APPLICATIONS OF DIGITAL SIGNAL PROCESSING

Edited by **Christian Cuadrado-Laborde**

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Contents

Preface

It is a great honor and pleasure for me to introduce this book "Applications of Digital Signal Processing" being published by InTech. The field of digital signal processing is at the heart of communications, biomedicine, defense applications, and so on. The field has experienced an explosive growth from its origins, with huge advances both in fundamental research and applications.

In this book the reader will find a collection of chapters authored/co-authored by a large number of experts around the world, covering the broad field of digital signal processing. I have no doubt that the book would be useful to graduate students, teachers, researchers, and engineers. Each chapter is self-contained and can be downloaded and read independently of the others.

This book intends to provide highlights of the current research in the digital signal processing area, showing the recent advances in this field. This work is mainly destined to researchers in the digital signal processing related areas but it is also accessible to anyone with a scientific background desiring to have an up-to-date overview of this domain. These nineteenth chapters present methodological advances and recent applications of digital signal processing in various domains as telecommunications, array processing, medicine, astronomy, image and speech processing.

Finally, I would like to thank all the authors for their scholarly contributions; without them this project could not be possible. I would like to thank also to the In-Tech staff for the confidence placed on me to edit this book, and especially to Ms. Danijela Duric, for her kind assistance throughout the editing process. On behalf of the authors and me, we hope readers enjoy this book and could benefit both novice and experts, providing a thorough understanding of several fields related to the digital signal processing and related areas.

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Part 1

DSP in Communications

1

Complex Digital Signal Processing in Telecommunications

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1. Introduction

1.1 Complex DSP versus real DSP

Digital Signal Processing (DSP) is a vital tool for scientists and engineers, as it is of fundamental importance in many areas of engineering practice and scientific research.

The "alphabet" of DSP is mathematics and although most practical DSP problems can be solved by using real number mathematics, there are many others which can only be satisfactorily resolved or adequately described by means of complex numbers.

If real number mathematics is the language of *real* DSP, then complex number mathematics is the language of *complex* DSP. In the same way that real numbers are a part of complex numbers in mathematics, *real* DSP can be regarded as a part of *complex* DSP (Smith, 1999).

Complex mathematics manipulates complex numbers – the representation of two variables as a single number - and it may appear that *complex* DSP has no obvious connection with our everyday experience, especially since many DSP problems are explained mainly by means of real number mathematics. Nonetheless, some DSP techniques are based on complex mathematics, such as Fast Fourier Transform (FFT), z-transform, representation of periodical signals and linear systems, etc. However, the imaginary part of complex transformations is usually ignored or regarded as zero due to the inability to provide a readily comprehensible physical explanation.

One well-known practical approach to the representation of an engineering problem by means of complex numbers can be referred to as the *assembling approach*: the real and imaginary parts of a complex number are real variables and individually can represent two real physical parameters. Complex math techniques are used to process this complex entity once it is assembled. The real and imaginary parts of the resulting complex variable preserve the same real physical parameters. This approach is not universally-applicable and can only be used with problems and applications which conform to the requirements of complex math techniques. Making a complex number entirely mathematically equivalent to a substantial physical problem is the real essence of *complex* DSP. Like complex Fourier transforms, complex DSP transforms show the fundamental nature of *complex* DSP and such complex techniques often increase the power of basic DSP methods. The development and application of *complex* DSP are only just beginning to increase and for this reason some researchers have named it *theoretical* DSP.

It is evident that *complex* DSP is more complicated than *real* DSP. Complex DSP transforms are highly theoretical and mathematical; to use them efficiently and professionally requires a large amount of mathematics study and practical experience.
Complex math makes the mathematical expressions used in DSP more compact and solves the problems which real math cannot deal with. Complex DSP techniques can complement our understanding of how physical systems perform but to achieve this, we are faced with the necessity of dealing with extensive sophisticated mathematics. For DSP professionals there comes a point at which they have no real choice since the study of complex number mathematics is the foundation of DSP.

1.2 Complex representation of signals and systems

All naturally-occurring signals are real; however in some signal processing applications it is convenient to represent a signal as a complex-valued function of an independent variable. For purely mathematical reasons, the concept of complex number representation is closely connected with many of the basics of electrical engineering theory, such as voltage, current, impedance, frequency response, transfer-function, Fourier and z-transforms, etc.
Complex DSP has many areas of application, one of the most important being modern telecommunications, which very often uses narrowband analytical signals; these are complex in nature (Martin, 2003). In this field, the complex representation of signals is very useful as it provides a simple interpretation and realization of complicated processing tasks, such as modulation, sampling or quantization.
It should be remembered that a complex number could be expressed in *rectangular*, *polar* and *exponential* forms:

$$a + jb = A\left(\cos\theta + j\sin\theta\right) = Ae^{j\theta} \,. \tag{1}$$

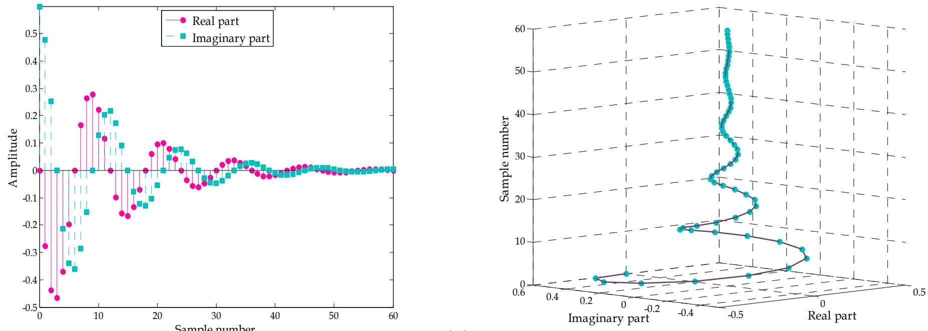
The third notation of the complex number in the equation (1) is referred to as *complex exponential* and is obtained after Euler's relation is applied. The exponential form of complex numbers is at the core of *complex* DSP and enables magnitude A and phase θ components to be easily derived.
Complex numbers offer a compact representation of the most often-used waveforms in signal processing – *sine* and *cosine* waves (Proakis & Manolakis, 2006). The complex number representation of sinusoids is an elegant technique in signal and circuit analysis and synthesis, applicable when the rules of complex math techniques coincide with those of sine and cosine functions. Sinusoids are represented by complex numbers; these are then processed mathematically and the resulting complex numbers correspond to sinusoids, which match the way sine and cosine waves would perform if they were manipulated individually. The complex representation technique is possible only for sine and cosine waves of the same frequency, manipulated mathematically by linear systems.
The use of Euler's identity results in the class of complex exponential signals:

$$x\left(n\right) = A\alpha^{n} = \left|A\right| e^{j\phi} e^{\left(\sigma_0 + j\omega_0\right)} = x_R\left(n\right) + jx_I\left(n\right) \,. \tag{2}$$

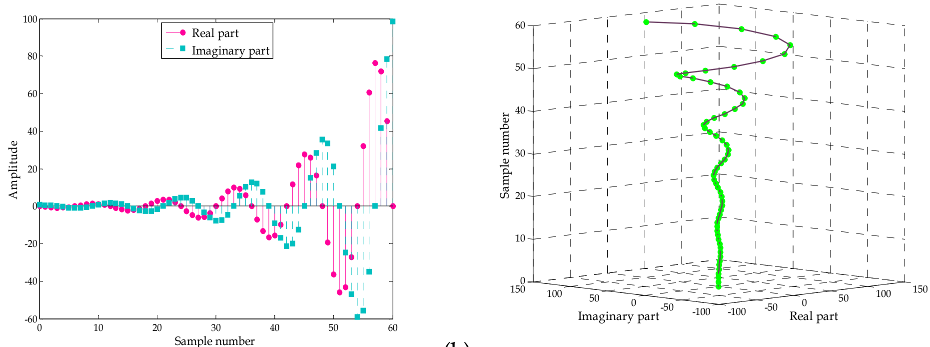
$\alpha = e^{\left(\sigma_0 + j\omega_0\right)}$ and $A = \left|A\right| e^{j\phi}$ are complex numbers thus obtaining:

$$x_R(n) = |A| e^{\sigma_0 n} \cos(\omega_0 n + \phi); \qquad x_I(n) = |A| e^{\sigma_0 n} \sin(\omega_0 n + \phi). \tag{3}$$

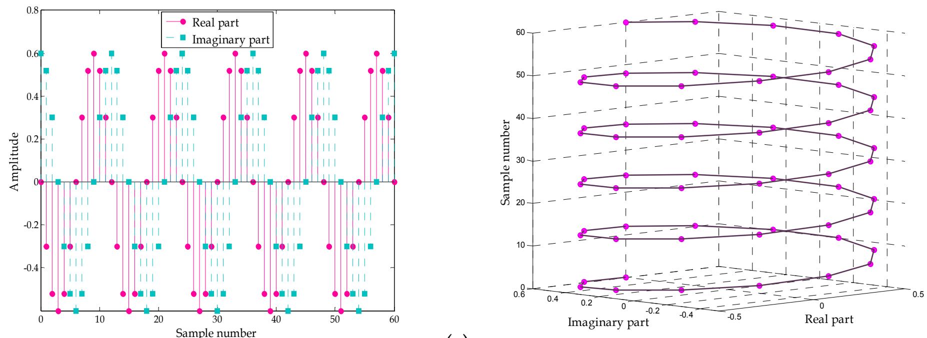
Clearly, $x_R(n)$ and $x_I(n)$ are real discrete-time sinusoidal signals whose amplitude $|A|e^{\sigma_0 n}$ is constant ($\sigma_0=0$), increasing ($\sigma_0>0$) or decreasing ($\sigma_0<0$) exponents (Fig. 1).



(a)



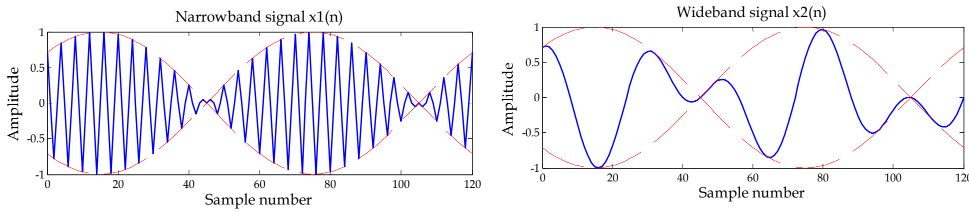
(b)



(c)

Fig. 1. Complex exponential signal $x(n)$ and its real and imaginary components $x_R(n)$ and $x_I(n)$ for (a) σ_0=-0.085; (b) σ_0=0.085 and (c) σ_0=0

The spectrum of a real discrete-time signal lies between $-\omega_s/2$ and $\omega_s/2$ (ω_s is the sampling frequency in radians per sample), while the spectrum of a complex signal is twice as narrow and is located within the positive frequency range only.
Narrowband signals are of great use in telecommunications. The determination of a signal's attributes, such as frequency, envelope, amplitude and phase are of great importance for signal processing e.g. modulation, multiplexing, signal detection, frequency transformation, etc. These attributes are easier to quantify for narrowband signals than for wideband signals (Fig. 2). This makes narrowband signals much simpler to represent as complex signals.



(a) (b)

Fig. 2. Narrowband signal (a) $x_1(n) = \sin(\pi/60n + \pi/4)\cos(\pi/2n)$;
wideband signal (b) $x_2(n) = \sin(\pi/60n + \pi/4)\cos(\pi/16n)$

Over the years different techniques of describing narrowband complex signals have been developed. These techniques differ from each other in the way the imaginary component is derived; the real component of the complex representation is the real signal itself.
Some authors (Fink, 1984) suggest that the imaginary part of a complex narrowband signal can be obtained from the first $x'_R(n)$ and second $x''_R(n)$ derivatives of the real signal:

$$x_I(n) = -x'_R(n)\sqrt{\frac{-x_R(n)}{x''_R(n)}} \; . \tag{4}$$

One disadvantage of the representation in equation (4) is that insignificant changes in the real signal $x_R(n)$ can alter the imaginary part $x_I(n)$ significantly; furthermore the second derivative can change its sign, thus removing the sense of the square root.
Another approach to deriving the imaginary component of a complex signal representation, applicable to harmonic signals, is as follows (Gallagher, 1968):

$$x_I(n) = \frac{-x_R(n)}{\omega_0}, \tag{5}$$

where ω_0 is the frequency of the real harmonic signal.
Analytical representation is another well-known approach used to obtain the imaginary part of a complex signal, named the *analytic* signal. An analytic complex signal is represented by its *inphase* (the real component) and *quadrature* (the imaginary component). The approach includes a low-frequency envelope modulation using a complex carrier signal – a complex exponent $e^{j\omega_0 n}$ named *cissoid* (Crystal & Ehrman, 1968) or *complexoid* (Martin, 2003):

$$x_R(n) \otimes e^{j\omega_0 n} \Rightarrow \quad x(n) = x_R(n)e^{j\omega_0 n} = x_R(n)[\cos\omega_0 n + j\sin\omega_0 n] = x_R(n) + jx_I(n) \,. \tag{6}$$

In the frequency domain an analytic complex signal is:

$$X_C\left(e^{j\omega n}\right)=X_R\left(e^{j\omega n}\right)+jX_I\left(e^{j\omega n}\right). \quad (7)$$

The real signal and its Hilbert transform are respectively the real and imaginary parts of the analytic signal; these have the same amplitude and π/2 phase-shift (Fig. 3).

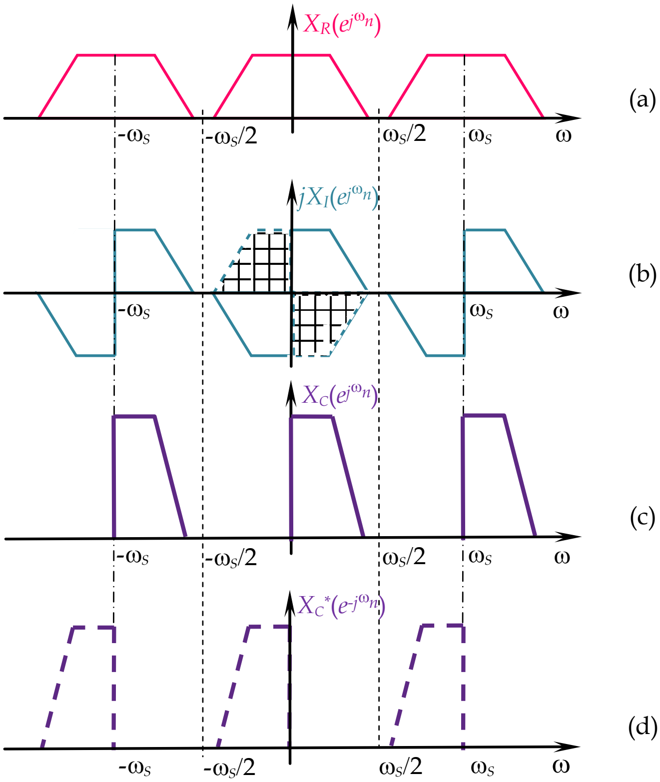


Fig. 3. Complex signal derivation using the Hilbert transformation

According to the Hilbert transformation, the components of the $X_R\left(e^{j\omega n}\right)$ spectrum are shifted by π/2 for positive frequencies and by –π/2 for negative frequencies, thus the pattern areas in Fig. 3b are obtained. The real signal $X_R\left(e^{j\omega n}\right)$ and the imaginary one $X_I\left(e^{j\omega n}\right)$ multiplied by j (square root of -1), are identical for positive frequencies and –π/2 phase shifted for negative frequencies – the solid blue line (Fig. 3b). The complex signal $X_C\left(e^{j\omega n}\right)$ occupies half of the real signal frequency band; its amplitude is the sum of the $X_R\left(e^{j\omega n}\right)$ and $jX_I\left(e^{j\omega n}\right)$ amplitudes (Fig. 3c). The spectrum of the complex conjugate analytic signal $X_C^*\left(e^{-j\omega n}\right)$ is depicted in Fig. 3d.