Recursive 디지털 필터 모델에 대한 역 필터링 기법

An inverse filtering technique for the recursive digital filter model

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요 약

본 논문에서는 디지털 필터 모델에 대한 역 필터링 기법을 제안한다. 이 기법은 안정한 non-causal IIR 역 필터를 안정한 causal 역 필터로 변환(근사)시키는 것이다. 실제로 이 역 필터에 대한 FIR 근사 방법을 제안한다. 전역통과 시스템에 대한 역 필터의 임펄스 응답은 그 시스템에 대한 임펄스 응답의 거울 영상(mirror image) 임을 알 수 있다. 특히 전역통과 시스템에 대한 임펄스 응답이 이러한 대칭성을 갖기 때문에, 제안한 기법은 다른 시스템 보다 전역통과 시스템에 더욱 유용하다. 제안한 역 필터링 기법을 설명하기 위하여, 네 개의 예제를 제시한다. 그들 중 둘은 전역통과 필터에 대한 것이며, 다른 두 개의 예제는 IIR과 FIR 필터에 대한 것이다. 또한 컴퓨터 시뮬레이션을 통하여 제안한 기법이 잘 동작함을 보인다.

Abstract

In this paper, an inverse filtering technique for the digital filter model is proposed. This technique enables us to obtain a stable non-causal IIR inverse filter by transforming (approximating) it to a causal stable inverse system. In practice, a causal FIR approximation to this inverse filter is proposed. It can be shown that the impulse response of the inverse filter for all-pass systems is simply the mirror image of the impulse response for the system. Specially, due to this symmetric property of the impulse response of all-pass systems, the proposed technique is more useful for all-pass systems than other systems. In order to illustrate the proposed inverse filtering technique, four examples are presented. Two of them are for all-pass filters. The other two examples are for IIR and FIR filters. Also, computer simulations demonstrate that the proposed technique works very well.

Key words : inverse filtering, recursive digital filter, all-pass systems

I. Introduction

The need to develop effective methods for deconvolving signals propagated through a channel is motivated by the necessity to estimate signals at the input to the channel. An application of the inverse filter design is in the recovery of a signal that has been transmitted through an imperfect transmission channel. The received signal, in general, will be different from the input as it will be distorted by the impulse response of the channel. To recover the original input, we need to pass the channel output through a system with an impulse response which is the inverse of the channel's impulse response. The output of the inverse system will be identical to the desired original input.

The purpose of this paper is to develop an inverse filter for the channel. Obtaining the inverse filter is a crucial step in the overall process for analyzing signals that have propagated through the channel. The inverse filter enables us to estimate input signals to the channel. This is done by processing the corresponding output signals through the inverse filter, a process that is known as deconvolution. If the parent causal

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system has a known impulse response and is excited by a causal input signal, then knowing the output signal, we can determine the samples of the input signal using a recursive relation without determining the inverse system. But this recursive method is impractical in points of view for real implementation and computational load.

The classical approach for obtaining the desired inverse filter requires computing all poles of inverse system, which corresponds to all zeros of the channel model. This approach is impractical for use when the channel is modeled using FIR filters that have a large numbers of filter coefficients. This is because this approach would require one to find the zeros of a large-order polynomial, which is clearly impractical. However, this problem is avoided when working with all-pass systems.

In this paper, an inverse filtering technique is suggested. Due to symmetric property of the impulse response of all-pass systems [1,2], the proposed technique is more useful for all-pass systems than other systems. It will be shown that the impulse response of the FIR approximation to the inverse filter for all-pass channels is simply the mirror image of the channel. Basically, this technique enables us to obtain a causal stable inverse filter. This is done by previously finding a stable non-causal IIR inverse filter, and then by transforming it to a causal stable inverse filter. In practice, a causal FIR approximation to this inverse filter is proposed.

The analysis of an inverse system is mentioned in Section 2. In Section 3, an inverse filtering technique for all-pass systems is presented with two examples corresponding to first- and second-order all-pass systems. In Section 4, this technique, for general systems, is extended and illustrated by two examples corresponding to IIR and FIR systems. The process of designing an inverse filter that is causal and stable is observed in Section 5. Some concluding remarks are included in Section 6.

II. Analysis of an Inverse System

1. Fundamentals of deconvolution

The output signal $y(n)$, in terms of the input signal $x(n)$ and the unit sample response $h(n)$ of a causal LTI (Linear Time-Invariant) system, is expressed by the convolution summation as follows:

$$y(n) = \sum_{k=0}^{\infty} x(k)h(n-k) \quad (1a)$$

or

$$y(n) = \sum_{k=0}^{\infty} h(k)x(n-k) \quad (1b)$$

Thus knowing the unit sample response of a causal discrete system, we can always compute the output sequence for any given sequence [2-5].

However, there are some cases when we already know the output waveshape and we would like to know what input waveshape caused it. This can be estimated as long as we know the $H(e^{jwT})$ of the system. The input waveshape can be determined by evaluating $X(e^{jwT}) = Y(e^{jwT})/H(e^{jwT})$ for $X(e^{jwT})$ and then taking the inverse Fourier transform of $X(e^{jwT})$ to get $x(n)$. This process is referred to as deconvolution, and there is need for caution in using it. Specifically, the complex division of the $Y(e^{jwT})$ data array by the $H(e^{jwT})$ data array may result in a divided-by-zero situation at some data points. We will need to take programming steps to avoid these divided-by-zero possibilities. Also, the above deconvolution procedure is sensitive to noise like a round-off noise. Noise components in the data being deconvolved may become greatly amplified in the results [2,6].

It is seen that this deconvolution process is just the opposite of that of convolution summation; hence it is called deconvolution. It can be said that convolution is the analysis phase of a problem and deconvolution is the synthesis phase.

2. LTI inverse systems

In some application, it is desirable to remove the inherent effect of a discrete process so that the original input signal to the process can be recovered. For example, in a digital communication system, the transmission channel may introduce error into the digital signal being transmitted. It would be useful if a process could be devised so that the error could be removed in order to recover the transmitted digital signal.

To accomplish this, we design a corrective system which, when cascaded with the original system, will generate an output signal identical to the original input signal. Such a system is said to be the inverse of the original system. In order for two systems connected in cascade to produce an output which is identical to the input signal, the overall unit sample response of the cascade connection must be a unit sample function.

Next, we address the definition of an LTI inverse system. A system is said to be invertible if distinct inputs lead to distinct outputs. In other words, a system is invertible if by observing its outputs, we can determine its inputs. As illustrated in Fig. 1 for the discrete-time case, we can construct an inverse system $h_i(n)$ which, when cascaded with the original system $h(n)$, yields an output $z(n)$ equal to the input of the first system, namely $x(n)$. Thus, the series interconnection in Fig. 1 has an overall input-output relationship that is the same as that for the identity system [2-3,7-9].

Since the overall impulse response in Fig. 1 is $h(n)*h_i(n)$, $h_i(n)$ must satisfy for it to be the impulse response of the inverse system:

$$h(n)*h_i(n)=\delta(n) \tag{2}$$

Most stable systems can be approximated as FIR systems whose have N_0 sample points, and a value for N_0 can always be chosen such that the unit sample response is negligibly small outside the time interval being considered [7]. Thus we will restrict ourselves to the case of an FIR system for all practical purposes.

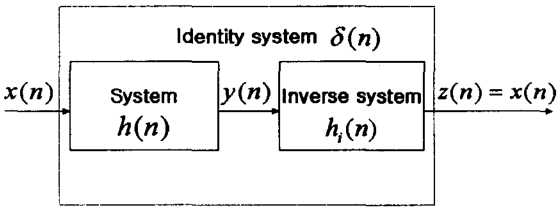


Fig. 1. Concept of the inverse system for discrete-time LTI system.

3. Condition for existing a causal inverse system

In some applications, such as deconvolution, it is useful for us to design the inverse of a given system. In this subsection, we briefly address the condition for the existence of a causal inverse system. A minimum-phase condition [2,5,10,11] is that there exists a causal, stable inverse system with transfer function $H_i(z)$ such that

$$H(z)H_i(z)=1 \tag{3}$$

Since $H_i(z)=1/H(z)$, it is clear that $H(z)$ must have all its poles and zeros inside the unit circle in order for a stable and causal inverse to exist. Thus the minimum-phase system has all its poles and zeros inside the unit circle.

Therefore, if a given system $H(z)$ satisfies a minimum-phase condition, then the causal, stable inverse of this system exists and can be implemented in the real world. Otherwise a causal, stable inverse of this system does not exist. It should be emphasized at this point a system (or sequence) can be causal but non-minimum-phase. However, all stable, minimum-phase systems (sequences) are causal. It has been shown that [5] any causal system can be represented as the cascade of a minimum-phase system with an all-pass system.

To illustrate, consider a non-minimum-phase system $H(z)$, with one zero outside the unit circle at $z=1/z_0$, $|z_0|<1$, and the remainder of its poles and zeros inside the unit circle. Then $H(z)$ can be expressed

$$H(z)=H_1(z)(z^{-1}-z_0) \tag{4}$$

where $H_1(z)$ is minimum-phase. Equivalently (4) can be expressed as

$$\begin{aligned} H(z) &= H_1(z)(z^{-1}-z_0)\frac{(1-z_0z^{-1})}{(1-z_0z^{-1})} \\ &= H_1(z)(1-z_0z^{-1})\frac{(z^{-1}-z_0)}{(1-z_0z^{-1})} \\ &= H_{\min}(z)\left[\frac{(z^{-1}-z_0)}{(1-z_0z^{-1})}\right] \end{aligned} \tag{5}$$

Since $|z_0|<1$, the factor $H_1(z)(1-z_0z^{-1})$ is minimum-phase and the factor $(z^{-1}-z_0)/(1-z_0z^{-1})$ is a first-order all-pass. The term $H_{\min}(z)=H_1(z)(1-z_0z^{-1})$ differs from $H(z)$ in that the zero of $H(z)$ that was outside the unit circle at $z=1/z_0$ is reflected inside the unit circle to $z=z_0$ in $H_{\min}(z)$. Clearly this example can be generalized to encompass general non-minimum-phase systems with rational system functions. Thus, any rational system function $H(z)$ corresponding to a causal system can be expressed in the form

$$H(z)=H_{\min}(z)H_{ap}(z) \tag{6}$$

where $H_{\min}(z)$ is the transfer function of a minimum-phase system and $H_{ap}(z)$ is that of an all-pass system. Any pole or zero of $H(z)$ that is inside the unit circle also appears in $H_{\min}(z)$. Any pole or zero of $H(z)$ that is outside the unit circle appears in $H_{\min}(z)$ in the conjugate reciprocal location; i.e., it is reflected about the unit circle. Thus we can form a minimum-phase system from a causal, non-minimum-phase system, keeping the magnitude of transfer function the same. This is achieved by reflecting zeros that are outside the unit circle into the unit circle.

III. Inverse Filtering Technique for All-pass Systems

In the dissertation [1], it was seen that the transfer function of an all-pass filter has all poles and zeros occurring in complex conjugate reciprocal pairs. Thus the inverse of an all-pass filter that is causal and stable has a transfer function whose poles are outside the unit circle, and whose zeros are inside the unit circle. Therefore, this inverse filter is an IIR filter that is either non-causal and stable, or causal and unstable. We now address the problem of designing a causal and stable FIR inverse filter, whose impulse response closely approximates that of the IIR non-causal and stable inverse filter. However, the output of this FIR inverse filter is delayed compared to its IIR counterpart.

We consider the modified inverse system shown in Fig. 2. Here the inverse for an all-pass system has a cascade form that is obtained in two steps. First, we find the IIR impulse response $h_{ai}(n)$ for the inverse of a given all-pass system, and approximate it as a FIR filter which has N_0 sample points, and then shift it by N_0 samples; in other words, the output of this stage is delayed by N_0 samples compared to its input, as indicated by $z(n) \approx x(n-N_0)$ in Fig. 2. Next, the output $x(n-N_0)$ of the modified inverse system $h_{ai}(n-N_0)$ is shifted backwards by N_0 samples as depicted in Fig. 2. Thus the final output of the inverse filter is an approximation to original input $x(n)$.

To illustrate, two examples (for 1st-order and 2nd-order all-pass filters) are considered in what follows.

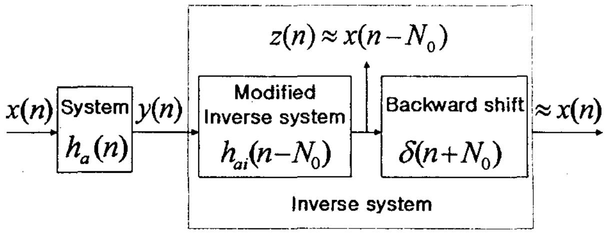


Fig. 2. Modified inverse filter for an all-pass system.

EXAMPLE 1 (for 1st-order all-pass filter):

Consider the first-order all-pass filter

$$H_a(z) = \frac{z^{-1}-a}{1-az^{-1}} = \frac{-a(z-a^{-1})}{z-a} \quad for\ |a| < 1 \tag{7}$$

This filter is causal and stable when the ROC (Region of Convergence) is $|z|>|a|$. Using the residue theorem, the impulse response is given by

$$h_a(n) = \frac{1}{2\pi j}\oint H_a(z) z^{n-1} dz = \begin{cases} -a & for\ n=0 \\ (a^{-1}-a)a^{n} & for\ n \geq 1 \end{cases} \tag{8}$$

Also, the inverse $H_{ai}(z)$ of $H_a(z)$ is

$$H_{ai}(z) = \frac{1-az^{-1}}{z^{-1}-a} = \frac{z-a}{-a(z-a^{-1})} \tag{9}$$

This inverse filter is non-causal and stable when its ROC is $|z|<|a^{-1}|$. Similarly, the impulse response of the inverse filter is

$$h_{ai}(n) = \frac{1}{2\pi j}\oint H_{ai}\left(\frac{1}{p}\right) p^{-n-1} dp = \begin{cases} -a & for\ n=0 \\ (a^{-1}-a)a^{-n} & for\ n \leq -1 \end{cases} \tag{10}$$

where $z=1/p$. From (9) and (10), it is apparent that $h_a(n)$ is symmetric with $h_{ai}(n)$ about the vertical axis. In general this is true because the transfer function of an all-pass filter has all-poles and zeros occurring in complex conjugate reciprocal pairs [1,2].

Simulation results for $a=0.7$ are shown in Fig. 3, for the case when the input is given by

$$x(n) = \cos(2\pi \cdot 60 \cdot nT) + \sin(2\pi \cdot 300 \cdot nT) \tag{11}$$

where $T=1/1000\,(sec)$ is the sampling interval. From plots shown in Fig. 3, it is observed that values of $h_a(n)$ and $h_{ai}(n-N_0)$ beyond about 10 samples are relatively very small. Thus it is assumed that $h_a(n) = h_{ai}(n-N_0) = 0$ *for* $n \geq 21$, i.e., $N_0 = 20$, which means that $h_a(n)$ and $h_{ai}(n)$ are treated as the impulse response of FIR filter with 20 coefficients.

When the inverse filtering scheme proposed in this paper, as depicted in Fig. 2, is applied to Example 1, the results shown in Fig. 3 are obtained. From Fig. 3, the plot in (a) shows $h_a(n)$ which is the impulse response of a first-order all-pass filter; (b) shows input $x(n)$; and (c) shows $y(n)$ which is the output of $h_a(n)$. Also (d) shows $h_{ai}(n-N_0)$ which corresponds to the impulse response of the inverse filter for $h_a(n)$; (e) shows $z(n)$, the output of the inverse filter, which corresponds to the N_0-points delayed version of input; and (f) shows the backward N_0-points delayed version of $z(n)$, which is the same as input $x(n)$.

In the horizontal axis of Fig. 3(c), the number of data points of $y(n)$ is sixty-nine because of $y(n) = x(n) * h_a(n)$. Similarly, in Fig. 3(e), the number of data points of $z(n)$ is eighty-eight because of $z(n) = y(n) * h_{ai}(n-N_0)$. In Fig. 3(f), the reason why the horizontal axis is started from 20-points is because the reconstructed input can be treated as the backward 20-points delayed version of $z(n)$.

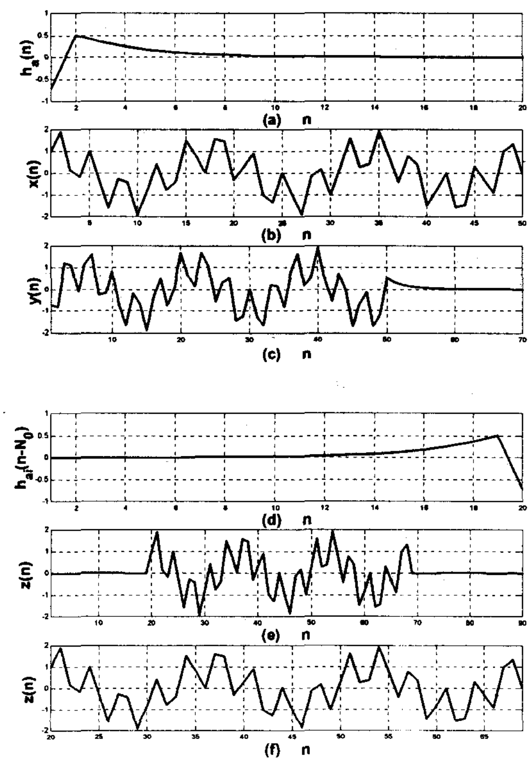


Fig. 3. Simulation results for Example 1. (a) $h_a(n)$, (b) $x(n)$, (c) $y(n)$, (d) $h_{ai}(n-N_0)$, (e) $z(n)$, (f) partially shifted version of $z(n)$

EXAMPLE 2 (for 2nd-order all-pass filter):

Consider a second-order all-pass filter [1]

$$H_a(z) = \frac{a^2 - 2az^{-1}\cos bT + z^{-2}}{1 - 2az^{-1}\cos bT + a^2 z^{-2}}, \quad \text{for } |a| < 1 \text{ and } bT \leq \pi \tag{12}$$

This filter is causal and stable when its ROC is $|z| > |a|$. As in Example 1, the impulse response is expressed as [1]

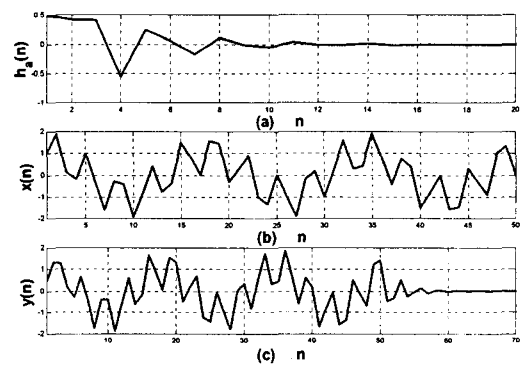
$$h_a(n) = \frac{1}{2\pi j}\oint H_a(z) z^{n-1} dz = \begin{cases} a^2 & \text{for } n = 0 \\ \dfrac{a^{n-2}}{\sin bT} h_0(n) & \text{for } n \geq 1 \end{cases} \tag{13}$$

where,

$h_0(n) = a^4 \sin[(n+1)bT] - 2\beta a \sin nbT + \sin[(n-1)bT]$ with $\beta = a\cos bT$. Also, $H_{ai} = 1/H_a(z)$ is non-causal and stable when the ROC is $|z| < |1/a|$.

Due to the symmetrical properties associated with $h_a(n)$ and $h_{ai}(n)$, the impulse response $h_{ai}(n-N_0)$ is obtained by simply "turning around" $h_a(n)$. Simulation results for $a = 0.7$, $bT = 0.7\pi$ and $N_0 = 20$ are shown in Fig. 4, for the case when the input $x(n)$ is as in (11).

As given at the end of Example 1, similar explanations for the simulation results shown in Fig. 4 can be applied to this example.



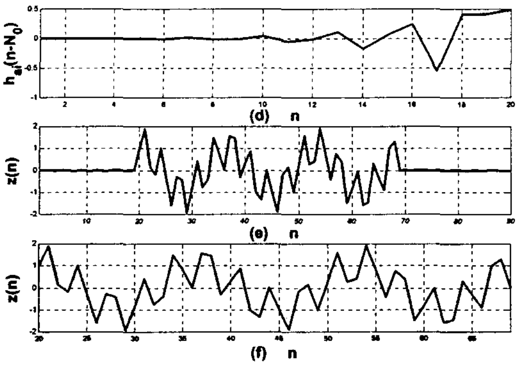


Fig. 4. Simulation results for Example 2.
(a) $h_a(n)$, (b) $x(n)$, (c) $y(n)$, (d) $h_{ai}(n-N_0)$,
(e) $z(n)$, (f) partially shifted version of $z(n)$

Ⅳ. Inverse Filtering Technique for General Systems

In the previous section we addressed the problem of inverse filtering when the original system is an all-pass system. We now consider the case when a system is not necessarily an all-pass system. In this case, the procedure for obtaining the desired inverse filter is best illustrated by two simple examples.

EXAMPLE 3 (for 2nd-order IIR filter):

Consider a second-order IIR filter whose transfer function is

$$H(z)=\frac{(z-5)(z-9)}{(z-\frac{1}{2})(z-\frac{1}{3})} \tag{14}$$

This filter is causal and stable when its ROC is $|z|>\frac{1}{2}$. Using the residue theorem, the impulse response is obtained as

$$h(n)=\frac{1}{2\pi j}\oint H(z)z^{n-1}dz$$
$$=\begin{cases}1 & for\ n=0\\ (459)\cdot(\frac{1}{2})^n-(728)\cdot(\frac{1}{3})^n & for\ n\geq 1\end{cases} \tag{15}$$

Next, the inverse $H_i(z)=1/H(z)$ is given by

$$H(z)=\frac{(z-\frac{1}{2})(z-\frac{1}{3})}{(z-5)(z-9)} \tag{16}$$

This inverse filter is non-causal and stable when its ROC is $|z|<5$. The corresponding impulse response is obtained via the residue method to be

$$h_i(n)=\frac{1}{2\pi j}\oint H_i(\frac{1}{p})p^{-n-1}dp$$
$$=\begin{cases}\frac{1}{270} & \text{for } n=0\\ (1.05)\cdot(\frac{1}{5})^{-n}-(\frac{552.5}{270})\cdot(\frac{1}{9})^{-n} & \text{for } n\leq -1\end{cases} \tag{17}$$

where $z=1/p$. Simulation results for $N_0=20$ are summarized in Fig. 5, for the case when the input is $x(n)$ in (11).

As given at the end of Example 1, similar explanations for the simulation results shown in Fig. 5 can be applied to this example.

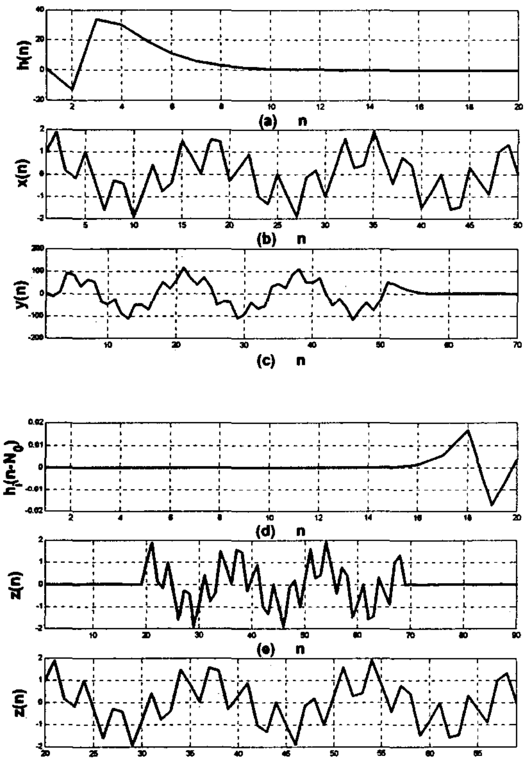


Fig. 5. Simulation results for Example 3.
(a) $h(n)$, (b) $x(n)$, (c) $y(n)$, (d) $h_i(n-N_0)$,
(e) $z(n)$, (f) partially shifted version of $z(n)$

EXAMPLE 4 (for 2nd-order FIR filter):

Consider a second-order FIR filter whose transfer function is

$$H(z)=\frac{2}{7}-\frac{15}{7}z^{-1}+z^{-2} \tag{18a}$$

$$= \frac{\frac{2}{7}(z-\frac{1}{2})(z-7)}{z^2} \tag{18b}$$

This filter is causal and stable when its ROC is $|z| > 0$. By inspection of (18a), the impulse response is

$$h(n) = \begin{cases} \frac{2}{7} & for\ n=0 \\ -\frac{15}{7} & for\ n=1 \\ 1 & for\ n=2 \\ 0 & for\ n \geq 3 \end{cases} \tag{19}$$

The transfer function of the inverse filter is

$$H_i(z) = \frac{\frac{7}{2}z^2}{(z-\frac{1}{2})(z-7)} \tag{20}$$

where $H_i(z) = 1/H(z)$. From (20) it is apparent that this inverse filter is non-causal and stable when its ROC is $\frac{1}{2} < |z| < 7$. Its impulse response is evaluated as follows:

(i) when $n \geq 0$

$$h_i(n) = \frac{1}{2\pi j}\oint \frac{\frac{7}{2}z^2 \cdot z^{n-1}}{(z-\frac{1}{2})(z-7)}dz$$

$$= Residue\ at\ z = \frac{1}{2}$$

$$= (-\frac{7}{26}) \cdot (\frac{1}{2})^n\ for\ n \geq 0 \tag{21}$$

(ii) when $n \leq -1$

$$h_i(n) = \frac{1}{2\pi j}\oint H_i(\frac{1}{p})p^{-n-1}dp$$

$$= \frac{1}{2\pi j}\oint \frac{p^{-n-1}}{(p-2)(p-\frac{1}{7})}dp$$

$$= Residue\ at\ p = \frac{1}{7}$$

$$= (-\frac{49}{13}) \cdot (\frac{1}{7})^{-n}\ for\ n \leq -1 \tag{22}$$

where $z = 1/p$. Using (19)-(22), simulation results for $N_0 = 20$ are summarized in Fig. 6, for the case when the input is $x(n)$ in (11).

As given at the end of Example 1, similar explanations for the simulation results shown in Fig. 6 can be applied to this example.

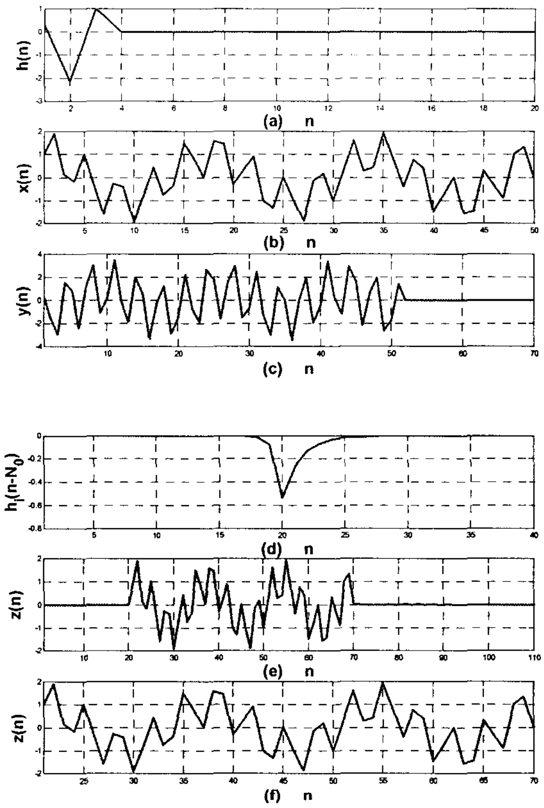


Fig. 6. Simulation results for Example 4. (a) $h(n)$, (b) $x(n)$, (c) $y(n)$, (d) $h_i(n-N_0)$, (e) $z(n)$, (f) partially shifted version of $z(n)$

V. Observations

From the previous two examples it is apparent that the process of designing an inverse filter that is causal and stable requires that we know the location of the poles and zeros of the original system. We can then choose an appropriate ROC with respect to those poles of $H_i(z)$ that lie outside the unit circle to obtain a non-causal, but stable filter. The introduction of an appropriate delay of N_0 samples then yield an FIR approximation to the desired inverse filter.

However, the proposed technique is essential to identify these zeros of $H(z)$ that lie outside the unit circle, since these zeros will become the poles of $H_i(z) = 1/H(z)$ that lie outside the unit circle. This basic problem is avoided when the channel is modeled by all-pass systems. This is because the poles of

$H_i(z)$ are simply the zeros of the all-pass section, which is known. Hence, for the case of all-pass systems, the design of the corresponding causal and stable FIR approximation to the inverse filter using the approach mentioned above becomes very straightforward.

Table 3. Mean square errors about four examples.

Examples	Mean square errors
Example 1	1.6305×10^{-6}
Example 2	2.5774×10^{-6}
Example 3	6.2174×10^{-11}
Example 4	1.4467×10^{-13}

Table 1 represents the mean square errors for given four examples. Where the error is defined as the difference between the reconstructed input by inverse filtering and the original input. Observing the plots shown in four figures corresponding to four examples and the mean square errors in Table 1, it can be recognized that the proposed inverse filtering technique works well. The tool used for computer simulations is signal processing tool box using MATLAB.

Ⅵ. Conclusion

This paper has developed an inverse filtering technique for deconvolving signals through the channel. Due to symmetric property of the impulse response of all-pass systems, this technique is more useful for all-pass systems than other systems. In order to illustrate the proposed inverse filtering technique, four examples were considered. Observing these examples, we found that this technique works very well. Also, we found that this technique is very useful for the channel modeled by all-pass systems.

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