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Popularly used Signal Processing and Analysis Techniques for Biomedical Signals

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***Abstract*— Human life is precious. It is often encountered by many diseases posing the physicians' new challenges every time in their treatment, such for example COVID-19. Unless the root cause of the disease is known clearly, it is difficult to plan for its treatment. A thorough understanding is possible only when the biological signals are observed and analyzed clearly. For this purpose, technology plays an important role in the handling of the signals. Biomedical signal acquisition, processing, and analysis are the prime stages in the diagnosis of any disease. In this paper, a few advanced signal processing methods have been discussed. Each method has merits and demits but, a particular method of interest depends on the computational time, algorithm implementation, and the device cost. The methods explained here are not just limited rather several algorithms are available in the literature. However, the methods explained in this paper gives a basic understanding of each technique with suitable examples.**

Keywords— biomedical signals; signal processing; analysis; artificial neural networks; fuzzy logic; diagnosis of disease

I. Introduction

All the living things in the world deliver biomedical signals [1]. The human body continuously produces signals of different types which may be electrical, chemical, or acoustic which conveys information about a person's biological system. But the signals have very low amplitude and therefore; to enhance their signal-to-noise ratio they need to be processed before using for the diagnosis of patients. A specific signal has to be extracted to get meaningful information that may indicate a disease from which the patient is suffering; thus helping the physicians to plan for the course of treatment. Therefore, the processing of biomedical signals is a crucial part to get the clinical information from the hidden signals, otherwise, the useful signals may get corrupted by the noise and artifacts. Biomedical signal processing has become an important step to extract clinical, biomechanical, or pharmaceutical relevant information for the improved medical diagnosis.

Biomedical signal processing is a huge area comprising of subjects like anatomy, statistics, calculus, computer, and circuit design [2]. When the computer was first used in the field of medical signal analysis, the whole world wanted to make everything automated. Since the physicians are responsible for the results, it is therefore desired that any new method should be used by the physician independently for diagnosing the problems and finding the solutions. Therefore, simple, accurate, and fast-acting signal processing techniques are required to access the medical signals and processing them for conclusive analysis. Some of the advanced signal processing techniques are explained in the below sections.

II. Methods

Biomedical signal processing incorporates the manipulation or transformation of the signals acquired from the human body to improve our understanding of information. It is mostly aimed to reduce the complexity of the signal into meaningful features that help in the medical diagnosis. The signal is trivial and exhibits different properties. There are many methods for processing the signal, but few popularly used methods are given below.

1. Filter processing
2. Statistical signal processing
3. Frequency domain analysis
4. Time-frequency analysis
5. Fuzzy logic
6. Artificial neural networks
7. Genomic methods

1. Filter processing

Filtering is a procedure in signal processing to remove the undesired components from the source signal. It is one of the primary classifications of signal processing. A desired frequency of the signal is selected from a signal range by using different types of filters. As per the output signal frequency, the filter can be four types.

i. High pass filter: This filter blocks the unidirectional and low frequencies and passes the frequency that is greater than the cutoff frequency.
ii. Low pass filter: Contrary to the HF filter, the LP filter eliminates the high frequencies of the signals and passes only the frequency that is lower than the cut off frequency.
iii. Bandpass filter: It passes a band of frequency and cuts the frequencies that are more or lower than the cut-off frequency.
iv. Bandstop filter: It stops a certain banded frequency and allows all the frequencies to pass through the filter.

Broadly, the filters can be categorized into linear/non-linear filters and analog /digital filters.

Linear / non-lnear filters

The filtering technique that gives the output signal from a time-varying input signal subject to the constraint of linearity is known as a linear filter. The linear filter can be analyzed using their transfer function in the frequency domain and also their impulse response in the time domain. To implement the linear filter in the time domain is unavoidably causal, needing a transfer function. An analog filtering element such as a resistor, capacitor, and inductor falls in this category. Also, the digital filter is made with a linear element included. Since the time-invariant filter is characterized by the response to the sinusoidal frequency, it is called a frequency filter. If the output of a filter is not a linear function of the input then it is known as a nonlinear filter. For any two input variables r and s if the input combination is αr + βs then the output αR + βS is not linear to input. The nonlinear filter can be continuous or discrete.

Nonlinear filters are used in many applications but removing noise is the widest field [3-6]. But it is harder to design the nonlinear filer compared to the linear filter because the mathematical tools used for signal processing is not used on it [7]. That is why many times it is needed to design another linear filter to remove the noise coming out with the output of a nonlinear filter.

Analog / digital filters

The analog filter is an electronic circuit made with a resistor, capacitor, inductor and it is the basic block of signal processing [8]. It can be classified as the active filter (that uses a capacitor, inductor, and sometimes resistor) and passive filter (uses active element like an amplifier, Op-amp, etc.)

Digital filter is the signal processing technique that does the mathematical functions on discrete or sampled data. This filter is usable in the same way as an analog filter and the signal can be modified by it.

Filters are widely used in radio, television, electronics and telecommunication, image processing, audio recording, computer graphics, radar, control systems, music synthesis.

2. Statistical signal processing

In this signal processing technique, a signal is processed as stochastic and the statistical properties of the signal are used for processing [9]. This signal processing technique is widely used in communication theory, seismology, medical diagnosis, array processing, and climate modeling. This technique is used for pattern recognition also.

Statistical signal processing is particularly an important area in the fields of medical diagnosis because the statistical data of a particular group of patients or diseases will give very important conclusions after processing them. It is an approach to utilize the statistical properties of the signal for the processing task. As the signals are random, the statistics can play an important role in processing them. Statistics can help to know the signal characteristics, parameters, and analyzing procedures.

The multiple sinusoids model is expressed as

$$y(t) = \sum_{k=1}^{M}\{A_k\,cos(\omega_k t) + B_k sin(\omega_k t)\} + n(t);\ t = 1, \dots\dots, n \quad (1)$$

Here, $\omega_k's$ represent the real radian frequencies of the signals, A_k'and B_k'represent the amplitudes of the signal, n(t)'s are the error random variables with mean zero and finite variance. Most of the periodic signals can be expressed by Equation (1). With the proper selection of M with the frequencies and amplitudes.

Almost every time the unknown parameters A_k', B_k', ω_k and M are unknown. Using one appropriate statistical method, first M is determined and then all other parameters are determined.

3. Frequency domain analysis

In frequency domain analysis, the signal is described and analyzed as a function of frequency rather than time. In physics, control systems, engineering, and statistics, the frequency domain is a term used to describe the analysis of mathematical functions or signals concerning frequency, rather than time [10].

In frequency domain analysis, the input signal is of sinusoidal form which changed in magnitude and phase at the output signal. The analysis is done based on the phase and magnitude change for a certain domain of frequencies. The time-domain signal (Figure 1) is converted to a frequency domain for the analysis. For example, a voice waveform in the time and frequency domain is shown below (Figure 2).

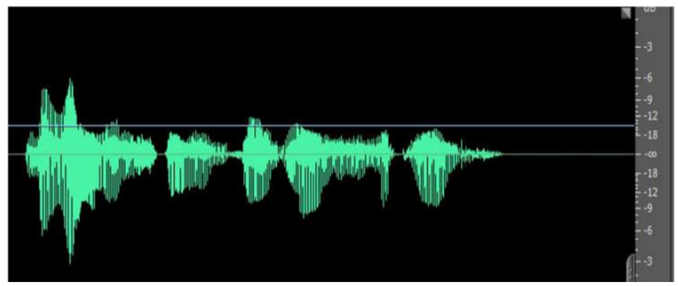

Figure 1. Voice Waveform in Time Domain

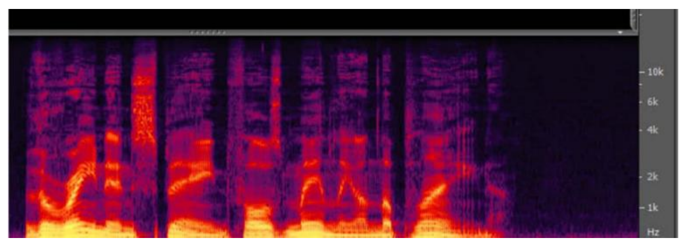

Figure 2. Voice Waveform in Frequency Domain

The main advantages of frequency domain signal analysis are-

- It ensures better signal identity and characteristics
- Signal adding and subtracting from the original signal is easier
- Easier to determine the transfer function
- Designing and adjusting the system becomes easy
- For nonlinear system also frequency domain analysis is easier

4. Time-frequency analysis

Time-frequency signal processing is the method or technique used to analyze the signal concerning both time and frequency at a time. This method is an alternative to the frequency and time-domain analysis (FTA). The FTA is very helpful for non-stationary signal processing. The dependency of the signal on the time spectrum is called non-stationarity.

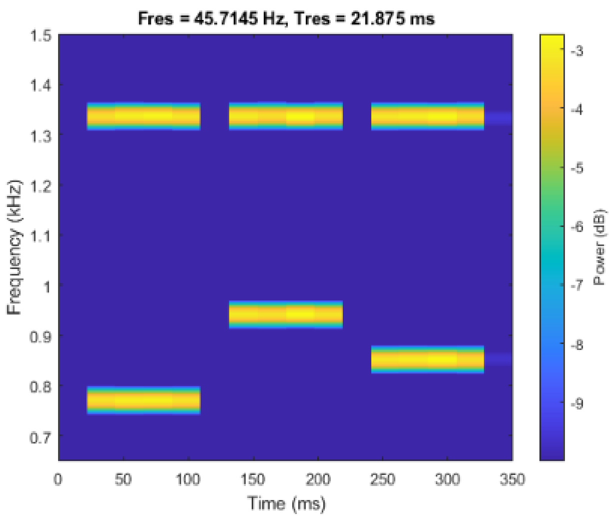

Figure 3. Time-Frequency Domain Signal

Figure 3 shows the Time-Frequency domain signal where the spectrogram indicates the power level of frequency. Blue color indicates lower power and yellow color indicates higher power. The horizontal yellow line indicates the signal with that frequency and at that time. The graph clearly shows that the lowest frequency comes first and the highest frequency comes middle. It means that the FTA will enable us to know about the priority of a frequency range from the data.

Short time Fourier frequency transform is mostly used as time-frequency analysis as it applies the Fourier transform for a specific portion of the signal. Also, the wavelet transforms, Wigner-Ville distribution, Gabor transform, Hilbert transform is used in the medical data processing and analysis. FTA is used for many applications of signal processing like denoising, detrending; also it can be used for recognition of particular systems, signal detection, and medical image segmentation. FTA is widely used in the diagnosis of heart disease [11].

5. Fuzzy logic

Fuzzy logic was first invented by Dr. Lotif Zadeh in the 1960s. It's an approach that uses a "degree of truth" for the analysis rather than the conventional true and false as in the Boolean logic. So fuzzy logic is the usable approach where the binary Boolean logic is a special case of it. Fuzzy logic is like the human brain; if the logic exceeds a certain threshold, it goes into higher truth. Hence, artificial intelligence (AI) also it is being used.

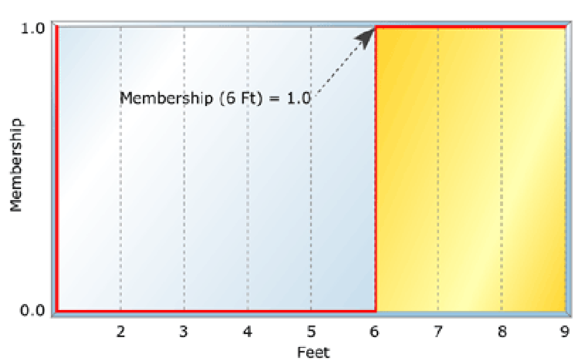

Figure 4a. Boolean variable for Tall

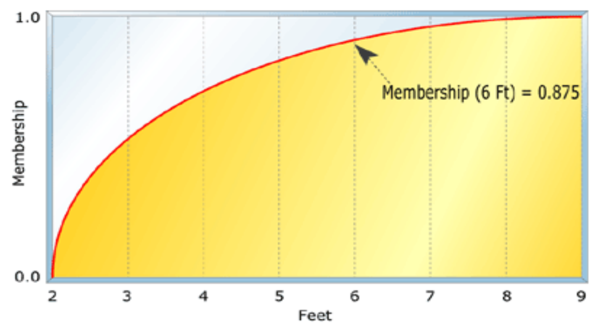

Figure 4b. Fuzzy variable for Tall

In Figure 4a, the Boolean logic is showed where tall is false for less than six feet and true for greater than six feet. But in Figure 4b, the tall is neither false nor true, it offers a varying degree of membership.

Fuzzy logic Architecture

The architectural view of fuzzy logic is presented in Figure 5.

Rule Base: The rules are contained here and the decision expert can control the decision-making process using the 'if-then' condition. The recent version of fuzzy logic has reduced many conditions for the operation.

Fuzzification: It helps the expert to convert the crisp number to fuzzy sets. Crisp inputs like pressure, room temperature are passed into the control system for text processing tasks.

Inference Engine: The rules need to implement the operation is determined by this. It determines the degree of match and based on the percentage of matched between inputs and rules, it selects the needed rules.

Defuzzification: In this step, the crisp value is found from the conversion of fuzzy sets.

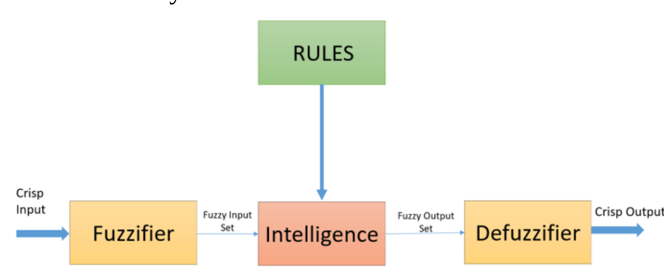

Figure 5. Fuzzy logic architecture

Advantages of Fuzzy Logic System:

- The fuzzy logic system is like human logic; very easy to understand.
- The sensor used for this system is inexpensive.
- The sensor used for this logic is less complex.
- Easy solution for complex issues.
- Precision inputs are not necessary

Disadvantages of Fuzzy Logic Systems:

- The accuracy of this system is sometimes not reliable.

- The result is based on assumption, so may not be widely accepted.
- The pattern recognition capability of the fuzzy logic system is not as like the neural network or machine learning.
- Extensive test equipment is needed for the validation of the system.

6. Fuzzy logic

Artificial Neural Network (ANN) is a computing system that was inspired by the biological neural network [12]. ANN is widely used for the prediction of protein secondary structure, voice recognition, cancer classification, gene prediction, etc. [13]. ANN is the process based on connected units or nodes known as an artificial neuron. Each of the connections can transmit the signal to the other neurons. The signal received by this transmission processes and the computation is done using some non-linear function of the inputs. The procedure is depicted in Figure 6. The connection is known as edges. Generally, the edges and neurons have weights that adjust as learning processes. The signal connection (increase or decrease) depends on the weight. Sometimes the neurons have a threshold that determines when the signal will be sent. Neurons may have some layers and different transformation is possible within different layers.

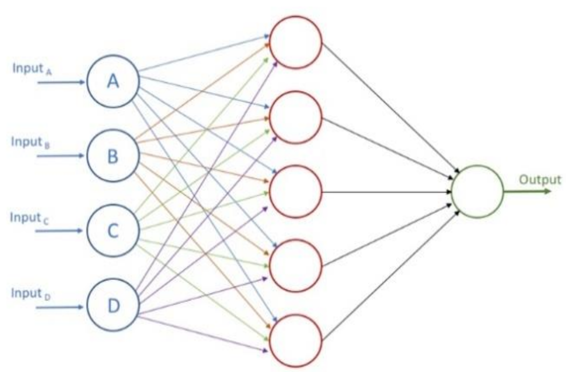

Figure 6. Artificial Neural Network

For neural network-based signal processing, training is done for the input, layer, operation, and output. Here weight plays a vital role to determine the output for the input. This training is given using some examples and based on the processing results and it gives the output in operation. After training, testing is done and experts have to know the performance of the network (accuracy, sensitivity, selectivity, etc.).

The general guidelines for designing an ANN is-

a. Perform knowledge of the required information needed to solve the problem and identifying the decision factors.
b. Remove the noise by using the Pearson correlation matrix or chi-square test to keep the correlated signal only.
c. Select the ANN learning method based on the features and decision making process.
d. Determine the training data set volume.
e. Determine the hidden layers.
f. Quantity of hidden nodes in the last hidden layers equal to the decision factors used by domain experts to solve the problem.

Advantages of Neural Network over other Processing techniques:

a. It can solve complex problems with nonlinear model learning.
b. ANN can be used as a general technique after learning from the explanations given in the training section.
c. ANN does not impose any restriction on the input variables.

7. Genetic Signal processing methods

Processing and analysis of the genomic data using digital signal processing (DSP), is referred to as Genomic Signal Processing (GSP) [14]. In GSP, generally, the signal is taken from the production of mRNA, and the protein carried out within the cell. So the DNA is an inevitable part of GSP, however, the GSP is used to understand the structure of DNA. Figure 7 represents the steps of GSP.

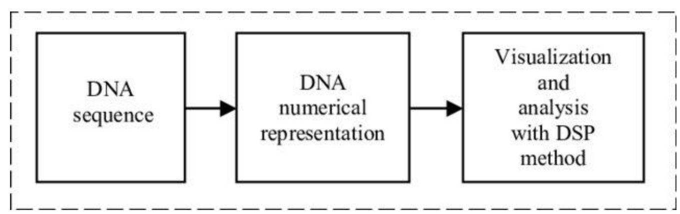

Figure 7. Genetic Signal Processing

III. SUMMARY

Human physiological processes generate signals from each event of body activity. This signal is an action potential which is in the form of electrical voltage generated from each cell. When this signal is analyzed, the on-going cell activity can be known. If abnormal activity is found then that will be treated by suitable medication to bring down to the normal state. But, the biomedical signals are small in amplitude (few micro-volts) and low frequency (few tens of Hz). Therefore, more efficient algorithms are required compared with those of non-biomedical signals. To do so, a few advanced signal processing algorithms are explained with examples. As digital devices such as computers, digitizing circuits, display devices, data storage, and transmission technology are progressing rapidly; signal processing and analysis techniques are also emerging exponentially thereby attracting many researchers in this area. These techniques are not only cost-effective, but their precision and accuracy have been much higher than the traditional methods. This has a great impact on the diagnosis and treatment of diseases making health care much reliable, faster, and low cost.

IV. Conclusion

Few commonly used advanced level biomedical signal processing techniques have been discussed in this paper. Each method has its own merits. The selection of a particular signal processing technique depends on the type of the signal, the data size, its complexity, and the domain of display for the further decision that could help in understanding the disease of the patient at the earliest. These are not only the methods explained here, but they help in understanding the structure of any algorithm required in the processing and analysis of clinical signals. It is like open-ended research wherein newer algorithms are being added often by the researchers.

Acknowledgment

For a better understanding of the techniques, the author has taken a few figures from the internet which are available as open articles.

CONFLICT OF INTEREST

The authors declare that they have no conflict of interest.

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