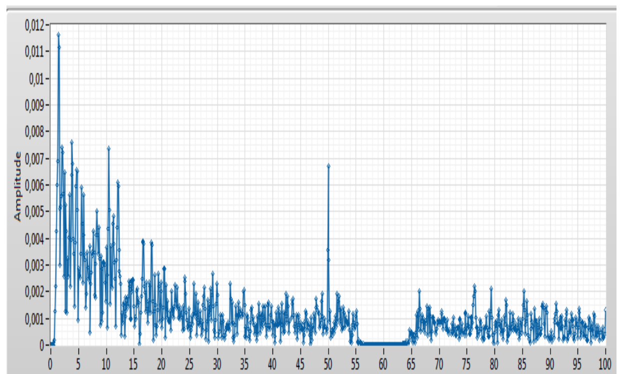

Figure-6. Spectral analysis of the output of the coherent averaging 10 times.

4. CONCLUSIONS

- It was established that for the acquisition of bio potential generated by the auditory nerve the most appropriate electrode location corresponds to the following: the reference electrode is placed on the nasion, the active electrode is located in the Centrex (Cx), and the electrode land is located in the front area of the skull (Fz 10-20).
- We have designed and implemented a computer-controlled audiometer that meets the functionality of audiometric equipment type 4, has been equipped with software for control and presentation of results, improving the efficiency of the testing process.
- An important design element of an audiometric device is the variation of signal strength as it determines the quality of the review, taking into account the characteristics of the signal to be delivered to the transducer it was found that the integrated circuit as the last step PGA2310 control voltage levels responsive to the desired shape, allowing the

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generation of various intensities at specific frequencies.

- The design process of the control software of the device is very delicate, the high degree of accuracy required when searching synchronization between the application of the acoustic stimulus to the patient and the start of data acquisition for each exam times. A slight variation in this item, causes a shift of the signal for the coherent averaging process, will result in a decreased level of the signal SNR.
- In tests, it was possible to determine if the patient is lying when intensity difference between the tone is applied and the minimum threshold of hearing at that frequency, exceeds 30 db. Given that the test lasts too high compared to the response of the patient to apply the stimulus.
- For the implementation of a computer monitor neuronal activity of the patient during an examination of pure-tone audio metry, very robust filtering methods are needed, since the characteristics of pitch modulation in these tests, coincides with the normal neuronal activity human brain (modulation with modulation frequency of 5 Hz to 20 Hz).

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