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An Approach to Active Sonar Suppression Using a Dynamic Interference Model

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FOR THE COMMANDER

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MASSACHUSETTS INSTITUTE OF TECHNOLOGY
LINCOLN LABORATORY

AN APPROACH TO ACTIVE SONAR SUPPRESSION USING A DYNAMIC INTERFERENCE MODEL

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ABSTRACT

This report summarizes preliminary work in the suppression of interfering active sonar for enhancement of underwater transients. The active sonar is modeled as an amplitude-modulated (AM) and frequency-modulated (FM) chirp signal. The estimation and suppression of the AM-FM chirp is performed in the framework of an analysis/synthesis technique that is based on a sine-wave representation of signals with components characterized by time-varying amplitudes, frequencies, and phases. Parameters of the chirp model, estimated within the sine-wave analysis, are used to reconstruct a chirp over short-time intervals, according to its dynamics. The contribution of the chirp is then subtracted from the sine-wave components of the received signal (i.e., the sum of interfering signal and desired transient), and the enhanced signal is obtained by sine-wave synthesis of the remaining contribution due approximately to the transient. Significant interference suppression is obtained because the chirp dynamics are preserved over short-time segments.

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1. INTRODUCTION

An important naval scenario involves a sonar operator who may be listening for neighboring submarines; the operator may be listening, in particular, for submarine-generated transients, for long periods of time under adverse conditions. These conditions include the presence of interfering signals that are often louder than the transient, thus reducing the sonar operator's ability to detect and discriminate submarine transients. Recently, not only have the submarine transients become quieter, but the interference has become stronger. Signal enhancement by interference suppression would improve the audibility of the discriminating signal and reduce the fatigue and stress of the sonar operator. Transient signals that help to identify the submarine are often atonal and rapidly varying, including sounds from changes in submarine movements (e.g., decompression) and engine speed, and internal sounds such as slamming doors. Interfering signals include ocean background noise, flow noise over hydrophones, natural biological and geological sounds, noise from surface ships, and active sonar.

This report summarizes preliminary work in the suppression of interfering active sonar, which is modeled as an amplitude-modulated (AM) and frequency-modulated (FM) chirp signal. Suppression is assumed to occur on a single channel in contrast to multichannel techniques, such as adaptive beamforming [1], that are not model-based approaches. The framework for the new approach is an analysis/synthesis technique that is based on a sine-wave representation of signals with components that are characterized by time-varying amplitudes, frequencies, and phases [2,3]. Sine-wave analysis is used to estimate chirp signal parameters, which are used to reconstruct a chirp over short-time intervals, according to its dynamics. The contribution of the chirp is then subtracted from the sine-wave components of the received signal (i.e., the sum of interfering signal and desired transient), and the enhanced signal is obtained by sine-wave synthesis of the remaining contribution due approximately to the transient. Significant interference suppression is obtained because the chirp dynamics are preserved over short-time segments.

Section 2 briefly describes state-of-the-art underwater active sonar suppression, Section 3 reviews sinusoidal analysis/synthesis, Section 4 describes the approach of the proposed work and preliminary results, and Section 5 presents conclusions and discusses future work.

2. STATE OF THE ART

Approaches to interference suppression attempt to remove the interfering signal without distorting the desired signal. In the case where the interference is a large active sonar from surface vessels (e.g., 50 dB above the transient), there is no existing technique that adequately satisfies these two constraints. In the context of single-channel techniques, at least two methods are being considered.

One method of suppression uses least-mean-square adaptive cancellation [4], which requires a reference signal. An adequate reference can be found as long as the source of interference is close to its destination, i.e., the listening submarine, and as long as the interference is stationary, e.g., the interfering engine tones emitted by the listening submarine. In the cancellation of active sonar, the reference signal may be known, as in the case of friendly interference. In this simple scenario and where the active sonar tone (CW) is steady, adaptive cancellation performs well. This approach has not been demonstrated, on the other hand, when the active sonar is rapidly changing (e.g., a linearly changing FM chirp) or when the source of interference has traversed a path that is not represented by a slowly varying linear filter.

An alternate approach under consideration uses phase-lock loops to estimate the chirp signal to be subtracted from the received signal. The phase-lock loop, which tracks the chirp phase in the time domain by using a causal signal prediction, is derived under the assumption of a slowly varying FM in the presence of random noise with certain statistical characteristics, e.g., white Gaussian [5]. The presence of a *transient*, the *signal of interest*, may then act as a nonrandom perturbation that violates the assumed noise model. Other problems concern the intermittency of the active sonar, possibly causing synchronization problems for the phase-lock loop, and a multipath that the sonar signal may traverse prior to reaching its destination. Multipath may also be a problem for adaptive cancellation.

3. SINUSOIDAL FRAMEWORK

The sine-wave representation of a signal is given by a sum of sine waves with time-varying amplitudes, frequencies, and phases [2,3]:

$$s(t) = \sum_{k=1}^{N} A_k(t)\cos[\theta_k(t)], \tag{1}$$

where the amplitudes and phases are denoted by $A_k(t)$ and $\theta_k(t)$, respectively. The time-varying frequency of each sine wave is given by the derivative of the phase and is denoted by $\omega_k(t) = \dot{\theta}_k(t)$. The components in Eq. (1) thus represent the amplitude and phase of each sine-wave component along the frequency trajectory $\omega_k(t)$. Although this model was originally formulated for speech signals, it is capable of representing many nonspeech signals such as music, underwater sounds, and those consisting of the sum of two or more components, such as speech that is added to music or noise.

Using the above model, an analysis/synthesis system has been developed. A sampled data notation is used because measurements are made using digitized sounds. In particular, the continuous time variable t is replaced by the integer-valued index $n = 0, 1, 2, ...$ where $n = t/T_o$ with T_o being the time sampling period. In the experimental work in this report, $T_o = 100$ μs, corresponding to a sampling rate of 10,000 samples/s and a 5-kHz signal bandwidth. On each analysis frame the sine-wave parameters are estimated at time samples $n = mQ$, where the frame number $m = 0, 1, 2...$, and where Q is the number of samples in the frame interval. The dependence of the sine-wave parameters on the discrete time variable n is therefore replaced by their dependence on the frame number m, e.g., $A_k(n)$ is replaced by $A_k(mQ)$ or for simplicity by $A_k(m)$. A 5- to 10-ms frame interval has been found to produce high-quality reconstruction for most signals of interest.

The analysis window (typically 25 ms) is placed symmetric relative to the origin, which is defined by the center of the current analysis frame; hence the window takes on values in the interval $-L/2 \leq n \leq L/2$ where L is the length of the window. A short-time Fourier transform (STFT) is then computed over this duration with a fast Fourier transform (FFT). The excitation frequencies $\omega_k(m)$ are estimated by picking the peaks of the uniformly spaced (FFT) samples of the STFT magnitude. The sine-wave amplitudes $A_k(m)$ and phases $\theta_k(m)$ at the center of each analysis frame are then given by the amplitude and phase of the STFT at the measured frequencies.

The first step in synthesis requires associating the frequencies $\omega_k(m)$ measured on one frame with those obtained on a successive frame. This process is accomplished with a nearest-neighbor matching algorithm that incorporates a birth-death process of the component sine waves, which are allowed to come and go in time. Amplitude $A_k(m)$ and phase $\theta_k(m)$ parameters are interpolated across frame boundaries at the matched frequencies to upsample to the original sampling rate. The amplitude is interpolated linearly and the phase is interpolated with a cubic polynomial, the latter using the methods described in McAulay and Quatieri [2,3]. The interpolated amplitude and phase components are used to form an estimate of the waveform according to Eq. (1). A block diagram of the baseline sine-wave analysis/synthesis system is presented in Figure 1. Experiments have shown that the synthetic signals (a large set of speech, music, and underwater sounds) produced by this system are essentially perceptually indistinguishable from the original.

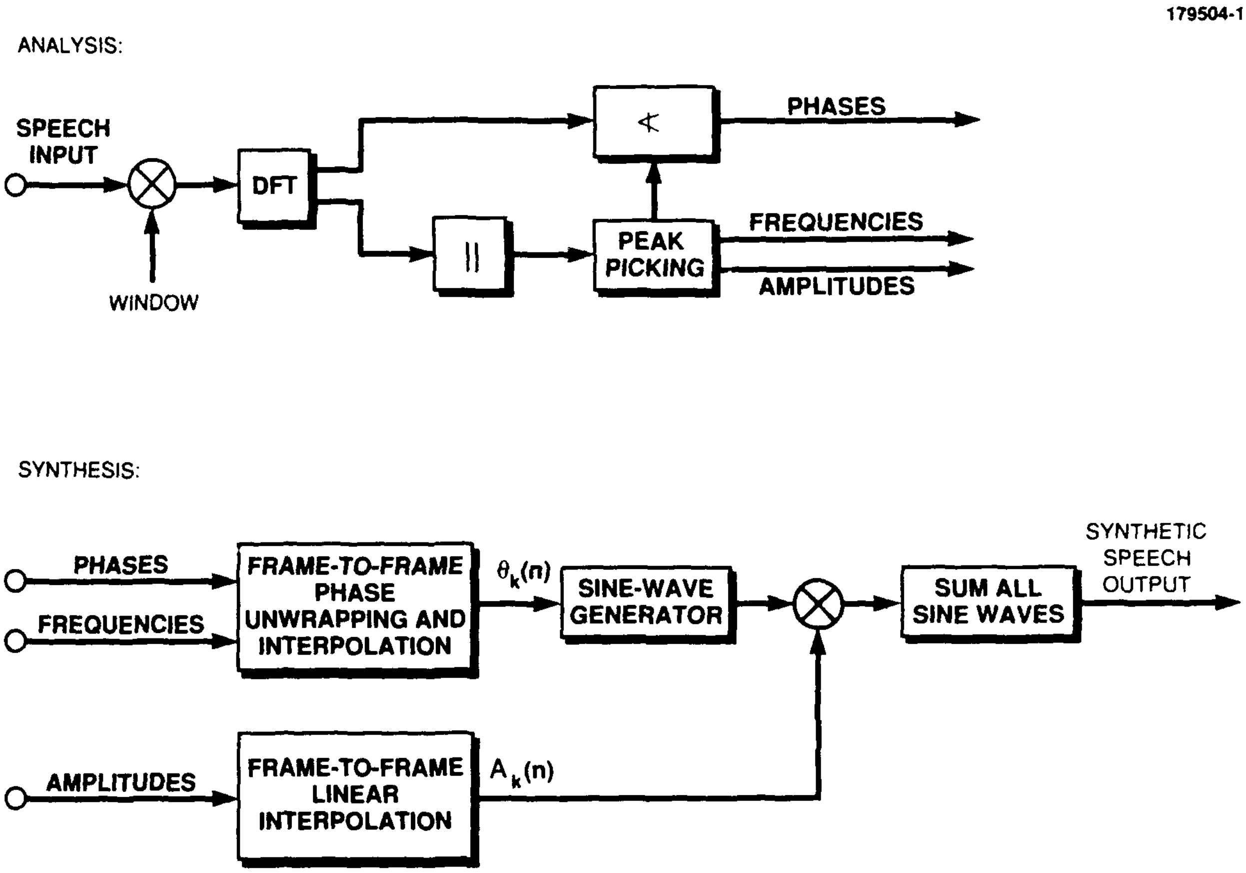

Figure 1. Block diagram of the baseline sinusoidal analysis/synthesis system.

4. APPROACH TO INTERFERENCE SUPPRESSION

Active sonar is typically generated as a periodically occurring chirp signal with an AM envelope and a phase derived from FM. The chirp signal is written as

$$c(t) = a(t)\cos[\phi(t)] \quad -T/2 \leq t \leq T/2, \tag{2a}$$

where T is the chirp duration; the phase signal is written as

$$\phi(t) = \int_{-T/2}^{t} \omega(\tau)d\tau + \phi(-T/2). \tag{2b}$$

where $\phi(-T/2)$ is the initial phase and where (in this report) the sweep frequency is assumed linear of the form

$$\omega(t) = \beta t + \omega_o, \tag{2c}$$

with β being the sweep rate and ω_o a frequency offset that can be considered the carrier frequency. Continuous time notation has been used because instantaneous frequency must be integrated. With this linear frequency assumption, the phase $\phi(t)$ takes on a quadratic form. The chirp amplitude $a(t)$ is likewise assumed piecewise linear. The received signal is the chirp interference periodically repeated, i.e., $\sum_k c(t - kR)$ with the chirp repetition rate R^{-1} added to the desired transient $d(t)$ as well as to other background interferences $e(t)$:

$$x(t) = d(t) + c(t) + e(t), \tag{3}$$

where for notational simplicity only the prototype chirp is assumed present. The approach to active sonar suppression is to reconstruct the interfering chirp $c(t)$ over the duration of the short-time analysis window by using estimated model dynamics, i.e., the linear sweep frequency $\omega(t)$ and linear amplitude $a(t)$, with some appropriate starting conditions, and then to subtract the reconstructed chirp from the received signal. The estimation and reconstruction take place in the context of the sine-wave analysis/synthesis system of Section 3. Ideally, this operation should preserve the perceptual quality of the desired transient.

4.1 Algorithm

The proposed algorithm to eliminate AM-FM chirp interference is illustrated in Figure 2. The input waveform consists of the desired transient $d(t)$ and a relatively large chirp interference

$c(t)$. In this report a Hamming analysis window is applied; its duration is 25 ms with a shift at a frame interval of 10 ms. The window duration was chosen as a compromise between time and frequency resolution with respect to the time varying chirp frequency [6]. The frame interval was chosen empirically to capture the time variations in both the chirp and transients of interest. In the analysis stage of the sine-wave analysis/synthesis, estimates are made of the amplitudes, phases, and frequencies of the sine waves of the received signal, as described in Section 3. The sine-wave parameters contain information about both the interfering and desired signals. Seventy sine waves were found adequate to represent the signals of interest.

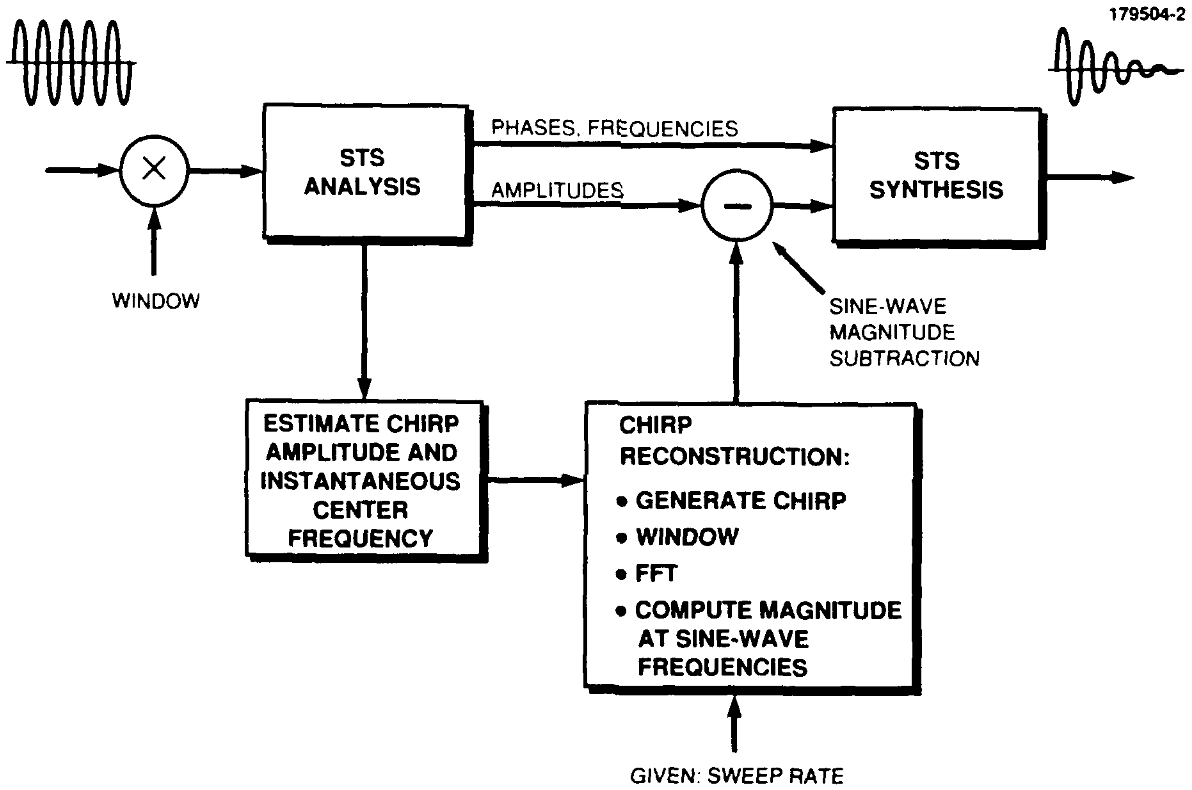

Figure 2. A proposed algorithm for AM-FM chirp interference suppression.

The analysis stage is also used to estimate parameters of the AM-FM chirp interference model, ideally including the frequency, amplitude, and phase at the center of the analysis window.

as well as the frequency sweep rate and amplitude trajectory. Because the chirp frequency changes linearly (and "slowly") over the duration of the symmetric analysis window, and because the chirp dominates the transient, the chirp parameters at the window center are assumed to correspond to the sine wave with the largest amplitude. More specifically, the chirp amplitude estimate at the center of the mth analysis frame is given by

$$\begin{aligned} \hat{a}(m) &= \max \ A_k(m) \ \text{with respect to } k \\ &= A_{k_{\max}}(m), \end{aligned} \tag{4}$$

where $A_k(m)$ are the measured sine-wave amplitudes in the model Eq. (1) and $k_{\max}$ denotes the frequency at which the maximum value in Eq. (4) occurs. The FFT length is chosen at 2048 to assure adequate resolution of both the chirp center frequency and sine waves that compose the underlying desired transient. The search interval for the maximum is constrained to equal the chirp frequency range, which is known a priori. The chirp phase at the center of the analysis frame is given by

$$\hat{\phi}(m) = \theta_{k_{\max}}(m), \tag{5}$$

where $\theta_k(m)$ are the sine-wave phases in Eq. (1). Therefore, with knowledge of the amplitude and frequency slopes in Eq. (2), an estimate of the chirp signal can be obtained, i.e., sufficient constraints exist to determine the linear amplitude and quadratic phase estimates. For the purposes of this report, the linear frequency sweep rate, i.e., β in Eq. (2c), is known a priori, and the chirp amplitude envelope is assumed constant and equal to the estimate in Eq. (4).

The chirp signal is reconstructed by projecting the chirp frequency forward and backward over the window duration $-L/2 \leq n \leq L/2$ from the center of the analysis window according to the sweep rate. Denoting by $\hat{\omega}_o$ the estimate of the center frequency and assuming a constant amplitude $\hat{a}(m)$ over the analysis window, the estimated chirp for the mth frame is given by

$$\hat{c}(n;m) = \hat{a}(m)\cos[\hat{\phi}(n;m)], \tag{6a}$$

where from Eqs. (2) and (5)

$$\hat{\phi}(n;m) = \beta\frac{n^2}{2} + \hat{\omega}_o n + \hat{\phi}(m) \quad -L/2 \leq n \leq L/2, \tag{6b}$$

where the time sampling interval has been normalized to unity. The chirp $\hat{c}(n;m)$ is windowed with the same window used in the original analysis

$$\tilde{c}(n;m) = w(n)\hat{c}(n;m). \tag{7a}$$

The Fourier transform magnitude of the chirp estimate is then computed at the sine-wave frequencies measured in analysis

$$|\tilde{C}_k(m)| = |\tilde{C}(\omega;m)|_{\omega=\omega_k}, \tag{7b}$$

where $\tilde{C}(m;\omega)$ is the Fourier transform of $\tilde{c}(n;m)$ and where ω_k are the sine-wave frequencies in Eq. (1). The chirp magnitude contribution Eq. (7b) is subtracted from the measured sine-wave amplitudes to obtain an estimate of the sine-wave amplitudes of the desired signal $d(n)$:

$$|\hat{D}_k(m)| = \begin{cases} A_k(m) - \alpha|\tilde{C}_k(m)| & \text{if } A_k(m) - \alpha|\tilde{C}_k(m)| \geq 0 \\ 0 & \text{otherwise,} \end{cases} \tag{8a}$$

and the phase estimate of the desired signal is left intact

$$\angle\hat{D}_k(m) = \theta_k(m). \tag{8b}$$

In Eq. (8), $D_k(m) = D(\omega;m)|_{\omega=\omega_k}$, where $D(\omega;m)$ is the Fourier transform of the desired transient over the analysis window. $\hat{D}_k(m)$ denotes the estimate of $D_k(m)$. The constant α allows a possible gain adjustment in the estimated chirp, e.g., normalization with respect to the energy in the received signal over each frame. An estimate of the transient is then found by performing sine-wave synthesis based on the amplitude and phase estimates in Eq. (8).

An implicit assumption in this algorithm is that the measured peak amplitudes are the sum of the peak amplitudes of the transient and the chirp, which generally is only an approximation due to the complex nature of the Fourier transform. In addition, because the measured phase is used in the reconstruction, some distortion will be introduced into the time structure of the estimated transient; in spectral regions where the chirp is much larger than the transient, the chirp phase dominates. Ideally, a complex subtraction should then be performed; however, such an operation may be sensitive to the perturbations in the initial frequency and phase measurements.

4.2 Preliminary Results

To test the algorithm, a transient was created in the laboratory by digitally recording the acoustic signal from the bounces of a can that was dropped on a table. A periodically occurring chirp interference signal was added digitally (see Figure 3). The chirp repeats every second with

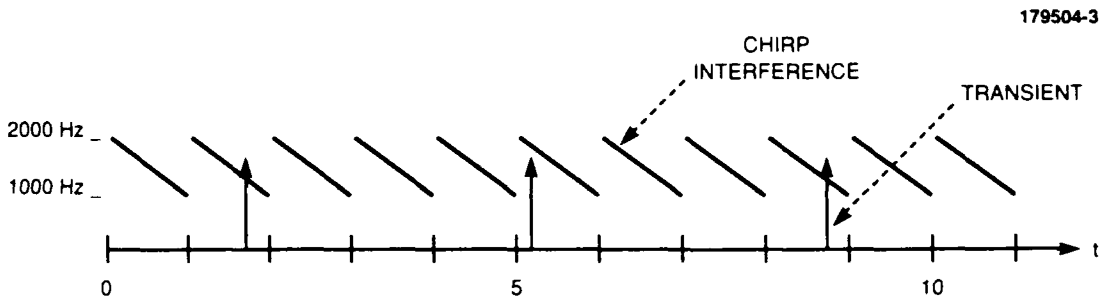

Figure 3. Test signal of three falling-can transients added to an interfering chirp.

a small gap between each, and chirp frequency runs from 2000 to 1000 Hz. The can is dropped roughly every 4 s with multiple bounces occurring per drop.

Figure 4 gives an example of chirp suppression using the algorithm of Figure 2. Figures 4(a) and (b) show one bounce of the falling can added to the chirp interference, both in the time and the frequency domains. As illustrated, the chirp dominates the transient; in the time domain the transient is a small glitch in the waveform, while in the frequency domain the transient appears as a perturbation on the dominant main- and sidelobes. Moreover, the bulk of the chirp spectrum is located at about the spectral peak of the falling can, thus providing roughly a worst-case situation. Figures 4(a) and (b) illustrate the enormity of the active sonar suppression problem.

Figures 4(c), (d), and (e) show the result of the processing. Figures 4(c) and (d) give the superimposed original (dashed) and estimated (solid) spectrum of the transient. Figure 4(e) shows the original (dashed) and estimated transient (solid; displaced for display purposes) in the time domain. There is good spectral and temporal resolution of the estimated transient and excellent reduction of the chirp sidelobes with about 44-dB interference suppression. This suppression measurement was made by computing the average energy in the residual waveform at the output of the suppression system in a region without the transient. The residual was compared to the energy in the interfering chirp when it passed through the sine-wave analysis/synthesis system with the suppression mechanism disengaged; this operation was performed to ensure the proper reference energy. It is clear from Figures 4(c) and (d) that the large sidelobes of the interfering chirp have been effectively eliminated. The spectral region of greatest transient distortion, manifested by a large notch, occurs at the chirp spectral peak where sensitivity to the subtraction is greatest. In informal listening, the audibility of the falling-can transient has been significantly enhanced and the perceptual character of the can has been essentially preserved. The remaining FM chirp residual perceptually takes on the form of a low-level background "whishing."

To contrast the new technique of suppression with a conventional approach, an adaptive notch filter was applied in the context of the sine-wave analysis/synthesis. In this procedure, the frequency derived from the maximum search in Eq. (4) is used as the center frequency of the notch

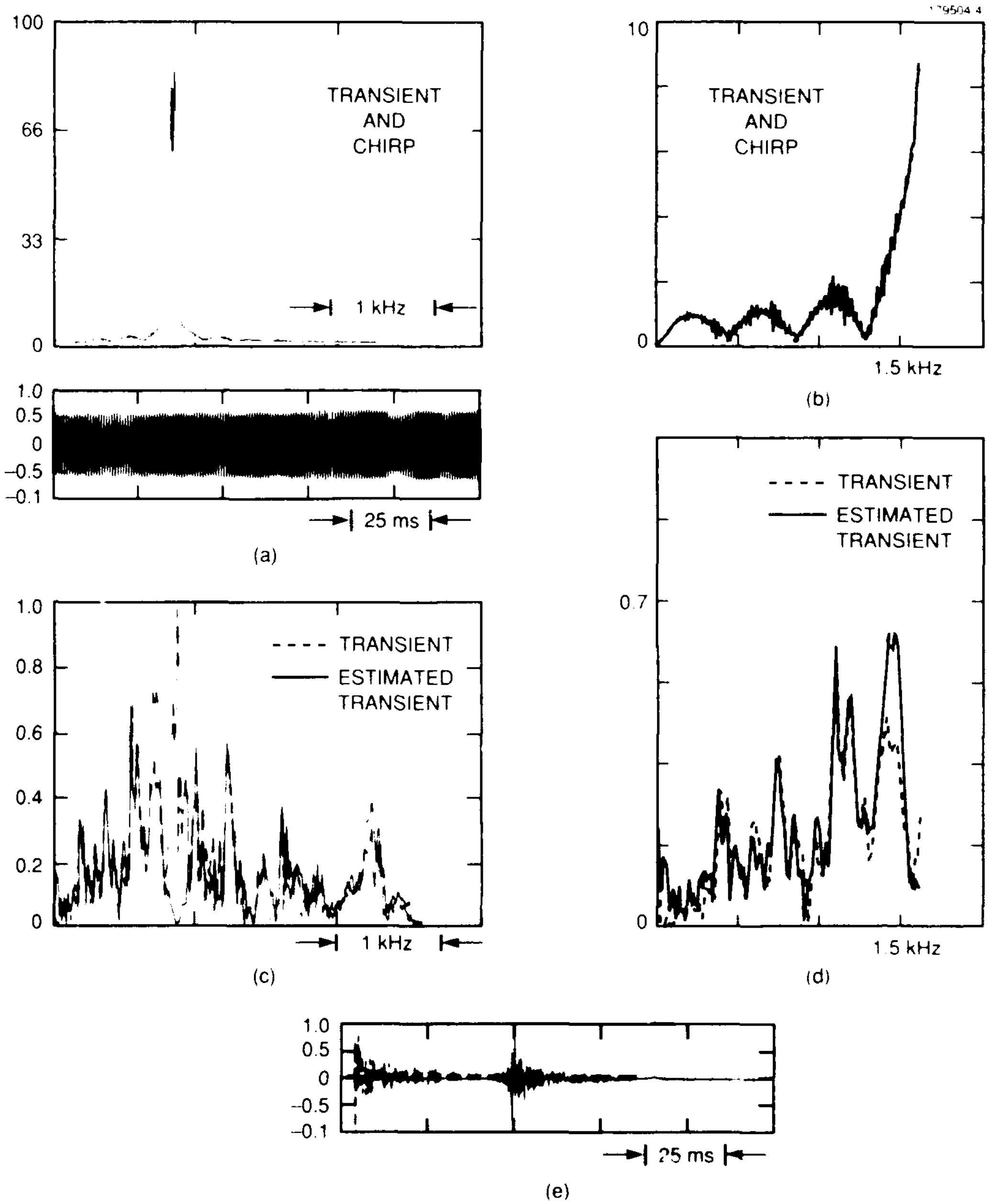

Figure 4. Example of AM-FM chirp interference suppression. (a) Transient plus chirp; (b) blowup of (a); (c) superimposed transient and estimate (in frequency); (d) blowup of (c), and (e) superimposed transient and estimated transient (in time).

filter. Different notch shapes were tested, all with about a 100- to 200-Hz notch width. Application of the notch filter does not account for the underlying chirp dynamics and, as a result, does not reduce sidelobes without perceptual distortion of the transient. With acceptable preservation of the quality of the transient, the chirp residual is quite large and perceptually annoying.

5. DISCUSSION AND FUTURE WORK

This report describes a new approach to suppress active sonar interference, one that relies on estimating the parameters of a dynamic model of the interference and performs in the framework of sine-wave analysis/synthesis. The algorithm accounts for the time variation of the chirp via the model dynamics over short-time segments and, hence, results in large sidelobe suppression. In contrast, applying an adaptive notch filter located at the estimated center frequency does not account for the underlying dynamics and the frame-to-frame sidelobe variation. This comparison indicates that viewing the interference as a deterministic noise source may be preferred over conventional statistical approaches to noise reduction where the interference is seen as a stationary random process [7]. These preliminary results represent the first stage of the effort. In developing the approach, the heuristics of the technique will be placed in a more analytic framework, and environmental effects such as multipath and ocean noise will be considered. Moreover, ways are being investigated to improve the basic technique, remove idealized assumptions and make it practical, and extend the approach to other classes of deterministic interference.

One improvement involves refining the estimated chirp phase, amplitude and frequency. The algorithm in Figure 2 uses only the spectral magnitude in doing suppression; zero spectral phase was assumed in the chirp reconstruction using Eq. (8b). Phase would require a complex spectral subtraction and thus, in the time domain, corresponds to a subtraction of the estimated chirp from the received signal. Although using phase can potentially improve suppression by giving a *more accurate reconstruction*, the possibility of drift in the signal time structure warrants careful consideration because the chirp is reconstructed by projection from an initial phase estimate. To avoid drift, an analysis-by-synthesis approach is being considered to refine the phase estimate. This approach is similar to a recently proposed method of reconstructing chaotic sequences with uncertain initial conditions [8]. A refinement of this type may help reduce spectral distortion of the transient in regions of the spectrum where the chirp dominates, such as at the spectral peak in Figure 4. Another related area of future work is the development of algorithms to estimate the FM sweep rate, which was assumed known, as well as the AM envelope, which was assumed constant. One approach being considered to estimate these dynamics is a new technique to track AM envelope and FM instantaneous frequency of chirp signals [9–11]. The method, using a three-point time-domain operator, provides the potential for a highly time-resolved AM and FM detector.

Another consideration is AM-FM chirp parameter estimation in the presence of disturbances, such as hydrophone flow and ocean noise, as well as the transient itself. The chirp center frequency and amplitude estimates are prone to error due to these background terms as well as inherent inaccuracies in the spectral peak-picking. A standard method to reduce the effect of such error is to average nearby points. For this method to work, the average must be taken over a short-time interval. For example, one might consider averaging estimates of the center frequency obtained over adjacent frames and then using this estimate in the algorithm of Section 3. Inherent in this procedure, however, is smearing the frequency estimate and possibly a less effective suppression. As an alternative, it is proposed to make multiple predictions of the chirp from both the past and future adjacent segments based on chirp parameters estimated over these segments and average the predictions, which may allow averaging over much longer time periods. Predictions may occur from points that are close in time as well as those that are closely aligned on the phase space trajectory of the AM-FM chirp dynamics and would allow a larger number of elements to be averaged than could be obtained solely from closely spaced points in time. Although the specifics are quite different, this notion is not unlike the approach recently taken by Farmer and Sidorowich [12] to reduce noise in nonlinear prediction, which appears to be particularly effective when the desired signal results from a chaotic dynamical system.

Clearly, the approach of viewing the interference as a deterministic noise is applicable to other forms of interference. Biologics, such as whale and dolphin signals, may be modeled by AM-FM signals, hence the current technique may be directly useful. More generally, other forms of underwater noise such as geologic and hydrophone flow, which appear to be random, may have an underlying deterministic nature. Such interference may be modeled, for example, by the output of a chaotic dynamical system that has recently provided the basis for modeling many apparently random signals [12].

Finally, in underwater signal reception beamformers are used to obtain direction information [1]. Each channel provides a time delay, and typically up to 50 channels may exist in forming the beam. A decision must be made as to whether suppression is to be applied before or after beamforming. Some timing inconsistency among channels may be introduced if suppression occurs prior to beamforming because the suppression may modify the phase characteristic of the transient (an important study in itself). In terms of implementation, suppression prior to beamforming requires up to 50 parallel channels, while suppression after beamforming must take place for each beam. In addition, the original model of the interference source may need altering according to beamforming characteristics.

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13. ABSTRACT (*Maximum 200 words*)

This report summarizes preliminary work in the suppression of interfering active sonar for enhancement of underwater transients. The active sonar is modeled as an amplitude-modulated (AM) and frequency-modulated (FM) chirp signal. The estimation and suppression of the AM-FM chirp is performed in the framework of an analysis/synthesis technique that is based on a sine-wave representation of signals with components characterized by time-varying amplitudes, frequencies, and phases. Parameters of the chirp model, estimated within the sine-wave analysis, are used to reconstruct a chirp over short-time intervals, according to its dynamics. The contribution of the chirp is then subtracted from the sine-wave components of the received signal (i.e., the sum of interfering signal and desired transient), and the enhanced signal is obtained by sine-wave synthesis of the remaining contribution due approximately to the transient. Significant interference suppression is obtained because chirp dynamics are preserved over short-time segments.

14. SUBJECT TERMS
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