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A NEW APPROACH TO TRANSMULTIPLEXER IMPLEMENTATION

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Abstract. A new approach to the transmultiplexer implementation is presented, which falls into the class of methods that do not use FFT-type processors. It is based on the concept of baseband analytic signal and on its successive allocation in the FDM format by appropriate digital interpolation and filtering. The main advantages of this method are the absence of digital modulators and the large transition bandwidths allowed to the digital filters that can, thus, be designed with a smaller number of coefficients.

Finally, some preliminary results are discussed showing the saving in the computational complexity achieved by this method with respect to other non-FFT approaches.

1. INTRODUCTION

In present telecommunication networks the two standard techniques of signal multiplexing, i.e. frequency-division (FDM) and time-division multiplexing (TDM), coexist and the conversion between the two signal formats is commonly achieved by back-to-back connection of two FDM and TDM terminals, regenerating the analog baseband version of each multiplexed signal. In recent years, however, considerable attention has been devoted to the implementation of the conversion process by digital signal processing at the multiplex level (without recovering the baseband analog signal) and this solution can be competitive or even more convenient in the next future due to the decreasing cost and the increasing processing rate of digital integrated circuits.

Many approaches to an all-digital TDM-FDM translator (transmultiplexer) have been proposed since Darlington's paper in 1970 (Ref. 1). They can be grouped in two general classes: FFT and non-FFT methods, depending respectively on the presence or absence of a FFT-type processor (Refs. 2-3).

This paper presents in Sect. 2 a new approach within the second class to a digital implementation of a TDM-FDM transmultiplexer, which avoids the use of digital modulators and is based upon the generation of a (complex) SSB baseband signal (analytic signal) and on its successive allocation in one of the multiplex format bands by digital interpolation and (complex) bandpass filtering. Finally, it is shown how this approach can lead to a convenient system implementation that even in a non-optimized version suggests a feasible solution to the TDM-FDM translation problem.

2. TDM-FDM CONVERSION THROUGH THE ANALYTIC SIGNAL

In the conversion in the TDM to FDM direction, L TDM signals (L even in all practical cases), each sampled at the frequency $f_s = 1/T_s$, have to be allocated in SSB/FDM format in the frequency band 0 to $Lf_s/2$. The final analog FDM signal is generated after the D/A conversion of a digital signal in the FDM format obtained by appropriate digital processing of the L input TDM signals. The digital FDM signal must be sampled at least at Lf_s. Of course, by appropriate bandpass D/A conversion the analog FDM signal can be allocated, if required, in any band $qLf_s/2$ to $(q+1)Lf_s/2$ (q integer). This is the case for example (Ref. 4) of the 60-108 kHz FDM primary group, which can be obtained by extracting the required band through bandpass D/A conversion of the digital FDM signal sampled at 112 kHz.

Consider a TDM signal $s_i(nT_s)$ whose spectrum is schematically illustrated in Fig. 1a. The associated sampled analytical signal $z_i(nT_s)$ is expressed by (Ref. 5)

$$z_i(nT_s) = s_i(nT_s) + j\,\hat{s}_i(nT_s) \qquad (1)$$

where $j = \sqrt{-1}$ and $\hat{}$ denotes the Hilbert transform operator. Its spectrum is shown in Fig. 1b. The complex signal

$$u_i(nT_s/L) \triangleq u_{Ri}(nT_s/L)+j\, u_{Ii}(nT_s/L) \qquad (2)$$

$$= \begin{cases} z_i(nT_s/L), & n=0,\pm L,\pm 2L,\ldots \\ 0 & \text{, otherwise} \end{cases}$$

obtained by inserting L-1 zeroes between two consecutive samples of $z_i(nT_s)$ has a baseband extending to $Lf_s/2$ and a spectrum shown in Fig. 1c for the case L = 4 (Ref. 6). Because of the presence of the negative frequency components $u_i(nT_s/L)$ is not a sampled analytical signal. The filtering with the complex bandpass filter $H_i^c(f)$ having the frequency response shown in Fig. 1d produces a new sampled analytic signal (Fig. 1e) $w_i(nT_s/L)$ at the sampling frequency Lf_s which can be put in the form

$$w_i(nT_s/L) \triangleq v_i(nT_s/L)+j\, \hat{v}_i(nT_s/L) \qquad (3)$$

Its real part $v_i(nT_s/L)$ gives a SSB signal allocated in one of the assigned bands of the FDM format (Fig. 1f).

This procedure allows the correct allocation in the FDM format only for the bands (even channels) $rf_s/2$ to $(r+1)f_s/2$, r=0,2,...,L-2. The allocation in the remaining bands, for r odd integer less than L (odd channels), can be obtained by considering the "shifted" sampled analytic signal

$$z_i'(nT_s) = (-1)^n \left\{ s_i(nT_s)+j\, \hat{s}_i(nT_s) \right\}$$
$$= s_i'(nT_s)-j\, \hat{s}_i'(nT_s) \qquad (4)$$

where by definition

$$s'(nT_s) \triangleq (-1)^n\, s_i(nT_s) \qquad (5)$$

The spectrum of $z_i'(nT_s)$ is that of Fig. 1b shifted to the left of $f_s/2$ (i.e. is nonzero for negative frequencies and zero for positive frequencies). Therefore, increasing the sampling frequency up to Lf_s through the insertion of L-1 zeroes in between two consecutive samples and the subsequent appropriate complex bandpass filtering defined above gives a complex signal whose real part is a SSB signal correctly allocated in one of the odd channels of the FDM format.

3. IMPLEMENTATION THROUGH REAL SIGNAL PROCESSING

The described method, as stated above, involves complex signals at various steps of the processing procedure, even though the input TDM signals and the output FDM signal are all real. It is possible, however, to simplify the method to obtain only the real output signal (without its imaginary part) and to describe it only through operations on real quantities.

Without loss of generality, we can suppose that the ith TDM signal $s_i(nT_s)$ should be allocated in the ith band of the FDM format, i.e. in the band $if_s/2$ to $(i+1)f_s/2$, i=1,2,...,L-2. The index i runs from 1 to L-2, because in all practical cases there are at least two empty guardbands at the limits of the frequency interval (Ref. 3)(essentially for implementation convenience), so that the actual number of signals to be allocated is L-2. Therefore, the complex bandpass filter $H_i^c(f)$ can be defined through its complex coefficients $h_i^c(nT_s/L)$ in the form

$$h_i^c(nT_s/L) \triangleq h_i(nT_s/L)+j\, \hat{h}_i(nT_s/L) \qquad (6)$$

where $h_i(nT_s/L)$ and $\hat{h}_i(nT_s/L)$ form a Hilbert transform pair. In other words, the $h_i(nT_s/L)$ are the coefficients of the ith real bandpass filter (whose frequency response for positive frequencies is still exactly that shown in Fig. 1d) and the $\hat{h}_i(nT_s/L)$ are those of its associated quadrature filter.

Then it is straightforward to show that for the even channels

$$v_i(nT_s/L) = \mathrm{Re}\left\{ w_i(nT_s/L) \right\}$$
$$= \mathrm{Re}\left\{ u_i(nT_s/L) * h_i^c(nT_s/L) \right\}$$
$$= u_{Ri}(nT_s/L) * h_i(nT_s/L) \qquad (7)$$
$$- u_{Ii}(nT_s/L) * \hat{h}_i(nT_s/L)$$

where $*$ denotes the discrete convolution operation, while for the odd channels (because of the minus sign in (4) in place of the plus sign in (1))

$$v_i(nT_s/L) = u'_{Ri}(nT_s/L) * h_i(nT_s/L) + u'_{Ii}(nT_s/L) * \hat{h}_i(nT_s/L) \quad (8)$$

where in this case the $u'_{Ri}(nT_s/L)$ and $u'_{Ii}(nT_s/L)$ are obtained, according to (2), from the shifted signal $z'_i(nT_s)$ given in (4).

The real digital FDM signal $y(nT_s/L)$ is finally obtained by summing all $v_i(nT_s/L)$

$$y(nT_s/L) = \sum_{i=1}^{L-2} v_i(nT_s/L) \quad (9)$$

The block diagram of the system configuration for the TDM to FDM conversion is shown in Fig. 2.

4. DESIGN CONSIDERATIONS AND FINAL COMMENTS

As is clear from Fig. 2, this method of TDM-FDM conversion is essentially concerned with the design of digital wideband Hilbert transformer at the sampling rate f_s and bandpass quadrature filters at the sampling rate Lf_s. The most stringent conditions on the overall conversion system computational complexity are posed by the design of the quadrature filters because they operate at the higher sampling rate and are bandpass filters of relatively narrow bandwidth. The method requires the design of a bank of such bandpass quadrature filters (or equivalently a bank of complex bandpass filters). However, their relatively large transition bandwidths (approximately $f_s/2$) can be conveniently exploited to achieve the specified frequency requirements with a much reduced number of filter coefficients. This property and the absence of any digital modulator make this method advantageous with respect to other non-FFT approaches, like a direct use of a bank of real bandpass filters, digital Weaver and Hartley modulators (Ref. 2). Moreover the relatively large transition bandwidths of the filters are also a distinguishing feature of the present approach with respect to the filter characteristics required by the methods employing FFT-type processors as those described in Ref. 2.

Simulations for the comparison of the proposed method with approaches using several different implementations of non-FFT methods were carried out for the case of the 60-108 kHz FDM primary group and have shown that the reduction in the computational complexity that can be achieved with the new method allows a saving in the arithmetic operations of the order of 25% (Ref. 7). In these simulations the computational complexity was still higher than that for the FFT approaches proposed in the literature (Ref. 2). However, the filters used in the simulation of the present method were not optimized either in terms of design or implementation techniques. The filter optimization is the current research topic in this field to arrive at the minimum computational complexity for a correct comparison with FFT approaches. To this end it is worth noting that the required bank of complex bandpass filters may be specified in terms of appropriate frequency translations of a suitable lowpass prototype, which is moreover a real filter if the frequency response of the complex bandpass filters is symmetrical with respect to its center frequency. In this case it is sufficient for the design of a single real lowpass prototype to satisfy specified characteristics in order to guarantee that all the complex filters of the bank will meet the required specifications for the transmultiplexer. This property can be conveniently exploited to make the design of the filter bank easier and to realize a more efficient system implementation.

Another main difference that distinguishes the present method (and in general all the non-FFT methods) from FFT approaches is the complete hardware separation of all the channels, a feature which can be advantageous in the system fault recognition and elimination.

Finally, it must be noted that the method applies equally well to the FDM to TDM conversion simply by reversing the block diagram of Fig. 2, substituting adding points with branch ones and sampling rate increase operations with sampling rate decrease ones.

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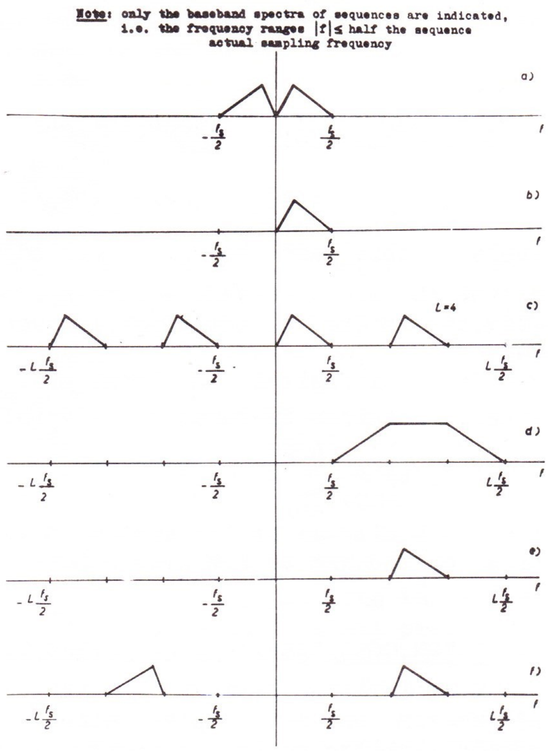

Fig. 1 - Frequency plots explaining the system operation for the TDM-FDM conversion

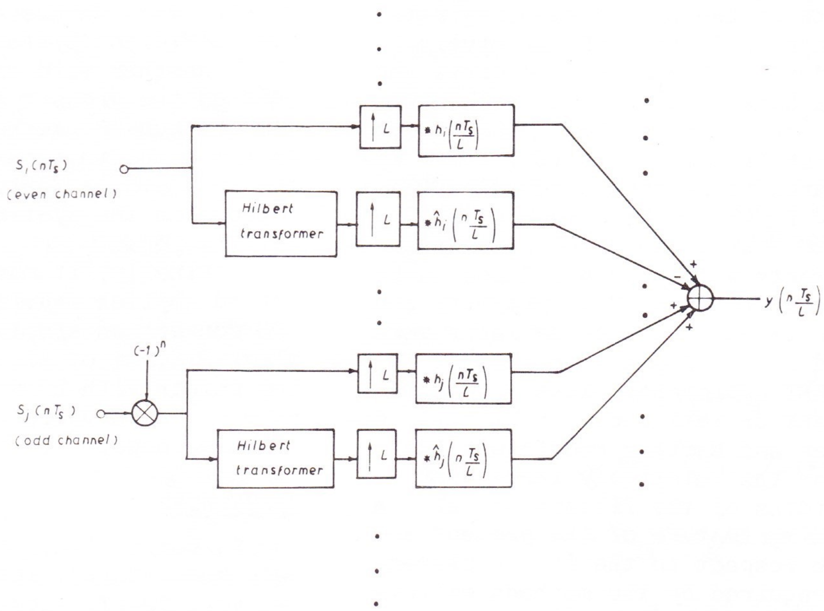

Fig. 2 - Block diagram of the system configuration for the TDM-FDM conversion