Fast Convergence Algorithm for Adaptive Noise Cancellers with 
SNR-Based Stepsize Control

Akihiko Sugiyama
Yahoo! JAPAN Research, Yahoo Japan Corporation
Kioi Tower, Tokyo Garden Terrace, 1-3 Kioicho, Chiyoda-ku, Tokyo 211–8666, JAPAN

Abstract
This paper proposes a fast convergence algorithm for adaptive noise cancellers with SNR-based stepsize control. SNR-based stepsize control reduces interference by the noise-cancelled signal in adaptation. A second SNR estimate initially controls the stepsize, followed by a first SNR estimate to promote coefficient growth. The numerator of the second SNR estimate is the noise-cancelled signal obtained as the subtraction result different from the primary signal in the conventional algorithm to accelerate initial convergence. Evaluations with clean speech and noise recorded at a busy station demonstrate that the time until SNR-estimate switchover is reduced by 91% compared to the conventional algorithm, leading to fast evolution of adaptive filter coefficients.

I. Introduction
Adaptive noise cancellers (ANCs) [1] are more suitable for signal enhancement than noise suppressors when SNR (signal-to-noise ratio) is relatively low. In human-robot communication and mobile phone handsets, ANCs have demonstrated high potentials [2, 3]. An adaptive filter in the ANC generates a noise replica to be subtracted from the primary microphone signal and the coefficients are updated with the subtraction result, or equivalently, the noise-cancelled signal, which consists of pure error and the target signal. The target signal is not correlated with the error and interferes coefficient adaptation, leading to distortions in the target signal and insufficient noise cancellation [2].

Undesirable effect of the interference can be reduced by controlling the coefficient adaptation stepsize based on an estimated SNR [3]. Serious interference is detected by a high SNR and the stepsize is set small to minimize the effect of interference. However, in initial convergence, the estimated SNR, defined as a power ratio of the subtraction result to the adaptive-filter output, equals infinity. Zero initial coefficients of the adaptive filter makes the denominator of the SNR estimate zero. Such a large value of the SNR estimate results in a near-zero stepsize and coefficients do not grow at all.

A solution is to use a second SNR estimate in the initial period, then, switch to the conventional (or first) SNR estimate [4]. The second SNR estimate is a power ratio of the primary input to the reference input. A non-zero denominator helps coefficients grow with time. Nevertheless, the instantaneous power of the primary input hardly becomes comparable to that of the reference input SNRs at high SNRs, leading to scarce coefficient adaptation. This undesirable situation continues because the second SNR estimate is independent of coefficient growth and does not benefit from coefficient adaptation. As a result, the initial start-up is slow for high SNRs and SNR-estimate switch-over may not happen.

This paper proposes a fast convergence algorithm for adaptive noise cancellers with SNR-based stepsize control. Use of the subtraction result as the numerator instead of the primary input makes the new second SNR estimate approach the true SNR like the first SNR estimate.

II. Noise Canceller with Two SNR Estimates
Figure 1 depicts a blockdiagram of an ANC with a first and a second SNR estimate. The noise cancelled signal \( e(k) \) is expressed by

\[
e(k) = x_P(k) - \hat{n}(k) = s(k) + \Delta n(k),
\]

\[
\Delta n(k) = n(k) - \hat{n}(k) = n(k) - \sum_{l=k-N+1}^{k} x_R(l) w(k, k - l),
\]

where \( x_P(k), x_R(k), n(k), \) and \( \hat{n}(k) \) are the primary- and the reference-microphone signals, noise to be cancelled in \( x_P(k) \), and a noise replica. \( w(k, i) \) is the \( i \)-th filter coefficient at time \( k \). Assuming good noise cancellation represented by \( \Delta n(k) = 0 \), \( e(k) \) can be regarded as a replica of the target signal.

1. First SNR estimate \( \sigma_1^2(k) \) in SNR1
With the replicas, \( \hat{n}(k) \) and \( e(k) \), of the noise and the target signal, a first SNR estimate, \( \sigma_1^2(k) \), is calculated by (3) as the output of SNR1 in Fig. 1.

\[
\sigma_1^2(k) = e^2(k)/\hat{n}^2(k),
\]

(1) and (2) indicate that \( e(k) = x_P(k) \) continues and no coefficient grows because all coefficients \( w(k, i) \) are initialized to zero. \( \sigma_1^2(k) \) does not initially work at all as an SNR estimate.

2. New Second SNR estimate \( \sigma_2^2(k) \) in SNR2
As a new second SNR estimate to replace the first SNR estimate initially, SNR2 of Fig. 1 calculates a power ratio \( \sigma_2^2(k) \) of the noise-cancelled signal to the reference-microphone signals.

\[
\sigma_2^2(k) = e^2(k)/x_R^2(k).
\]
$\sigma^2_1(k)$ and $\sigma^2_2(k)$ are different only in the denominator. $\hat{n}^2(k)$ in (3) increases from zero with adaptation whereas $x_n^2(k)$ in (5) does not change with coefficient growth. $\sigma^2_2(k)$ is initially smaller and suitable for coefficient growth. $\sigma^2_1(k)$ is initially too big to grow coefficients, however, is more accurate after convergence. Thus, $\sigma^2_2(k)$ is initially used and switched to $\sigma^2_1(k)$ when their averages coincide [4]. The SNR estimate $\sigma^2_{2\text{conv}}(k)$ or $\sigma^2_{2\text{prop}}(k)$, whichever is in use, is converted to a stepsize $\mu(k)$ and used in coefficient adaptation [3].

The second SNR estimate $\sigma^2_{2\text{conv}}(k)$ in [4] is

$$\sigma^2_{2\text{conv}}(k) = \frac{x^2_{\text{prop}}(k)}{x^2_{R}(k)}, \quad (5)$$

whose numerator is the primary input $x_{\text{prop}}(k)$. Because $x_{\text{prop}}(k)$ does not change with coefficient adaptation, $\sigma^2_{2\text{conv}}(k)$ of the proposed algorithm is superior to $\sigma^2_{2\text{conv}}(k)$ in [4] and provides quick initial coefficient growth.

### III. Evaluations

Evaluations were performed using clean female speech and noise recorded at a busy train station, both sampled at 8 kHz. Convolution of the noise and an impulse response of length $N = 1024$ identified in a room with a smartphone handset was added to the speech for the primary signal. Parameters for coefficient adaptation were set as in [3]. SNR estimates were averaged over 128 samples for changeover decision. Figure 2 illustrates the primary signal $x_{\text{prop}}(k)$ and the reference signal $x_R(k)$.

The conventional SNR estimate $\sigma^2_{\text{conv}}(k)$ [4] in gray and the new SNR estimate $\sigma^2_{\text{prop}}(k)$ in black are compared in Fig. 3. $\sigma^2_{\text{conv}}(k)$ and $\sigma^2_{\text{prop}}(k)$ use the primary input $x_{\text{prop}}(k)$ and $e(k)$ respectively as the numerator of $\sigma^2_{2\text{conv}}(k)$. Thanks to the benefit from growing coefficients, the changeover for $\sigma^2_{\text{prop}}(k)$ takes place at $k_{\text{prop}} = 5930$ whereas the changeover is at $k_{\text{conv}} = 66506$ for $\sigma^2_{\text{conv}}(k)$. Time until the changeover is 91% shorter with the proposed algorithm than the conventional algorithm.

Shown in Fig. 4 are normalized coefficient error vector (NCEV) $\| h - w(k) \|^2 / \| h \|^2$ where $h$ and $w(k)$ are the impulse response and the coefficient vector. NCEV is successfully reduced in both the conventional [4] and the proposed algorithms. An early changeover from $\sigma^2_{2\text{conv}}(k)$ to $\sigma^2_{2\text{conv}}(k)$ with the proposed algorithm contributes to quick coefficient growth. Time for NCEV to reach $-5$ dB with the proposed algorithm is 50% shorter than that with the conventional algorithm [4].

### IV. Conclusion

A fast convergence algorithm for adaptive noise cancellers with SNR-based stepsize control has been developed. The numerator of the second SNR estimate has been changed from the primary signal in the conventional algorithm to the noise-cancelled signal to accelerate initial coefficient growth. Evaluations with speech and recorded noise have demonstrated that the time until SNR-estimate changeover is reduced by $91\%$ compared to the conventional algorithm.

### References


