

# *A Fast and Soft Voice Activity Detector Using GARCH Model and Markov Model Chain*

Sadegh Rezaei and Rasool Tahmasbi

## **ABSTRACT**

This paper presents a robust algorithm for Voice Activity Detector (VAD) based on GARCH (Generalized Autoregressive Conditional Heteroscedasticity) Model, Variance Gamma Distribution (VGD) and binary Markov model.

GARCH models are new statistical methods that are used especially in economic time series. There is a consensus that speech signals exhibit variances that change through time. GARCH models are a popular choice to model these changing variances.

Speech signal is assumed to have VGD because the VGD has heavier tails than the Gaussian Distribution (GD) and the distribution of noise signal is assumed to be Gaussian.

In the proposed method, heteroscedasticity will be modeled by GARCH and then the parameters of the distributions will be estimated recursively. Finally, using a binary Markov model and comparing it with an adaptive threshold, leads to the derivation of soft and hard detection.

The simulation results show that the proposed VAD is able to operate down to -5 dB and in nonstationary environments.

## **KEYWORDS**

Estimation Theory, GARCH Model, Heteroscedasticity, Markov Model Chain, Probability Distribution, Voice Activity Detection.

## **1. INTRODUCTION**

Voice activity detection (VAD) refers to the ability of distinguishing voice from noise and is an integral part of a variety of speech communication system, such as speech coding [1], speech recognition, audio conferencing, hands-free telephony [2], speech enhancement [3], wireless communication [4], [5] and echo cancellation.

During the last years, numerous researchers have studied different strategies for detecting speech in noise and the influence of the VAD on the performance of speech processing systems. Sohn [1] proposed a robust VAD algorithm based on a statistical Likelihood Ratio Test (LRT) involving a single observation vector. Later, Cho [6] suggested an improvement based on a smoothed LRT.

It has been shown recently [7]–[8] that incorporating long-term speech information to the decision rule reports benefits for speech/pause discrimination in high noise environments. For example, Ramírez [9] proposed an LRT involving multiple and independent observations.

In [10] the method is a little different. But they incorporate long-term information, too: the signal is first

decorrelated using an orthogonal transformation and then a Hidden Markov Model (HMM) is employed. Also, they assumed that the distribution of speech is Laplacian, because, in [11], it is shown that speech signal has Laplacian Distribution (LD). In this paper, we assume that the speech signal has a Variance Gamma Distribution (VGD), since it is a generalization of LD and in specific case ( $\lambda = 1$ ) VGD becomes LD.

Tahmasbi and Rezaei [12] used a GARCH model with an empirical adaptive threshold function. They showed that the empirical adaptive threshold function has better results than of [10].

Note that approaches of [1] and [9] are performed in frequency domain, but [10] is performed in time domain. However it takes time to decorrelate signal via orthogonalization. Our proposed method is performed in time domain; like [10], it is necessary that the signal be uncorrelated. To decorrelate signals, we used Generalized Autoregressive Conditional Heteroscedasticity (GARCH) model, which could model both noise and speech heteroscedasticity. However, estimating of GARCH parameters are time-consuming. So, a predefined estimation of the parameters is presented to solve this

S. Rezaei is with the Department of Statistics, Ahvaz University, Ahvaz, Iran (e-mail: [srezaei@aut.ac.ir](mailto:srezaei@aut.ac.ir)).

R. Tahmasbi, is student of Mathematical Statistics, Amirkabir University of Technology, Tehran, Iran (e-mail: [rtahmaseby@yahoo.com](mailto:rtahmaseby@yahoo.com)).

problem.

In section 2, the GARCH model is introduced and is shown that it can model heteroscedasticity and also it is shown that every GARCH series are uncorrelated. In section 3, we review elements of Markov Model. Section 4 presents statistical aspects of speech and noise and adaptive threshold function. In section 5, algorithm of the proposed method is presented. And section 6 illustrates the experimental results of proposed method.

## 2. GARCH MODEL

GARCH models are new statistical methods that are used especially in economic time series. GARCH stands for Generalized Autoregressive Conditional Heteroscedasticity. Loosely speaking, you can think of heteroscedasticity as time-varying variance (i.e., volatility). Conditional implies a dependence on the observations of the immediate past, and autoregressive describes a feedback mechanism that incorporates past observations into the present. GARCH then is a mechanism that includes past variances in the explanation of future variances. More specifically, GARCH is a time series modeling technique that uses past variances and past variance forecasts to forecast future variances.

*Definition:* Let  $(Z_t)$  be a sequence of i.i.d. random variables such that  $Z_t$  have standard Gaussian distribution.  $(Y_t)$  is called GARCH(q,p) process if

$$Y_t = \sigma_t Z_t \quad t \in \mathbb{Z} \quad (1)$$

where  $(\sigma_t)$  is a nonnegative process such that

$$\sigma_t^2 = \alpha_0 + \alpha_1 Y_{t-1}^2 + \dots + \alpha_q Y_{t-q}^2 + \beta_1 \sigma_{t-1}^2 + \dots + \beta_p \sigma_{t-p}^2 \quad (2)$$

and

$$\alpha_0 > 0; \alpha_i \geq 0, i = 1, \dots, q; \beta_i \geq 0, i = 1, \dots, p. \quad (3)$$

For  $p = 0$  the process reduced to ARCH(q) (Autoregressive Conditional Heteroscedasticity of order q).

In ARCH(q) processes, the conditional variance is specified as a linear function of past sample variances only, whereas the GARCH(p,q) process allows lagged conditional variances to enter as well. This corresponds to some sort of learning mechanism [13].

To review elementary aspects of GARCH model, denote  $E_{t-1}[\cdot]$  as conditional expectation while condition is on the past information up to time  $t-1$  (which is denoted by  $\phi_{t-1}$ . I.e.  $\phi_{t-1} = \sigma\{Z_{t-1}, Z_{t-2}, \dots\}$ , where  $\sigma\{\cdot\}$  is the sigma field generated by  $\{\cdot\}$ ). So

$$E_{t-1}[\cdot] = E[\cdot | \phi_{t-1}] \quad (4)$$

and conditional variance is

TABLE 1  
MEAN OF GARCH PARAMETERS

	SNR(db)				
	15	10	5	0	-5
$\hat{\alpha}_0$	0.0001	0.0003	0.0006	0.0010	0.0015
$\hat{\alpha}_1$	0.3325	0.2490	0.1449	0.0672	0.0249
$\hat{\beta}_1$	0.6654	0.7316	0.8209	0.8958	0.9462

TABLE 2  
VARIANCE OF GARCH PARAMETERS

	SNR(db)				
	15	10	5	0	-5
$\hat{\alpha}_0$	0.0001	0.0001	0.0001	0.0001	0.0001
$\hat{\alpha}_1$	0.0216	0.0208	0.0098	0.0047	0.0008
$\hat{\beta}_1$	0.0232	0.0211	0.0147	0.0096	0.0034

TABLE 3

SNR	PROBABILITY OF TRUE DETECTION (%)					
	15	10	5	0	-2	-5
P	98.52	97.61	97.04	94.28	90.74	75.91

TABLE 4

SNR (db)	PROBABILITY OF TRUE DETECTION (%)				
		Sohn	Gazor	Ramirez	Proposed
	5	91.05	94.21	95.35	97.01
0	84.62	79.36	87.85	94.34	
-2	73.28	69.38	80.26	90.69	

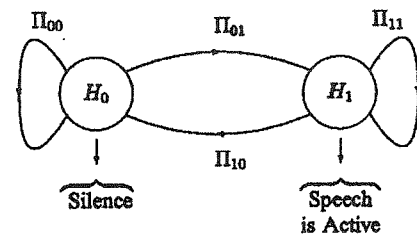


Figure 1: Binary Markov Model for VAD.

$$V_{t-1}[\cdot] = V[\cdot | \phi_{t-1}] \quad (5)$$

Now suppose  $(y_t)$  is a GARCH process and

$$\sum_{i=1}^q \alpha_i - \sum_{j=1}^p \beta_j < 1$$

So,

$$E(y_t) = E(\sigma_t Z_t) = E(\sigma_t) E(Z_t) = 0 \quad (6)$$

$$\begin{aligned} V(y_t) &= E(y_t^2) = E(\sigma_t^2 Z_t^2) = E(\sigma_t^2) \\ &= \alpha_0 + \sum_{i=1}^q \alpha_i E(y_t^2) + \sum_{j=1}^p \beta_j E(\sigma_t^2) \end{aligned}$$

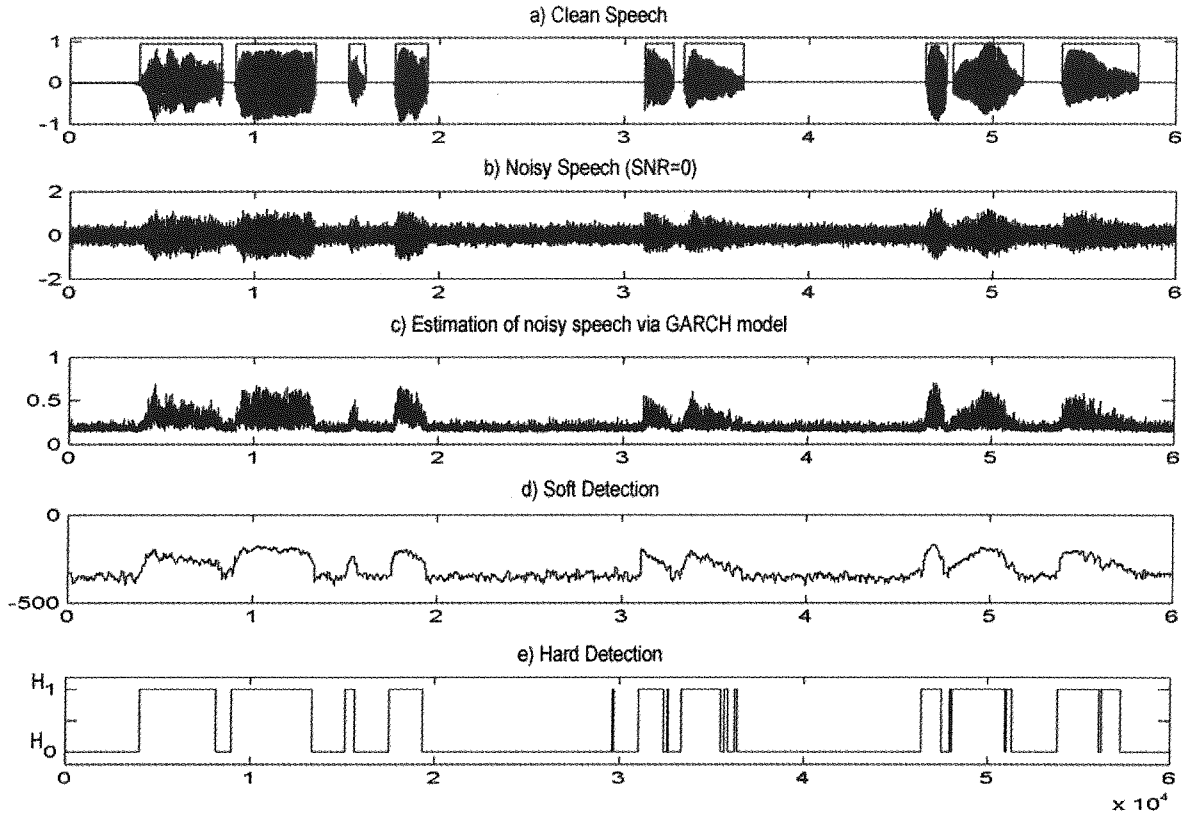


Figure 2: Results of the proposed VAD with 0 db SNR. (a) Clean speech. (b) Noisy speech with zero SNR. (c) Estimation of noisy speech via GARCH Model (d) Soft detection. (e) Hard detection.

$$\Rightarrow V(y_t) = \frac{\alpha_0}{1 - \sum_{i=1}^q \alpha_i - \sum_{j=1}^p \beta_j} \quad (7)$$

and its conditional expectation and variance is:

$$E[y_t | \phi_{t-1}] = E[\sigma_t Z_t | y_{t-1}, y_{t-2}, \dots] = \sigma_t E[Z_t] = 0 \quad (8)$$

$$\begin{aligned} V[y_t | \phi_{t-1}] &= E[y_t^2 | y_{t-1}, y_{t-2}, \dots] \\ &= E[\alpha_0 + \sum_{i=1}^q \alpha_i y_{t-i}^2 + \sum_{j=1}^p \beta_j \sigma_{t-j}^2 | y_{t-1}, y_{t-2}, \dots] \\ &= \alpha_0 + \sum_{i=1}^q \alpha_i y_{t-i}^2 + \sum_{j=1}^p \beta_j \sigma_{t-j}^2 = \sigma_t^2. \end{aligned} \quad (9)$$

As you can see, the mean and variance of a GARCH process are constant but its conditional variance is changing over time. So, in comparison with Autoregressive Moving Average (ARMA) process, GARCH could model the time-variation of variance (i.e., volatility).

Other interesting properties of GARCH process are:

- i)  $Cov(Y_t, Y_{t-k}) = 0$ , for  $k > 0$ ;
- ii)  $(Y_t)$  has heavier tails than the Gaussian Distribution (GD).

Note that property (i) together with normality

assumption of noises ensure that the  $(Y_t)$  are independent.

As [14] discussed when the mean level of a series stayed close to zero over the entire period and changes in variance (volatility) occurred, then this series could model through GARCH. It is clear that the speech signals have these properties (Fig 2.b) and several examples showed that it could be modeled through GARCH(1,1) and for example [15] used a GARCH(1,1) for modeling speech. So, we used a GARCH(1,1) to model the speech signal. Therefore,

$$\hat{\sigma}_t^2 = \hat{\alpha}_0 + \hat{\alpha}_1 Y_{t-1}^2 + \hat{\beta}_1 \hat{\sigma}_{t-1}^2 \quad (10)$$

where  $\hat{\alpha}_0$ ,  $\hat{\alpha}_1$  and  $\hat{\beta}_1$  should be estimated. (We call *garchfit* in Matlab to calibrate GARCH model).

The mean and variance of these three parameters for 100 different speeches with different SNR are shown in Table 1 and 2. As you see, the estimation values are related to the amount of SNR, i.e., less SNR more  $\beta_1$  and less  $\alpha_1$  and visa versa.

From now on, let us denote  $x(t)$  as estimation of noisy speech via GARCH. I.e.  $x(t) = \hat{\sigma}_t$ .

### 3. BINARY MARKOV MODEL

As the correlation of signal samples in consecutive frames is strong, the sequences of frame hypothesis states

can be modeled as a first-order Markov process, which simply assumes that the current state only depends on the current data and the previous state. The use of Markov model can also prevent clipping effect of weak speech, because when decision is made, the previous state is also taken into consideration.

Here, we use a two-state discrete (binary) Markov model to estimate the probability of Voice Being Active (VBA) at each time frame. As shown in Fig.1, these states are  $H_0$  (Silence) and  $H_1$  (Speech, i.e., VBA). To obtain this, we consider that the sequence of hypothesis is a binary Markov process with the following probability transition matrix:

$$\Pi = \begin{bmatrix} \Pi_{00} & \Pi_{10} \\ \Pi_{01} & \Pi_{11} \end{bmatrix} \quad (11)$$

where  $\Pi_{00}$  denotes the probability of silence when there was silence in the previous time instance,  $\Pi_{10}$  denotes the probability of silence when there was speech in the previous time instance and so on. In HMM, this matrix is often determined by  $\Pi_{ij} = T_{ij} / \sum_q T_{iq}$ , where  $T_{pq}$  is a measured number of transitions from  $p$ th state to  $q$ th state [10].

#### 4. STATISTICAL ASPECTS

After estimating noisy speech through GARCH model, the result is a series of data that have heavier tails than the Gaussian and are independent - by normality assumption of noises. In such situation some distribution like VGD is used [17]-[18].

For distinguishing speech or silence we do as follows:

Assume that  $X_t$  is an  $m$ -dimensional vector of data which is estimation of noisy speech via GARCH model at time  $t$ , i.e.,

$$X_t = [x(t), x(t-1), \dots, x(t-m+1)]^T \quad (12)$$

where  $(.)^T$  denotes the transposing operation.

Then, we use the below hypothesis via binary Markov model:

$$\begin{cases} H_0 : X_t \text{ is silence} \\ H_1 : X_t \text{ is speech} \end{cases} \quad (13)$$

Before starting the next section, we review the elementary features of the distribution of speech and noise, prior and posterior voice activity probabilities.

#### A. Distribution of speech

A random variable  $X$  is said to be Variance Gamma Distribution (VGD) with parameters  $\lambda, \alpha, \beta$  and  $\mu$  if its density is given by

$$f_X(x) = C \cdot |x - \mu|^{\lambda-0.5} K_{\lambda-0.5}(\alpha|x-\mu|) e^{\beta(x-\mu)} \quad (14)$$

with

$$C = \frac{\gamma^{2\lambda}}{\sqrt{\pi} \Gamma(\lambda) (2\alpha)^{\lambda-0.5}}, \quad \gamma^2 = \alpha^2 - \beta^2 \quad (15)$$

$K_\lambda(\cdot)$  is the modified Bessel function of the third kind (see [19], Ch. 11) and  $\Gamma(\cdot)$  is gamma function. The parameter domain is restricted to  $\lambda > 0$  and  $\alpha > |\beta|$ . If  $\beta = 0$  then the distribution is symmetric and  $\alpha, \lambda$  are shape parameters. The moment generating function of  $X$  is given by

$$L(z) = e^{\mu z} \left(\frac{\gamma}{\gamma_z}\right)^{2\lambda}, \quad z \in R \quad (16)$$

where  $\gamma_z = \sqrt{\alpha^2 - (\beta + z)^2}$ . Therefore, we obtain the centralized moments by  $L^{(n)}(0) = E(X^n)$ :

$$E(X) = \mu + 2 \frac{\beta\lambda}{\gamma^2} \quad (17)$$

$$V(X) = \frac{2\lambda}{\gamma^2} \left(1 + 2\left(\frac{\beta}{\gamma}\right)^2\right) \quad (18)$$

In [20] the estimation of the parameters of VGD via moment matching method is presented. Denote  $\nu$  and  $\kappa$  as skewness and kurtosis, respectively [20]. So their estimation is:

$$\hat{\lambda} = \frac{3}{\kappa - 3} \quad (19)$$

$$\hat{\beta} = \frac{\nu}{(\kappa - 3) \sqrt{V(X)}} \quad (20)$$

$$\hat{\alpha} = \sqrt{\frac{6}{(\kappa - 3)V(X)} + \frac{\nu^2}{(\kappa - 3)^2 V(X)}} \quad (21)$$

$$\hat{\mu} = E(X) - \frac{\nu \sqrt{V(X)}}{\kappa - 3} \quad (22)$$

Assume that the vector  $Y$  is noisy speech and the vector  $X$  is its estimated GARCH model and  $\alpha_0, \alpha_1, \beta_1$  are brought out from Table 1 (or by *garchfit* command in Matlab).

$$x(1) = \sqrt{|y(1)|};$$

For  $t = 2$  to  $N$  (length of  $Y$ )

$$x(t) = \text{sqr}t(\alpha_0 + \alpha_1 y^2(t-1) + \beta_1 x^2(t-1)); \quad \text{Eq. (10)}$$

$$\hat{\mu}_t = \text{Mean of } X_t; \quad \text{for } t > m \text{ use (30)}$$

$$\hat{\sigma}_t^2 = \text{Variance of } X_t; \quad \text{for } t > m \text{ use (31)}$$

$$\hat{\alpha}_t = \text{sqr}t\left(\frac{2\lambda}{\hat{\sigma}_t^2}\right); \quad \text{Eq. (28)}$$

$$\text{compute } L_m(X_t) \quad \text{Eq. (43)}$$

$$\text{compute } P_{t|t} = \frac{L_m(X_t)P_{t|t-1}}{L_m(X_t)P_{t|t-1} + (1 - P_{t|t-1})} \quad \text{Eq. (42)}$$

$$\text{Soft\_Detection}(t) = \log(P_{t|t}(X_t)) - m \log(C); \quad \text{Eq. (49)}$$

Next  $t$

Figure 3: Proposed VAD algorithm

In [20], it is shown that if  $\nu$  (skewness) is close to zero, then these estimators could successfully obtain good approximation to  $\lambda, \alpha, \beta, \mu$ . On the other hand, in [11] it is shown that speech signal has a symmetric distribution ( $\nu = 0$ ). So

$$\hat{\beta} = 0 \quad (23)$$

$$\hat{\mu} = E(X) \quad (24)$$

$$\hat{\lambda} = \frac{3}{\kappa} - 3 \quad (25)$$

$$\hat{\alpha} = \sqrt{\frac{2\hat{\lambda}}{V(X)}} \quad (26)$$

As you can see, estimating of  $\lambda$  is related to  $\kappa$  (kurtosis). To simplify the computations, we assumed a fixed value for  $\lambda$ . For example, [10] assumed  $\lambda = 1$  (LD is specific case of VGD), but one could consider all of the above equations.

Therefore, by assuming that speech distribution is symmetric, then for a fixed  $\lambda$ , the estimation of the parameters via the moment matching method is as follows:

If the vector  $X_t$  was estimated at time  $t$ , then the estimation of  $\mu_t$  and  $\alpha_t^2$  is

$$\hat{\mu}_t = \bar{X}_t = 1/m \sum_{i=0}^{m-1} x(t-i) \quad (27)$$

$$\hat{\alpha}_t = \sqrt{2\lambda / \hat{\sigma}_t^2} \quad (28)$$

$$\hat{\sigma}_t^2 = V(X_t) = 1/(m-1) \left[ \sum_{i=0}^{m-1} x^2(t-i) - m\hat{\mu}_t^2 \right] \quad (29)$$

and can be recursively computed via

$$\hat{\mu}_t = \hat{\mu}_{t-1} + R_t / m \quad (30)$$

$$\hat{\sigma}_t^2 = \hat{\sigma}_{t-1}^2 + R_t / (m-1) [T_t - R_t / m - 2\hat{\mu}_{t-1}] \quad (31)$$

$$R_t = x(t) - x(t-m) \quad (32)$$

$$T_t = x(t) + x(t-m) \quad (33)$$

Note that  $R_t$  and  $T_t$  is defined to shorten above equations.

#### B. Distribution of Noise

We assume that the noise components are Gaussian. Therefore, its PDF is given by

$$f_X(x) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{-\frac{1}{2\sigma^2}(x-\mu)^2} \quad (34)$$

If the vector  $X_t$  was observed, then the estimation of

$$\mu_t \text{ and } \sigma_t^2 \text{ are} \quad (35)$$

$$\hat{\mu}_t = \bar{X}_t$$

$$\hat{\sigma}_t^2 = V(X_t) \quad (36)$$

and can be recursively computed as same as above

recursive estimations, i.e.,

$$\hat{\mu}_t = \hat{\mu}_{t-1} + R_t / m \quad (37)$$

$$\hat{\sigma}_t^2 = \hat{\sigma}_{t-1}^2 + R_t / (m-1) [T_t - R_t / m - 2\hat{\mu}_{t-1}] \quad (38)$$

### C. Prior and Posterior Voice Activity Probabilities

The adaptive VAD algorithm starts with GARCH Model and then parameters of speech and noise are estimated. So, the prior probability,  $P_{t|t-1}$  of VBA at time  $t$  is provided. i.e.,

$$P_{t|t-1} = P(\text{speech is presented at time } t \text{ | observation up to time } t-1). \quad (39)$$

Note that a prior probability of VBA is initialized at  $P_{1|0} = 1/2$  at time  $t = 1$ . Based on these assumptions the hypothesis (13) could be revised as follows:

$$\begin{cases} H_0 : X_t \text{ is silence with the prior} \\ \text{probability } (1 - P_{t|t-1}) \\ H_1 : X_t \text{ is speech with the prior} \\ \text{probability } P_{t|t-1} \end{cases} \quad (40)$$

Now the objective is to estimate recursively the posterior probability  $P_{t|t}$

$$P_{t|t} \equiv P(\text{speech is presented at time } t \text{ | observation up to time } t). \quad (41)$$

This is soft decision rule that estimates the probability of VBA in each time instance, where  $P_{t|t}$  and  $P_{t|t-1}$  are estimated probability of VBA, with and without current information.

Based on the Baye's rule, we can easily derive soft VAD rule as follows:

$$P_{t|t} = \frac{L_m(X_t)P_{t|t-1}}{L_m(X_t)P_{t|t-1} + (1 - P_{t|t-1})} \quad (42)$$

where  $L_m(X_t)$  is the Likelihood ratio at time  $t$ , i.e.,

$$L_m(X_t) = \prod_{i=0}^{m-1} \frac{P_{H_1}(x(t-i)|H_1)}{P_{H_0}(x(t-i)|H_0)} \quad (43)$$

where  $P_{H_1}(\cdot|H_1)$  and  $P_{H_0}(\cdot|H_0)$  are probability distribution function (PDF) of speech and noise, respectively. A hard decision rule is then given by

$$H(t) = \begin{cases} H_1 & \log(P_{t|t}(X_t)) \geq \eta \\ H_0 & \log(P_{t|t}(X_t)) < \eta \end{cases} \quad (44)$$

A prediction of the prior probability for next time instance,  $P_{t+1|t}$  is required to perform the soft decision. This prediction is easily obtained based on the assumed

Markov Model,

$$\begin{pmatrix} 1 - P_{t+1|t} \\ P_{t+1|t} \end{pmatrix} = \begin{pmatrix} \Pi_{00} & \Pi_{10} \\ \Pi_{01} & \Pi_{11} \end{pmatrix} \begin{pmatrix} 1 - P_{t|t} \\ P_{t|t} \end{pmatrix} \quad (45)$$

That is,

$$P_{t+1|t} = \Pi_{01}(1 - P_{t|t}) + \Pi_{11}P_{t|t} \quad (46)$$

### D. Adaptive Threshold

As discussed before, speech is active at time  $t$  if  $\log(P_{t|t}(X_t)) \geq \eta$  (based on HMM). It is better, however, to compare  $\log(P_{t|t}(X_t))$  with a threshold function,  $\eta_t$  at each time  $t$ , i.e.,

$$X_t = \begin{cases} \text{speech} & \log(P_{t|t}(X_t)) \geq \eta_t \\ \text{Silence} & \log(P_{t|t}(X_t)) < \eta_t \end{cases} \quad (47)$$

It is natural to assume that the threshold function,  $\eta_t$  be a function of parameters of the speech distribution and  $m$ . An appropriate approach is to define

$$\eta_t = \tau + m \log(C) \quad (48)$$

where  $\tau$  is a constant threshold and  $C$  is defined in (15) and its parameters had been estimated by  $X_t$  at time  $t$ . Note that (48) is obtained, empirically. So, speech is active if,

$$\log(P_{t|t}(X_t)) - m \log(C) \geq \tau \quad (49)$$

The left side of (49) can be viewed as a criterion for detecting silence or speech activity and is a criterion for soft detection. This criterion will be compared with  $\tau$  and hard detection will be derived.

## 5. PROPOSED VOICE ACTIVITY DETECTION ALGORITHM

For implementation of the proposed VAD, its algorithm is presented and shown in Fig 3.

This algorithm can divide into two parts: algorithms for  $t \leq m$  and for  $t > m$ . Since for  $t \leq m$ , the length of vector  $X_t$  is less than  $m$ , so we can not use recursive Eq. (30-33) and (37-38) for estimating  $\hat{\mu}_t$ ,  $\hat{\sigma}_t^2$  and  $\hat{\alpha}_t^2$ .

And should use (27-29) and (35-36) for  $X = [x(t), x(t-1), \dots, x(2), x(1)]$ . Also, for computing soft detection, we use information up to time  $t (< m)$ . On the other hand, since log-likelihood is motivated by length of segment, so we multiply log-likelihood by  $m/t$ . But for  $t > m$  since length of  $X_t$  is  $m$  so, use recursive (30-33) and (37-38).

## 6. EXPERIMENTAL RESULTS

In this section, the results of the proposed method are presented. The speech signals are obtained from <http://www.dailywave.com>. Fig. 2(a) shows the clean speech together with manually marked clean speech sample. Fig. 2(b) shows the noisy speech with Gaussian noise and its SNR is equal to 0. GARCH estimation of noisy speech is shown in Fig. 2(c). Its estimated GARCH parameters are  $\hat{\alpha}_0 = 0.002$ ,  $\hat{\alpha}_1 = 0.12$  and  $\hat{\beta}_1 = 0.83$ . As discussed before, Table 1 and 2 show the mean and variance of GARCH parameters for different noisy speeches with Gaussian noise. Since estimation of GARCH parameters is related to SNR, so it is possible to use these approximated parameters instead of using an algorithm for finding exact estimation of the parameters.

The left side of (49) is computed and shown in Fig. 2(d). This criterion is compared with  $\tau = -300$  and hard decision is derived. The hard detection is shown in Fig. 2(e).

To evaluate the performance of the proposed VAD, the speech and silence intervals marked manually then the hard decision of the VAD is compared with the manually marked intervals. For Fig 2, the Probability of True Detection (PTD) is equal to 0.94 and the results of the proposed method with different SNR are presented in Table 3. The PTD is more than 96 percent when SNR is greater than 2, and in very noisy cases (SNR= -5) is near 75 percent which are good results. So, the presented algorithm can be viewed as a robust algorithm.

The PTD, as evaluation criteria for the performance of the previous works and presented method are given in Table 4 with different SNR. As you see, the performance of the proposed method is more than Ramirez [9], Gazor [10] and Sohn [1]. Also, to prove the superiority of our algorithm, we examine our algorithm with non-Gaussian noises.

As same as Fig. 2, the results of proposed method with noises of  $t$  distribution (with 4 degree of freedom) and

Beta Distribution (with parameters  $\alpha = 2.5$  and  $\beta = 1.5$ ) are shown in Fig. 4 and 5, respectively. Since Beta random numbers are not zero-mean, so we subtract their mean and then add these noises to clean speech signal. Also Fig. 6 shows the results of the proposed method with colored noise. To generate colored noise we passed white noise through a finite impulse response.

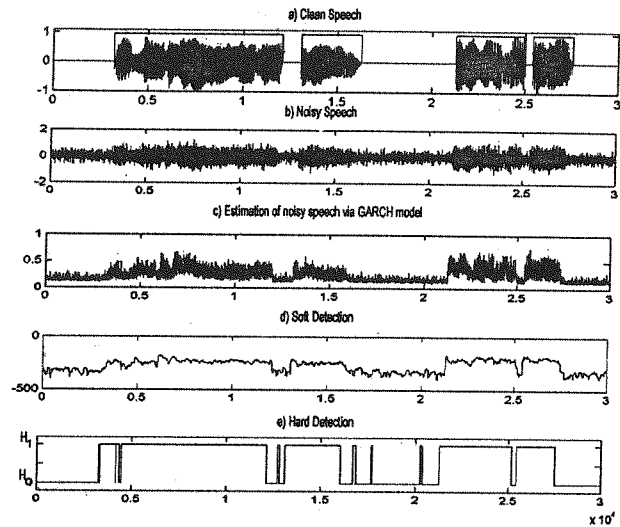


Figure 4: Results of the proposed VAD for  $t$  Distribution noise. (a) Clean speech. (b) Noisy speech. (c) Estimation of noisy speech via GARCH Model. (d) Soft detection. (e) Hard detection.

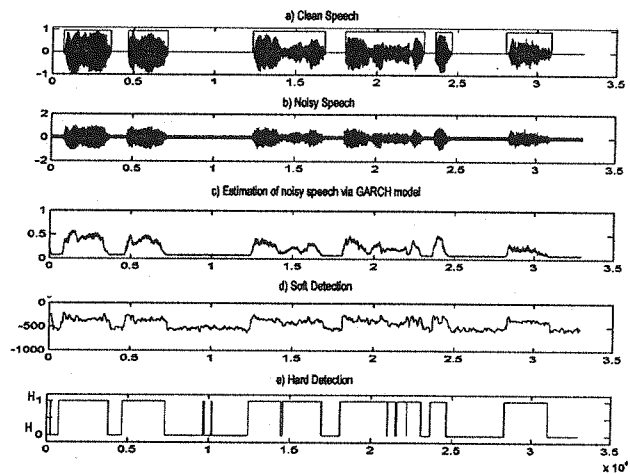


Figure 5: Results of the proposed VAD with Beta Distribution noise. (a) Clean speech. (b) Noisy speech. (c) Estimation of noisy speech via GARCH Model. (d) Soft detection. (e) Hard detection.

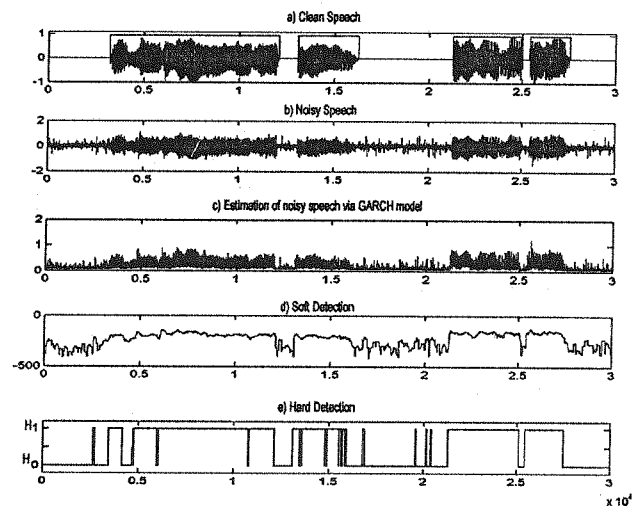


Figure 6: Results of the proposed VAD with colored noise. (a) Clean speech. (b) Noisy speech. (c) Estimation of noisy speech via GARCH Model. (d) Soft detection. (e) Hard detection.

Note that for Fig. 3 and 5, SNR is 5db and for Fig 4 it is 10 db.

The PTD for these noisy speeches are 0.96, 0.94 and 0.93, respectively.

## 7. CONCLUSION

The objective of this paper is to exploit the properties of new statistical tools such as GARCH model and heavy tailed distribution for finding a robust algorithm for VAD in the presence of high level of noise. The results show that the performance of the presented VAD has been improved when we take advantage of the adaptive threshold function.

The complexity of the proposed algorithm is very low due the fact that it is performed in time domain and estimations can be computed recursively.

## 8. REFERENCES

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