The Methods Of Synthesis And Matched Processing The Normal System Of Orthogonal Circle M-Invariant Signal

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ABSTRACT:

There is scientific work containing the recurrence method of synthesis the new class of orthogonal circle m-invariant signals; designed effective algorithms of fast-direct computing m-convolution in time domain; engineer methods of design economic scheme of decoders for optimal receiving in aggregate of suggested signal.

I. INTRODUCTION

The developments of signal theory are, today, very specially updated due to advantage achievements achieved in technology producing computing technique facilities, measure technique and information transmission, digital signal processing theory, technique and other fields as well. Multiform, that might be, of signal theory applications seems to be opposite science directions aiming to normalize signal requirements which as information carrier body in different applied systems. Perfection and accuracy of mathematic models are agreed to different requirements of real information processing and transmission [1].

In the signal theory and its properties orthogonal systems are of special interests. The main reason for these special interests is capability of improving system information characteristics of orthogonal signals. Essentially, the system of orthogonal functions is an optimum class of carrier signals. The use of orthogonal signal class allows to establish information transmission system with very highly interference voiding, almost approaching to marginal values. Analyzing the multi-channel information transmission systems shown that the use of orthogonal signals will have highly interference voiding, good energy characteristics and highly bandwidth usage factor. Periodical nature of signal classes is not only easy to form themselves but also reduce considerably memory capacity of decoding or digital signal processing circuits. The last worth matter to say is invariant nature m of synthesized signal class. When processing complex signal classes (i.e. Psuedo Noise)[2], the main reason that limit popularly use them in the real is complex technique level in forming optimum processing schemes, in particular, adaptive processing at receiver if the length of these signal classes $N=m^n$ are considerable.

Originated from the above mentioned matters, the problem put forward is to find out a synthesis method for Amplitude-phase-modulation signal classes which has ideally relative characteristic (orthogonal) allowing to use of fast digital convolution computing algorithm to minimize technique complex level in case of entire decoding. After general synthesis of fast decoding algorithms allows to suggest basically modern proposals to construct fast digital convolution computing algorithm for synthesized signal class. That is:

1. Analyzing structure characteristics of base functions allows to use fast transformation algorithm, orthogonal, in particular the base functions Valenkin – krestienson[3].
2. Considering structure characteristics of special discrete signal classes, i.e. the signals Sirtev, Haffmer, etc.
3. Considering cyclic shift mode of signal processing in which always has extra computation (repeated coincide) not dependence to pre-shown detail structure of the signal (the coincide of bunch of in put signals due to cyclic shift mode generating).

The article also investigates problems of constructing coordinated algorithms to enable fast digital convolution computing operation. Base on the above mentioned matters the article have provided a method to synthesize orthogonal signal class, period, special structure suitably coping to the base of fast discrete orthogonal transmissions (FDOT). With this class of signal, cyclic convolution vector coincides to the convolution vector m and in special case when m=2, coincides to binary convolution.

II. PHYSIC NATURE AND CHARACTERISTIC OF VECTORS.

Approaching to the purpose and requirements introduced, the article has investigated physic nature and characteristics of cyclic convolution vectors – Z and convolution vector m - L those are defined as:

+ Cyclic convolution:

$$Z(\tau) = \sum_{i=0}^{N-1} X(i)h(\tau - i);\tau = \overline{0, N-1}. \qquad (1)$$

+ Convolution m:

$$l(\tau) = \sum_{i=0}^{N-1} X(i).h(\tau \underset{m}{\Theta} i);\tau = \overline{0, N-1} \qquad (2)$$

wherein: $X(i), \overline{i = 0, N-1}$, N=m^m - are input stream.

$h(i), \overline{i = 0, N-1}$, N=m^m - impulse respond of the filter. m, the order of shifting.

In the matrix form, (1) and (2) may be in the form:

$$\begin{cases} Z = X.C \\ L = X.M \end{cases} \qquad (3)$$

wherein, X – vector of in put stream, C- cyclic matrix (period N), M – shift m matrix
in which each next stream of matrix is the result of shifting – m of previous stream.
To have Z ≡ L, it is previously that

$$C \equiv M \qquad (4)$$

Carefully Investigating the structure of cyclic and shifting m matrixes has found series individual natures of them and clearly proved an axiom about structure of generating code sequence (GCS) $\{h(i)\}; i = \overline{i = 0, N-1}$.

Axiom: *Cyclic Matrix C and shifting matrix M of order N=m^n is formed in the base GCS $\{h(i)\}; i = \overline{i = 0, N-1}$, N=m^n by way of shifting correlatively N or m coincided elements if elements of $\{h(i)\}$ satisfy the condition:*

$$h(im^{K-1}) = h(im^{K-1} - lm^{n-K}), K = \overline{1, n-1};$$
$$l = \overline{1, m^{n-k} - 1}$$
$$h(im^{n-1}), i = \overline{0, m-1}, \text{ unconstraint.} \qquad (5)$$

The limited system (5) determines the law of division GCS into φ subset of similar elements in which, φ = n(m-1)+1.

Beside the invariant characteristic with shifting m, GCS must has ideal correlation characteristic (orthogonal):

$$R(\tau) = \sum_{i=0}^{N-1} h(i)h(\tau + i) = 0, \tau = \overline{1, N-1} \qquad (6)$$

The formulas (5) and (6) are mathematical model of synthesis method GCS. This model, in general, is not programming model but the model for collection method. Researching the structure characteristics of GCS and characteristics of correlative matrix allowing to introduce a synthesis algorithm GCS, invariant with shifting m which have ideal correlative characteristic AKF=0.

With m=2, we have root solution:

$$X_1 = \alpha$$
$$X_2 = \pm X_1$$
$$X_3 = -X_2 \pm 2X_1$$

$$\begin{cases} X_i = -(2^{i-3}X_2 + 2^{i-4}X_3 + \ldots + X_{i-1}) \pm 2^{i-2}X_1 \\ \ldots\ldots \\ X_n = -(2^{n-3}X_2 + 2^{n-4}X_3 + \ldots + X_{n-1}) \pm 2^{n-2}X_1 \\ X_{n+1} = -\left(2^{n-2}X_2 + 2^{n-3}X_3 + \ldots + 2X_{n-1} + X_n\right) \end{cases}$$

$$X_4 = -2X_2 - X_3 \pm 4X_1$$
$$X_5 = -4X_2 - 2X_3 - X_4 \pm 8X_1$$
$$\ldots$$

For simplicity in expression the synthesis algorithm, we assume the following symbols:

$$h(2^{K-1}) = h(2^{K-1} + 2^K l) = X_K, K = \overline{1, n-1},$$
$$l = \overline{1, 2^{n-K} - 1}$$
$$h(i2^{n-1}) = X_{n+1}, i = \overline{0,1}, \text{ unconstraint.} \qquad (8)$$

For an example: m=2, n=3, N=2^3=8 we have limited system (5):

$h(1)=h(3)=h(5)=h(7)=X_1$
$h(2)=h(6)=X_2$
$h(0)=X_3$; $h(1)=X_4$.

One of the solutions from system (7) is:

$X_1=\alpha$, $X_2=-\alpha$, $X_3=3\alpha$, $X_4=-\alpha$; with α is any integers number. With $\alpha=1$ we have:

$$\{GCS\}=\{h(i)\}=\{3,1,-1,1,-1,1,-1,1\}$$

For this way, each of GCS is a modulation function of signal, it allows to establish a normal system of N dimensions of orthogonal, period and invariant signals with shifting m (length N). By inversing phase GCS we will have a biorthogonal system. This is multi-dimension signal class having optimum packet factor. With this class of signal, peak factor $1< v<N/3$.

III. CONCLUSION

Synthesized signal class allow us to compute convolution not only by FDDT algorithm but also by fast direct in time domain. There are two proposed algorithms fast direct computing vektor Z in time domain FDM - 1 and FDM -2

The essence of FDM-1 is cyclic shifting operation, parallel by the time, all positions of cyclic convolution vector with respect to the period nature, times use of sub-total (medium calculation results) the products of invariant m of generating code sequences (GCS). The complex of this method is:

$$Q_{FDM-1}=[(m+1)N-m]_{\oplus}+[(m+2)N-3m]_{(.)} \quad (9)$$

The essential of FDM-2 is making twice the received in put signal to produce two period of itself. Then, the signals of whole normal system (N) are serial terms in the bound of 2 cycles of any signal in the system. Therefore, to decode normal system of cyclic shift signal instead of N channel filter we use single channel filter fitting for any signal of system N signal. Effective of this method shown that, to compute each of cyclic convolution operations it must be done:

$$Q_{FDM-2}=[n(m-1)+1]_{(.)}+[(2n+1)(m-1)]_{\oplus} \quad (10)$$

Then, with the method of direct convolution computing, normal system of signal requires:

$$Q_{DM}^{(N)}=N^2{}_{(.)}+N(N-1)_{\oplus} \quad (11)$$

To insist advantage and economic effectiveness of algorithms FDM-1 and FDM-2 we introduce the concept: economic effectiveness:

$$\gamma_{AND}=N^2/Q_{(.)} \qquad \gamma_{OR}=N(N-1)/Q_{\oplus} \quad (12)$$

Performing engineering method in design special decoding circuits, economic with the time in the form of serial and parallel (shifting operation) for entirely decoding orthogonal, period and invariant m signals. After analysis series of schemes, we insist the serial decoding circuits has highest economic effective, with *m not* so great, increasing in technique complex (the number of AND and OR elements), allow to reduce personal by rate: $\log_2 N$, in which, $N=2^n$, means that $\gamma \approx N^2/\log_2 N$.

References

[1] Kotelnikov UA. Theory of Potencjal anti-interference.

[2] Sverdlic MV. Oftimal diserit signals.

[3] Varakin LE. Theory of Pseudo- Noise Signals