Journal of Circuits, Systems, and Computers
Vol. 28, No. 1 (2019) 1950009 (8 pages)

DOI: 10.1142/S0218126619500099

Performance and Analysis of Transmultiplexers Using Decimator and Interpolator*

S. Arunkumar†

Department of Electronics and Communication Engineering,
PSNA College of Engineering and Technology,
Dindigul 624622, Tamil Nadu, India
murphy_arun@yahoo.co.in

P. Ganesh Kumar

Department of Information Technology,
KLN College of Engineering,
Pottapalayam, Sivagangai 630611,
Tamil Nadu, India
ganesh_me@yahoo.com

Received 3 May 2017
Accepted 25 March 2018
Published

This paper deals with the smart applications of multirate digital signal processing. The two major operations are accomplished in consumer electronics and communication engineering. The process of reducing the sampling frequency of a sampled signal is called decimation. In the usage of decimating filters, only a portion of the out-of-pass band frequencies aliases into the pass band, in systems wherein different parts operate at different sample rates. A filter design, tuned to the aliasing frequencies all of which can otherwise stealth into the pass band, not only provides multiple stop bands but also exhibits computational efficiency and performance superiority over the single stop band design. The proposed method of transmultiplexer using decimation and interpolation filters analysis procedure is not only efficient but also opens up a new vista of being simple and elegant to compute for the desired over and above transmultiplexer.

Keywords: Decimator; interpolator; SSB modulator; SSB demodulator; FDM signal; TDM signal; integrator; differentiator.

1. Introduction

Transmultiplexer is an equipment that transforms signals derived from frequency-division multiplex equipment, such group or supergroups, to time-division

*This paper was recommended by Regional Editor Piero Malcovati.
† Corresponding author.

multiplexed signals. The optimal design of M-channel perfect-reconstruction finite impulse response (FIR) quadrature mirror filter (QMF) banks is based on the orthogonal matrix concept.[1] In Ref. 2 the ~~the~~ authors proposed for any aliasing (overlapping) problems that occurred by reducing the sampling process in adaptive FIR filter, in order to ensure convergence of the adaptive FIR filter it is necessary to attenuate the high-frequency sigma–delta ($\Sigma\Delta$) quantization noise. In Ref. 3 the authors discussed the problem of multirate signal processing over arbitrary fields. Multistage decimation filters[4,5] are also widely used to decimate signals oversampled by $\Sigma\Delta$ analog-to-digital (A/D) converters.[6] In Ref. 7 the authors had put forward the design of a 2D FIR compaction filter followed by a 2D FIR filter bank (FB) that packs the maximum energy of the input process into a few subbands. The energy compaction property of the 2D compaction filter is extremely good for higher filter orders and it converges to the ideal optimal solution as the order tends to infinity. The design procedure is very straightforward and involves a 2D spectral factorization. In Ref. 8 the authors proposed the schemes with the goal of increasing the rejection of the $\Sigma\Delta$ quantization noise around the folding bands and reducing the pass-band drop of the designed decimation filters with respect to classic cascaded integrator–comb (CIC) structures. In Ref. 9 the authors offered a design approach that exploits trade-offs involving filter complexity, power consumption and sensitivity with respect to capacitance ratio errors. In Ref. 10 the authors showed that the decimation is done in two or three stages to reduce the hardware complexity and power dissipation. In Ref. 11 the authors came up with a novel design approach for the processor that employs an iterative procedure to jointly optimize the interpolation, decimation and estimation tasks for reduced-rank parameter estimation. In Ref. 12 the authors proposed a wideband low-distortion sigma–delta ADC with residue number system (RNS)-based decimation filter for wireless local area network (WLAN) standard. In Ref. 13 the authors proposed that the frequency response of their filters are better than other filters in canonical signed digit (CSD) coefficient non uniform filter bank transmultiplexer (NUFBTMUX) obtained by rounded and compared with those of continuous coefficient NUFB TMUX. In Ref. 14 the authors declared that approximate frequency-domain magnitude constraints are derived for filter bank system (FBS) prototype filters (PFs) such that a prescribed signal-to-distortion ratio of the FBS output signal is ensured. In Ref. 15 the authors came up with a new approach to implement computationally efficient reconfigurable FBs. If the coefficients of a finite impulse response filter are decimated by M, i.e., if every Mth coefficient of the filter is kept unchanged and remaining coefficients are replaced by zeros, a multiband frequency response will be obtained. In Ref. 16 the author discussed the recent computer simulations in order to evaluate the performance of the designed fixed-point filters. In Ref. 17 the authors proposed a modified Newton's algorithm to design oversampled single-prototype causal FIR nearly-perfect reconstruction (PR) DFT-modulated filter banks allowing low system delays. In Ref. 18 a least-mean-square error approach is introduced to increase the length of the designed lower-ordered PF with the PR property unchanged. In Ref. 19 the authors proposed a new metric for estimating the power dissipation of a filter structure from its architecture, accounting for dissipation both in the adder cells and the flip-flops. In Ref. 20 a closed-form equation is derived which relates the mean-squared error (MSE) to the frequency response of the interpolation filter. In Ref. 21 the authors proposed the design of new selective CIC filter functions. The authors proposed a matrix form of the Farrow transfer function and used it to derive state-space transformations between the Lag range and Farrow structure.[22] The theoretical design equations and impulse response coefficients, as well as the frequency response characteristics of the novel filter functions, are presented.

This paper is organized as follows. Section 2 describes the details of the transmultiplexer. Section 3 discusses the major issues related to the proposed design. Finally, Sec. 4 draws the conclusion.

2. Transmultiplexer

Transmultiplexer for TDM-to-FDM conversion: The input signal for FDM-to-TDM conversion, the input signal $\{x(n)]$, is a time-division multiplexed signal consisting of L signals, which are separated by a commutator switch. Each of these L signals is then modulated at different frequencies to obtain an FDM signal for transmission. In a transmultiplexer for FDM-to-TDM conversion, the composite signal is separated by filtering into L signal components which are then time-division multiplexed.

Telephony: Single-sideband transmission is used with channels spaced at a nominal 4-kHz bandwidth. Totally 12 channels are usually stacked in frequency to form a basic group channel, with a bandwidth of 48 kHz. Larger bandwidth FDM signals are formed by frequency translation of multiple groups into adjacent frequency bands.

FDM-to-TDM conversion: The analog FDM signal is passed through an A/D converter. The digital signal is then demodulated to baseband by means of single-side band demodulators. The output of each modulator is decimated and fed to commutator of the TDM system.

The 12-channel FDM signal is sampled at the Nyquist rate of 96 kHz and passed through an FB demodulator. The basic building block of the FDM demodulator consists of a frequency converter, a low-pass filter and a decimator, as illustrated in Fig. 1. Frequency conversion can be efficiently implemented by use of the polyphase filter structure. Thus the basic structure for the FDM-to-TDM conversion has the form of a DFT filter bank analyzer (Fig. 2). Since the signal in each channel occupies a 4-kHz bandwidth, its Nyquist rate being 8 kHz, hence the polyphase filter output can be decimated by a factor of 12. Consequently, the TDM commutator is operating at a rate of 96 kHz.

TDM-to-FDM conversion: The 12-channel TDM signal is demultiplexed into 12 individual signals, where each signal has a rate of 8 kHz. The signal in each channel is

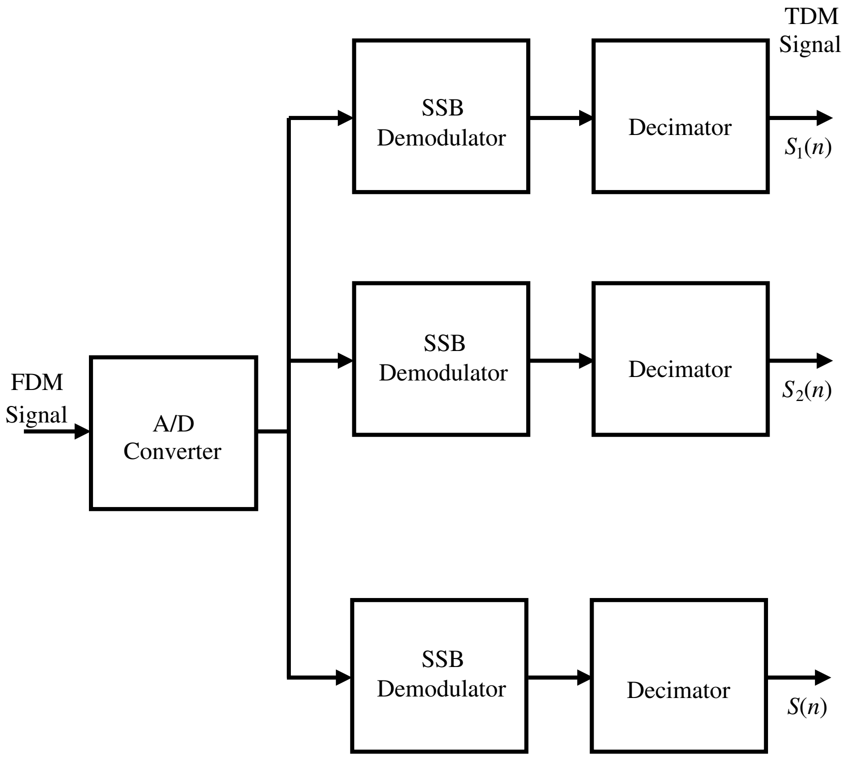

Fig. 1. Transmultiplexer setup.

interpolated by a factor of 12 and frequency converted by a single-sideband modulator. The signal outputs from the 12 single-sideband modulators are summed up and fed into the D/A converter. Thus we obtain the analog FDM signal for transmission. As in the case of FDM-to-TDM conversion, the interpolator and the

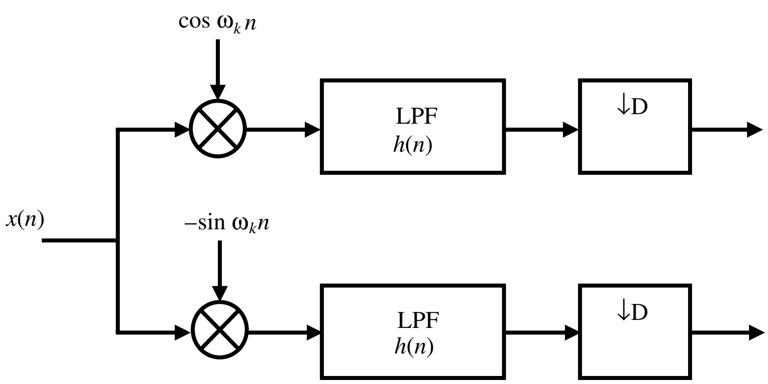

Fig. 2. Block diagram of FDM-to-TDM transmultiplexer.

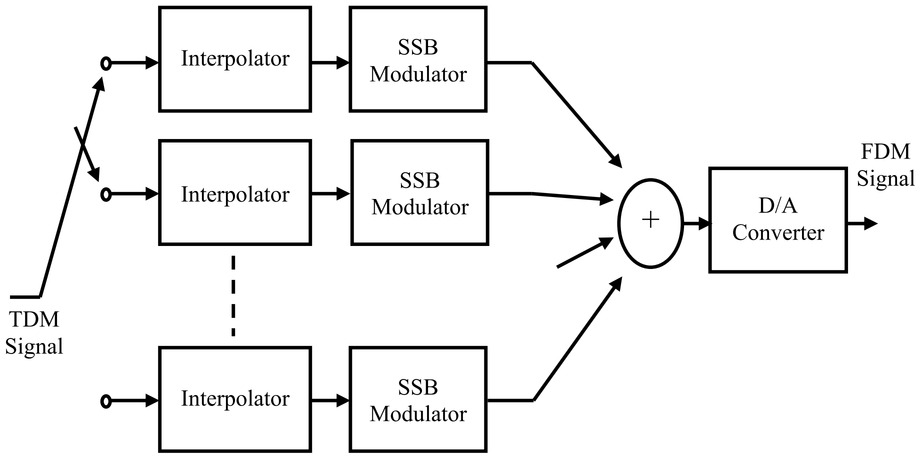

Fig. 3. Block diagram of TDM-to-FDM transmultiplexer.

modulator filter are combined and efficiently implemented by use of a polyphase filter (see Fig. 3). The frequency translation can be accomplished by the discrete Fourier transform (DFT).

3. TDM versus FDM

Time-division and frequency-division multiplexing accomplished the same case by different means. Indeed, they may be classified as dual techniques. The individual TDM channels are assigned to distinct time slots but jumbled together in the frequency domain; conversely, the individual FDM channels are assigned to distinct frequency slots but jumbled together in the time domain.

First and foremost, TDM involves simpler instrumentation. FDM requires an analog subscriber modulator, bandpass filter and demodulator for every message channel, all of which are replaced by the TDM commutator and demodulator switching circuits. And TDM synchronization is but slightly more demanding than that of suppressed-carrier FDM. Second, TDM is vulnerable to the usual causes of cross-talk in FDM, namely, imperfect bandpass filtering and nonlinear cross modulation. However, TDM cross-talk immunity does depend on the transmission bandwidth and the absence of delay distortion. In tetra groups of FDM channel, it's frequency range, bandwidth and number of channels as depicted in Table 1.

Table 1. FDM hierarchy.

Designation	Frequency range (L4 system)	Bandwidth	Number of voice channels
Group	60–108 kHz	48 kHz	12
Supergroup	312–552 kHz	240 kHz	60
Mastergroup	0.564–3.084 MHz	2.52 MHz	600
Jumbogroup	0.5–17.5 MHz	17 MHz	3,600

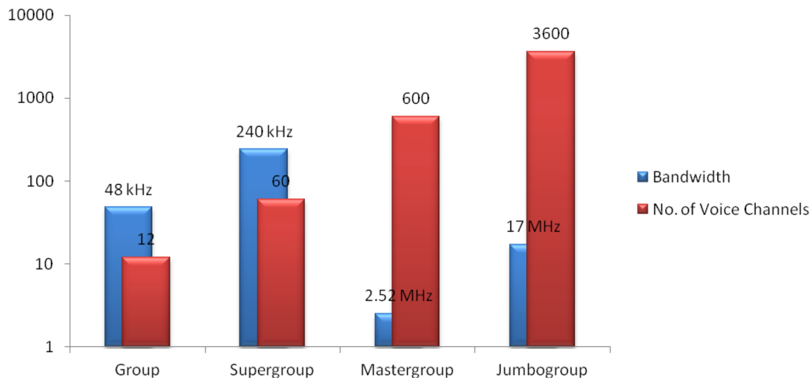

Fig. 4. Histogram for FDM hierarchy.

Third, the use of submultiplexers allows a TDM system to accommodate different signals whose bandwidths or pulse rates may differ by more than an order of magnitude. Finally, TDM may or may not be advantageous when the transmission medium is subject to fading. Rapid wideband fading might strike only occasional pulses in a given TDM channel, whereas all FDM channels would be affected. But slow narrowband fading wipes out all the TDM channels, whereas it might hurt only one FDM channel. In this work, jumbo group has highest voice channel among those the other groups is shown in (Fig. 4).

4. Conclusion

It is comfortable to design and analyze the implementation of transmultiplexer by multirate DSP systems. Decimation filters have been widely employed in consumer electronics to simplify the analog front-end, and to improve the overall digital audio systems like Karaoke and DVD player performances. In this paper efficient design techniques for transmultiplexer decimation and interpolation filters have been shown. TDM synchronization is but slightly more demanding than that of suppressed-carrier FDM. Performance of transmultiplexer by FIR filter is elegant one, but not opted in IIR filter. The proposed method would be useful in that it can improve the transmultiplexer performance of the comb FIR filter in multirate DSP system technology. In future research, finite word length must be considered in transmultiplexer.

Acknowledgments

The authors would like to thank the Editor-in-Chief and the reviewers for their valuable comments and constructive criticisms on the paper which made this final contribution considerably easier for the readers to grasp.

References

1. M. Abo-Zahhad and M. A. Sabah, Design of selective M-channel perfect reconstruction FIR filter banks, *Electron. Lett.* **32** (1999) 1223–1225.
2. E. Pfann, R. W. Stewart and M. W. Hoffman, Oversampled sigma-delta LMS adaptive FIR filters, *IEE Proc., Vis. Image Signal Process.* **147** (2000) 385–392.
3. S. Sarkar and H. V. Poor, Multirate signal processing on finite fields, *IEE Proc., Vis. Image Signal Process.* **148** (2001) 254–262.
4. J. H. Reed, *Software Radio: A Modern Approach to Radio Engineering* (Prentice Hall PTR, 2002).
5. R. E. Crochiere and L. R. Rabiner, *Multirate Digital Signal Processing* (Prentice Hall PTR, 1983).
6. F. Harris, *Multirate Signal Processing for Communications* (Prentice Hall PTR, 2004).
7. N. Vijayakumar and K. M. M. Prabhu, Two-dimensional FIR compaction filter design, *IEE Proc., Vis. Image Signal Process.* **148** (2001) 173–181.
8. M. Laddomada and M. Mondin, Decimation schemes for $\Sigma\Delta$ A/D converters based on Kaiser and Hamming sharpened filters, *IEE Proc., Vis. Image Signal Process.* **151** (2004) 287–296.
9. A. Petraglia, J. S. Pereira and F. A. P. Barugui, Low-sensitivity IIR switched-capacitor decimation filters using delay lines with staggered T/H stages, *IEE Proc., Circuits Devices Syst.* **153** (2006) 193–198.
10. T. K. Shahana, B. R. Jose, R. K. James and K. Poulose Jacob, GUI based decimation filter design tool for multi-standard wireless transceivers, *Proc. IET-UK Int. Conf. Information and Communication Technology in Electrical Sciences* (2007), pp. 600–605.
11. R. C. de Lamare and R. Sampaio-Neto, Adaptive space-time reduced-rank estimation based on diversity-combined decimation and interpolation applied to interference suppression in CDMA systems, *IET Signal Process.* **3** (2009) 150–163.
12. R. J. Babita, T. K. Shahana and P. Mythili, Wideband low-distortion sigma-delta ADC for WLAN with RNS based decimation filter, *Proc. IET-UK Int. Conf. Information and Communication Technology in Electrical Sciences* (2007), pp. 546–552.
13. V. J. Manoj and E. Elizabeth, Design of non-uniform filter bank transmultiplexer with canonical signed digit filter coefficients, *IET Signal Process.* **3** (2009) 211–220.
14. D. Alfsmann and H. G. Gockler, Frequency-domain magnitude constraints for oversampling complex-modulated NPR filter bank system design ensuring prescribed signal-to-distortion ratio, *Electron. Lett.* **46** (2010) 205–206.
15. R. Mahesh and A. P. Vinod, Low complexity flexible filter banks for uniform and non-uniform channelization in software radios using coefficient decimation, *IET Circuits Devices Syst.* **5** (2011) 232–242.
16. M. Laddomada, Fixed-point design of generalized comb filters: A statistical approach, *IET Signal Process.* **4** (2010) 158–167.
17. J.-Z. Jiang, P.-L. Shui and Z.-J. Zhang, Design of oversampled DFT-modulated filter banks via modified Newton's method, *IET Signal Process.* **5** (2011) 271–280.
18. H. Liu, Z. Yang and Z. Cao, Design perfect reconstruction cosine-modulated filter bank by variable neighbourhood search-least-mean-square error, *IET Signal Process.* **6** (2012) 273–280.
19. K. Mondal and S. Mitra, Non-recursive decimation filters with arbitrary integer decimation factors, *IET Circuits Devices Syst.* **6** (2012) 141–151.

20. E. R. Pelet, Effect of interpolation filter in digital quadrature amplitude modulation signal resampling, *IET Commun.* **6** (2012) 2833–2839.
21. B. P. Stosic and V. D. Pavlovic, Design of new selective CIC filter functions with pass-band-droop compensation, *Electron. Lett.* **52** (2016) 115–117.
22. D. Lamb, L. F. O. Chamen and V. H. Nascimento, Efficient filtering structure for spline interpolation and decimation, *Electron. Lett.* **52** (2016) 39–41.