

DIGITAL SIGNAL PROCESSING - APPLICATIONS IN MEDICINE

Paulo S. R. Diniz

Program of Electrical Engineering, COPPE/EE/Federal University of Rio de Janeiro, Brazil

David M. Simpson and A. De Stefano

Institute of Sound and Vibration Research, University of Southampton, UK

Ronaldo C. Gismondi

School of Medicine, State University of Rio de Janeiro, Brazil

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Summary

This chapter presents a brief introduction to some of the main concepts of digital signal processing, which is a technology widely used to solve current real life problems. Digital signal processing is one of the main tools in modern technology since it relies on

the fast developing digital integrated circuit technology, which in turn allows for high performance computing machines at relatively low cost. Each concept discussed is followed by its application in the analysis, synthesis and enhancement of medical signals. The correct analysis of measured signals from a patient relies on the removal of contamination from clinically relevant signals and correct measures of parameters related to these signals. In other cases, diagnosed impairments of a patient can be compensated by properly shaped signals synthesized via external equipments. Currently many types of electronic equipment in a wide variety of application fields utilize digital signal processing technology.

1. Introduction

Most natural phenomena occur continuously in time, such as the variations of temperature of the human body, forces exerted by muscles, or electrical potentials generated on the surface of the scalp. These are analog signals, being able to take on any value (though usually limited to a finite range). They are also continuous in time, i.e. at all instants in time their value is available. However, analog, continuous-time signals are not suitable for processing on the now ubiquitous computer-type processors (or other digital machines), which are built to deal with sequential computations involving numbers. These require digital signals, which are formed by sampling the original analog data.

The theory of sampling was developed in the early 20th century by Nyquist [1] Shannon and others, and revolutionized signal processing and analysis [2]-[6] especially from the 1960s onwards, when the appropriate computer technology became widely available. The rapid development of high-speed digital integrated circuit technology in the last three decades has made digital signal processing the technique of choice for many most applications, including multimedia, speech analysis and synthesis, mobile radio, sonar, radar, seismology and biomedical engineering. Digital signal processing presents many advantages over analog approaches: digital machines are flexible, reliable, easily reproduced and relatively cheap. As a consequence, many signal processing tasks originally performed in the analog domain are now routinely implemented in the digital domain, and others can only feasibly be implemented in digital form. In most cases, a digital signal processing system is implemented using software on a general-purpose digital computer or digital signal processor (DSP). Alternatively, application specific hardware usually in the form of an integrated circuit can also be employed.

In this chapter we will discuss a number of fundamental principles and basic tools for digital signal processing. These will be illustrated with examples from medical applications, where signal analysis has been widely applied in patient monitoring, diagnosis and prognosis, as well as physiological investigation and in some therapeutic settings (e.g. muscle and sensory stimulation, hearing aids).

We will first discuss the principles of sampling. Here it is demonstrated that certain signals (those limited in the range of frequencies they contain), can be fully represented by a sequence of samples. These digital signals coincide with the original analog (and continuous-time) signals at predefined time instants. By interpolation, the continuous-time signal can be recovered (without error!) from the sequence of samples.

Next we will discuss digital filters, which are one of the main tools employed in signal processing, as they suppress certain frequency bands, and enhance others. This is particularly useful in reducing noise and other sources of interference, which are an almost constant problem in medical applications. In these signals the most common sources of noise are electromagnetic interference at 50/60 Hz (and in higher frequency ranges), and contamination by other, unwanted physiological signals (e.g. electrical signals from muscles may obscure signals of neural origin). Patient motion often results in short-lasting artifacts in recorded data, and filters can again be used to mitigate these. Digital filters can also be employed to describe the relationship between physiological signals, such as that between blood pressure and blood flow. As such, they are also useful in characterizing physiological systems.

In Fourier analysis, a signal is split into constituent sinusoidal oscillations (we will assume that readers are familiar with the continuous time Fourier transform), and the discrete Fourier transform is an extension developed for the analysis of digital signals. For the current work in particular, it provides a very convenient tool for specifying and describing digital filters, and it also provides the basis for the sampling theorem.

Many signals, including most from a biomedical origin, can be classified as random: repeated recordings result in signals that are all different from each other, but share the same statistical characteristics. The power spectrum reflects some of the most interesting of these characteristics. The interpretation and estimation of the power spectral density will be addressed in the final part of this chapter.

2. Digital Signal Processing of Continuous-Time Signals

A typical digital signal processing system includes the following subsystems, as illustrated in Figure 1.

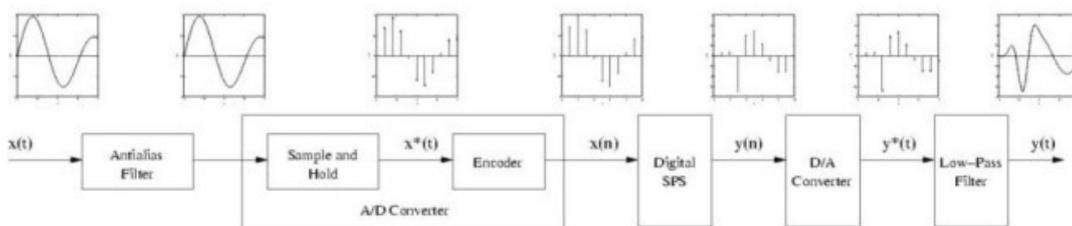


Figure 1. Architecture of a complete DSPPS used to process an analog signal

- A/D Converter – transforms the analog input signal into a digital signal. It does so by acquiring samples from the signal at (normally) equally spaced time intervals and converting the level of these samples into a numeric representation that can be used by a digital signal processing system. In accordance with the sampling theorem, a low-pass (anti-alias) filter is usually required prior to A/D conversion.
- Digital Signal Processing - the digital signal processing system (DSPPS) performs arithmetic operations on the input sequence. In a typical application, the desired signal features are enhanced in output signal, and unwanted components (such as noise and artifact) are suppressed. Further analysis of the output signal may be carried out in order

to extract information. In a medical application, this could for example be to determine the average heart-rate, the peak blood pressure, or an index of muscle activity. If an output signal is required (e.g. in a hearing aid), the next two steps are carried out.

- D/A Converter - converts the DSPS output into analog samples that are equally spaced in time.
- Lowpass Filter - converts the analog samples into a continuous-time signal. This step is equivalent to an interpolation operation between the discrete analog samples produced in the previous step.

Example 1: A hearing aid

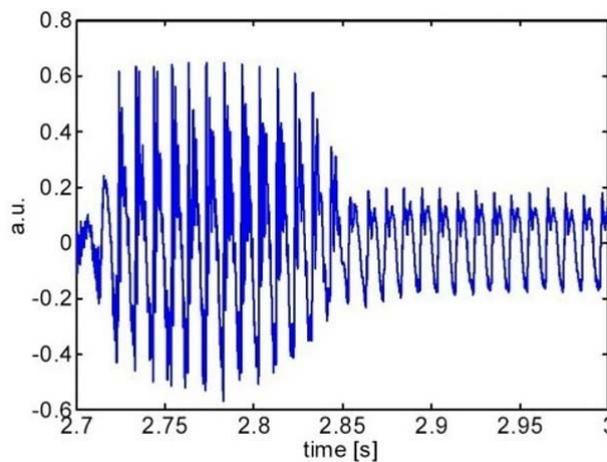


Figure 2. A short segment of speech signal that may form the input to a hearing aid. The segment displayed corresponds to the sound /um/.

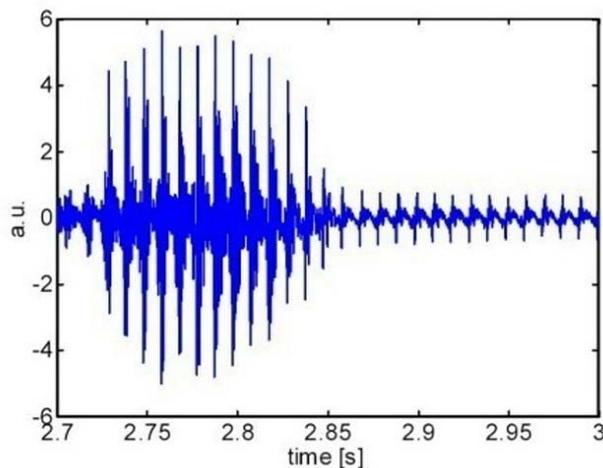


Figure 3. The output of the digital filter (digital signal). The frequencies around 1500 Hz, where the patient showed the greatest loss in hearing, have been most amplified.

Further details on the filter used may be found in an example below.

A hearing aid may be used to illustrate the operation of a DSPS: the sound is picked up by the microphone (Figure 2), converted into an electric signal, and then digitized

(Figure 3). The digital signal is then filtered to selectively amplify those frequency bands in which the patient shows the most severe hearing loss. Further processes (Figure 4) may also be applied, including amplitude compression, in which the system gain is reduced when the amplitude exceeds some pre-defined threshold values, in order to avoid excessive loudness to the ear. Finally the processed digital signal is converted back to analog form in the digital-to-analog (D/A) converter, and delivered to the ear via the earphone.

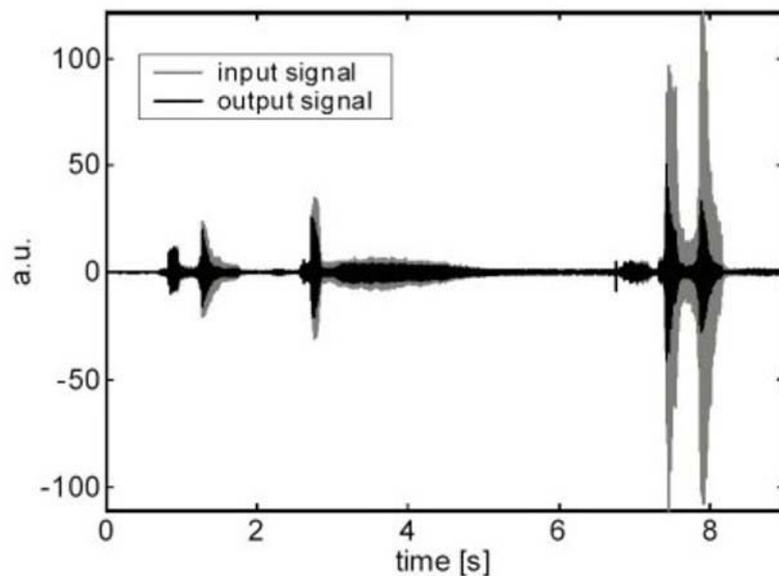


Figure 4. In a further stage of signal processing in hearing aids, the amplitude-range may be compressed. Thus the gain of the system is progressively reduced when the sound-volume exceeds a specified level, as is the case near the end of this recording. A small delay in adapting the gain to the signal-levels may also be noted, as the average of the most recent amplitudes drives the gain control.

In signal analysis the periodic signals play a major role since they serve as basic signals from which many other signals can be constructed and analyzed. Let's take as example the sinusoid $\sin \omega_0 t$, that is a periodic signal since $\sin(\omega_0 + 2\pi)t = \sin \omega_0 t$.

There are periodic signals that are harmonically related, meaning that they consist of a set of periodic signals whose fundamental frequencies are all multiples of a single positive frequency ω_0 . For example, the periodic signals $\sin k \omega_0 t$, for any integer k , is called the k -th harmonic of $\sin \omega_0 t$.

Also in signal analysis it is often performed a linear transformation of the signal from the time domain to the frequency domain and vice versa, depending on the domain in which either the relevant information is exposed in a clearer way or the mathematical manipulations are simpler.

In the particular case of periodic signals, they have quite compact representation in the frequency domain where only the fundamental frequency information ω_0 and their harmonics are required, whereas in the time domain they are represented by a

continuous function of time.

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Biographical Sketches

Paulo S. R. Diniz is a full Professor at the Federal University of Rio de Janeiro (UFRJ), teaching at Polytechnic School and COPPE. He received the Electronics Eng. Degree (Hons) from UFRJ, M. Sc. from COPPE/UFRJ, and the Ph.D. from Concordia University in Canada. He has received some awards for best papers and research achievements, including the Fellow award from the Institute of Electrical and Electronic Engineers (IEEE) in the USA. He also holds a *Dosentti* position at Helsinki University of Technology, in Finland. His research interests include adaptive systems, digital signal processing, communications, and electronic circuits, among others. He teaches wireless communications, digital communications, adaptive filtering, digital signal processing, filter banks and wavelets etc. He has over 200 publications in journals (mostly IEEE journals), international conferences, and book chapters. He has written two textbooks, namely: 1-Adaptive Filtering: Algorithms and Practical Implementations, Kluwer Academic Publishers, 2nd edition, 2002; 2- Digital Signal Processing: System Analysis and Design, Cambridge University Press, 2002 (with E. A. B. da Silva and S. L. Netto).

David Simpson obtained a BSc in Biomedical Electronics from the University of Salford (UK) and a

PhD in Electrical Engineering from Imperial College of Science, Technology and Medicine, University of London (UK). He was a visiting lecturer in the Biomedical Engineering Program at the Federal University of Rio de Janeiro (Brazil) from 1989 – 1998, and a research fellow in Medical Physics at the University of Leicester (UK). Currently he is a lecturer in Biomedical Signal Processing at the University of Southampton. His research interests focus on the processing of physiological signals, in particular those related to blood flow control and cerebral electrical activity (EEG and evoked potentials), with a focus on statistical aspects of signal analysis. He is a member of the Institute of Physics and Engineering in Medicine (UK), the IEEE Signal Processing Society (USA) and the Brazilian Biomedical Engineering Society. He has published over 100 articles in conferences and journals.

Antonio De Stefano is Distance Learning Coordinator for the Master Training Packages in Biomedical Signal Processing at the ISVR in the University of Southampton, UK. He received the Electronics Eng. Degree, the specialization in Biomedical Sciences and the CEng qualification from the University Federico II in Naples (Italy), and PhD from the ISVR in the University of Southampton (UK). His research interests include biomedical signal processing, implementation of distance learning packages for biomedical subjects, wavelet based noise reduction and image enhancement, techniques for EMG analysis during walking of pathological children, and mechanical models for speech design. He has over 20 publications in journals and international conferences.

Ronaldo Curi Gismondi is an Associate Professor at the State University of Rio de Janeiro (UERJ), teaching at School of Medicine. He received the Medical Degree (Hons) from Federal Fluminense University (UFF - 1980), M.Sc.(1991) and D.Sc.(1999) from Biomedical Engineering Program of COPPE/UF RJ. His research interests include artificial neural networks applications in health systems modeling, also telemedicine and information technology applied to medical education. He has over 50 publications in journals, and book chapters. He is, also, an active physician in Niteroi, near Rio de Janeiro, as a General Practitioner.