A NEW LATTICE LP-BASED POST-FILTER FOR ADAPTIVE NOISE CANCELLERS IN MOBILE AND VEHICULAR APPLICATIONS

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Abstract *- Adaptive Noise Cancellation (ANC) is a well-known technique for background noise reduction in automobile and vehicular environments. The noise fields in automobile and other vehicle interior obey the diffuse noise field model closely. On the other hand, the ANC does not provide sufficient noise reduction in the diffuse noise fields. In this paper, a new multistage post-filter is designed for ANC as a solution to diffuse noise conditions. The designed post-filter is a single channel Linear Prediction (LP) based speech enhancement system. The LP is performed by an adaptive lattice filter and attempts to extract speech components by using intermediate ANC signals. The post-filter has no processing delay which is suitable for speech communication systems. We have evaluated the performance of proposed system in various real-life noise fields, recorded in an automobile environment. The experimental results using various quality measures show that the proposed method is superior to both the adaptive noise canceller and LP-based speech enhancement systems.*

Keywords *- Adaptive Noise Cancellation (ANC), automobile and moving vehicle speech enhancement, diffuse noise field, lattice filter, noise reduction.*

I. INTRODUCTION

Performance of speech processing systems degrades dramatically in noisy and reverberant environment such as the interior of a moving vehicle [1-4]. In such situations, speech recognition systems cannot work efficiently and hands-free cellular telephones cannot provide satisfactory quality of communication [3]. Therefore, noise reduction is a necessary part of current speech processing systems in mobile and vehicular applications.

Studies of the background acoustic noise characteristics in automobiles show that it is non-stationary and lowpass with the dominant noise energy in the frequency band below *1000 Hz* [3]. This background noise is generated outside the cabin of automobiles and penetrates to the inside. Therefore, high correlation between the outside and the inside noises is expected [5]. Most of the investigations of speech enhancement in automobiles have employed Adaptive Noise Cancellation (ANC) technique to improve the speech quality by removing this correlation [1]. However, since various sources like mechanical friction within the engine compartment, air flow, road noise, street noise and etc, generate automobile noise field through many paths, there is no such high correlation between the inside and outside noises in all frequencies [5] and they are correlated only at lower frequencies. Experiments have shown that the noise field is approximately diffuse [2, 4]. In this case, ANC and other multi-microphone speech enhancement techniques cannot work efficiently [6, 7] (if they are employed alone and without any further process).

Authors in [1] introduced this challenge and applied an array of N microphones as a solution. Kuo *et al.* [3] put a microphone next to each noise source and proposed a multi-reference ANC to compensate the effect of multi-source multi-path phenomena. It can be easily demonstrated that the speech and noise components of the ANC output are approximately separated in the frequency domain. Thus, speech enhancement can be performed by an appropriate post-filtering. Abutalebi *et al.* [2, 4] proposed a hybrid system that integrates a Subband Adaptive Filter (SAF) and a Wiener filter. They used a Voice Activity Detector (VAD) to control both the adaptation in the SAF and the noise spectrum estimation in the Wiener filter. Accuracy of the VADs degrades in the noisy environment; also, the VAD implementation makes the system more complex. As a remedy, in this research, we have designed a simple yet efficient Linear Prediction (LP)-based post-filter to eliminate the VAD block.

Speech enhancement based on LP was introduced by Kawamura [8, 9] and continued by Rank [10]. This method reduces prediction coefficients to near zero for white signals. On the other hand, speech is an AR process which can be linearly predicted by its past samples. Thus, white noise can be removed from noisy speech using this technique [8, 9]. The effective implementation of linear predictors is based on lattice filters which has a direct relation to a physical model of the vocal tract [10]. This method cannot work efficiently in colored noisy environment. To alleviate this problem we propose a multi-stage structure and change the updating algorithm of PARtial CORrelation (PARCOR) coefficients such that they consider speech components of noisy signals.

In Section II, we study the properties of noise fields in automobile environment, then the performance of ANCs are calculated according to these properties. Section III explains

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the lattice linear predictors and its problems for speech enhancement. Section IV gives the proposed multistage filtering approach for noise suppression. Experimental results are given in Section V and finally, we summarize our works and conclude in Section VI.

II. ANC AND ITS CHALLENGE IN VEHICULAR ENVIRONMENT

Fig. 1 shows the overall structure of conventional adaptive noise canceller. An adaptive filter is employed to remove the correlation between ANC inputs i.e. reference signal, $\zeta_2(n)$, and desired signal, $d(n)$. The reference input is not correlated with speech components of desired signal, $s(n)$, so $s(n)$ appears in the ANC output, $e(n)$, directly. On the other hand, in an ideal nonreverberant noisy situation which is generated by single source, noise components of two channels are correlated to each other and are eliminated from ANC output. We will demonstrate that in the case of automobile and vehicular environment, noise components are correlated only at lower frequencies; therefore ANC cannot eliminate higher frequency noise components.

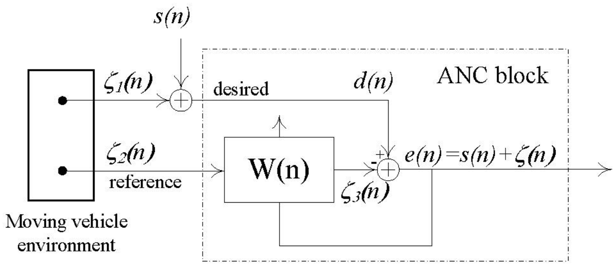

Fig. 1. Conventional ANC structure

A. ANC Noise Reduction Factor

For analyzing the noise reduction ability of ANCs in various conditions, a quantitative measure is needed. Several authors [1, 2, 4] have used Noise Reduction (NR) factor, that is defined by

$$NR(f)=\frac{S_{\zeta_1\zeta_1}(f)}{S_{\zeta\zeta}(f)} \tag{1}$$

where S denotes the cross power spectrum density. $\zeta(n)$, $\zeta_1(n)$ and other signals are named according to Fig. 1. When adaptive filter converges to its optimum solution (Wiener solution), we have [1, 11]

$$w_{opt}(f)=\frac{S_{\zeta_1\zeta_2}(f)}{S_{\zeta_2\zeta_2}(f)} \tag{2}$$

$$S_{\zeta\zeta}(f)=S_{\zeta_1\zeta_1}(f)-\left|w_{opt}(f)\right|^2\cdot S_{\zeta_2\zeta_2}(f)\cdot \tag{3}$$

By combining (1), (2) and (3), NR factor can be written as

$$NR(f)=\frac{1}{1-\gamma^2_{\zeta_2\zeta_1}(f)} \tag{4}$$

where $\gamma^2_{\zeta_2\zeta_1}(f)$ is the Magnitude Square Coherence (MSC) between two inputs $\zeta_2(n)$ and $\zeta_1(n)$, defined by Eq. (5) [1].

$$\gamma^2_{\zeta_2\zeta_1}(f)=\frac{\left|S_{\zeta_2\zeta_1}(f)\right|^2}{S_{\zeta_2\zeta_2}(f)\cdot S_{\zeta_1\zeta_1}(f)} \qquad 0\le\gamma^2_{\zeta_2\zeta_1}(f)\le 1 \tag{5}$$

As a result of equation (4), in a noisy environment, ANC can only eliminate the noise components with high MSC.

B. The Performance of ANC in an Automobile Environment

In an ideal diffuse noise field, received signal in each point is the sum of numerous noise signals with equal amplitudes which come from various directions with equal probability [1]. In a moving vehicle environment, reflection of a noise signal from a hard surface, such as a window, can be considered as a new noise signal located on the opposite side of the reflective boundary. Therefore, when a sufficient number of noise sources are considered, automobile noise field approaches diffuse noise model. The MSC of two signals, recorded in the ideal three-dimensional diffuse sound field, can be expressed as Eq. (6) [4].

$$\gamma^2_{\zeta_2\zeta_1}(f)=\frac{\sin^2(2\pi fd/c)}{(2\pi fd/c)^2}=\operatorname{sinc}^2\left(\frac{2fd}{c}\right) \tag{6}$$

where d is the microphone spacing and c is the sound velocity.

Fig. 2 shows the MSC of various real-life noise fields recorded in automobile environment (solid lines) and ideal diffuse noise field (dashed lines). As depicted in this figure, the curves of the recorded noises are extremely close to the ideal curve and this is consistent with our diffuse noise assumption. Noise signals are recorded in the following situations:

1- Sitting in the shopping mall (Sit-Mall)
2- Inside a working car parked next to a high way (HWY)
3- Inside a moving car with open windows (CarWopen)
4- Inside a moving car with closed windows (CarWclose)

This data were collected during the research reported in [2]. The sampling rate and microphone spacing are *16 KHz* and *38 mm* (a typical value for boomless headset), respectively.

According to the figures and Eq. (4), the most important deficiency of ANCs is that they can only eliminate the noise components in low frequencies. By this explanation, and considering that most of the speech energy content lays in the low frequencies, it is clear that a post-filtering stage is necessary to improve the ANC noise reduction ability.

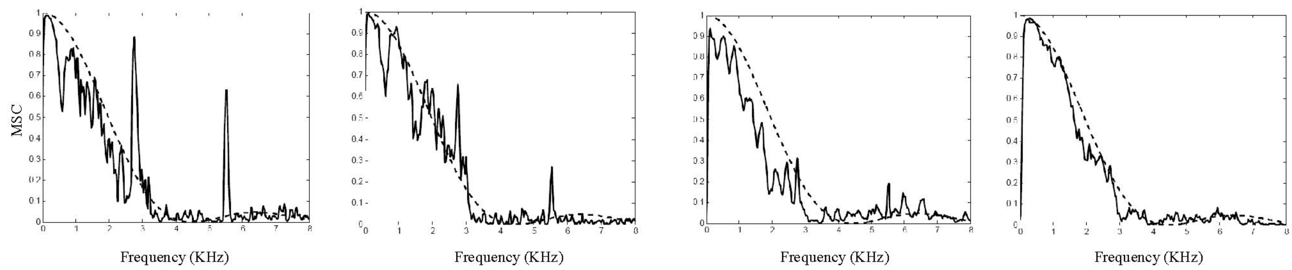

Fig. 2. Magnitude Square Coherence (MSC) of the theoretical diffuse noise field (dashed lines) and real-life automobile environment (solid lines) for (a) Sit-Mall (b) HWY (c) CarWOpen (d) CarWClose

III. LP-BASED SPEECH ENHANCEMENT AND ITS PROBLEMS IN VEHICULAR ENVIRONMENT

In this section, we investigate the suppression of white noise by applying a linear predictor. Predictor coefficients are determined in such a way that the prediction residual is minimized and whitened [8]. Therefore, white noise components do not change the filter coefficients, and directly appear in residual signal. On the other hand, if the number of predictor coefficients is sufficient (i.e., *512* or more taps in *16 kHz* sampling rate), linear predictor becomes a comb filter for speech signal and predictor output will be approximately equal to speech. This results in a simple white noise suppression technique [8-10]. It is considerable that by using low order predictors, just a spectral envelope of speech signal is formed and predictor cannot extract speech from noisy signal perfectly. The lattice structure has most commonly been used for implementing linear predictors in the context of speech processing applications [11].

A. Lattice Filter as a Linear Predictor

Forward and backward are two distinct forms of linear predictors. In the forward linear predictor of order m, the goal is to estimate present sample of $x(n)$ in terms of a linear combination of its past samples $x(n-1)$, $x(n-2)$, …, $x(n-m)$. The difference between $x(n)$ and estimated $x(n)$ is called forward prediction error and is denoted by $f_m(n)$, where m is the order of the predictor. Furthermore, $x(n-m)$ can be obtained as a linear combination of its future samples $x(n)$, $x(n-1)$, …, $x(n-m+1)$. It is called backward linear predictor and Backward prediction error, $b_m(n)$, is the distance between $x(n-m)$ and predicted $x(n-m)$ [11].

Lattice filter is an implementation of the order-update equations, which calculates the m^{th} order forward and backward prediction errors from the forward and backward prediction errors of order $m-1$. These order-updating equations are as follows [8, 10].

$$f_0(n) = b_0(n) = x(n) \tag{7}$$

$$f_m(n) = f_{m-1}(n) + \gamma_m^f(n) b_{m-1}(n-1) \tag{8}$$

$$b_m(n) = b_{m-1}(n-1) + \gamma_m^b(n) f_{m-1}(n) \tag{9}$$

In these equations, $\gamma_m^f(n)$ and $\gamma_m^b(n)$ are the PARCOR coefficients. Since they correspond to the correlation that remains between the forward and backward prediction errors, they are called Partial Correlation coefficients. Equations (10) and (11) are the theoretical expression for computing PARCORs [8, 10].

$$\gamma_m^f(n) = -\frac{E\{f_{m-1}(n) b_{m-1}(n-1)\}}{E\{b_{m-1}^2(n-1)\}} = -\frac{p_m(n)}{g_m^b(n)} \tag{10}$$

$$\gamma_m^b(n) = -\frac{E\{f_{m-1}(n) b_{m-1}(n-1)\}}{E\{f_{m-1}^2(n)\}} = -\frac{p_m(n)}{g_m^f(n)} \tag{11}$$

$g_m^b(n)$ and $g_m^f(n)$ are theoretically equal, but since we must approximate the expectation of these values, different symbols are considered in our notation. Commonly, the expectation operator is replaced by lowpass filtering of instantaneous values. We have applied a one-pole recursive lowpass filters as below.

$$p_m(n) = \lambda p_m(n-1) + f_{m-1}(n) b_{m-1}(n-1) \tag{12}$$

$$g_m^f(n) = \lambda g_m^f(n-1) + f_{m-1}^2(n) \tag{13}$$

$$g_m^b(n) = \lambda g_m^b(n-1) + b_{m-1}^2(n-1) \tag{14}$$

Fig. 3 shows a speech enhancement system based on lattice LP, called Lattice Linear Predictor (LLP) in the rest of this paper. In this system, first forward LP error is calculated which is an estimation of the noise, and then by subtracting estimated noise from noisy speech, speech components are extracted.

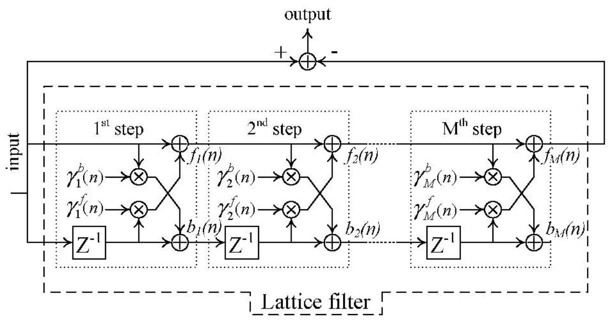

Fig. 3. Lattice Linear Predictor (LLP) as a speech enhancement system

The following problems cause this structure to be inadequate for appropriate noise reduction:

1) This method will encounter a difficulty, if the additive noise is colored.
2) The LLP structure cannot estimate clean speech signal perfectly, even in the presence of white noise.

In the next section, this system is utilized as a post-filter for Adaptive Noise Canceller and a hybrid system is proposed to alleviate both the ANC and LLP problems.

IV. PROPOSED POSTFILTER FOR ANC

As mentioned before, the NR of the ANC is decreased in high frequencies. Since noise fields of vehicular environments are practically lowpass, remaining noise components of ANC output, $\zeta(n)$, is much whiter than original noise components, $\zeta_1(n)$. LLP can eliminate the white noise components, so by inserting LLP as a post filter for ANC, reasonable performance is expected.

Although, $\zeta(n)$ is not completely white and some modifications are required to make the post-filter robust against colored noises. In this research, we propose a method that computes LLP PARCORs according to the speech components of LLP input, instead of LLP input oneself. To achieve this, following modifications are done.

1- $\gamma_1^f(n)$ and $\gamma_1^b(n)$, which were computed according to the equations (15) and (16), must be calculated as equations (17) and (18), respectively.

$$\gamma_1^f(n) = -\frac{E\{e(n)e(n-1)\}}{E\{e^2(n-1)\}} = -\frac{p_1(n)}{g_1^b(n)} \tag{15}$$

$$\gamma_1^b(n) = -\frac{E\{e(n)e(n-1)\}}{E\{e^2(n)\}} = -\frac{p_1(n)}{g_1^f(n)} \tag{16}$$

$$\gamma_c^f(n) = -\frac{E\{s(n)s(n-1)\}}{E\{s^2(n-1)\}} = -\frac{p_c(n)}{g_c(n-1)} \tag{17}$$

$$\gamma_c^b(n) = -\frac{E\{s(n)s(n-1)\}}{E\{s^2(n)\}} = -\frac{p_c(n)}{g_c(n)} \tag{18}$$

2- $f_0(n)$ and $b_0(n)$ must be set to $s(n)$ rather than $e(n)$ at the initialization of the updating equations Eq.(7).

In the rest of this section we review the implementation issues of these equations. To apply the first set of modifications, it is sufficient to compute $p_c(n)$ and $g_c(n)$ as

$$p_c(n) = E\{s(n)s(n-1)\} = E\{e(n)e(n-1)\} - E\{\zeta(n)\zeta(n-1)\} \tag{19}$$

$$g_c(n) = E\{s^2(n)\} = E\{e^2(n)\} - E\{\zeta^2(n)\}\cdot \tag{20}$$

The first term of these equations are known, but their second term must be calculated. We name these second terms as $p_{cs}(n)$ and $g_{cs}(n)$. From Fig. 2 we know

$$\zeta(n) = \zeta_1(n) - \zeta_3(n). \tag{21}$$

Thus, substituting (21) into (19) gives

$$p_{cs}(n) = E\{\zeta(n)\zeta_1(n-1)\} - E\{\zeta(n)\zeta_3(n-1)\}\cdot \tag{22}$$

According to the principle of orthogonality, when ANC converges to its optimum solution (Wiener solution), we have

$$E\{\zeta(n)\zeta_3(n-1)\} = 0\cdot \tag{23}$$

Combining equations (21), (22) and (23) gives

$$p_{cs}(n) = E\{\zeta_1(n)\zeta_1(n-1)\} - E\{\zeta_3(n)\zeta_1(n-1)\}\cdot \tag{24}$$

Note that speech is uncorrelated to noise i.e. $E\{s(n-1)\zeta_3(n)\}=0$, so (24) can be rewritten as

$$\begin{aligned} p_{cs}(n) &= E\{\zeta_1(n)\zeta_1(n-1)\} - E\{\zeta_3(n)(\zeta_1(n-1)+s(n-1))\} \\ &= E\{\zeta_1(n)\zeta_1(n-1)\} - E\{\zeta_3(n)d(n-1)\}\cdot \end{aligned} \tag{25}$$

Characteristics of various points in an ideal diffuse noise fields are identical [1], so by the diffuse model assumption of vehicular noise fields, for the first term of Eq. (25) we have

$$E\{\zeta_1(n)\zeta_1(n-1)\} = E\{\zeta_2(n)\zeta_2(n-1)\}\cdot \tag{26}$$

Hence, final equation for $p_{cs}(n)$ is given by

$$p_{cs}(n) = E\{\zeta_2(n)\zeta_2(n-1)\} - E\{\zeta_3(n)d(n-1)\} \tag{27}$$

With similar calculations the final expression for $g_{cs}(n)$ is obtained as

$$g_{cs}(n) = E\{\zeta_2^2(n)\} - E\{\zeta_3(n)d(n)\}\cdot \tag{28}$$

Substituting (27) and (28) into (19) and (20), respectively, gives the final expression for $p_c(n)$ and $g_c(n)$.

$$p_c(n) = E\{e(n)e(n-1) - \zeta_2(n)\zeta_2(n-1) + \zeta_3(n)d(n-1)\} \tag{29}$$

$$g_c(n) = E\{e^2(n) - \zeta_2^2(n) + \zeta_3(n)d(n)\} \tag{30}$$

As described in the previous section, expectation is evaluated by lowpass filtering of instantaneous values. The first modification is obtained by using equations (29) and (30).

The goal of second modification is to set the initial prediction error to $s(n)$, but it is not realizable because it requires some information about the clean speech which we do not have. To apply this conversion, we consider a multistage structure where in the $(i+1)^{th}$ stage the enhanced signal of i^{th} stage is used as the initial prediction error instead of $s(n)$. Hence, in each stage better noise reduction is done, and initial prediction errors of the next stage become closer to the clean speech signal. The multistage structure is shown in Fig. 4. Equations (29) and (30) are used to compute the initial PARCORs, $\gamma_1^f(n)$ and $\gamma_1^b(n)$, for all stages. The experimental results show that increasing the number of stages up to 4, improves the speech quality in the intense noisy environment. This structure solves the first problem of LLP mentioned above (in the sub-section III-A). The second problem of LLP is that clean speech signal cannot be extracted completely even if the noise components are perfectly white. For example, an unvoiced phoneme which is similar to noise, is not predictable in this structure. Consequently, LP-based speech enhancement systems must make a compromise between the portion of remaining background noise and eliminating speech components [8, 9]. To manage this tradeoff, we define a new parameter, λ, and calculate the LLP output of the i^{th} stage, $\hat{s}_i(n)$, by Eq. (31), where $f_M(n)$ and $\hat{s}_{i-1}(n)$ are forward LP error and predictor input, respectively.

$$\hat{s}_i(n) = \hat{s}_{i-1}(n) - \lambda f_M(n) \tag{31}$$

This parameter is set to 0.6 in our experiments.

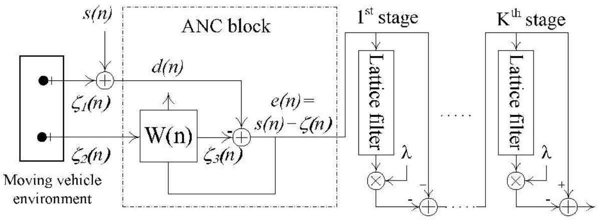

Fig. 4. Adaptive Noise Canceller integrated with K-stage lattice LP post-filter (ANCLLP-K)

V. EXPERIMENTS

In this section, we conduct various objective experiments to compare the previous and proposed methods.

A. Experimental Conditions

We used *4* set of dual channel noise signals, recorded in the real-life automobile environment. One channel was applied as a reference input of the ANC and other one was added to speech and employed as a desired input of ANC. Sit-Mall, HWY, CarWOpen and CarWClose situations described in section II, were chosen as the noise material. A test set of *10* sentences (*5* males and *5* females) from the FARSDAT database [12] were selected as the speech material.

Adaptive noise canceller was parameterized using *512* taps and Least Mean Square (LMS) adaptation algorithm with constant step-size set to *0.01*. Also, we made use of lattice filters with the order of *512*. LLP and ANC are lattice linear predictor and adaptive noise cancelation based speech enhancement systems. ANCLLP-i is a combination of ANC followed by multistage post-filter with *i* stages, which is depicted in Fig. 4.

B. Experimental Results

For objective evaluation of speech quality, we employed the Perceptual Evaluation of Speech Quality (PESQ) [13], SNR and segmental SNR measures in *0*, *5* and *10 dB* input SNRs. High correlations with subjective listening tests were reported using the PESQ score [13] and our informal listening tests were consistent with PESQ values. This score is mapped to a subjective Mean Opinion Score (MOS) scale, in the range of *-0.5* to *4.5*. Fig. 5 demonstrates the results of our evaluations with PESQ score.

For all input SNR levels, LLP and ANC provide the worst results while the proposed methods have improved the performance of these structures. In the case of intense noise conditions, ANCLLP-4, that uses the *4* stages lattice predictor as a post-filter, is the most effective method. In the higher input SNR levels using *2* or *3* stages results in remarkable noise reduction. Furthermore, this method does not distort speech signal very much. As the number of stages increases, the speech signal becomes more distorted such that the effect of this distortion is more than the effect of noise reduction, so in the case of higher input SNRs, increasing the number of stages more than 3, reduces the PESQ value.

Table 1 and 2 shows the evaluation results using segmental SNR and SNR. As it is shown, the proposed methods outperform the ANC and LPF in *0* and *5 dB* input SNRs, but the results of this measure are not consistent with PESQ and informal subjective quality assessment in the case of *10* dB input SNR. As described in [8], the LP-based enhancement filtering provides the improvement, in the sense of SNR measure, only in low SNR conditions.

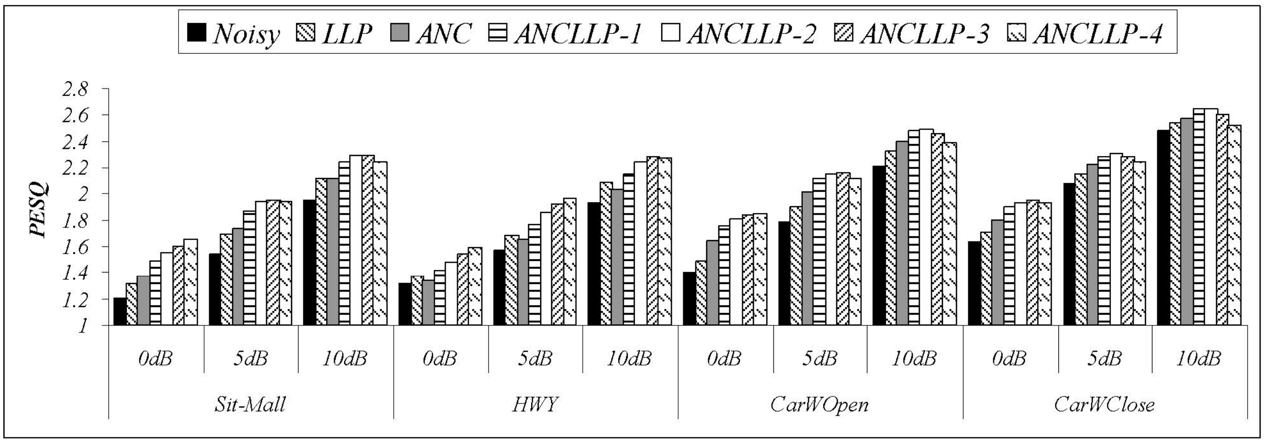

Fig. 5. Quality measure evaluation with PESQ in various situations for 0, 5 and 10 dB input SNRs

Table 1. Results of objective quality assessment using segmental SNR measure

Seg. SNR	Sit-Mall			HWY			CarWOpen			CarWClose		
	0dB	***5dB***	***10dB***	***0dB***	***5dB***	***10dB***	***0dB***	***5dB***	***10dB***	***0dB***	***5dB***	***10dB***
NOISY	*-3.3552*	*0.16311*	*3.9778*	*-3.2846*	*0.23741*	*4.0595*	*-2.9698*	*0.57033*	*4.3926*	*-2.504*	*1.0642*	*4.9399*
ANC	*-3.1089*	*0.54068*	*4.3521*	*-2.6815*	*0.90758*	*4.6506*	*-1.7321*	*2.1498*	*6.5399*	*-1.0633*	*2.6697*	*7.0051*
LLP	*-2.2065*	*0.69381*	*3.4757*	*-2.4412*	*0.62766*	*3.5956*	*-2.6698*	*0.51738*	*3.6585*	*-2.4593*	*0.76522*	*3.9667*
ANCLLP-1	*-1.9375*	*1.4131*	*4.6743*	*-1.6632*	*1.6625*	*4.9217*	*-1.1243*	*2.4103*	*6.2282*	*-0.64037*	*2.6844*	*6.4188*
ANCLLP-2	*-1.5931*	*1.3566*	*4.0106*	*-1.263*	*1.6911*	*4.355*	*-1.1076*	*2.0002*	*5.1295*	*-0.75078*	*2.1413*	*5.19*
ANCLLP-3	*-1.5073*	*0.97538*	*3.1741*	*-1.1601*	*1.3735*	*3.5398*	*-1.2449*	*1.4417*	*4.0407*	*-0.97328*	*1.542*	*4.0828*
ANCLLP-4	*-1.5816*	*0.56508*	*2.4493*	*-1.1502*	*1.014*	*2.7925*	*-1.4386*	*0.9241*	*3.1356*	*-1.2193*	*0.99566*	*3.1632*

Table 2. Results of SNR measure

SNR	Sit-Mall			HWY			CarWOpen			CarWClose		
NOISY	***0dB***	***5dB***	***10dB***	***0dB***	***5dB***	***10dB***	***0dB***	***5dB***	***10dB***	***0dB***	***5dB***	***10dB***
ANC	*0.52644*	*5.5295*	*10.449*	*1.1057*	*6.0383*	*10.825*	*1.9689*	*7.0582*	*12.651*	*2.523*	*7.2095*	*12.804*
LLP	*1.9625*	*6.0326*	*9.6395*	*1.5002*	*5.8478*	*9.7441*	*0.7539*	*5.3457*	*9.5712*	*0.26041*	*5.0112*	*9.445*
ANCLLP-1	*1.8502*	*6.3832*	*10.701*	*2.6856*	*7.1883*	*11.371*	*2.8977*	*7.5064*	*12.432*	*2.9535*	*7.3175*	*12.453*
ANCLLP-2	*2.3927*	*6.3851*	*9.8758*	*3.2813*	*7.2425*	*10.551*	*2.9562*	*7.0323*	*10.97*	*2.8762*	*6.8451*	*11.09*
ANCLLP-3	*2.4442*	*5.7893*	*8.6182*	*3.4037*	*6.6989*	*9.1915*	*2.7954*	*6.3344*	*9.4568*	*2.6298*	*6.1669*	*9.5755*
ANCLLP-4	*2.2196*	*5.0746*	*7.437*	*3.2777*	*5.9964*	*7.9035*	*2.488*	*5.5666*	*8.0948*	*2.3183*	*5.4477*	*8.1994*

VI. CONCLUSION

In vehicular applications, there are uncorrelated noise components at the inputs of adaptive noise cancellers. This degrades the performance of ANC extremely. Also, LLP cannot suppress the colored noise components. In this paper, we apply a new multistage LLP as a post-filtering method for ANC to alleviate the problems of them. The adaptation algorithms for LLP are modified such that they can eliminate colored noises too. Evaluation results show that the multistage post-filter with *4* stages in the intense noisy environments and with *2* or *3* stages in mild situations results in the highest speech quality.

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