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TELEVISION SIGNAL DEGHOSTING BY NONCAUSAL RECURSIVE FILTERING

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Abstract- **A signal corrupted by a ghosting channel can be deghosted by the appropriate inverse system. If the channel is nonminimum-phase, the inverse system has a noncausal impulse response. These can be implemented with recursive digital filters provided there are system elements to time-flip the signal. Such noncausal recursive filtering has advantages over strictly causal recursive filtering in its ability to equalize the phase as well as the magnitude response of the channel. It also has advantages over nonrecursive nonminimum-phase deghosting in that it requires much lower filter order.**

I. CHANNEL MODEL

Television ghosting is caused by reflective objects in the broadcast environment such as buildings and mountains. Due to the longer transmission path length, these reflectors introduce delay, and due to the electromagnetic properties, they may filter the transmitted signal. Such reflections are demodulated and displayed at the receiver along with the direct path signal. Most ghosts are seen as attenuated and shifted copies of the original image.

The impulse response of a ghosting channel is typically modelled as a finite number of weighted, delayed impulses, equivalent to a tapped delay line filter [1]. The time-sampled received signal, r[n], is represented by the discrete-time convolution

$$r[n] = s[n] * g[n] \qquad (1\text{ a})$$

where s[n] is the original video signal, and g[n] is the discrete-time channel impulse response, given by

$$g[n] = b_0\,\delta[n] + \sum_{i=1}^{N} b_i\,\delta[n-d_i]. \qquad (1\text{ b})$$

Here, b_i is a complex-valued reflection coefficient associated with the i^{th} reflector, and d_i is the number of samples of delay of the i^{th} ghost relative to the direct path signal. (This somewhat idealized model for g[n] may be expanded to incorporate filtering reflectors [2] and ghost delays of nonintegral sampling periods [3].) The zeros of G(z), the z-transform of g[n], may lie entirely within the unit circle in which case the channel is termed minimum-phase. Otherwise the channel is termed nonminimum-phase [4].

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II. THE DEGHOSTING SYSTEM

Deghosting is the process of ghosting channel equalization to mitigate the noticeable effects of the ghosts. Given (1) as our ghosting model, an approach to deghosting is filtering by the exact inverse, D(z)=1/G(z). In principle, this scheme would provide exact cancellation of all ghosts. In practice, the schemes used to estimate g[n] are not perfect due to noise and time varying ghosts, and the exact inverse might not be desirable if there is a strong additive noise component or zeros on the unit circle. In our discussion, however, we will assume that the desired equalizer is the ideal inverse system.

A causal, recursive filter can be used to realize or approximate D(z) if the channel is minimum-phase. If G(z) is nonminimum-phase, D(z) has poles which contribute to noncausal terms in the impulse response. For this case we propose the use of recursive filtering to implement these noncausal terms.

Specifically, consider the factorization of the ghosting channel

$$G(z) = G_i(z) \cdot G_o(z) \qquad (2)$$

where the zeros of $G_i(z)$ are inside the unit circle, and the zeros of $G_o(z)$ are outside the unit circle. We assume there are no zeros on the unit circle. Then the inverse system can be decomposed as a product,

$$D(z) = D_c(z) \cdot D_{ac}(z) \qquad (3\text{ a})$$

where

$$D_c(z) = 1 / G_i(z) \qquad (3\text{ b})$$

and

$$D_{ac}(z) = 1 / G_o(z), \qquad (3\text{ c})$$

or sum,

$$D(z) = E_c(z) + E_{ac}(z) \qquad (4\text{ a})$$

where

$$E_c(z) = N_c(z) / G_i(z) \qquad (4\text{ b})$$

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and

$$E_{ac}(z) = N_{ac}(z) / G_o(z). \quad (4\text{ c})$$

The poles of $D_c(z)$ and $E_c(z)$ are inside the unit circle; their impulse responses, $d_c[n]$ and $e_c[n]$, are strictly causal and infinite in length. The poles of $D_{ac}(z)$ and $E_{ac}(z)$ are outside the unit circle and consequently these systems correspond to stable, anticausal infinite impulse responses.

Since we are assuming real-time processing, these anticausal IIR systems cannot be realized exactly. One approach is to approximate such a system with a finite- length anticausal impulse response, which can then be implemented in real-time by incorporating a time delay equal to the impulse response length, and using a nonrecursive filter [5]. An alternative approach is based on time-flipping of the signal and anticausal impulse response, $h_{ac}[n]$. Specifically, with y[n] denoting the desired output of the anticausal filter and x[n] the input so that from

$$y[n] = h_{ac}[n] * x[n], \quad (5)$$

it follows that

$$y[-n] = h_f[n] * x[-n] \quad (6\text{ a})$$

where

$$h_f[n] = h_{ac}[-n]. \quad (6\text{ b})$$

The impulse response $h_f[n]$ is causal and infinite in length and can be implemented with a causal recursive filter. The task is now transformed into implementing $H_f(z)$, which has all its poles inside the unit circle since $H_f(z) = H_{ac}(1/z)$, and time-flipping the input and output signals. Since x[n] is of indefinite duration, it must be time-reversed in blocks [6,7]. We shall refer to this procedure as piecewise time reversal. It consists of approximating (6) by applying recursive filtering to time-reversed blocks of x[n]. The time reversal is accomplished with an element called a flip buffer. With this element, the system block diagrams for (3) and (4) are given in Figs. 1 and 2, respectively. Implementation or approximation of (6) will be called noncausal recursive (NCR) filtering.

III. IMPLEMENTATION

Since $E_c(z)$ and $E_{ac}(z)$ from (4) include numerator polynomials, any filters implementing them would need to provide for these with additional multipliers and adders. Consequently, implementation of D(z) by (3) is preferred.

Choice of Filter Form for $D_c(z)$ and $D_{ac}(1/z)$. The deghosting filters could be implemented by a variety of filter forms. However, the parallel filter would require the feedforward multipliers of each stage even though each filter is all-pole. Consequently, the direct form or cascade filter forms should be considered instead.

The cascade filter, employs two feedback coefficients to effect each pole pair. Consequently, it would require a total of d_N multipliers, since D(z) has d_N poles, where d_N is the delay of the N^{th} ghost. The direct form filter requires fewer because of the nature of the systems $1/G_i(z)$ and $1/G_o(z)$ which, in nonminimum-phase ghosting situations, comprise many zero valued coefficients which do not employ any multipliers at all.

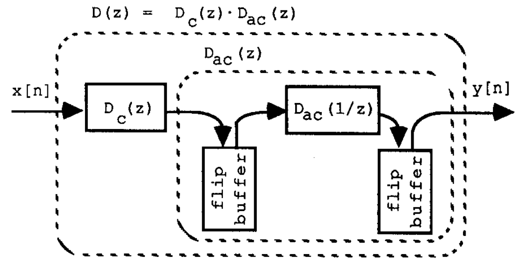

Fig. 1. Product form of noncausal recursive deghoster.

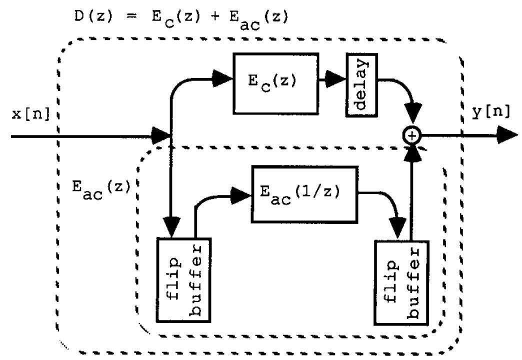

Fig. 2. Sum form of noncausal recursive deghoster. Note extra delay on top branch.

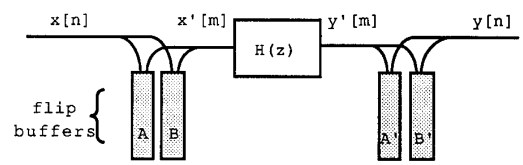

Fig. 3. Flip buffers straddling recursive filter H(z), which filters x'[m], the piecewise-time-reversed version of x[n].

In addition to the fact that it will require more multipliers, there is a more serious drawback to the use of the cascade filter. Since deghosting poles will typically lie quite near the unit circle, each stage will peak up a narrow band of frequencies and attenuate a broad region. Several such regions of adjacent stages may attenuate data in that part of the spectrum to a point that the data path has insufficient precision to represent all the information. In our simulation, as few as three successive stages caused problems even when four-byte floating-point arithmetic was used for NCR deghosting.

As shown in Fig. 1, the system $D_c(z)$ can continuously filter the incoming signal. However, in implementation of the anticausal system $D_{ac}(z)$, comprising flip buffers and the filter $D_{ac}(1/z)$, the following issues arise due to piecewise time reversal.

Time Reversal of the Signals. The input signal is broken up into segments of length F, called *flip blocks*, which overlap some number of samples, O. Due to the overlap, a new block begins every F-O samples of the input signal.

There must be two or more pairs of flip buffers, as shown in Fig. 3. These system elements could be realized by a chain of bidirectional shift registers; they form a first-in-last-out register so that the signal x'[m] contains time-flipped segments of x[n]. The filter H(z) is the, causal, stable recursive filter equal to $D_{ac}(1/z)$ (or $E_{ac}(1/z)$ if the sum form is employed). At the output, flip buffers process the sequence y'[m] by flipping it again and removing the overlapped portion of the flip blocks to form y[n].

FIR Nature of Impulse Response Resulting from Piecewise Time Reversal. For the effective impulse response of the system to be anticausal and infinite, the entire sequence x[-n] must be filtered by the recursive filter H(z). However, with piecewise time reversal, it is not the case that x[n] for all $n > n_0$ precedes sample $x[n_0]$ into the filter. Assuming the state of the filter is cleared before the introduction of the each flip block, the effective length of the impulse response at $y[n_0]$ is equal to the number of temporally contiguous samples which preceded $x[n_0]$ into the filter.

In any block processing scheme, the samples in the filtered block can be considered to have been filtered by an FIR filter of length equal to the smaller of the following: the length of the impulse response of the processing filter, or, the distance the sample was from the beginning of the block. In applications where the processing filter is FIR, the overlap length is chosen to be the same length; samples in the overlap region are excluded from the final output because they are not filtered by the whole impulse response. In our case, the use of a recursive filter provides an infinite impulse response at the block level, so the distance that a sample was from the beginning of the block determines the length of the effective FIR deghosting response.

The overlap length should consequently be chosen as the same value as the minimum length of an acceptable FIR approximation the infinite anticausal response. That is, since the samples in the overlapped portion of the blocks are excluded from the output, the samples in the nonoverlapped portion are filtered by an effective impulse response of at least the overlap length.

Choice of Overlap and Flip Buffer Lengths. As mentioned, the overlap length, O, should be chosen to provide an adequate approximation of the infinite anticausal response. If it is too short, artifacts will appear in the video; they will be of greatest magnitude in the portion near the overlap region. In the deghosted signal, they will appear along a single or adjacent horizontal lines; this spatial correlation will increase their perceptibility. Consideration of the anticausal portion of several deghosting impulse responses suggests that 2000 samples is a reasonable working figure for O.

Since the filter must process samples in the overlap region twice as they appear at the head of one block and tail of another, the filter H(z) must work at a shorter period per sample than the filter implementing $D_c(z)$ or $E_c(z)$. The consequence for choosing an excessive overlap length is a requirement for a very fast clocking rate for the filter H(z).

The flip buffer length, F, may be chosen bearing the following in mind: the delay is proportional to $2 \cdot (F-O)$ because the entire block must be flipped both at the input and output. Also, processing rate requirements for the filter H(z) are relaxed by choosing F large compared to O.

If the choice for O is too small for a given anticausal deghosting impulse response, artifacts will appear at the edge of the output of each flip buffer, that is, each F-O samples. Consequently, the larger F is, the less frequently these artifacts will appear. Alternately, F-O can be chosen to span exactly one field of video so that the edge of the flip block occurs such that any artifacts appear off screen.

IV. SIMULATION RESULTS

The NCR deghosting described here was simulated by computer processing. Color composite NTSC format video was used.

Photo 1 shows the original image. Photo 2 shows ghosting by the nonminimum-phase impulse response $g[n] = .43\delta[n] + .30\delta[n-24] + .26\delta[n-36]$. Photo 3 demonstrates the success of NCR deghosting. Photo 4 shows artifacts which crop up for a choice of F and O which are too short for this particular ghosting situation.

V. COMPARISON TO OTHER APPROACHES

Causal Recursive Equalizing. It should be noted that even when the ghosting channel is nonminimum-phase, the magnitude response can be equalized by a causal all-pole filter. However, the overall phase response is not compensated. Specifically, the cascade of such an equalizer with the channel constitutes an all-pass filter. The resulting nonlinear phase response degrades the transmitted video by smearing vertical edges and distorting the hue-bearing information in color TV signals. NCR filtering, on the other hand, can be thought of as both magnitude and phase equalization for ghosting channels.

Noncausal Nonrecursive Filtering. Although the effective length of the deghosting impulse response is not infinite because of piecewise time reversal, it is still advantageous to use the NCR filter rather than a nonrecursive filter. Each scheme relies on the fact that the infinite anticausal impulse response can be truncated at some point. For typical ghosting situations, the deghosting filter may comprise 100 poles and may be truncated to 2000 samples without introducing noticeable visual artifacts.

In such a situation, the nonrecursive filter would require at most 2000 multipliers to implement its approximation to the ideal deghosting impulse response. By comparison, the recursive filter would need at most 100 multipliers to implement the 100 poles. There is a requirement for the overlap buffer to be 2000 samples long; since only samples in the nonoverlapped portion of the flip block become part of the output and all sampled in this portion are filtered by an effective impulse response length of at least O.

Not only, then, does the NCR filter work with fewer multipliers, its approximation of the ideal response improves throughout the block since the effective length of the impulse response grows, approaching the ideal. Also, improving the approximation of the nonrecursive filter corresponds to adding multipliers; improving the NCR filter corresponds to lengthening O, that is, merely adding delay elements.

However, due to the fact that NCR filtering must wait until the whole flip buffer is full before it processes a flip block and because it must wait until the whole block it processed before it shifts any block as output, there is a delay of 2·(F-O). This is, in general, much longer than the delay encountered by use of the nonrecursive filter.

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Photo 1: Original Image

Photo 2: Image distorted by nonminimum-phase ghost; g[n]= .43 δ[n] + .30 δ[n] + .26 δ[n]

Photo 3: Deghost of photo 2 by noncausal recursive filtering.

Photo 4: Artifacts introduced from deghost of photo 2 when F and O are too short.