World Academy of Science, Engineering and Technology
International Journal of Electrical and Computer Engineering
Vol:15, No:3, 2021

A Review in Advanced Digital Signal Processing Systems

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***Abstract*—**Digital Signal Processing (DSP) is the use of digital processing systems by computers in order to perform a variety of signal processing operations. It is the mathematical manipulation of a digital signal's numerical values in order to increase quality as well as effects of signals. DSP can include linear or nonlinear operators in order to process and analyze the input signals. The nonlinear DSP processing is closely related to nonlinear system detection and can be implemented in time, frequency and space-time domains. Applications of the DSP can be presented as control systems, digital image processing, biomedical engineering, speech recognition systems, industrial engineering, health care systems, radar signal processing and telecommunication systems. In this study, advanced methods and different applications of DSP are reviewed in order to move forward the interesting research filed.

***Keywords*—**Digital signal processing, advanced telecommunication, nonlinear signal processing, speech recognition systems.

I. Introduction

ANY variable in time or space that can be measured is called a signal. For example, velocity is a quantity that varies per unit time and its value can be measured. The amount of speed at specific intervals can be measured and recorded as signals. The set of numbers that result from recording speed at different time intervals together form a signal. Quantities such as acceleration, temperature, humidity are also variable per unit time and can also be measured. Therefore, by sampling these quantities in different time units, a signal can be formed. Fig. 1 (a) shows the signal in a discrete form and Fig. 1 (b) shows the signal continuously form of the signals [1]. The horizontal time axis and the vertical axis also display the amount of signal intensity

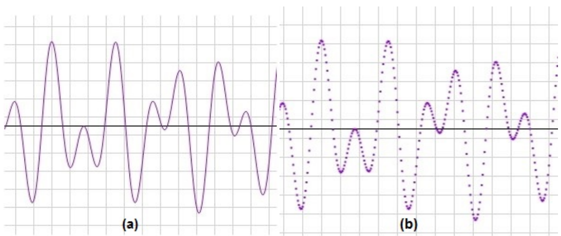

Fig. 1 The signal in a discrete form and the continuously form signal [1]

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An advanced method is presented [2] in order to increase security of CPU manufacture by preventing the capturing data on computer or smartphones by attackers. A review in recent development of network threat and security measures is presented [3] to classify the presented research works and suggest some future research trends. Advanced image processing systems is reviewed [4] to introduce new techniques in the image processing systems. Application of the Secure Socket Layer in the Network and Web Security is investigated [5] to increase the security measures in the web of data. Signals processed in this way are a sequence of numbers that represent examples of a continuous variable in areas such as time, space, or frequency. Signal processing is a science that analyzes signals. It is an important part of the process of searching for life in a place beyond the earth. DSP to improve an electrocardiogram signal display is shown in Fig. 2 [6].

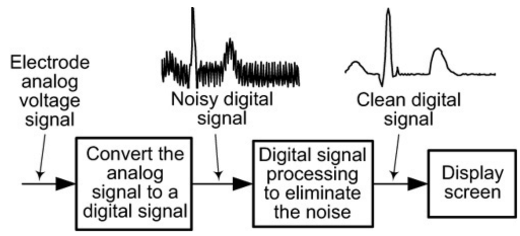

Fig. 2 DSP to improve an electrocardiogram signal display [6]

Amplitude, frequency and time are the most important variables of the signals which are analyzed. Domain of frequency is defined to provide the range of sound frequency as low or high frequency system. There are a number of high-frequency conversions in the frequency domain. Capstram, for example, takes a signal by the Fourier transform to the frequency domain, logarithms it, and then takes another Fourier transform. This action determines the harmonic structure of the initial spectrum. It has applications in the image processing for digital cameras, video processing to interpret moving images, wireless communication, array processing from an array of sensors, radar signal processing for further detection and financial signal processing for financial data using signal processing techniques specifically to predict goals and automatic driving. A very important part of communications between satellites, video, radio and wireless systems is done by signal processing as application of the signal analysis in the telecommunication systems and networks. Thus, the process can provide efficient data processing and transmission [7]. When you are on the go and need the Internet or GPS information to navigate, signal processing is a technology that converts and analyzes signals behind the scenes to help us communicate and learn technologies such as cell phones, WiFi, GPS devices, radar, sonar, radio and cloud computing infrastructure, etc. that have become part of everyday life [8], [9].

II. Different Categories for the Signal Processing Methods

A. Analog Signal Processing

Analog signal processing is for signals that have not been digitized, such as radios, telephones, radars, and older television systems. This includes nonlinear and linear electronic circuits. Linear circuits such as passive filter filters, collector filters, integral catchers, and delay lines. Nonlinear circuits include voltage controlled oscillators and phase locked loops [10].

B. Digital Signal Processing (DSP)

It is the use of digital processing systems by computers or more specialized digital signal processors to perform a variety of signal processing operations. DSP can include linear or nonlinear operators. The nonlinear DSP processing is closely related to nonlinear system detection and can be implemented in time, frequency and space-time domains.

C. Continuous Signal Processing in Time

The processing of continuous signals in time is for signals that change with continuous amplitude changes (without considering some intermittent points). Signal processing methods include time amplitude, frequency amplitude and mixed frequency amplitude. This technology mainly discusses modeling of continuous linear systems with constant time, aggregation of zero system state response, system function tuning and continuous filtering in time of definite signals [11].

D. Processing Discrete Signals in Time

Discrete signal processing is for signals that are sampled only at discrete points in time, and are quantized in time but not in quantities. Continuous analog signal processing is a technology based on electronic devices based on sampling and storage circuits, multiplexers and analog delay lines. This technology is a former example of DSP (referred to in the next section), and is still used in advanced GHz signal processing. The concept of processing discrete signals over time also refers to concepts and principles that provide a mathematical basis for processing digital signals, regardless of the quantization error [12].

E. Nonlinear Signal Processing

Nonlinear signal processing involves the analysis and processing of signals generated by nonlinear systems, which can be in the time domain or frequency. Nonlinear systems can produce complex behaviors such as branching, chaos theory, and harmonics that cannot be investigated by linear methods [13].

III. Signal Modeling Methodology

Signals are usually transmitted from the time or place domain to the frequency domain by converting Fourier. The Fourier transform converts temporal or spatial information into amplitude and phase components of each frequency. In some applications, how the phase changes relative to the frequency can be significant. When the phase is insignificant, the Fourier transform is often converted to the power spectrum, which is the second largest power of each frequency component [14].

A. Signal Population

In the population stage, a sample of the signal was recorded in the form of voltage. At this point, the sampled voltage must be converted to digital (binary number). For this purpose, analog-to-digital converters are used. For example, an 8-bit analog-to-digital converter produces a number between 0 and 255 for its input. We assume the converter input is in the range of 0 to 5 volts. This means that for the 0 V input, the output of the binary number converter will be 0 and for the 5 V input, the output of the binary number converter will be 255. Obviously, increasing the number of converter bits will increase the multiplication accuracy [15].

B. Signal Sampling

Sampling and converting analog-to-digital signals for signal acquisition and reconstruction, which includes the physical measurement of signals, their storage or transmission as digital signals, and the reconstruction of the original or approximate signal in subsequent possible applications. Sampling means reading the value of the input signal at specified intervals and moving it to the next step for a while. In this study, sampling is done in two stages, discretization and quantification [16]. Discretization means that the signal is divided into equal time intervals and each interval is represented by a measured amplitude. Quantification means that each measurement range is approximated by a value of a finite set. Rounding real numbers to integers is an example of this. Once the input signal is ready and filtered, the analog signal digitization begins. For example, when the 44,000 samples of the input signal per second need to be taken, the value of the analog signal at intervals of 0.00002 seconds should be saved and move it to the next step. For example, when a person speaks into a microphone, assuming a sampling rate of 44,000 samples per second, the digitizing system reads the output voltage of the microphone, which is amplified and filtered, every 0.00002 seconds, and multiplies its value by a multiplier [17].

IV. Signal Processing Methodology

Signal processing plays a key role in capturing and processing sound from the environment in order to enhance and amplify what the user uses the device. In this process, the sound is converted from analogue to digital with the least possible delay and then back to analogue and is directed to the ear.

When the signal is processed, first in different areas including time, frequency, etc. are checked to perform operations such as noise cancellation or extraction of important and required features in the signal or packaging of their information. After the pre-processing operation on the signal processing operations, the trick is ready for its information to be entered into classification and detection algorithms for classification and diagnosis. As a result, any noises in the sound signals are eliminated to increase the quality of communication systems [18].

The core of auditory rehabilitation technology consists of four coordinated parts: microphone, processor, receiver and power source. Theoretical analyzes and inferences in DSP are usually performed on time-discrete signal models without amplitude error (quantification error) "generated" during the abstract sampling process. Numerical methods require quantified signals, such as signals generated by an ADC. The processed result may be a frequency spectrum or a set of statistical indicators. But, it is another quantified signal that is converted to analogue by a Digital-to-Analog converter (DAC).

To digitally analyze and manipulate an analogue (continuous) signal, it must be digitized by an analogue-to-digital converter (ADC). Sampling is usually done in two stages, discretization and quantification. Discretization means that the signal is divided into equal time intervals and each interval is represented by a measured amplitude. Quantification means that each measurement range is approximated by a value of a finite set. Rounding real numbers to integers is an example of this. The most common purpose for analyzing signals in the frequency domain is to analyze the signal properties. By studying the frequency spectrum, the engineer can determine which frequencies are present in the input signal and which are absent. Frequency domain analysis is also called spectral analysis [19].

Common math operations include fixed point representation, floating point, real or complex number of numbers, multiplication and addition will be applied in the parameters of signals to be analyzed. Some other common used operations are supported by the buffer circle and look-up tables by the hardware. Examples of these algorithms are Fast Fourier Transform (FFT), FIR filters, IIR filters, and adaptive filters [19].

Prepare the input signal: We suppose the output of the sensor that generates the signal is voltage. For example, when a person speaks into a microphone, the microphone output changes voltage over a period of time, corresponding to the sound that is produced. For example, the output of a sensor used to record an ECG is changing at very small voltage intervals (millivolts). In the next steps of the digitization process, when we use an ADC, we practically need voltages in the range of 5 to 0 volts. But as we have seen, the output of some sensors (such as ECG sensors) is in the millivolt range. So we need a way to amplify the output of the sensors and prepare them for input into ADCs. This step of digitizing the input is called input signal preparation, in which we use amplifiers to increase/decrease the voltage gain. It should be noted that today there is a built-in signal amplifier system on sound cards and in our normal work we do not need to install an external amplifier to the sound card. In Windows operating system, volume controller is used in order to obtain the signal factors by the sound card.

A. Signal Filtering

Digital telephone lines are capable of carrying signals in the range of 0 to 3400 Hz. Therefore, signals that have a frequency outside this range must be filtered before they can be digitized. Filtering, especially in non-real time operations, can also be achieved in the frequency domain, in which the filter is applied in the frequency domain and then the result is returned to the time domain (or place/space). This can be an effective implementation and will necessarily respond to any filter, including excellent approximations of brick wall filters [20], [21]. In fact, after the input signal is prepared (amplified), it enters the filtering system to remove signals outside the frequency range of that application, from the system. Digital filters are available in both FIR and IIR. While FIR filters are always stable, IIR filters have feedback loops that may become unstable and fluctuate. The Z-converter provides a tool for analyzing the stability of IIR filters. This converter is similar to the Laplace transform, which is used to design and analyze similar IIR analog filters [22]. Filter design is especially important because if the signal is not filtered properly, it will malfunction. Due to the fact that the design of analog filters is beyond our scope of expertise, so it can be obtained from mentioning information about them. However, when describing how to design digital filters, it can show the parameters that must be considered to design a suitable filter. This is also possible by designing new filters. The process of applying filter to the digital signals is shown in Fig. 3 [23].

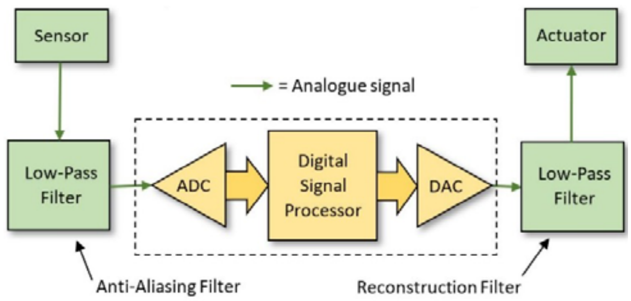

Fig. 3 The process of applying filter to the digital signals [23]

B. Multiply the Signal

Two or more signals are multiplied together in order to generate new signals from the values of the original signals in digital signal processing operations. The operation is applied to the continuous-time or discrete-time signals to create new values from the original signals [24].

Theoretical analyzes and inferences in DSP are usually performed on time-discrete signal models without amplitude error (quantification error) "generated" during the abstract sampling process. Numerical methods require quantified signals, such as signals generated by an ADC. The processed result may be a frequency spectrum or a set of statistical indicators. But often, it is another quantified signal that is converted to analog by a DAC [25].

V. Signal Processing Applications

Application of signal processing in different engineering and medical systems are expanded in the recent years. It can be applied to the engineering field as equipment maintenance as well as quality control. Also, it can be applied to improvement of healthcare systems in the hospitals to increase safety of patients. The signal processing systems are also applied to the language reorganization systems to detect and translate the different languages for people from different nationalities. Different applications of the DSP are shown in Fig. 4 [26].

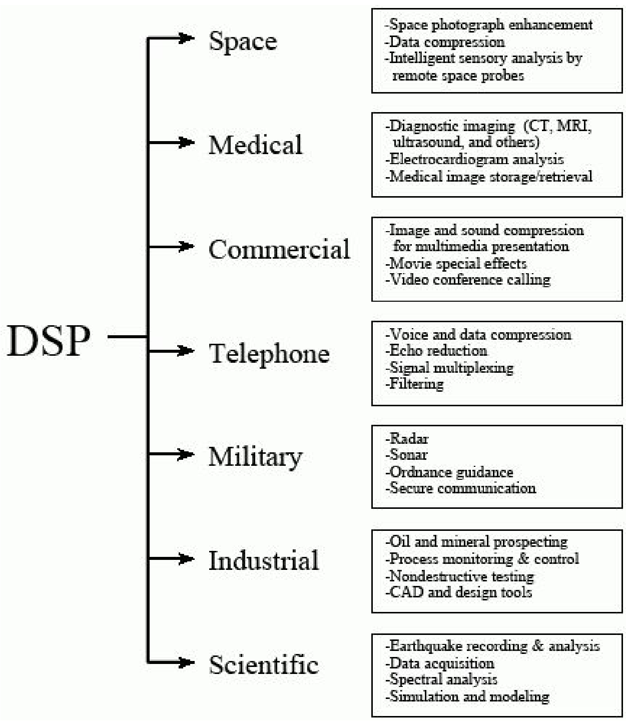

Fig. 4 Different applications of the DSP [26]

A. Speech Recognition Systems

From the beginning of intelligent human life on earth, he always wanted to get to know his surroundings by recognizing sounds. However, this effort was not limited to swimming in the sounds of the environment in humans, and the reaction of the environment to the human voice has always been important.

The challenge is to eliminate the noise which is in an audio file, or the noise that may occur when we speak into the microphone due to the environment. For this purpose, we need to process the digital signal on the computer system and after detecting the noise using a method to remove them. It is possible to design a software which types the words spoken in the microphone. Speech processing is a science that uses signal processing methods to do this. In this way, signal processing, including speech recognition, has always been very important in order to provide the desired input to computers by voice (as is common among humans) [27], [28]. Because the speed of data transfer to computers could be greatly increased. Speech recognition is a vital signal processing technology that is at the same time very easy to understand its overall function. Application of the signal processing method in the speech recognition systems is presented [29] to increase performances of computers in the communication systems. Application of sound detection system in the social networks is investigated [30] to increase quality of social networks for providing qualified communication system. In order to delete the noise from the digital signals of speech, advanced signal processing system is developed [31].

B. Radar and Space Signals Monitoring

Application of the signal processing technology in the radar signals monitoring is presented [32] to increase the effects of radars in the target detection. Advanced radar signal processing using programmable logic technique is investigated [33] in order to develop the detection power in the radar systems. The developed Fourier analysis method in the signal processing techniques is presented [34] to improve the quality of optical communication systems. Radar detection in the moments space of the scattered signal parameters is developed [35] to increase the detection rate in the signal processing methods. Radar Signal Processing for Sensing in Assisted Living is investigated [36] in order to analyze the challenges associated with the real-time implementation of signal processing/classification algorithms. Data Acquisition and Signal Processing System for CW Radar is presented [37] to develop simple and effective data acquisition and signal processing algorithm for radar system.

C. Industrial and Manufacturing Engineering

The ultrasonic signal in the gas flow meter system is analyzed by [38] to increase efficiency of gas transfers in the gas pipeline. To measure the viscosity of liquids in the engineering process, application of signal process techniques using the ultrasonic method is presented [39]. Applications of series modeling techniques in the signal processing methods are presented in the cutting zone prediction for the CNC machining operations [40], GPS and communication devices [41] and machine learning applications [42]. Application of signal processing systems in the sound of radial compressors is investigated [43] to decrease the cost maintenance in the power plants. The working condition of mixing and grinding machine is analyzed using the application of signal processing systems in the sound analysis [44]. To decrease the cost of engine repairing in different engineering field, an advanced engine fault detection systems is developed [45].

To increase efficiency in pavement methods of roads, application of signal processing techniques in sound analysis is presented by [46]. Advanced bowel sound analysis in the image processing systems applications is presented [47] to enhance quality control systems.

D. Health Care Systems

In the medical treatment of patient, sound analysis for the hart disease is investigated by [48]. Heart sound analysis system is developed by [49] to improve the quality of sound detection systems in the health care industries. English speech sound detection system is developed by [50] to increase the quality of sound detection technologies. Heart sound and lung sound of patients are analyzed by using the developed system in the research work [51] to improve quality of signal processing techniques in medical devices. Different sound signals of heart sound are analyzed and classified [52] to increase quality of medical care in hospitals.

VI. Conclusion

Signal processing applications include audio signal processing, audio compression, digital image processing, video compression, speech processing, speech recognition, digital telecommunications, digital instrument combinations, radar and sonar signal processing, financial signal processing, seismology and biopharmaceutical. DSP is the use of digital processing systems by computers or more specialized digital signal processors to perform a variety of signal processing operations. Specific examples include speech coding and transmission on digital mobile phones, room audio correction in hi-fi and amplification applications, weather forecasting, economic forecasting, seismic data processing, industrial process analysis and control, medical imaging Such as CAT and MRI scans, MP3 compression, computer graphics, image manipulation, sound interference and equalizer, and sound effects units. The use of digital computing in signal processing has many advantages over analog processing in many areas such as error detection and correction in communication as well as data compression. DSP can be used on both streaming and static data. High Accuracy, flexibility, ease of data storage, time sharing are some advantages of the DSP systems. But, the system complexity as well as power consumption is some disadvantages of the DSP systems. The developed DSP systems can increase the power of sending and receiving data in order to develop the telecommunication systems. Also, the quality of health care systems in hospital can increased to improve the safety of people in the society.

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