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The Wheatstone bridge-based analog adaptive filter with application in echo cancellation

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Abstract In this paper, a new analog adaptive filter is introduced with application in adaptive echo cancellation namely, the Wheatstone bridge-based analog adaptive filter (WAAF). It is proved the WAAF is a variable weight analog IIR filter. IIR filter weights vary with gate-source voltage control of a MOSFET transistor in triode region. The best balance point control of the WAAF is achieved using least mean square (LMS) algorithm. It is proved that analog LMS algorithm converges faster than digital LMS adaptive filter. The superiority of the proposed WAAF is observed in the designing process, computational cost, convergence speed and real time operation. Also, experimental results show ability of the proposed WAAF in the hybrid circuit of the telephone echo cancellation.

Keywords Wheatstone bridge · Analog-LMS algorithm · Analog-IIR filter · Hybrid echo cancellation

1 Introduction

Wheatstone bridges are used in a variety of applications to accurately measure an unknown resistance by nulling the bridge difference voltage to zero. It offers a good method for measuring small resistance changes accurately and allows a differential measurement that offers a higher common-mode noise rejection than in a single-element measurement [1]. Therefore, it is used for sensing temperature, strain, pressure, fluid flow, and dew point humidity. Today Wheatstone bridge is a basic structure for measurement, control devices, in pressure and temperature transducers, and other physical measuring systems.

In [2], the first flexible strain sensor based on pentacene semiconductors, employing a transistor-like Wheatstone bridge configuration presented. An operational transconductance amplifier with n output stages allows to cascade n Wheatstone-bridge-like sensors [3]. Also, in [4] new circuit topology based on duality of well-known voltage-mode Wheatstone bridge presented. This topology can implement ratio metric current measurement which is a vital aspect of current-mode instrumentation and signal processing. Another study on Wheatstone bridge is development of a new artificial neural network-based virtual fault detector for detection and identification of faults in DAS-connected Wheatstone bridge-oriented transducers of a computer-based measurement system [5].

Another application of Wheatstone bridge is seen in hybrid circuit in telecommunication systems. Figure 1 shows a simple hybrid circuit. Hybrid circuit is a simple Wheatstone bridge that ideally separates 2 lines side of 4 lines side to separate transmit signal from receive signal [6]. Echo is generated if unbalancing occurred in the Wheatstone bridge. Echo is incorporation of receive signal with transmit signal. One method for echo suppression is utilization of RC-networks to balance the Wheatstone bridge. Figure 2 includes a simple RC-balance circuit for balancing of the Wheatstone bridge. ZB in Fig. 1 must replaced by RC-balance. Subscriber line impedance varies in different frequencies of received signal. In the Fig. 1, subscriber line impedance is signed by Zi. RC balance will not balance the Wheatstone bridge completely and DSP

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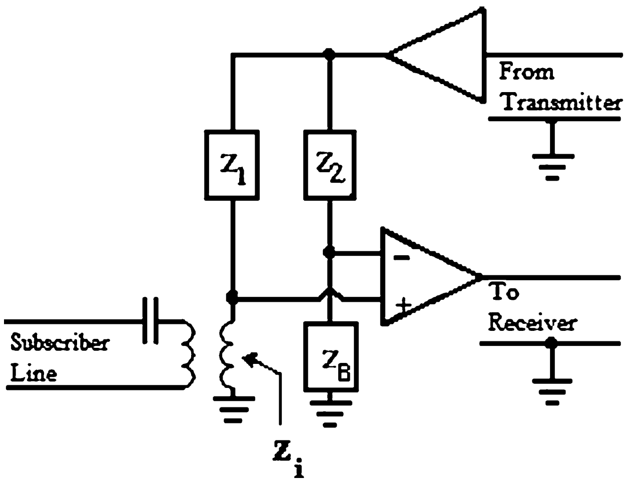


Fig. 1 Hybrid circuit

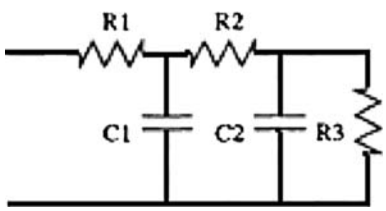


Fig. 2 Simple RC balance

processors will need to attenuate the echo by digital techniques [7–9].

In the 1960s, it was found that adaptive filter is able to reduce the echoes caused by the electrical coupling in the 4 wire to 2 wire hybrids. Adaptive filter estimate the impulse response of this hybrid circuit, and subtract an estimation of the echo from the return signal. Also, it must include some properties as fast convergence rate, high tracking capability for rapid estimation of echo channel. This purpose may be performed using FIR or IIR structures and learning scheme as least mean square (LMS) or its variants [10].

High power consumption of analog to digital converter (ADC) limits the high-speed operation and often makes to design analog adaptive filters. Also, low power, relatively low area, operating on binary signals above 10 Gb/s, low power delay elements and moderate linearity are other causes of choosing analog adaptive filters [11].

Most of analog adaptive filters are based on FIR filters. The implementation of broadband, low power delay elements is a key challenge in the design of these analog FIR filters. Clocked switched capacitors can be used to provide the delays, but considerable power is required to recover and distribute a low-jitter clock for filters above 10 Gb/s. More often, continuous-time delays are provided by active filters, passive devices or a combination of the two [12].

With these notes found that all today techniques to attenuate echo are based on adaptive filters that mounted parallel to hybrid circuit. Hybrid circuit is a Wheatstone bridge that is adaptive capable. In this paper a Wheatstone bridge-based analog adaptive filter (WAAF) presented. Hybrid circuit will replace by proposed WAAF as shown in Fig. 3. In this scheme, a new analog adaptive filter is constructed as it is substituted both hybrid circuit and adaptive filter.

The paper is organized as follows. The architecture of the proposed analog adaptive filter is explained in Sect. 2. Section 3 is devoted to adaptive echo cancellation using the proposed WAAF and in final section conclusions are presented.

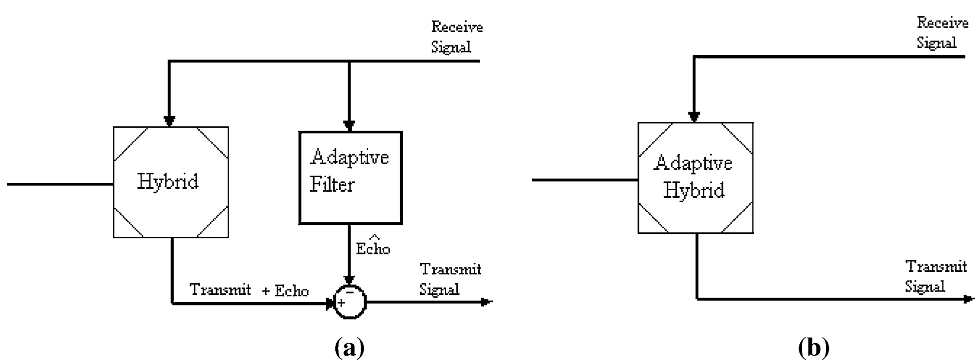
2 The proposed analog adaptive filter

A simple gradient search algorithm that is analog implement capable is LMS. This algorithm is used in the proposed analog adaptive filter to find optimum weights. In digital LMS algorithm, adaptation equation to find optimum weights computes as follow [13]:

$$\begin{aligned} w[n+1] &= w[n] + \mu x[n]e[n] \quad n = 0,1,2,3,\ldots, \quad w[0] = 0 \\ w[n+1] &= \mu \sum_{k=0}^{n} x[k]e[k] \end{aligned} \tag{1}$$

where $w[n]$ are the weighting coefficients of the linear combiner, $x[n]$ is a reference input, $e[n]$ is the error, and μ

Fig. 3 **a** Conventional method for echo cancellation. **b** The proposed scheme



is the step size. $w[0]$that chooses zero value, are initial value of the weights. By reducing the ADC's sampling time to zero, in an analog system, according to [10] Eq. 1 is converted to Eq. 2:

$$w_{(t)} = w_{(0^-)} + \mu \int_{0^+}^{t} x_{(t)} e_{(t)} \mathrm{d}t, \quad w_{(0^-)} = 0$$
$$w_{(t)} = \mu \int_{0}^{t} x_{(t)} e_{(t)} \mathrm{d}t \tag{2}$$

Sigma in Eq. 1 converted to integration at Eq. 2 and all n indexes converted to t index. The initial value of weights is shown by $w_{(0^-)}$and supposed that it is equal to zero. Equation 2 shows that implementation of analog LMS needs an analog integrator and an analog multiplier. A simple analog FIR or IIR filter needs to weighting coefficient. All last related works on analog adaptive filters was on FIR filters. In such filters, weighting coefficient usually is gm parameter of a simple amplifier. Figure 4 shows a single weight FIR filter and adaptation of this weight.

Figure 5 shows block diagram of WAAF. In this block diagram, there are four resistors. To balancing this bridge it is needed that weighting coefficient chooses from resistor families. gm may not chooses as weighting coefficient, because it is from admittance families ($\mathrm{gm} = \frac{i}{v} = \frac{1}{R}$). So in Fig. 5, R_2 must be a voltage controlled resistor (VCR). FET families can be as a VCR in triode region. Drain-source resistor in this region, computes as follow:

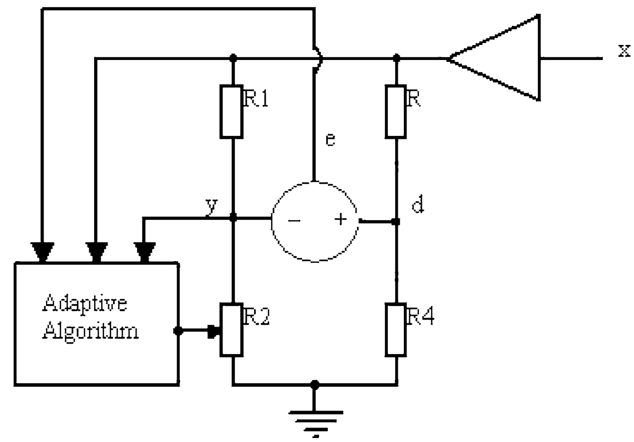


Fig. 5 Block diagram of WAAF

$$i_{\mathrm{D}} = k_n' \frac{W}{L}\left[(v_{\mathrm{GS}} - v_{\mathrm{T}})v_{\mathrm{DS}} - \frac{1}{2}v_{\mathrm{DS}}^2\right] \quad \text{if } |v_{\mathrm{DS}}| < < 1,$$
$$r_{\mathrm{DS}} = \frac{v_{\mathrm{DS}}}{i_{\mathrm{D}}}, \quad k'\frac{W}{L} = k \tag{3}$$
$$r_{\mathrm{DS}} \approx \frac{1}{k(v_{\mathrm{GS}} - v_{\mathrm{T}})}$$

where i_{D} is drain-source current, k_n' is process transconductance parameter, W is MOSFET channel width, L is MOSFET channel length, v_{GS} is gate-source voltage, v_{T} is threshold voltage and v_{DS} is drain-source voltage. If R_2 replaced by a MOSFET, y-signal computes by Eq. 4:

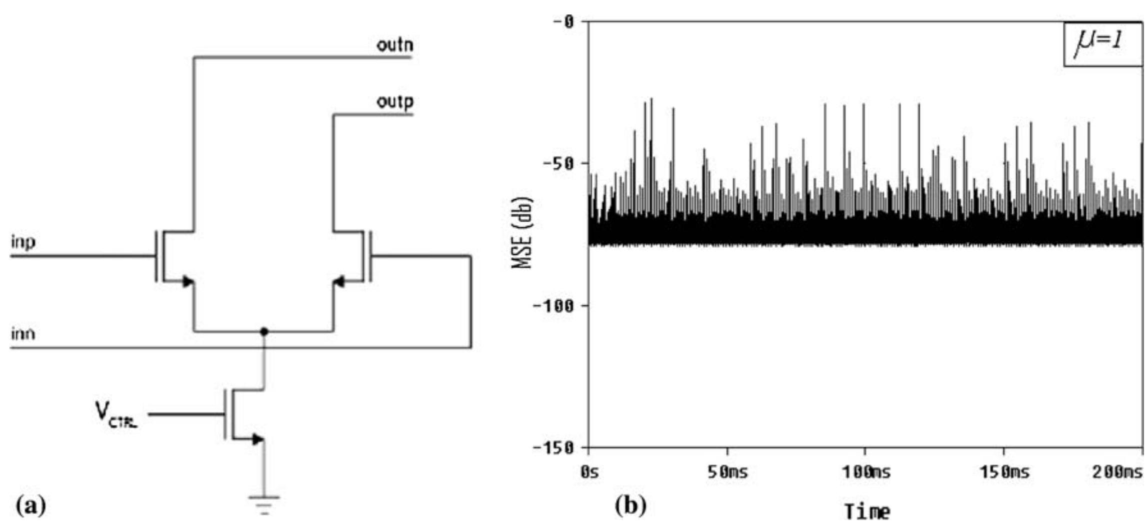


Fig. 4 **a** Single tap FIR filter [11]. gm will controlled by variation of V_{CTRL}. Also output equation will compute as follow: $I_{\mathrm{outn}} = -I_{\mathrm{outp}} = \mathrm{gm}\frac{(V_{\mathrm{inn}} - V_{\mathrm{inp}})}{2}$ and $\mathrm{gm} = k'\frac{W}{\sqrt{2}L}(V_{\mathrm{CTRL}} - VT)$ (it is supposed that all transistors are same). **b** Adaptation of this single tap FIR filter (simulation in SPICE)

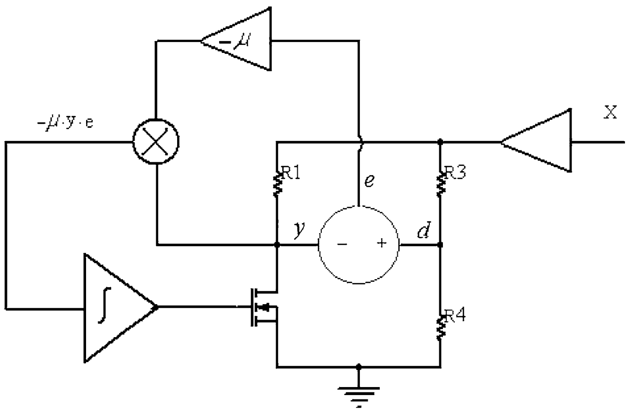


Fig. 6 The proposed WAAF (Fig. 5 with more details)

$$y_{(t)} = x_{(t)}\frac{r_{\mathrm{DS}}}{r_{\mathrm{DS}} + R_1} \tag{4}$$

In Eq. 4, r_{DS} can replace by Eq. 3:

$$y_{(t)} = x_{(t)}\frac{1}{1 - R_1 k v_{\mathrm{T}} + R_1 k v_{\mathrm{GS}}} \tag{5}$$

Equation 5 can simplify as follow:

$$\begin{aligned}(1 - R_1 k v_{\mathrm{T}} + R_1 k v_{\mathrm{GS}})y_{(t)} &= x_{(t)}\\ y_{(t)} &= x_{(t)} + (R_1 k v_{\mathrm{T}} - R_1 k v_{\mathrm{GS}})y_{(t)}\\ y_{(t)} &= x_{(t)} + (R_1 k v_{\mathrm{T}} + R_1 k(-v_{\mathrm{GS}}))y_{(t)}\end{aligned} \tag{6}$$

d-signal can computes by Eq. 7:

$$d_{(t)} = x_{(t)}\frac{R_4}{R_3 + R_4} \tag{7}$$

So error signal that is shown by $e_{(t)}$, computes as follow:

$$e_{(t)} = d_{(t)} - y_{(t)} \tag{8}$$

By attention to Eq. 6, it is found that adaptive weight ($-v_{\mathrm{GS}}$) multiplied to y-signal. So this WAAF is an IIR filter and by attention to Eq. 2, adaptive weight shall computes as follow:

$$-v_{\mathrm{GS}(t)} = \mu\int_0^t y_{(t)}e_{(t)}\mathrm{d}t \quad \Rightarrow \quad v_{\mathrm{GS}(t)} = -\mu\int_0^t y_{(t)}e_{(t)}\mathrm{d}t \tag{9}$$

Equation 9 shows analog-LMS adaptation for an analog IIR filter. By attention to Eq. 5 that is written for enhancement type NMOS, if $1 - R_1 k v_T > 0$, this analog-IIR filter is stable. Because in enhancement type NMOS, to induce channel, positive value of v_{GS} needed. So $1 - R_1 k v_T + R_1 k v_{\mathrm{GS}} \neq 0$ and this IIR filter is stable. By all illustrated, the proposed WAAF must be as shown in Fig. 6.

In Fig. 6 showed an analog multiplier to multiply error signal and y-signal together. Also μ is shown as a gain of an amplifier. Figure 12 in simulations shows that variation of μ parameter will gives different convergence speed. And best choice for this parameter is around ten. But amplification can limit the maximum work frequency. So μ or part of this parameter can be removed and goes into integrator coefficient. Integrator block has an integration coefficient. Usually integration coefficient has large value. Because integration circuit will made by RC circuits and these circuits have a $\frac{1}{RC}$ term for integration coefficient. In this paper integration coefficient chooses equal to 10^4. In this paper, all simulations are done in SPICE and MATLAB.

3 Adaptive echo cancellation

As illustrated in Section 1, in adaptive echo cancellation (AEC) two main problems are mentioned a follows,

(1). System identification of the echo path in order to cancel the echo
(2). Reliable and fast control of the adaptation by a double-talk detector (DTD) in order to avoid a divergence of the adaptive system identification algorithm during presence of speech in other side of the receiving line [14].

Double-talk problem occurs when there is a signal from another side of the line for send on hybrid circuit. So send signal is made from echo and desired-send signal.

$$e_{(t)} = d_{(t)} + \xi_{(t)} - y_{(t)} \tag{10}$$

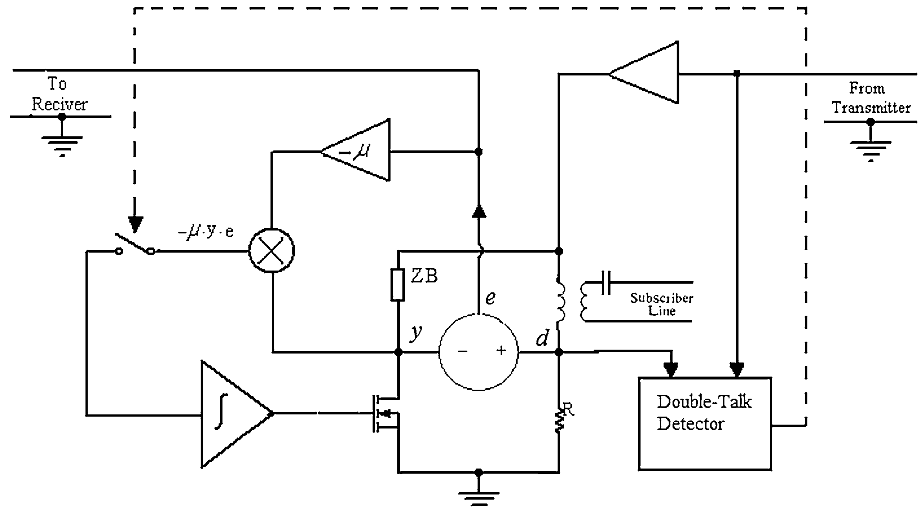
In Eq. 10, signal of $e_{(t)}$ is that of will be send. When echo attenuates, deference between $d_{(t)}$ signal and $y_{(t)}$ signal is equal to zero and send signal will made by $\xi_{(t)}$. $\xi_{(t)}$is the desired-send signal. $e_{(t)}$ signal is used in adaptation algorithm too. So adaptation algorithm must be stopped during $\xi_{(t)}$ is not equal to zero. In adaptive echo cancellation to solve this problem used of DTD to find double-talk time. So many DTD algorithms introduced in past years and still continued, like that is described in [14]. In this paper, to show ability of the proposed adaptive hybrid, supposed that an ideal DTD is provided. So the proposed adaptive hybrid is shown in Fig. 7.

Because the specification of speech signal is time variable, so the proposed echo canceller must be tested in non-stationary. Sinusoidal chirp tracking is a standard method. Because the chirp sinusoid represents a well-defined form of non-stationarity. The chirped input signal is given by:

$$S(k) = \sqrt{P_{\mathrm{s}}}e^{j[(2\pi f_{\mathrm{c}} + \psi k/2)k + \phi]} \tag{11}$$

where $\sqrt{P_{\mathrm{s}}}$denotes the signal amplitude, f_{c} is the center frequency, ψ is the chirp rate and ϕ is an arbitrary phase

Fig. 7 The proposed adaptive hybrid circuit



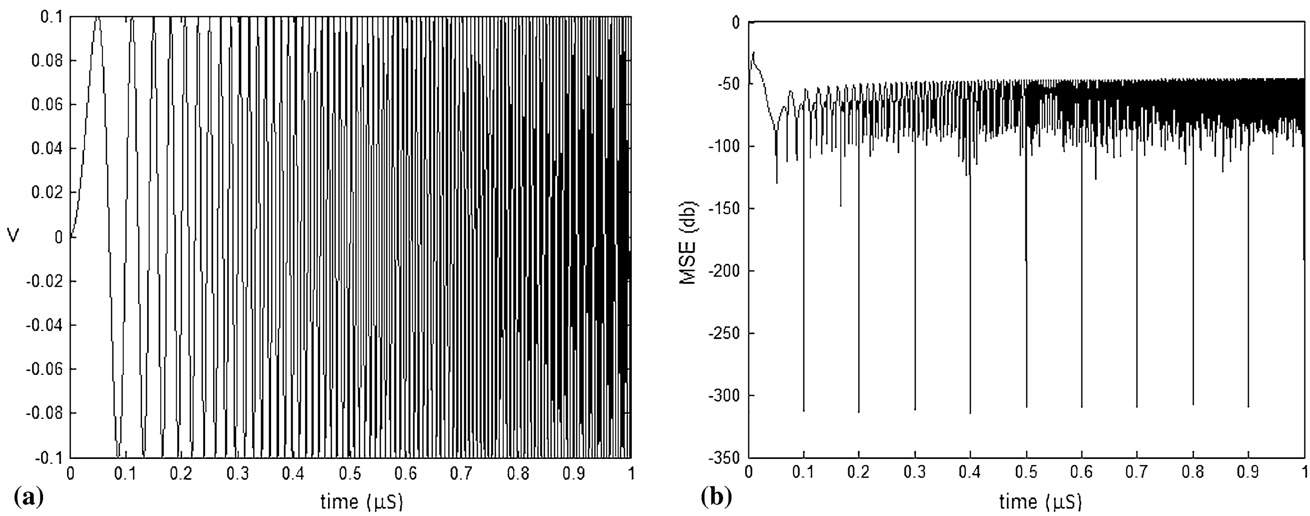


Fig. 8 **a** Test chirp signal. **b** Tracking of the proposed circuit ($\mu = 10$ and optimum-$R = 1$ KΩ). This figure shows that the proposed adaptive filter can track signal perfectly (simulation in MATLAB)

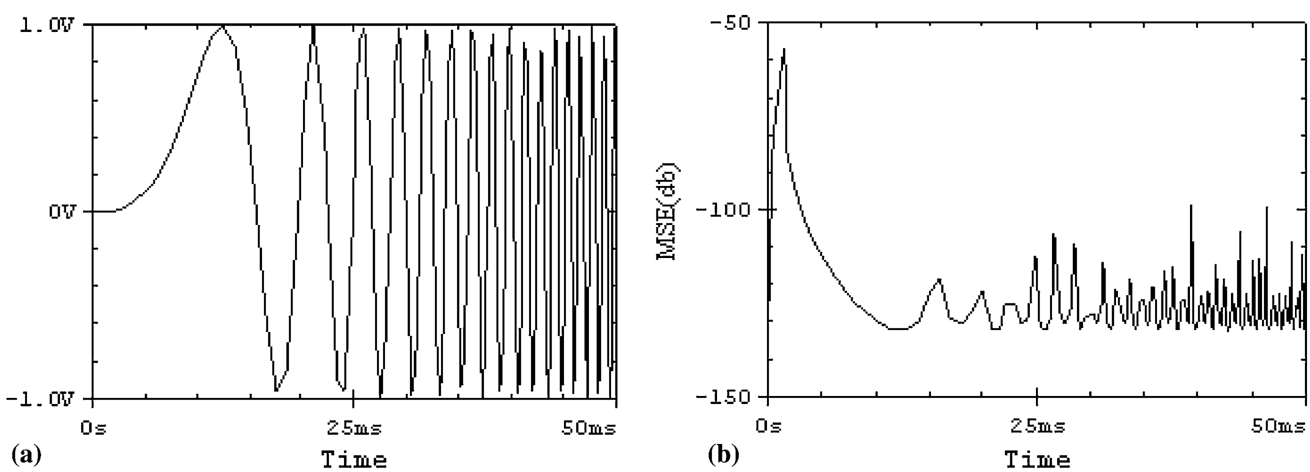


Fig. 9 **a** Test chirp signal. **b** Tracking of the proposed circuit ($\mu = 100$ and optimum-$R = 1$ KΩ). This figure shows that the proposed adaptive filter can track signal perfectly (simulation in SPICE)

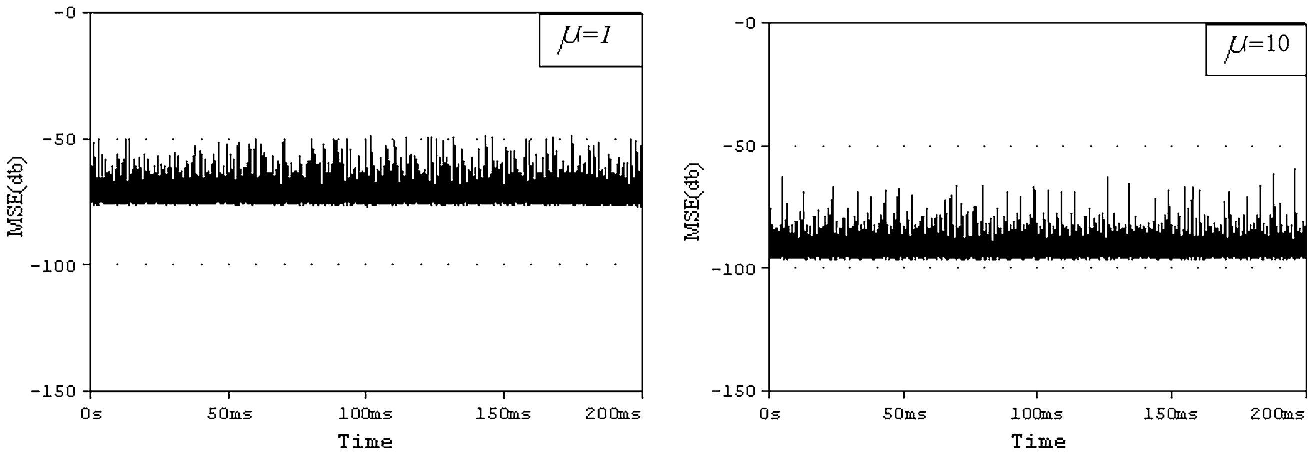


Fig. 10 Effect of step size over MSE. Input signal is $\mathrm{Sin}(2\pi 10^7 t)$ (simulation in SPICE)

shift. The signal $S(k)$ is deterministic but nonstationary because of the chirping. Figure 8 shows ability of the proposed adaptive filter in chirp tracking in channel by MATLAB simulation. In addition, Fig. 9 shows it by SPICE simulation.

Effect of step size (μ) is shown in following figure in tracking of sinusoidal signal to form of $\mathrm{Sin}(2\pi 10^7 t)$ for input signal and optimum $R = 1\ \mathrm{K}\Omega$. By attention to Eq. 3, it can not be expected that the proposed adaptive filter shows behavior as conventional LMS. Because of there is a nonlinearity term in Eq. 3 that with approximate, this term removed.

Figure 10 shows that for sinusoidal input signal, increasing μ will give better MSE. The same feature is seen

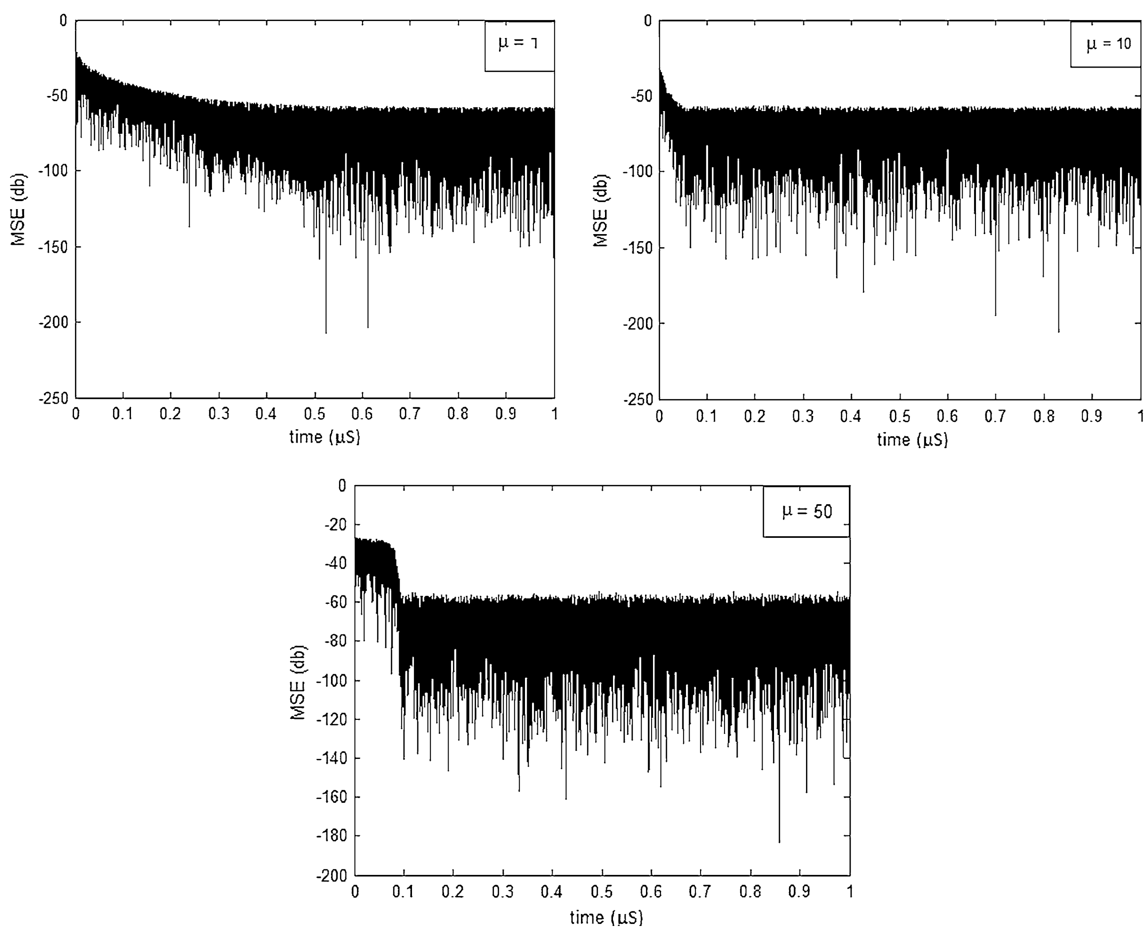


Fig. 11 Effect of step size over MSE. Input signal is white noise (simulation in MATLAB)

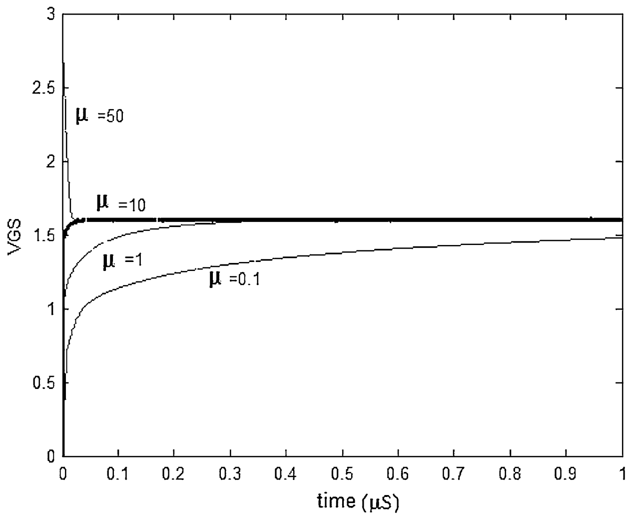


Fig. 12 Effect of variation of μ on weight convergence speed. Input signal is white noise (simulation in MATLAB)

in the result tested in MATLAB with sinusoidal input signal. But it is not expected of conventional LMS algorithm. This behavior occurred because of approximate in Eq. 3. But this behavior was not seen by white noise input signal. This test simulated in MATLAB. By seen Fig. 11, it is found that increasing of μ just changes convergence speed on input white noise.

Figures 11 and 12 shows variation of convergence speed due to variation of μ. In enhancement type MOSFET, until v_{GS} is less than v_T, r_{DS} shows infinite value. So in the first adaptation step, error signal and y-signal have large values. And if μ has large value too, by first step, v_{GS} goes upper than v_T. This behavior showed in Fig. 12 the best choice of μ is around ten.

The goal of an echo canceller is to completely remove any emanating signal from impedance mismatching. Real systems are incapable of perfect cancellation for a variety of reasons. It is important, therefore to be able to quantify actual adaptive echo canceller performance and to understand system and physical limitations that effect cancellation so that an adaptive echo canceller can be optimized for its environment. The most common measurement of adaptive echo canceller performance is echo return loss enhancement (ERLE) [15] that can be expressed as:

$$\mathrm{ERLE}_{(\mathrm{db})} = 10\log_{10}\left(\frac{E\left[d_{(t)}^2\right]}{E\left[e_{(t)}^2\right]}\right). \tag{12}$$

The ERLE is simply to the ratio of signal to the 'noise' that could not be cancelled from the signal. On the other hand, ERLE is defined in decibels as the average power of the reference signal $d_{(t)}$ over the average power of the error signal $e_{(t)}$. In this paper we evaluate the performance of echo canceller by ERLE simulation. Figure 11 shows the captured results. This figure shows that the proposed adaptive filter able to attenuate echo as well. Also, by comparing the proposed adaptive hybrid by digital technique that is described in [15] it will be found more ability in convergence speed and ERLE by the proposed adaptive hybrid.

Figure 13 shows that the proposed adaptive hybrid has an ERLE in 55 (db). And this value is good for an echo canceller. Also, convergence of the proposed adaptive filter is achieved in a few micro seconds.

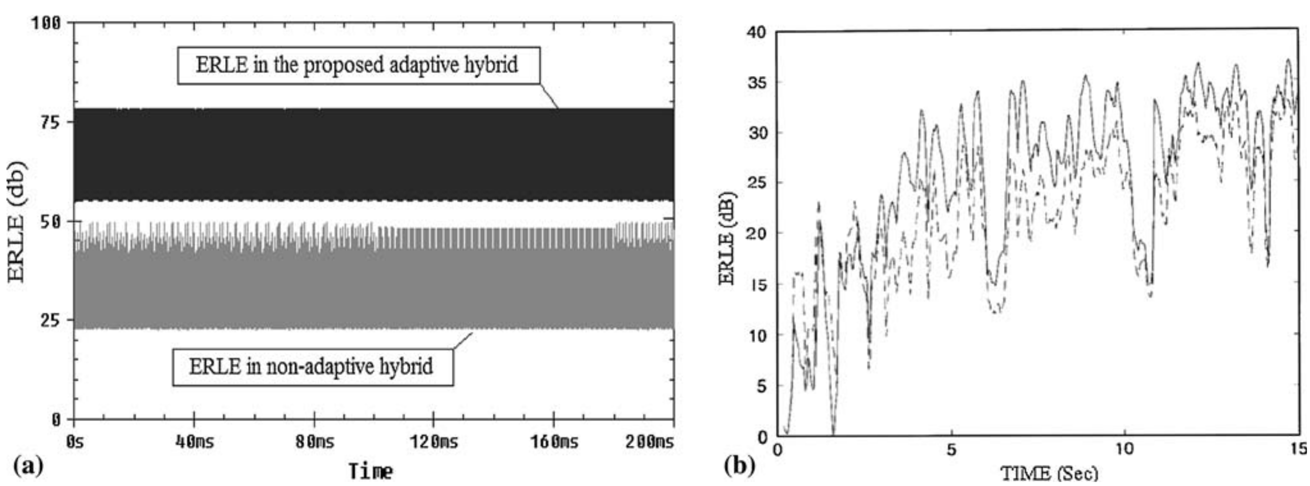


Fig. 13 **a** ERLE in the proposed adaptive hybrid circuit with $\mu = 10$ and in non adaptive hybrid (simulation in SPICE). **b** ERLE in digital algorithm that in [15] described. *Solid line* fast LMS/newton algorithm; *dashed line* normalized LMS

4 Conclusion

Some problems in digital echo suppression enforce us to presentation new analog adaptive filter special to echo suppression in the telephone line. The proposed WAAF could cancel echo on telecommunication systems. One branch of Wheatstone bridge include a FET as voltage controlled resistor for adaptive weight in the proposed circuit and this made to have an IIR filter. To find optimum weigh an analog-LMS on an analog IIR filter used. This single weight analog-IIR filter always can be stable. This adaptive filter in this application is called adaptive hybrid. Obtained results satisfied us and prove ability of this method.

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