System of Transformations for DSP design

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Abstract: Many mathematical transformations are used in engineering. Mathematical transformations are of paramount importance in the design of digital signal processing algorithms. Effective engineering can only be done if it is scientifically supported. Digital signal processing algorithms are mostly designed and verified in the complex frequency domain and in the time domain. However, distortion of some planes may be justified to achieve effective results. In this article, a complete transformation system is presented, taking care to create a transparent and uniform notation system. The article also presents the relationships between each transformation and the mapped planes. Additionally,the steps of DSP algorithm design implemented on each plane, in the case of algorithms with direct structure and wave digital structure.

Keywords: Digital Signal Processing; DSP; Transformation; complex frequency; time domain

1 Introduction

In some fields, "engineering knowledge" is a prerequisite. In electrical engineering, knowledge of Ohm's law is trivial, but knowledge of Maxwell's equations is also expected. Knowledge and use of basic transformations are also trivial in many cases, such as the Fourier and Laplace transforms. However, specialized fields interpret and apply these transformations somewhat differently. This is because in some cases these transformations and the calculations related to them [1] [2] cannot be defined in finite form.

1.1 Well known formulas – Laplace and Fourier

A well-known transformation for engineering calculations in classical design processes is the Laplace transform [3]. It is used to transform functions from the time domain to the complex frequency domain, and its inverse is used to transform functions from the complex frequency domain to the time domain.

In general, the transformation equation is:

$$X(s) = \int x(t) \cdot e^{-s \cdot t} \cdot dt \tag{1}$$

In relation (1), “s” denotes the complex independent frequency variable in complex frequency plane:

$$s = \sigma + j \cdot \omega \tag{2}$$

The search function (sometimes called “mother function”) in transformation (1) can also be written in the following form:

$$e^{s \cdot t} = e^{\sigma \cdot t} \cdot e^{j\omega \cdot t} \tag{3}$$

In the quasi-stationary case, when the effects of on- and off-state phenomena are negligible, the value of σ=0 in equation (3), so the search function simplifies to the expression $e^{j\omega \cdot t}$, so the transformation equation (1) becomes as follows:

$$X(\omega) = \int x(t) \cdot e^{-j \cdot \omega \cdot t} \cdot dt \tag{4}$$

The resulting relation (4) is called the Fourier transform.

In engineering practice, the aforementioned transforms (1) and (4) can be effectively used for the design and analysis of linear networks, since the convolution operations performed in the time domain are simplified to simple multiplication in the complex frequency domain.

1.2 Description of harmonic signals as the sum of two rotating vectors (phasors)

The basic operation of Fourier analysis (4) can be easily interpreted as follows: every harmonic oscillation on the complex frequency plane is formed by the sum of two vectors [7] rotating in opposite directions but with the same absolute value of the angular frequency ω (5) (6).

Description of the cosine function with unit amplitude:

$$\cos(\omega_1 \cdot t) = \frac{1}{2} \cdot e^{j \cdot \omega_1 \cdot t} + \frac{1}{2} \cdot e^{-j \cdot \omega_1 \cdot t} \tag{5}$$

The sum of the two rotating vectors provides the real value of the cosine function at each instant of time. This is illustrated in Fig. 1.

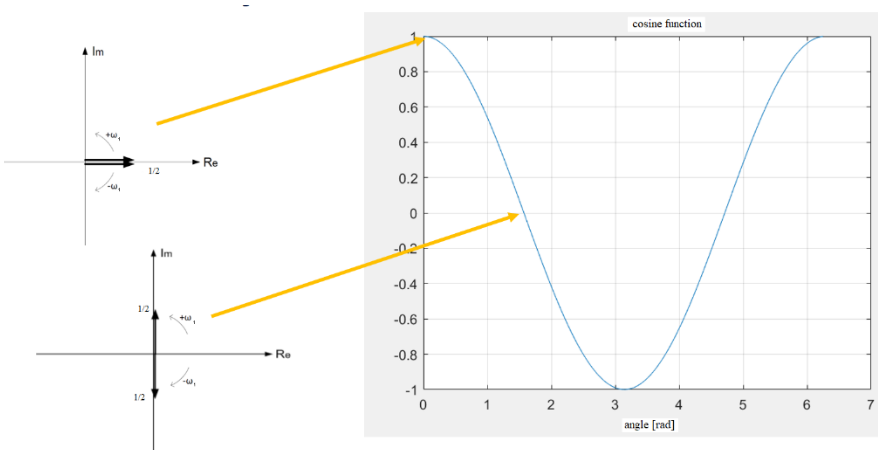

Figure 1
Cosine function and two rotated vector (5)

A sinusoidal signal of unit amplitude can also be described by the sum of two rotating vectors:

$$\sin(\omega_1 \cdot t) = -j \cdot \frac{1}{2} \cdot e^{j \cdot \omega_1 \cdot t} + j \cdot \frac{1}{2} \cdot e^{-j \cdot \omega_1 \cdot t} \tag{6}$$

The rotating vector decomposition, as a mathematical model, well confirms the negative spectrum obtained during Fourier analysis.

Fig 2 illustrates how the spectrum of an examined cosine function is obtained during the analysis:

$$\frac{A}{2} e^{-j\omega_1 t} + \frac{A}{2} e^{j\omega_1 t} = A \cdot \cos(\omega_1 \cdot t)$$

$$X(\omega) = \int_{-\infty}^{+\infty} \left(\frac{A}{2} e^{-j\omega_1 t} + \frac{A}{2} e^{j\omega_1 t}\right) \cdot e^{-j\omega t}\, dt$$

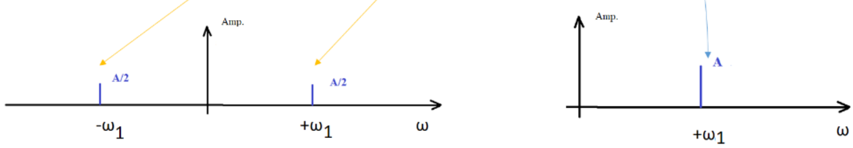

Figure 2
Cosine function and its real and calculated spectrum

“A” is the amplitude of the cosine function in Fig2.

2 Transformations for continuous systems

In the case of continuous systems, the Laplace and inverse Laplace transforms (1) have provided an almost complete toolkit for linear network design. Tasks that are difficult to perform in the time domain – mainly due to the high computational requirements and the accumulation of computational inaccuracies – for example, to induce convolution, we transform the weight function characterizing the network to the complex frequency plane, thus obtaining the complex transfer function of the network.

However, for example, in the case of robotics and mechatronics systems, description and modeling in the time domain is common [4] [5] [6] [13] [18].

Given the weight function of a linear system, after fixing the initial conditions, the output signal response to an input signal can be calculated using convolution (7) [8].

$$Out(t) = \int_{-\infty}^{+\infty} In(\tau) \cdot k(t-\tau) \cdot d\tau \tag{7}$$

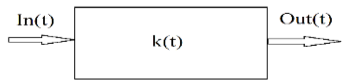

Figure 3
Interpretation of convolution in time domain

In the figure above, k(t) symbolizes the weight function of the linear system.

Transforming the weight function describing the input signal and the operation of the network in the time domain to the complex frequency plane, we obtain:

$$\mathscr{L}\{In(t)\} = In(s)$$

$$\mathscr{L}\{k(t)\} = H(s) \tag{8}$$

In expression (7), the Laplace transform of the weight function is H(s), which represents the complex transfer function of the network.

Based on all this, the output response signal can be calculated as follows (9):

$$In(s) \cdot H(s) = Out(s) \tag{9}$$

Based on the above, the transformation system and planes required for the design of continuous-time systems are illustrated in the following figure:

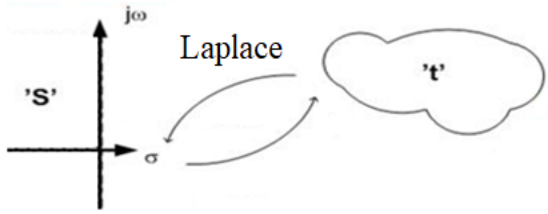

Figure 4
Transformation systems of analogue linear circuits

3 Discrete systems

The basis of the operation of discrete-time systems is that they interpret the characteristics of their environment in the time domain only at the sampling times. No information is available between the sampling times. We can obtain a sampled, discrete-time system from continuous time by sampling.

3.1 Sampling and the impact of sampling

Sampling from a continuous-time signal can be interpreted as taking a measurement at intervals of "T", i.e. taking a sample from it. "T" is called the sampling time and its reciprocal is called the sampling frequency "fs".

Sampling is performed according to Shannon's sampling theorem – if we want to process baseband signals. According to this, the minimum value of the sampling frequency (fs) is given by the following rule:

$$f_s \geq 2 \cdot f_{max} \tag{10}$$

In relation (9), the value of fmax refers to the highest frequency component occurring in the signal to be processed.

As a result of sampling, only discrete time instants are interpreted in the time domain, and no information is available between the sampling time instants. As a result, the imitating spectral components appear above the frequencies fmax/2, which is indicated by the periodicity of the "S" plane.

In relation (9), the value of fmax refers to the highest frequency component occurring in the signal to be processed.

As a result of sampling, only discrete time instants are interpreted in the time domain, with no information available between the sampling time instants. As a result, overlapping spectral components appear above frequencies fmax/2, which is indicated by the periodicity of the "S" plane.0

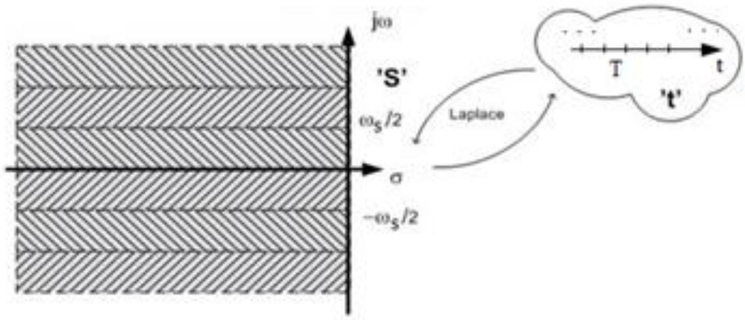

Figure 5
After sampling, the "S" plane becomes periodic

In Figure 5, meaning $\omega_s = 2 \cdot \pi \cdot f_s$.

3.2 Discrete transformations

The Laplace transform relation (1) defined in continuous time now becomes as follows in the discrete case (11):

$$X(s^*) = \sum_{n=-\infty}^{+\infty} x(n) \cdot e^{-s \cdot n \cdot T} \tag{11}$$

After sampling, the samples are only available at the time of sampling.

In relation (11), the index "n" denotes the sample number [7].

Let us use the following formal substitution option (12) and thus the complex variable "z" is introduced:

$$z = e^{s \cdot T} \tag{12}$$

By formally introducing the complex variable "z", we arrive at the "Z" transformation (13). "z" and the plane thus mapped have also undergone a normalization, so that the "z" plane becomes a plane free of physical dimensions.

$$X(z) = \sum_{n=-\infty}^{+\infty} x(n) \cdot z^{-n} \tag{13}$$

The basic transformation scheme of discrete systems is summarized in the following figure:

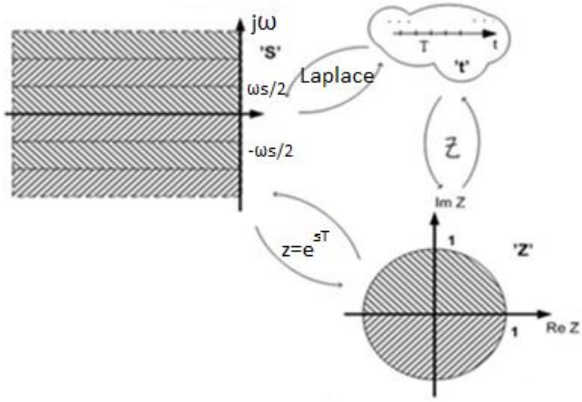

Figure 6
The basic transformation scheme of discrete systems

It should be noted here that the summation (11) and (13) can be performed in a finite time window, in which case the problem of windowing must be addressed [11].

4 Discrete systems transformations – DSP design planes

After sampling, the periodicity of the "s" plane does not allow the application of traditional circuit design methods. However, all important information is found in the range ω=0 … ω=ωs/2.

Eliminating this periodicity is an effective solution for making traditional design methods applicable once again. We must define a transformation that eliminates periodicity. However, the conditions for using traditional design methods are the followings:

- The transformation should be origo-preserving;
- The transformation should be axis-preserving, meaning that the values located on the ω axis of the "s" plane should be transformed to an axis of the new plane;
- The transformation shifts the value ωs/2 to positive infinity, while the value - ωs/2 to negative infinity.

The above conditions are satisfied by the following relation:

$$\psi = \frac{1-e^{-s\cdot T}}{1+e^{-s\cdot T}} \qquad (14)$$

The new plane, eliminating periodicity, is therefore denoted by "Ψ".

It should be noted here that this transformation formula (14) also eliminates physical dimensions when moving from the "s" plane to the "Ψ" plane, in contrast to the formula $\Psi = \frac{2}{T} \cdot \frac{1-e^{-st}}{1+e^{-st}}$ which preserves the [1/s] dimension in relation to the independent variable.

The independent complex variable of the new complex plane thus created is:

$$\psi = \alpha + j\Omega$$

Checking the origo-preserving:

In case of s=0 the $\psi = \frac{1-e^{-0 \cdot T}}{1+e^{-0 \cdot T}} = 0$, so origo-preserving is fulfilled.

Let us examine the case when σ=0 on the "s" plane. Then the transformation (14) becomes the following (15):

$$\psi = \frac{1-e^{-j\omega \cdot T}}{1+e^{-j\omega \cdot T}} \tag{15}$$

Let's multiply both the numerator and denominator of equation (15) by (16):

$$e^{j\omega \cdot \frac{T}{2}} \tag{16}$$

Then this is the result (17):

$$\psi = \frac{1-e^{-j\omega \cdot T}}{1+e^{-j\omega \cdot T}} \cdot \frac{e^{j\omega \cdot \frac{T}{2}}}{e^{j\omega \cdot \frac{T}{2}}} = \frac{e^{j\omega \cdot \frac{T}{2}} - e^{-j\omega \cdot \frac{T}{2}}}{e^{j\omega \cdot \frac{T}{2}} + e^{-j\omega \cdot \frac{T}{2}}} \tag{17}$$

Based on the previously defined relationship (5):

$$\cos\left(\omega \cdot \frac{T}{2}\right) = \frac{1}{2} \cdot e^{j\omega \cdot \frac{T}{2}} + \frac{1}{2} \cdot e^{-j\omega \cdot \frac{T}{2}} \tag{18}$$

$$\sin\left(\omega \cdot \frac{T}{2}\right) = j \cdot \left(-\frac{1}{2} \cdot e^{j\omega \cdot \frac{T}{2}} + \frac{1}{2} \cdot e^{-j\omega \cdot \frac{T}{2}}\right) \tag{19}$$

By applying (18) and (19), equation (17) becomes as follows after the conversions:

$$\psi = \alpha + j\Omega = j \cdot \frac{\sin\left(\omega \cdot \frac{T}{2}\right)}{\cos\left(\omega \cdot \frac{T}{2}\right)} \tag{20}$$

It can be seen from relation (20) that the quotient of the sinus and cosine functions always yields a real value, so this quotient, when multiplied by "j", always yields a purely imaginary result, i.e. the real part of the "Ψ" plane, "α", will always be zero. This proves that transformation (14) is axis-holding, i.e. it transforms the jω axis of the "s" plane into the jΩ axis of the "Ψ" plane.

Based on the above and using the known $tg(x) = \frac{\sin(x)}{\cos(x)}$, the axis keeping between "ω" and "Ω" is fulfilled according to the relation (15), the "jω" axis of the "s" plane is transformed to the "jΩ" axis of the "Ψ" plane (21).

$$\Omega = tg(\omega \cdot T/2) \tag{21}$$

It is advisable to check the elimination of periodicity, that is, whether the value ωs/2 goes to positive infinity, while the value - ωs/2 goes to negative infinity.

We can do this with a case study by substituting half of the sampling frequency, which in this case is π/T, insert into equation (21).

The expression (21) is hereinafter called the tangent transformation, and the relationship between jω and jΩ is illustrated in the following figure.

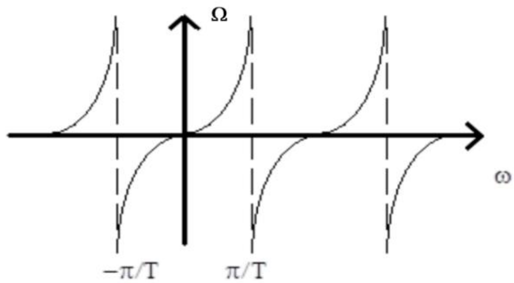

Figure 7
Tangent transformation, connects Ω and ω

The transformation between the "Ψ" and "z" planes is determined from the results so far (12) and (15):

$$\psi = \frac{1-z^{-1}}{1+z^{-1}} \tag{22}$$

Thus, the entire transformation system is available for DSP algorithm design, with the help of which both direct-structure and wave-digital-structure algorithms can be designed.

The relationships between the entire transformation system and the individual planes are illustrated in the following figure.

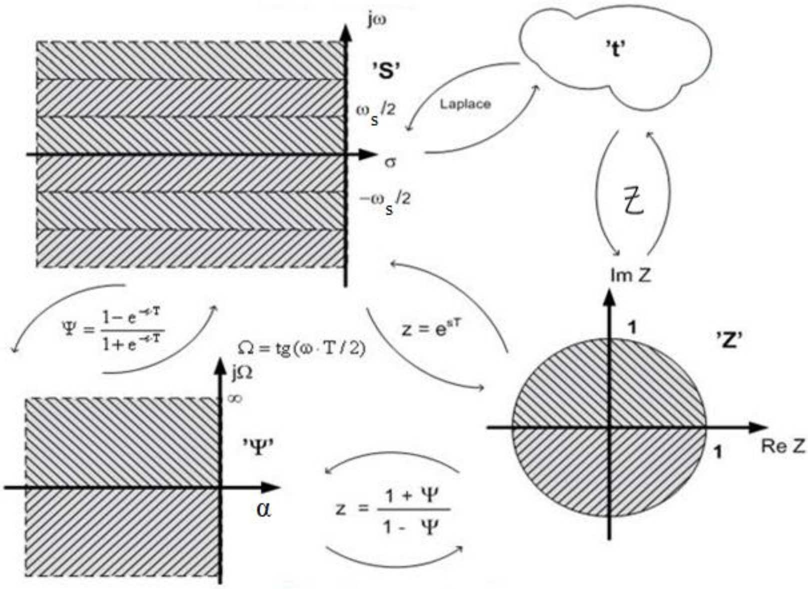

Figure 8

Systematisation of transformations

The design steps to be performed on each plane and examples of the application of transformations are presented in the next chapter.

5. DSP algorithm design steps at each transformation plane

After the entire transformation system has been built, this article organizes the planes on which the main design steps can be implemented. The main steps are numbered in the Figure 9.

I should note here that DSP algorithms are primarily studied in the context of infocommunications and electrical engineering, but at the same time, the application of this system in robotics research [5], [6] may also be promising. Due to space limitations, the article does not discuss this aspect, but its importance is undeniable nowadays.

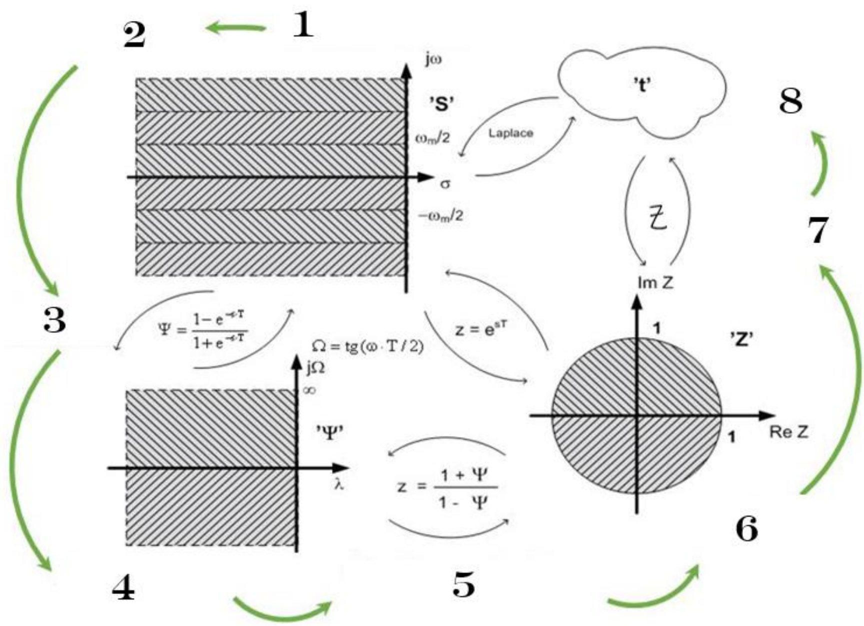

Figure 9
DSP design step marking

The numbers indicated in the figure above will be used in the following for the design steps.

The design steps are interpreted in two design methodologies:

• Design of direct structure algorithms;

• Design of wave digital algorithms.

In the case of the above two methodologies, there are common steps, which are discussed in the next subsection.

Here I must draw the attention of the Dear Reader that in the case of robotic systems, and machine learning modeling is often done in the time domain [17] but this does not exclude the applicability of the described transformation system.

The examples presented in this paper primarily discuss the applicability of the transformation system from an info-communication and circuit design perspective.

5.1 Common design steps

The design often starts from the requirement given on the complex frequency plane ('S' plane 1st tepp at Figure 9.) In the case of filter design, if the task is not to create a low-pass filter algorithm, then it is necessary to convert it into a low-pass requirement by a so-called "type transformation" [19], a general example of which is illustrated in the following figure:

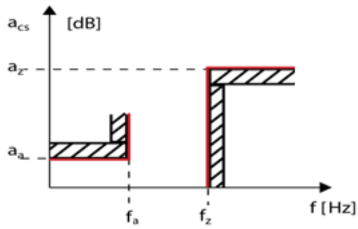

Figure 10.
Low-pass requirement on frequency domain

In the above example, the requirement is defined for a quasi-stationary state, in which case σ=0.

We need to define the operating frequency of the DSP algorithm, i.e. the sampling frequency of the system, the method and effect of which are given in subsection 3.1 of this article. Based on this, it can be seen that the 'S' plane becomes periodic, which affects the requirement shown in Figure 10 in the following way:

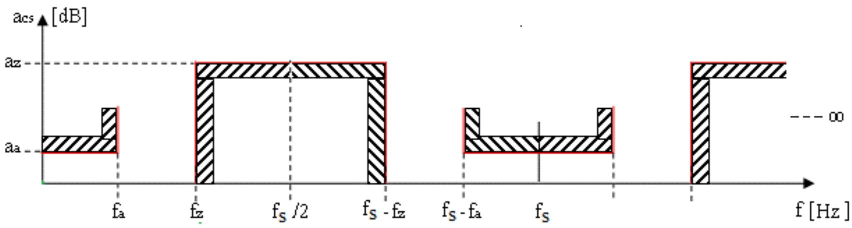

Figure 11
Low-pass requirement on frequency domain after sampling

In order to proceed with the design, periodicity must be eliminated by applying transformation (14), which in this case can be simplified by tangent transformation (21). This is illustrated in the following figure:

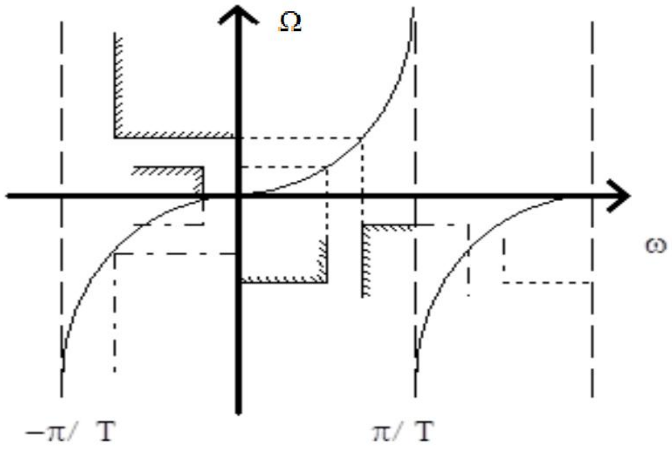

Figure 12
Eliminate the periodicity of “S”, using tangent transformation

The transformation (21) is presented in a different, more visual way in the following figure:

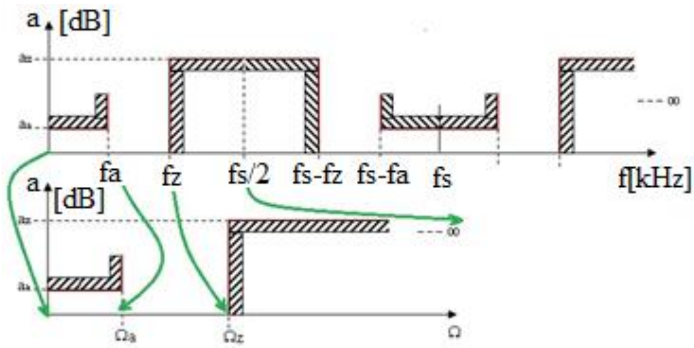

Figure 13
Eliminate the periodicity of “S”, using tangent transformation, the periodic free physical dimensionless requirement is available at ‘Ψ” – in this simplified case-study at ‘Ω’.

The transformation shown in Figures 12 and 13 does not modify the attenuation values of the requirement. The transformed requirement only contains numbers without physical dimension. After this requirement transformation, the further steps depend on whether we are designing a direct structure algorithm or a wave digital algorithm.

The task of this article is to systematize the transformations, so the further design steps are only at the list level in the next two subsections.

5.2 Direct structure DSP design

Starting from the requirement transformed to the ‘Ψ’ plane, the task (Fig 9) is to determine a transfer function defined on the “z” plane.

To do this, in the first step, a transfer function satisfying the requirement of the ‘Ψ’ plane (Fig. 13.) is required (Fig. 9. step 4).

Then, in the 5th step, applying equation (23), which can be expressed from equation (22), we obtain the transfer function H(z).

$$z = \frac{1+\Psi}{1-\Psi} \tag{23}$$

The expected form for H(z) will be an N-th degree polynomial/polynomial form (24).

$$H(z) = \frac{d_0+d_1 \cdot z^{-1}+d_2 \cdot z^{-2}+\cdots+d_n \cdot z^{-N}}{1-c_1 \cdot z^{-1}-c_2 \cdot z^{-2}-\cdots-c_n \cdot z^{-N}} \tag{24}$$

The next step (Fig 9. step 6) takes place on the “z” plane, where the transfer function (24) is transformed into a graphical signal flow diagram.

The interpretation of H(z) is then:

$$H(z) = \frac{Yout(z)}{Xin(z)} \tag{25}$$

The general signal flow diagram simplified for the case of a 2nd degree solution:

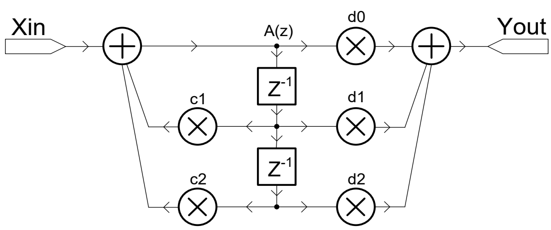

Figure 14

Transforming the transfer function H(z) (24) into a signal flow diagram in the 2nd order case

At this step, it should be noted that in the special case, when the multiplier constants of the feedback branch are zero, the feedback disappears. This means that the denominator of equation (24) becomes a constant one:

$$H(z) = d_0 + d_1 \cdot z^{-1} + d_2 \cdot z^{-2} + \cdots + d_n \cdot z^{-n} \tag{26}$$

This is actually the discrete-time convolution of equation (7) [8], where the constants “d” are samples of the algorithm’s weight function. With the disappearance of the feedback branches, we get the classic FIR structure:

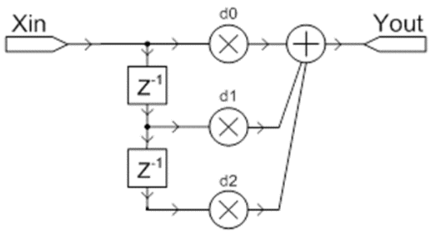

Figure 15

Transforming the transfer function H(z) (26) into a signal flow diagram in the 2nd order case

Step 7: Transform the DSP signal stream diagram from the “z” plane to the time domain. The values of the multiplier constants remain unchanged during the transformation.

In step 8, it is recommended to perform a time domain analysis of the algorithm [14] [16] and then transform the result to the complex frequency plane. This allows us to verify whether the designed algorithm can satisfy the requirement specified in step 1, as well as the stability of the algorithm [12].

5.3 Wave digital design

Starting from the requirement transformed to the ‘Ψ’ plane, the goal is to design a reference circuit that can satisfy this (Fig 10) requirement. The reference circuit will be passive, consisting exclusively of passive components [9].

A passive reference network is typically composed of resistance and lossless reactance components (inductance and capacitance) [10] [15].

The individual components are connected by gate-by-gate coupling and the components of the reference circuit are described by voltage waves, where the forward wave at the gate is (27):

$$A(\psi) = U(\psi) + I(\psi) \cdot R_G(\psi) \tag{27}$$

The reflected wave on the gate is (28):

$$B(\psi) = U(\psi) - I(\psi) \cdot R_G(\psi) \tag{28}$$

In equations (27) and (28), R_G is the gate resistance, which is a numerical value without physical dimension on the “Ψ” plane.

The connection of the individual components is matched with so-called adaptors, which provide either series or parallel connection.

As step 5 of Fig.9, the individual components described by voltage waves are transformed to the “z” plane using the transformation (23) derived from (22).

In step 6, a DSP signal flow diagram is generated from the components transformed to the “Z” plane.

In step 7, we transform the DSP signal flow diagram from the “Z” plane to the time domain. During the transformation, the values of the multiplication constants will remain unchanged.

In step 8, it is advisable to perform a time domain analysis of the algorithm [9] [11], and then transform the result to the complex frequency plane. This allows us to check whether the designed algorithm can satisfy the requirement specified in step 1.

In the design method, we take care to keep passivity at each step. A specific wave digital filter design is presented in a previous paper [20]

5.4 Uniform marking system

At the beginning of the article, it was promised to develop a uniform and transparent notation system and use it consistently. Here, as a summary, the notations used and their meanings are given.

A	forward voltage wave
A(Ψ)	forward voltage wave on Ψ plane
B	backward (reflected) wave
B(Ψ)	backward (reflected) wave on Ψ plane
fs	sampling frequency [1/s ≡ Hz]
H(s)	Transfer function {in complex frequency domain}
k(t)	weight function {in continuous time domain}
ω	angle frequency [rad/s]
Ω	distorted frequency at ‘Ψ’ plane, with no physical dimension
Ψ	Complex frequency at ‘Ψ’ plane, with no physical dimension
s	Comlpex frequency at ‘S’ plane
t	time [s]
T	Sampling time [s], reciprocal of fs
z	indeppendent comlex physical dimension free variable on “Z” plane

Conclusions

The aim of this paper was to organize the transformations that serve as the basis for the design of DSP algorithms in a transparent manner and to create a transformation unit using a clear notation system.

The article shows that uniform transformations can be used for the design of either direct structures or wave digital structures, although the design steps are different. It can be a precursor for the design of fuzzy systems and for the research of OSSEFS algorithms [21].

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