APPLICATION OF THE ANALYTIC SIGNAL METHOD TO THE 60- CHANNEL TRANSMULTIPLEXER DESIGN

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ABSTRACT

The application of a digital per-channel transmultiplexer system, based on the generation of a voiceband analytic signal and on its interpolation and band-pass filtering, to the 60-channel standard group is presented.

The design of the required complex digital filters to meet the CCITT specifications is in particular discussed, together with the corresponding computational complexity.

1. INTRODUCTION

Digital signal processing in comunications is becoming very attractive due to the increased processing rate, the decreasing cost and the suitability for high-density integration of digital circuits. The conversion between frequency-division-multiplexing (FDM) and time-division-multiplexing (TDM) and viceversa at the multiplex signal level (transmultiplexer) represents in particular an interesting application, where many of the theoretical and practical properties of the digital signal processing must be exploited in order to meet the very stringent requirements of telecommunications equipment.

Many approaches have been proposed in the last decade and up-to-date developments have been reported in the recent literature[1]. Here we consider the application of the analytic signal method[2] to the conversion between two 30-channel PCM groups and the 60-channel FDM supergroup. The block diagram of the general TDM-FDM conversion system, based on the analytic signal method, is shown in Fig. 1. The signal $s_i(nT)$,

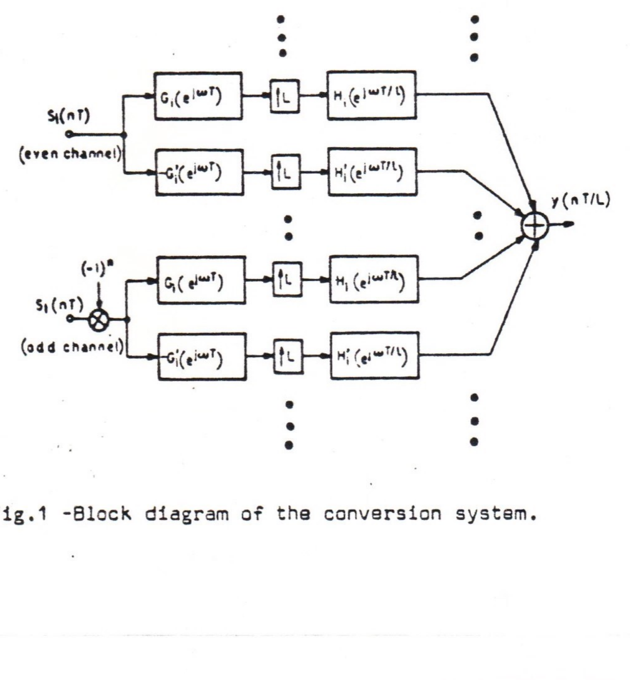

Fig.1 -Block diagram of the conversion system.

n integer, of the ith TDM channel, sampled at the frequency $1/T$ = 8kHz (or its alternate-sign-inverted version for the odd channels) is supplied to the input of two real digital filters G_i (exp(jωT)) and $-G_i'$ (exp (jωT)), that are the real part and the negative of the imaginary part of the complex filter ($j=\sqrt{-1}$)

$$\bar{G}_i(\exp(j\omega T)) = G_i(\exp(j\omega T)) + j\,G_i'(\exp(j\omega T)) \quad (1)$$

This complex filter is so defined as to produce at its output the alternate sampled analytic signal[2]

$$s_i(nT) = (-1)^{in}\left[s_i(nT) + j\,\hat{s}_i(nT)\right] \quad (2)$$

where $\hat{}$ denotes the Hilbert transform operator.

The sampling frequency of the outputs of the two filters G_i (exp(jωT)) and $-G_i'$ (exp(jωT)) is then increased by a factor L, so that the new sampling frequency L/T is compatible with the FDM-format digital output signal. In the 60-channel case the FDM supergroup occupies the frequency range 312-552 kHz. An appropriate sampling frequency for this application is therefore L/T = 576 kHz, giving an interpolation factor L = 72.

The real filters H_i (exp(jωT/L)) and H_i' (exp (jωT/L)), too, are respectively the real part and imaginary part of a complex filter

$$\bar{H}_i(\exp(j\omega T/L)) = H_i(\exp(j\omega T/L)) + jH_i'(\exp(j\omega T/L)) \quad (3)$$

able to retain only the replica of the input signal correctly allocated in one of the FDM-format bands.

This conversion system allows wide transition bands to the high rate filters, whose realization heavily influences the overall system computational complexity.

2. DIGITAL FILTER DESIGN

Two complex filters $\bar{G}_i$ (exp(jωT)) and $\bar{H}_i$ (exp (jωT/L)) have to be designed. They have to comply with the specifications for transmultiplexer system (CCITT recommendations G791, G792, G793): the sums of their in-band deviations must not exceed 0.3 dB, while the stopband attenuation for each filter must be not less than 65 dB.

A FIR filter design is convenient for both the interpolating filters $\bar{H}_i$(exp(jωT/L)) and the filters $\bar{G}_i$ (exp(jωT)). Furthermore it is not possible to design the complex filters through the separate designs of the real and imaginary parts, respectively, because the in-band signal phase addition and adjacent-band signal phase cancellation mechanisms require similar filter responses in the passband and stopband and mainly in the transition bands.

Therefore the design of the $\overline{G}_1(\exp(j\omega T))$ has been carried out[3] using a FIR minimum phase complex filter design, for the even channels. The filter for the odds channels has a frequency response exactly shifted by $f_s/2$ and can be obtained by multiplying the complex impulse response of the first filter by the sequence $(-1)^n$. The design, exploiting the limited extention (300 to 3400 Hz) of the frequency band of the telephone signals, leads to a 28-order filter (29 coefficients), having the following characteritics: in-band deviation of 0.19 dB, out-of-band minimum attenuation of 65.73 dB, with a 15 bit coefficient accuracy [3].

The $\overline{H}_1(\exp(j\omega T/L))$ can be designed as linear-phase complex filters, due to their small contribution to the group delay, and must have a maximum in-band deviation of 0.11 dB. Because of the high interpolation factor (L=72) it is convenient to split them in two or more interpolation stages. As shown in[2] all the interpolation filters can be obtained through the design of a low-pass prototype. Therefore two or more low-pass prototypes have been designed to meet the preceding specifications by a modified version of the Remez exchange algorithm[4]. The results of the interpolation filter design are reported in Table 1.

L_1	L_2	N_1	I.B.	S.B.	N_2	I.B.	S.B.	$R_{MP}\times10^3$
8	9	39	0.032	68.6	27	0.043	66.11	2280
6	12	29	0.034	68.11	47	0.014	76.02	2776
3	24	15	0.023	71.55	119	0.034	68.25	3232
4	18	19	0.036	67.56	71	-0.036	67.78	2696
12	6	59	0.03	69.19	23	0.0033	88.38	3016
18	4	89	0.029	69.58	11	0.042	66.35	2680

Table 1 -Results for the $\overline{H}i$ filters (S.B, I.B. in dB.)

3. REALIZATION COMPLEXITY

The multiplication rate for each signal path, R_{Mp}, is given by the relation

$$R_{Mp} = N f_s + \sum_{k=1}^{K} \lceil N_k/L_k \rceil \prod_{j=1}^{k} L_j f_s \qquad (4)$$

Where N is the number of the coefficients of the low-rate minimum-phase filters, N_k is the number of the coefficients of the k -th stage channel filter interpolating by the factor L_k, K is the number of interpolation stages for the channel filter, $f_s = 1/T$ is the sampling frequency of the imput TDM signal and $\lceil x \rceil$ indicates the smallest integer equal or greater than x. As shown in the previous section N = 29 for the low-rate filters. The values of N_k and L_k for the considered K interpolation stages are indicated in table 1. Using these numbers, the multiplication rates R_{Mp} shown in the same Table have been evaluated for each solution according to (4).

Table 1 shows that two stages with interpolation factor $L_1 = 8$ and $L_2 = 9$, respectively, appear as the most appropriate solution from a computational point of view to the implementation of the channel filters of Fig. 1.

4. CONCLUSION

A transmultiplexer structure for directly interfacing two 30 channel TDM groups with a 60 channel FDM supergroup has been discussed, based on a recently proposed method. The preliminary results here reported show the feasibility of the analytic signal method to the implementation of the 60-channel transmultiplexer.

The computational complexity shown in Table 1 allows to use presently available low-cost digital integrated circuits. Other advantages of this approach are the system structure modularity and the use of FIR filters, with their inherent high noise immunity, the absence of limit cycles and of stability problems and their simple hardware realization, throughout the whole conversion system.

Further improvements of this structure are expected to be obtained from a minimum-phase design of both low-rate and high-rate interpolation filters.

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