Sidelobe Suppression in Low and High Time–Bandwidth Products of Linear FM Pulse Compression Filters

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Abstract —The peak sidelobe suppression of unweighted linear FM surface acoustic wave filters limits the dynamic range of pulse compression systems. Using a discrete inverse Fourier transform "sampling technique" and complex Fresnel integral algorithms, this paper extends previous results of other authors from a time–bandwidth product of 50 up to the high value of 720. In the present work, the weighting is applied in the frequency domain, employing an external Hamming weighting function. The output waveforms are determined for different sampling rates. The results show that a peak sidelobe suppression of −38 dB from the main lobe is achieved for high time–bandwidth product ($TB = 720$) at a sampling rate of 512 with broadening in the main lobe, while it is −41 dB for a low time–bandwidth product of $TB = 50$. Also, the paper contains charts showing the sidelobe suppression of unweighted and Hamming externally weighted linear FM pulse compression filters at different values of time–bandwidth products TB (50,100,250,370,510,720) with different central frequencies, dispersion times, and bandwidths B. The skirt steepness, sidelobe ripple rejection, Gibbs ripples of the wave spectrum, reduction of the insertion loss, and suppression of Fresnel ripples are also compared.

I. Introduction

THE MINIMUM INSERTION loss and maximum accuracy may be achieved by acoustic surface wave linear FM chirp filters of large time-bandwidth product (TB). The large TB product is necessary to implement long-range, high-range-resolution compression radar systems for the detection, identification, and tracking of high-speed airborne objects. The minimum insertion loss incurred during expansion increases the dynamic range and signal sensitivity [1]. Moreover, lower sidelobes can be realized by weighting the signal in either the frequency or time domain with broadening in the main lobe. Weighting in the two domains is closely related. The peak sidelobe suppression of unweighted linear FM filters limits the dynamic range of pulse compression systems [2]. A widely used weighting function is the Hamming function, which theoretically has the largest sidelobe suppression [3]–[6].

Pulse compression using dispersive delay lines is commonly used to improve the performance of pulsed radar systems. SAW linear FM is gaining widespread use in radar systems to increase the range and target resolution. Moreover, it offers a convenient way for the realization of internally time domain weighted compression filters of low time–bandwidth product [4], [5]. There are still situations that demand the use of an external weighting. This, for example, may apply to systems where a single SAW device has to be time shared for chirp generation and compression. This paper provides numerical results on the obtainable suppression using the discrete inverse Fourier transform "sampling technique" and the complex Fresnel integral algorithms. The output waveforms are determined for different sampling rates N (64,128,256,512,1024) for the weighted and unweighted filter. A comparison between low and high time–bandwidth product is discussed. The skirt steepness, sidelobe ripple rejection, and the Gibbs ripples of the wave spectrum, as well as the reduction of the insertion loss and the suppression of Fresnel ripples, are compared.

The discussion in Section II contains analyses of rectangular linear FM pulse compression filters (magnitude and phase of wave spectrum and insertion loss). Section III continues with the sidelobe suppression of weighted and unweighted output waveforms.

II. Rectangular Linear FM

The rectangular envelope linear FM pulse with center frequency f_0, bandwidth B, and duration T of the chirped signal is defined in [7] as

$$S(t) = e^{j2\pi\left(f_0 t + \frac{B}{2T}t^2\right)}, \qquad |t| \leqslant T/2 \tag{1}$$

where the chirp magnitude is unity. Here, for a low TB product filter, the center frequency is 100 MHz, the bandwidth 46.7 MHz, and the time chirp duration 1.05 μs. For a high TB product filter, the values are 300 MHz, 120 MHz, and 6 μs, respectively. The output wave spectrum is calculated by applying the Fourier transform

$$S(f) = \int_{-\infty}^{+\infty} S(t)e^{-j2\pi ft}\,dt. \tag{2}$$

The wave spectrum will be

$$S(f) = \sqrt{\frac{T}{2B}}\left[Z(u_2) - Z(u_1)\right]e^{-j\frac{\pi T}{B}(f-f_0)^2} \tag{3}$$

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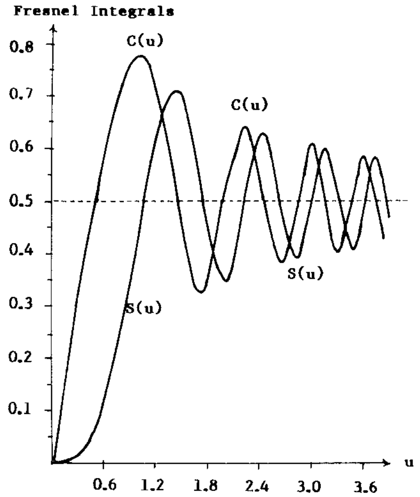


Fig. 1. Fresnel integrals.

where the complex Fresnel integral $Z(u)$ is [8]

$$Z(u)=\int_0^u \cos\left(\frac{\pi}{2}x^2\right)dx+j\int_0^u \sin\left(\frac{\pi}{2}x^2\right)dx \tag{4}$$

with the arguments u_1 and u_2 given by

$$u_1=-2(f-f_0)\sqrt{\frac{T}{2B}}-\sqrt{\frac{TB}{2}} \tag{5}$$

$$u_2=-2(f-f_0)\sqrt{\frac{T}{2B}}+\sqrt{\frac{TB}{2}}. \tag{6}$$

$Z(u)$ is a function of Fresnel integral cosine $C(u)$ and Fresnel integral sine $S(u)$, where

$$C(u)=\int_0^u \cos\left(\frac{\pi}{2}x^2\right)dx \tag{7}$$

$$S(u)=\int_0^u \sin\left(\frac{\pi}{2}x^2\right)dx. \tag{8}$$

The Fresnel ripple values of these functions are high at small argument (u) and small at high values of (u). They are convergent functions and tend to 0.5 at very high values of (u), as shown in Fig. 1. The tapered cosine squared function in [4]–[7] increases the argument of the Fresnel integral functions, where small Fresnel ripples are achieved. The magnitudes of the wave spectra $S(f)$ are shown in Figs. 2(a) and 2(b). The maximum variation of Gibbs ripples is ± 1.3 dB for $TB=50$ and ± 1.6 dB for $TB=720$. The transition bandwidth is 7 MHz and the side ripple level is 0.4 dB at $TB=50$ and 7 MHz and 0.2 dB at $TB=720$.

At the center frequency $f=f_0$, the magnitude of the wave spectrum is $H(f_0)=\sqrt{T/2B}$, and the phase depends on two angles values. The first is $-\frac{\pi T}{B}(f-f_0)^2$, which is zero at $f=f_0$; the second is

$$\tan^{-1}\left[\frac{\int_0^{\sqrt{TB/2}}\sin\left(\frac{\pi}{2}x^2\right)dx-\int_0^{\sqrt{-TB/2}}\sin\left(\frac{\pi}{2}x^2\right)dx}{\int_0^{\sqrt{TB/2}}\cos\left(\frac{\pi}{2}x^2\right)dx-\int_0^{\sqrt{-TB/2}}\cos\left(\frac{\pi}{2}x^2\right)dx}\right] \tag{9}$$

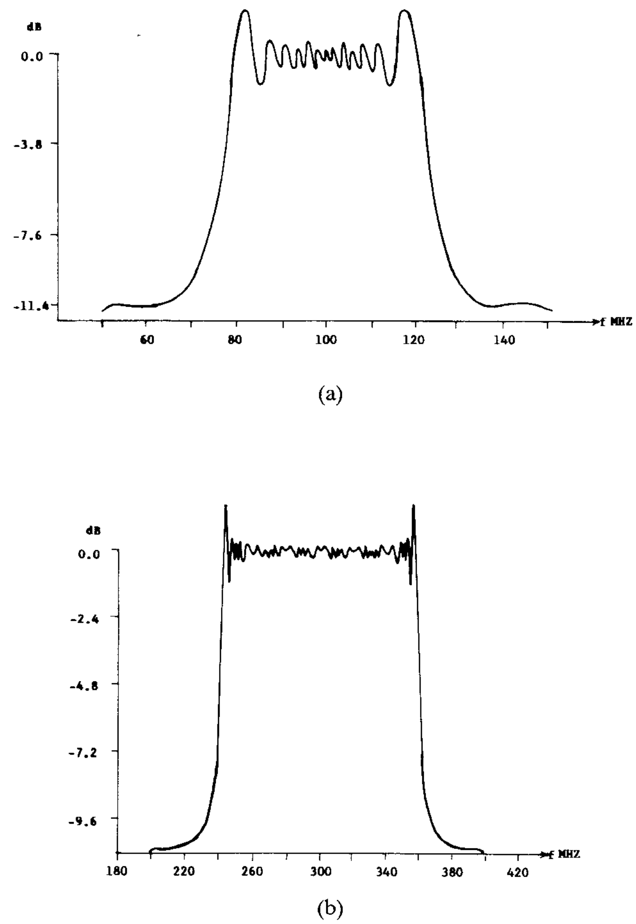


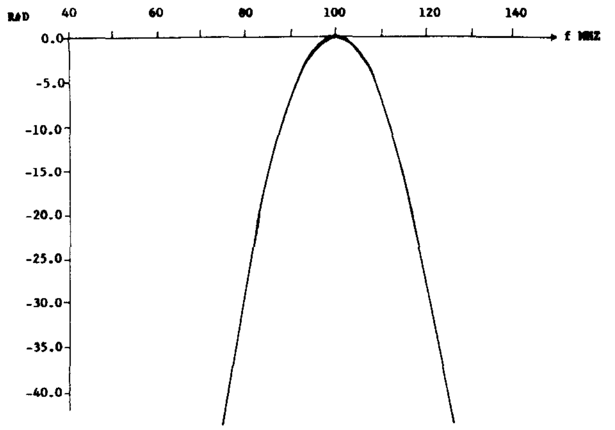
Fig. 2. The frequency response of a rectangular envelope for (a) $TB=50$ and (b) $TB=720$.

where the product TB is the compression ratio. The lower the value of TB, the smaller the value of the second angle. Figs. 3(a) and 3(b) show the phases of the wave spectrum for TB values of 50 and 720, respectively. The phase is approximately equal to zero at and near f_0. Good linearity of the phase characteristics is achieved at the frequencies far from the center frequency because the second value is very small with respect to the first angle value.

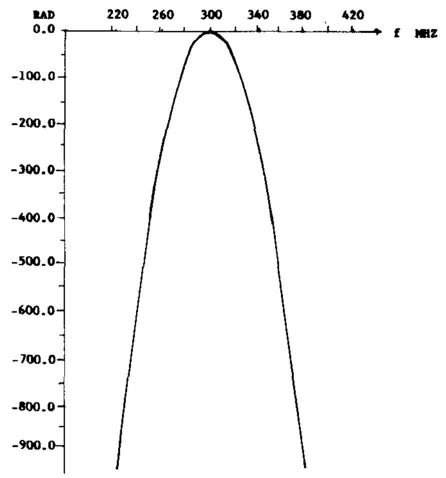
The insertion loss of surface wave filters is due to several contributions: (1) Loss of the input transducer, and (2) loss at the output transducer, (3) surface wave propagation loss, and (4) loss due to beam spreading. The first two are the most significant while the last two are relatively small [9]. The insertion loss can be calculated from the following equation [8]:

$$IL=10\log_{10}|S(f)|^2. \tag{10}$$

As shown in Figs. 4(a) and 4(b), the maximum variation of Fresnel ripples is ± 3.0 dB for $TB=50$ while it is ± 2.5 dB for $TB=720$. Also, the reductions of the insertion loss at the two passband edges are -10 dB and -7 dB for $TB=50$ and $TB=720$, respectively.

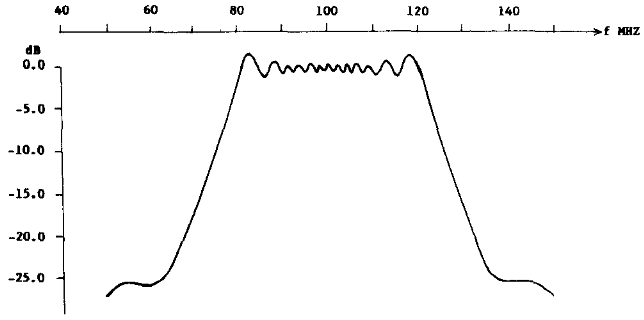


(a)

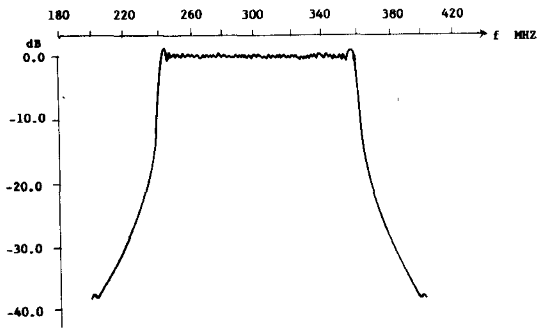


(b)

Fig. 3. The phase of the frequency response of a rectangular envelope for (a) $TB = 50$ and (b) $TB = 720$.



(a)



(b)

Fig. 4. The insertion loss of a rectangular envelope for (a) $TB = 50$ and (b) $TB = 720$.

TABLE I
SIDELOBE SUPPRESSION, IN dB

Sampling Rate N	No Weighting $TB = 50$	No Weighting $TB = 720$	External Weighting $TB = 50$	External Weighting $TB = 720$
64	−13.94	−18.52	−31.04	−28.17
128	−13.15	−18.43	−41.02	−35.64
256	−13.16	−17.93	−40.91	−36.4
512	−13.17	−18.02	−41.05	−38.0
1024	−13.17	−18.02	−41.05	−38.0

III. SIDELOBE SUPPRESSION IN TIME DOMAIN

The unweighted output time waveform of the rectangular linear FM pulse compression filter can be calculated as in [5] by the following:

$$G(t) = F^{-1}[S(f)\cdot S^*(f)] \tag{11}$$

while the weighted output, employing Hamming external weighting functions in the frequency domain, is

$$G_w(t) = F^{-1}[S(f)\cdot S^*(f)\cdot W(f)]. \tag{12}$$

Here, F^{-1} denotes the inverse Fourier transform, $S(f)$ is the linear FM filter frequency response, $W(f)$ is the Hamming weighting function [3], and $S^*(f)$ is the matched filter frequency response, which is the complex conjugate function of $S(f)$, i.e.,

$$S^*(f) = \sqrt{\frac{T}{2B}}\,[Z^*(u_2) - Z^*(u_1)]\,e^{j\frac{\pi T}{B}(f-f_0)^2} \tag{13}$$

where the complex conjugate Fresnel integrals $Z^*(u)$ are

$$Z^*(u) = \int_0^u \cos\left(\frac{\pi}{2}x^2\right)dx - j\int_0^u \sin\left(\frac{\pi}{2}x^2\right)dx. \tag{14}$$

The Hamming weighting function $W(f)$ is

$$W(f) = 0.08 + 0.92\cos^2\left[\frac{\pi(f-f_0)}{B}\right], \qquad |f-f_0| < \frac{B}{2}. \tag{15}$$

Using the discrete inverse Fourier transform "sampling technique" and the complex Fresnel integral algorithms, the output waveforms are determined for different sampling rates N (64, 128, 256, 512, 1024) of weighted and unweighted filters having $TB = 50$ and 720. The peak sidelobe suppression of the weighted filter is −38 dB from the main lobe at sampling rate 512 with broadening in the main lobe at high TB, while the suppression is −41 dB at low TB. Table I compares the sidelobe suppression at different rates N. Figs. 5(a) and 5(b) show the calculated output unweighted waveforms for low and high TB at sampling rate 512. The weighted waveforms are presented in Figs. 6(a) and 6(b) at the same sampling rate.

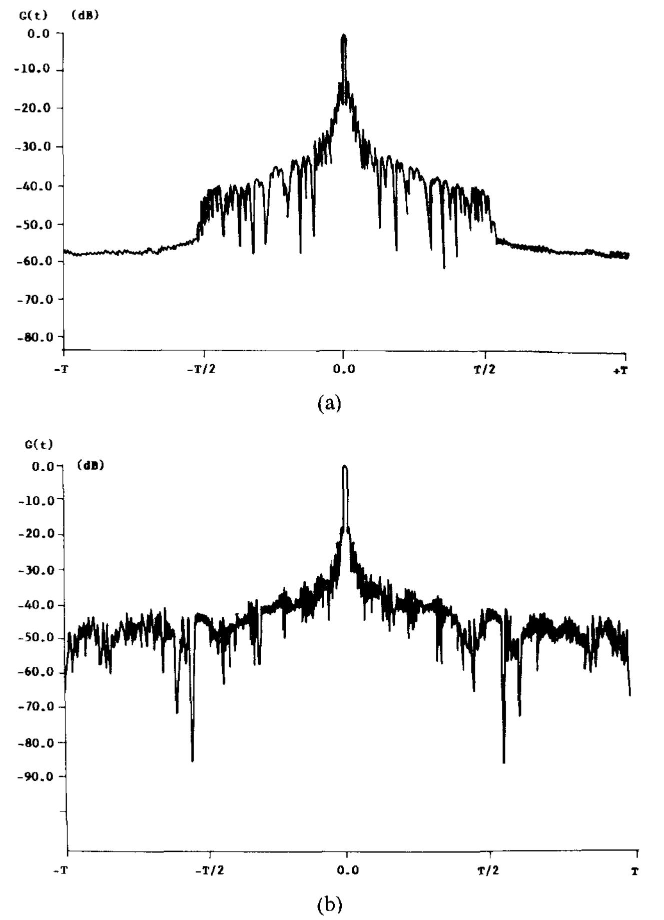


Fig. 5. The unweighted output time waveform of a rectangular envelope ($N=512$) for (a) $TB=50$ and (b) $TB=720$.

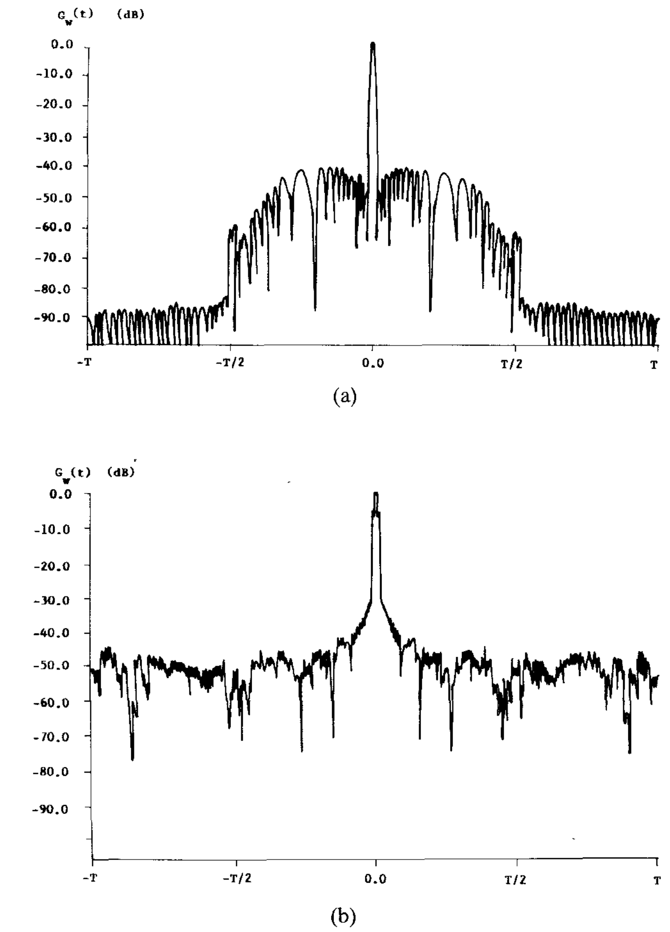


Fig. 6. The weighted output time waveform of a rectangular envelope ($N=512$) for (a) $TB=50$ and (b) $TB=720$.

The sidelobe suppressions of weighted and unweighted linear FM filters take the same values at sampling rates $N=512$ and 1024 for low and high-bandwidth products, as shown in Table I. Table II compares the sidelobe suppressions of weighted and unweighted filters at central frequencies f_0 (100 and 300 MHz), dispersion times T (0.85, 1.06, 2.12, 3.06, 4.24, and 6 μs), and frequency bandwidths B (46.7 and 120 MHz), giving the time-bandwidth products TB ranging from 50 to 720. Fig. (7) shows that the sidelobe suppression of weighted filters at $TB=510$ is lowered at -31 dB and returns again to increase to -38 dB at $TB=720$. Also, the unweighted sidelobe suppression at $TB=720$ is -18 dB, greater than that at $TB=510$, which is -13.4 dB.

TABLE II

f_0 (MHz)	T (μs)	B (MHz)	TB	Sidelobe Suppression (dB) No Weighting	Sidelobe Suppression (dB) External Weighting
100	1.06	46.7	50	−13.7	−41.0
300	0.85	120.0	100	−13.9	−39.0
300	2.12	120.0	250	−13.8	−39.8
300	3.06	120.0	370	−14.0	−40.0
300	4.24	120.0	510	−13.4	−31.0
300	6.0	120.0	720	−18.0	−38.0

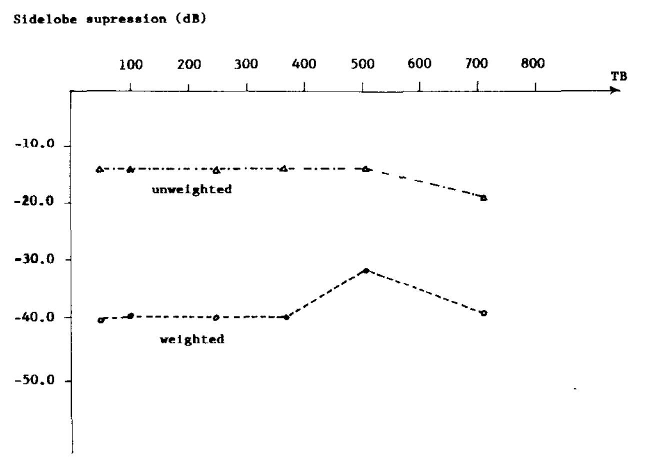


Fig. 7. Sidelobe suppression of weighted and unweighted filters versus TB product.

IV. Conclusions

Sidelobe suppressions of -41 dB and -38 dB are achieved for very high frequency linear FM pulse compression weighted filters having low ($TB=50$) and high ($TB=720$) time–bandwidth products, respectively. The weighting is applied in the frequency domain, employing external Hamming weighting functions. Moreover, better skirt steepness, reduced side ripples, and small Gibbs ripples in the filter frequency response, as well as low insertion loss and flat Fresnel ripples, are achieved in the high TB product.

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