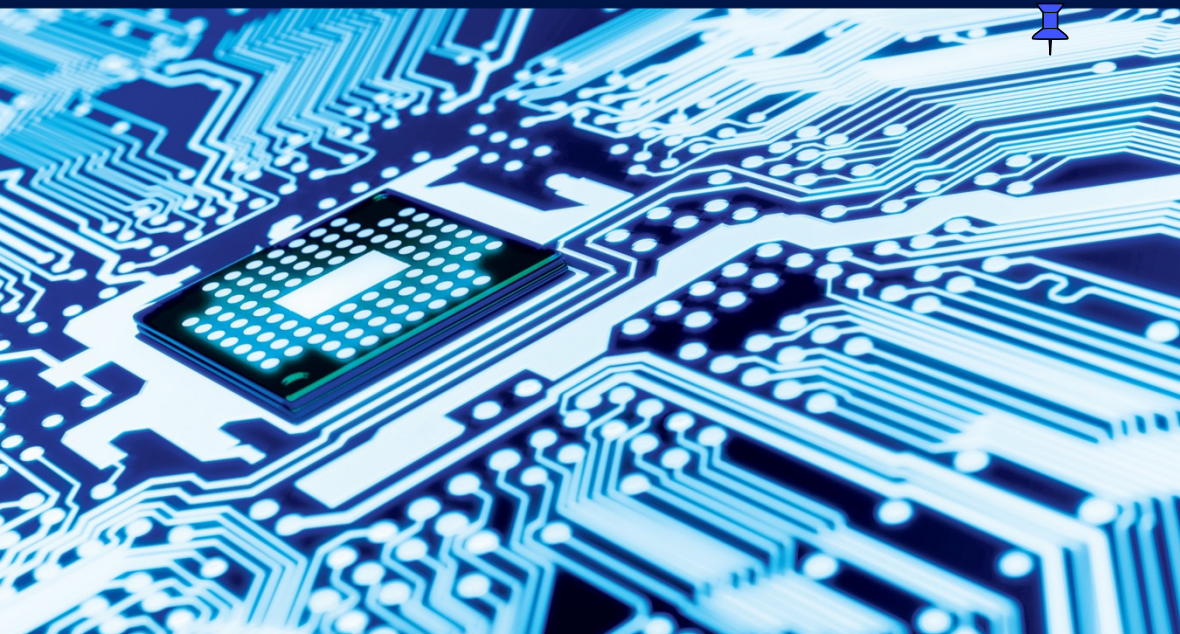


ELECTRONICS ENGINEERING SERIES



Fundamentals of Electronics 2

*Continuous-time Signals
and Systems*

Pierre Muret

ISTE

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*Continuous-time
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Preface

Today, we can consider electronics to be a subject derived from both the theoretical advances achieved during the 20th Century in areas comprising the modeling and conception of components, circuits, signals and systems, together with the tremendous development attained in integrated circuit technology. However, such development led to something of a knowledge diaspora that this work will attempt to contravene by collecting both the general principles at the center of all electronic systems and components, together with the synthesis and analysis methods required to describe and understand these components and subcomponents. The work is divided into three volumes. Each volume follows one guiding principle from which various concepts flow. Accordingly, Volume 1 addresses the physics of semiconductor components and the consequences thereof, that is, the relations between component properties and electrical models. Volume 2 addresses continuous time systems, initially adopting a general approach in Chapter 1, followed by a review of the highly involved subject of quadripoles in Chapter 2. Volume 3 is devoted to discrete-time and/or quantized level systems. The former, also known as sampled systems, which can either be analog or digital, are studied in Chapter 1, while the latter, conversion systems, we address in Chapter 2. The chapter headings are indicated in the following general outline.

Each chapter is paired with exercises and detailed corrections, with two objectives. First, these exercises help illustrate the general principles addressed in the course, proposing new application layouts and showing how theory can be implemented to assess their properties. Second, the exercises act as extensions of the course, illustrating circuits that may have been described briefly, but whose properties have not been studied in detail. The

first volume should be accessible to students with a scientific literacy corresponding to the first 2 years of university education, allowing them to acquire the level of understanding required for the third year of their electronics degree. The level of comprehension required for the following two volumes is that of students on a master's degree program or enrolled in engineering school.

In summary, electronics, as presented in this book, is an engineering science that concerns the modeling of components and systems from their physical properties to their established function, allowing for the transformation of electrical signals and information processing. Here, the various items are summarized along with their properties to help readers follow the broader direction of their organization and thereby avoid fragmentation and overlap. The representation of signals is treated in a balanced manner, which means that the spectral aspect is given its proper place; to do otherwise would have been outmoded and against the grain of modern electronics, since now a wide range of problems are initially addressed according to criteria concerning frequency response, bandwidth and signal spectrum modification. This should by no means overshadow the application of electrokinetic laws, which remains a necessary first step since electronics remains fundamentally concerned with electric circuits. Concepts related to radio-frequency circuits are not given special treatment here, but can be found in several chapters. Since the summary of logical circuits involves digital electronics and industrial computing, the part treated here is limited to logical functions that may be useful in binary numbers computing and elementary sequencing. The author hopes that this work contributes to a broad foundation for the analysis, modeling and synthesis of most active and passive circuits in electronics, giving readers a good start to begin the development and simulation of integrated circuits.

Outline

1) Volume 1: Electronic Components and Elementary Functions [MUR 17].

- i) Diodes and Applications
- ii) Bipolar Transistors and Applications
- iii) Field Effect Transistor and Applications
- iv) Amplifiers, Comparators and Other Analog Circuits

2) Volume 2: Continuous-time Signals and Systems.

i) Continuous-time Stationary Systems: General Properties, Feedback, Stability, Oscillators

ii) Continuous-time Linear and Stationary Systems: Two-port Networks, Filtering and Analog Filter Synthesis

3) Volume 3: Discrete-time Signals and Systems and Conversion Systems [MUR 18].

i) Discrete-time Signals: Sampling, Filtering and Phase Control, Frequency control circuits

ii) Quantized Level Systems: Digital-to-analog and Analog-to-digital Conversions

Pierre MURET
November 2017

Introduction

This volume is dedicated to the study of linear and stationary systems in which time is considered as a continuous variable, as well as certain extensions in the case of nonlinear systems. It is mainly centered on single-input and single-output systems but a method capable of generalizing studies to linear or nonlinear multi-input and multi-output systems is also addressed. Generally, in order to highlight the properties of these systems, one must necessarily rely on the analysis of electrical signals that either characterize their response to an excitation signal or their natural (or proper) response. The former output signal is dependent on the input signal and is called forced response, whereas their natural response is independent of the excitation signal placed on their input. Therefore, it is essential to begin with the representations of signals by forming a close correlation between the time domain and the frequency domain, which are connected by the Fourier transform or decomposition into Fourier series. It is then natural to customize the study to the case of stationary systems, for which the forced response is invariant under time translation of the signal applied on input, and which, in addition, follow the principle of causality. The unilateral Laplace transform then proves to be useful and it leads us to the notion of transfer function or transmittance, together with the Fourier transform in the case of finite energy signals. The properties of these two types of transforms and their application to the case of electronic systems are covered in the first part of Chapter 1 while the consequences of causality are addressed in Chapter 2.

The second part of Chapter 1 is dedicated to the study of feedback and its applications, and then to the different methods for studying the stability of the systems, or to means able to control their instability, as is the case for

oscillators. A system is stable if, after a finite life span excitation, it finally returns to its previous idle state, namely without any variation of electrical quantities, and it is unstable otherwise. In the early stages of electronics, feedback was paramount and it led to much progress and the development of a multitude of applications, which are reviewed here. The mathematical tools constituted by the time–frequency transforms mentioned earlier or representations in the complex plane are then used to address problems of system stability, including the case of those that incorporate a feedback loop, known as looped systems. The extension to state variables and state representation, which is based on the decomposition of the response of a system into a set of first-order differential equations, is then addressed. The previous concepts finally make it possible to detail the different ways for analyzing oscillators' operation, which initially can be considered as linear systems at the limit of stability, but which in practice are always subject to a limitation of the amplitude that requires nonlinearity to be taken into account. The transition from predictable operation to a chaotic regime is presented in the case of a model system.

In Chapter 2, the properties of stable electronic systems are particularized to the case of networks and specially quadripoles. The different representations of networks in the form of quadripoles are discussed, as well as all notions of impedance or admittance deriving therefrom. Some are measurable, thus experimentally feasible, while others are fictional, such as image impedances, but open a highly fruitful scope of application, which is the subject of the last section of this chapter. The concepts of matching, whether power or impedance matching, are detailed, as well as their consequences and rules to apply in practice in order to optimize the operation of electronic assemblies and to best take advantage of the components that are included.

The last part of Chapter 2 is devoted to stable systems that can be analyzed as analog filters, namely satisfying the principle of causality, of which the general consequences are presented. There are either circuits incorporating one or more active devices such as operational amplifiers or passive circuits, limited here to non-dissipative cases. The synthesis of these analog filters is thorough, and can be used to determine the value of all the components of a filter based on imposed criteria, most often a template in the frequency domain. Two topics are presented; on the one hand for active filters and on the other hand for non-dissipative passive filters. In the second case, the method using effective parameters is an exact method, but not

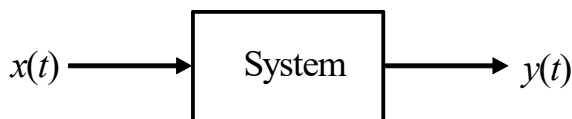
covering all the applications, while the method of image parameters is suitable to most requirements, with a deviation from the template that can be minimized. The ways to make adjustments and all circuits necessary for the practical implementation of the filters are detailed. Examples are given for each important case, based on the transfer functions calculated by means of software programs (here, MATLAB). The different possible choices for the computational functions are presented in relation to the criteria to be verified. In the case of the synthesis based on image parameters, formulas allowing the calculation of all elements are demonstrated. Although the case of systems with distributed (or scattered) elements, essential when the wavelength becomes comparable to the dimensions of the circuit, is not explicitly addressed, the description of the quadripoles using s -parameters, as detailed in Chapter 2, easily adapts.

Continuous-time Systems: General Properties, Feedback, Stability, Oscillators

The linear and stationary systems that concern us here deliver output signal $y(t)$ when input signal $x(t)$ is applied to them, solution to a *real* and linear ordinary differential equation, where t represents the time variable:

$$a_n \frac{d^n y}{dt^n} + a_{n-1} \frac{d^{n-1} y}{dt^{n-1}} + \dots + a_1 \frac{dy}{dt} + a_0 y = b_m \frac{d^m x}{dt^m} + b_{m-1} \frac{d^{m-1} x}{dt^{m-1}} + \dots + b_1 \frac{dx}{dt} + b_0 x$$

which can also be seen as a linear application:



Function $\exp(\alpha t)$, with *real or complex* α , is of special importance since it is the specific function of the system's differential equation, which means that if $x(t) = \exp(\alpha t)$, the output signal is also proportional to $\exp(\alpha t)$. It is this fundamental property that warrants the approaches discussed in the following sections 1.1 and 1.2. Another method, based on the state-space form, also applicable to nonlinear systems, is presented in sections 1.4.5 and 1.5.

1.1. Representation of continuous time signals

These signals are real electrical quantities and thus measurable functions of time variable t , which itself is a continuous variable. They are also referred to as analog signals. An additional representation is formed by the frequency spectrum.

1.1.1. Sinusoidal signals

In general, any real sinusoidal signal of angular frequency ω_1 and constant frequency f_1 ($\omega_1 = 2\pi f_1$) is written as $y(t) = A \cos(\omega_1 t + \varphi_1)$, once a time and phase origin has been selected. But in complex numbers, this can also be written as:

$$y(t) = A \frac{\exp[j(\omega_1 t + \varphi_1)] + \exp[-j(\omega_1 t + \varphi_1)]}{2}$$

Both exponential terms with imaginary exponent appear with the same A coefficient and are always complex conjugates, two conditions that are required if $y(t)$ is real. The two vectors corresponding to images on the complex plane rotate in opposite directions; thus, frequency $-f_1$ is consistently found at the same time as frequency f_1 .

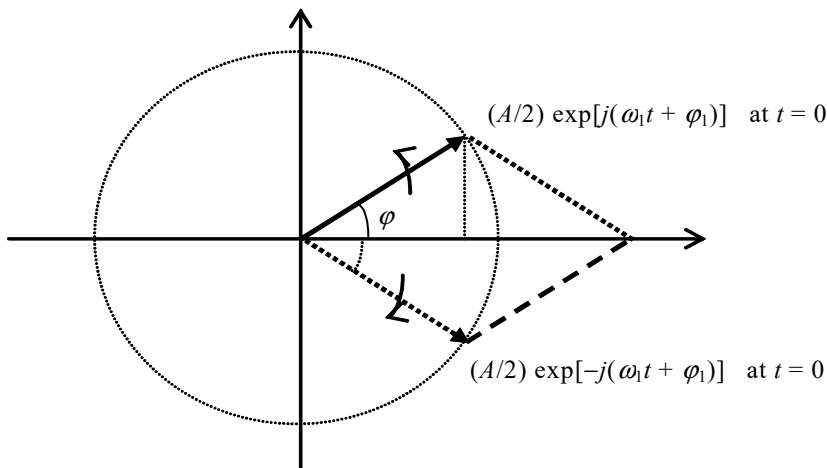


Figure 1.1. Representation of a sinusoidal signal on the complex plane

The spectral or frequency representation is thus formed simply by two lines of amplitude $A/2$ at frequencies f_1 and $-f_1$, and phase lines φ_1 and $-\varphi_1$ at these same frequencies.

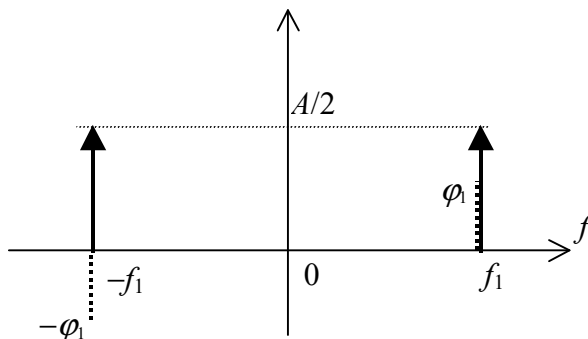


Figure 1.2. Spectrum of a sinusoidal signal (amplitude solid line, phase dotted)

Indeed, sinusoidal signals of the same frequency form a two-dimensional vector space for which a basis is provided by $\exp[j\omega_1 t]$ and $\exp[-j\omega_1 t]$ ($\cos[\omega_1 t]$ and $\sin[\omega_1 t]$ form another basis). Thus, we can write:

$$y(t) = c_1 \exp[j2\pi f_1 t] + c_{-1} \exp[-j2\pi f_1 t]$$

with

$$c_1 = \frac{A}{2} \exp(j\varphi_1)$$

and

$$c_{-1} = \frac{A}{2} \exp(-j\varphi_1)$$

where c_1 and c_{-1} are complex conjugates.

However, here only the first of these terms will be considered, with the second provided by complex conjugates. This leads to rotating vector or Fresnel representation: concerning the instantaneous values, only $A \exp[j(\omega_1 t + \varphi_1)]$ is used in the complex plane (or rather $\frac{A}{\sqrt{2}} \exp[j(\omega_1 t + \varphi_1)]$ if these values are considered to be root mean square (rms) quantities for

power calculations). Again, $y(t)$ is found in the first case by projection on the real axis, that is, by taking the real part of symbolic representation to within a coefficient of 2.

1.1.2. Periodic signals

From sinusoidal signals, the case of periodic signals $y_T(t)$ with period T equal to $1/f_1$ can be generalized by performing the development as a Fourier series. Periodic signals of period T also constitute a vector space but of dimension $2N$ if signal reconstitution requires N sinusoidal signals of harmonic frequencies $f_1, 2f_1, 3f_1, 4f_1, \dots, Nf_1$. The series' convergence to $y_T(t)$ is made certain if N approaches infinity:

$$y_T(t) = \lim_{N \rightarrow \infty} \sum_{n=-N}^N c_n \exp \left[j \frac{2\pi n t}{T} \right]$$

where the coefficients are calculated by Fourier series decomposition:

$$c_n = \frac{1}{T} \int_{t_0}^{t_0+T} y_T(t) \exp \left[-j \frac{2\pi n t}{T} \right] dt$$

Since $y_T(t)$ is real, c_n and c_{-n} are complex conjugates (same module and opposite phase). Hence, the even and odd symmetries respectively for module spectrum $|c_n|$ and for that of argument $\text{Arg}\{c_n\}$.

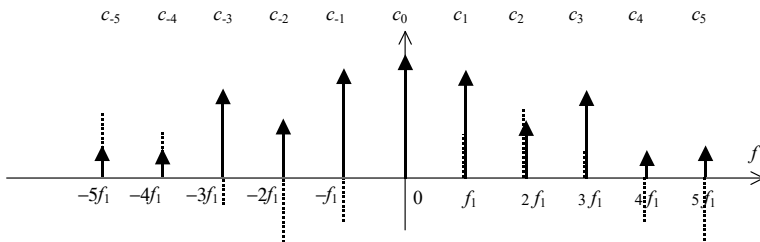


Figure 1.3. Spectrum of a periodic signal of repetition frequency f_1 (modules in bold and arguments in dotted lines)

By merging the conjugated terms, the real series can be written as:

$$y_T(t) = c_0 + \lim_{N \rightarrow \infty} 2 \sum_{n=1}^N |c_n| \cos \left[\frac{2\pi n t}{T} + \phi_n \right] \text{ where } \phi_n = \text{Arg}\{c_n\}$$

1.1.2.1. Power of a periodic signal

Power (average energy over time) is calculated by Parseval's rule, which shows that this energy is independent of time or frequency representation (to within the factor R or $1/R$) and is obtained by a scalar product of the signal by itself:

$$S = \frac{1}{T} \int_{t_0}^{t_0+T} [y_T(t)]^2 dt = \sum_{n=-\infty}^{\infty} |c_n|^2$$

No $c_n c_{n'}$ term with $n \neq n'$ appears since the basis of the vector space is orthogonal (scalar products of all basis vectors are zero unless $n = n'$). It should be noted that $|c_n|^2 = c_n \overline{c_n}$ and in the frequent event where power is calculated from a complex voltage U or a current I , using $\sum_{n=-\infty}^{\infty} U_n \overline{U_n}$ or alternatively $\sum_{n=-\infty}^{\infty} I_n \overline{I_n}$ if U_n and I_n represent, respectively, the $u(t)$ and $i(t)$ complex Fourier series decomposition coefficients.

1.1.3. Non-periodic real signals and Fourier transforms

If the signals are non-periodic, it can be assumed that the period T of the signals approach infinity on condition that $\lim_{T \rightarrow \infty} T c_n$ is convergent (the signals have to approach zero for $t \rightarrow \pm\infty$), replacing discrete variable n/T by continuous variable f (frequency) and thus defining the Fourier transform of $y(t)$ by : $\lim_{T \rightarrow \infty} T c_n = \int_{-T}^T y(t) \exp[-j2\pi ft] dt$ thus:

$$\text{FT}[y(t)] =$$

$$Y(f) = \int_{-\infty}^{\infty} y(t) \exp[-j2\pi ft] dt = \int_{-\infty}^{\infty} y(t) \cos[j2\pi ft] dt - j \int_{-\infty}^{\infty} y(t) \sin[j2\pi ft] dt$$

The symmetry properties are the same as for c_n since $y(t)$ is assumed to be a real function:

$$Y(-f) = \overline{Y(f)}$$

By changing variable t to $-t$, only the sine term changes sign thus providing:

$$\text{FT}[y(-t)] = \overline{Y(f)}$$

Obtaining $y(t)$ by means of the reverse FT calculated from the Fourier series by approaching the limit $y(t) = \lim_{T \rightarrow \infty} \sum_{n=-\infty}^{\infty} Tc_n \exp\left[j \frac{2\pi nt}{T}\right] \frac{1}{T}$ by replacing $\lim_{T \rightarrow \infty} Tc_n$ by $Y(f)$, n/T by f , $1/T$ by df and the total by an integral:

$$\text{FT}^{-1}[Y(f)] = y(t) = \int_{-\infty}^{\infty} Y(f) \exp[j2\pi ft] df$$

Other properties of the FT are as follows:

– The FT and the reverse transform are linear applications:

$$\text{If } y(t) = ay_1(t) + by_2(t) \quad \text{FT}[y(t)] = a\text{FT}[y_1(t)] + b\text{FT}[y_2(t)]$$

$$\text{If } Y(f) = \alpha Y_1(f) + \beta Y_2(f) \quad \text{FT}[Y(f)] = \alpha\text{FT}[Y_1(f)] + \beta\text{FT}[Y_2(f)]$$

– Derivation and integration of $y(t)$:

$$\begin{aligned} \text{If } Y(f) &= \text{FT}[y(t)], \quad \text{FT}\left[\frac{dy}{dt}\right] = j\omega Y(f) = j2\pi f Y(f) \quad \text{and} \quad \text{FT}\left[\int y(t) dt\right] \\ &= \frac{Y(f)}{j\omega} = \frac{Y(f)}{j2\pi f} \quad (\text{integration by parts of the definition where } y(t) \text{ is replaced} \\ &\text{by } dy/dt). \end{aligned}$$

– Delay theorem:

$$\text{If } Y(f) = \text{FT}[y(t)], \quad \text{FT}[y(t - t_0)] = \exp(-j2\pi ft_0) Y(f)$$

The phase alone is modified (phase delay if $t_0 > 0$ is a time delay) and not the transform modulus.

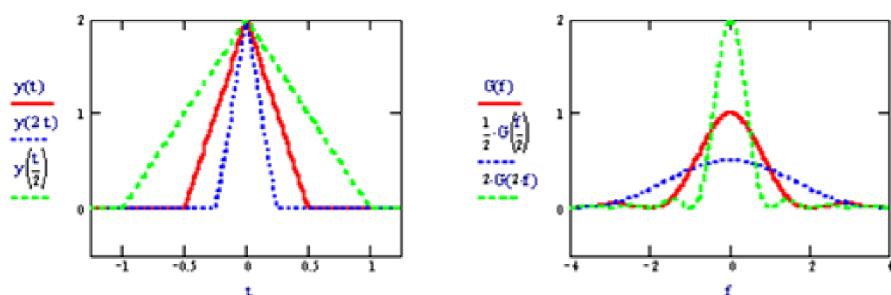


Figure 1.4. Triangular signals (left) and their spectrum (FT) (right). For a color version of this figure, see www.iste.co.uk/muret/electronics2.zip

– Similarity and dilatation/contraction of time/frequency scales:

If $Y(f) = \text{FT}[y(t)]$, $\text{FT}[y(\alpha t)] = \frac{1}{|\alpha|} Y\left(\frac{f}{\alpha}\right)$ (obtained by changing variable $t' = \alpha t$ in the definition, α real), as illustrated in Figure 1.4.

– Ordinary product of two functions and convolution product

$$\text{If } Y(f) = \text{FT}[y(t)] \text{ and } X(f) = \text{FT}[x(t)], \quad Z(f) = \text{FT}[y(t) x(t)] = \int_{-\infty}^{\infty} Y(f - \nu) X(\nu) d\nu = \int_{-\infty}^{\infty} Y(\nu) X(f - \nu) d\nu$$

and $\text{FT}^{-1}[Y(f) X(f)] = \int_{-\infty}^{\infty} y(t - \tau) x(\tau) d\tau = \int_{-\infty}^{\infty} y(\tau) x(t - \tau) d\tau$ or alternatively

$$\text{FT}\left[\int_{-\infty}^{\infty} y(t - \tau) x(\tau) d\tau\right] = \text{FT}\left[\int_{-\infty}^{\infty} y(\tau) x(t - \tau) d\tau\right] = Y(f) X(f)$$

– Wiener–Kinchine and Parseval theorems:

$$\text{FT}^{-1}\left[|Y(f)|^2\right] = \text{FT}^{-1}\left[Y(f) \overline{Y(f)}\right] = \int_{-\infty}^{\infty} y(\tau) y(-t + \tau) d\tau = \int_{-\infty}^{\infty} y(t) y(t - \tau) dt, \quad \text{the}$$

autocorrelation function of $y(t)$ (after reversing the names of variables t and τ) or rather $\text{FT}\left[\int_{-\infty}^{\infty} y(t) y(t - \tau) dt\right] = |Y(f)|^2$.

The autocorrelation function $C_{yy}(\tau) = \int_{-\infty}^{\infty} y(t)y(t-\tau)dt$ measures the degree of resemblance between the function and delayed function. Unlike the convolution product, the integration variable operates with the same sign in both factors under the integral symbol.

$$\text{FT} \left[\int_{-\infty}^{\infty} y(t)y(t-\tau)dt \right] = |Y(f)|^2 \text{ is the Wiener-Kinchine theorem stating that}$$

the FT of the autocorrelation function of $y(t)$ is equal to the squared modulus of the FT of $y(t)$. This autocorrelation function may be calculated not only for known (or determined) signals but also for random signals such as noise defined only by a density of probability.

$$\text{FT}^{-1} \left[|Y(f)|^2 \right] = \int_{-\infty}^{\infty} |Y(f)|^2 \exp(j2\pi f \tau) df = \int_{-\infty}^{\infty} y(t)y(t-\tau)dt \quad \text{is rewritten}$$

simply for $\tau = 0$:

$$\int_{-\infty}^{\infty} |Y(f)|^2 df = \int_{-\infty}^{\infty} [y(t)]^2 dt$$

This is the Parseval theorem that allows us to perform the energy calculation (clearly visible in the second member to within R or $1/R$ coefficient if y is, respectively, a current or a voltage) both in the time domain from $y(t)$ and in the frequency domain from $Y(f)$. Thus, we can proceed to $\int_{-\infty}^{\infty} |Y(f)|^2 df$, which is an energy with $|Y(f)|^2$ a spectral energy density in J/Hz; $|Y(f)|$ a spectral density of current or voltage within R or $1/R$ factor in A/Hz^{1/2} or in V/Hz^{1/2}.

1.2. Representations of linear and stationary systems and circuits built with localized elements

1.2.1. Representation using ordinary differential equation

Electric or electronic circuits built with localized elements are those featuring elements in which instantaneous currents (and voltages) are the same irrespective of the location considered in a conductor. Accordingly, it can be assumed that the wavelength of these currents, voltages and