ISSN 2229-5518

Phonemic Restoration of The Musical Sound With The Ephraim And Malah Noise Suppressor

Goutam Goyal, M.E., Biomedical Engineering Research Group , Singhania University Rajasthan , India, Email gautamgoyal05@yahoo.com ,
Dr. K.K. Dhawan, Ph.D. Director, Shekhawati Group Of College Shekhawati, Rajasthan, India ,Email dr.kk_dhawan@yahoo.com ,
Dr. S.S. Tiwari , Ph.D. , Managing Director , Sensors Technology Pvt. Ltd. , Gwalior , India , Email sst@sensorstechnology.com ,

Abstract-Under certain conditions, sounds actually missing from a speech signal can be synthesized by the brain and clearly heard. This illusory phenomenon, known as the phonemic restoration effect, reveals the sophisticated capability of the brain underlying robust speech perception in noisy situations often encountered in daily life. This paper presents a study of the noise suppression technique proposed by Ephraim and Malah. This technique has been used recently for the restoration of degraded audio recordings because it is free of the frequently encountered 'musical noise' artifact. It is demonstrated how this artifact is actually eliminated without bringing distortion to the recorded signal even if the noise is only poorly stationary.

1. INTRODUCTION

This perceptual restoration effect clearly indicates that the sounds we hear are not copies of physical sounds. The brain fills the gaps with the sounds that should exist in the portions masked by noise bursts based on the information in the remaining speech signal. What we perceive is the result of such unconscious interpretation.

The phonemic restoration effect involves two aspects: apparent continuity and increased intelligibility. The former aspect is not restricted to speech. The apparent continuity of a sound disrupted by an extraneous noise can be observed for various types of sounds, including music (16), environmental sounds, and pure tones. The phonemic restoration effect can be thought of as a special case of more general auditory continuity illusion (13). The necessary condition for the auditory continuity illusion is that the acoustic (spectral, temporal, and spatial) characteristics of the interrupting sound must be sufficient to mask the interrupted sound if the two sounds were presented simultaneously (13,17). This condition is known as the

Masking potential rule.

The first cue is a phenomenon called co articulation. Speech signal is produced by the movement of articulator organs such as lips, a tongue, and a jaw, all of which cannot move freely and abruptly. Instead, these organs move smoothly in a highly cooperative manner, resulting in the articulation of adjacent phonetic segments interacting with each other. This is co articulation. As a consequence of co articulation, information for a phonetic segment distributes over time in the range of a few hundred milliseconds, overlapping with that for adjacent segments. The brain can exploit such acoustic redundancy of speech signal in the phonemic restoration (19). Another cue is semantic context provided by sentences. In some cases, semantic information that appears after the disrupted portion can affect the perceptual restoration of the disrupted portion in an apparently retroactive fashion (20).

Semantic context seems to work when the masking potential rule is met (21) Speech signal is redundant not only in the time domain but also in the frequency domain. Perceptual restoration of missing information can occur also in the frequency domain (23). For example, a speech stimulus consisted of two widely separated narrow bands of speech is not very
intelligible. However, when noise was introduced in the spectral gap separating the two speech bands, intelligibility increases significantly. This spectral restoration is analogous
to the phonemic restoration in time domain: both represent mechanisms for minimizing interference when extraneous sounds replace portions of speech. These phenomena, as well as the phonemic restoration effect, demonstrate the sophisticated capabilities of the brain exploiting various types of redundancy in speech
signal to realize stable recognition of linguistic messages even under noisy situations in the real world.

At present, the noise reduction techniques used for the restoration of degraded audio recordings are based on ***short-time spectral attenuation.*** In such techniques, the attenuation that is to be applied to each one of the short-time Fourier transform coefficients is estimated by the ***noise suppression rule*** VI, (8), (11). One artifact that has been widely reported concerning the use of short-time spectral attenuation techniques is that the noise remaining after the processing has a very unnatural disturbing quality (9), (10), (12). This comes from the fact that the magnitude of the short-time spectrum ***IX(p, Wk)l*** exhibits strong fluctuations in noisy areas, which is a well known feature of the periodogram (2). After application of the spectral attenuation, the short-time magnitude spectrum in the frequency bands that originally contained noise now consists of a succession of randomly spaced spectral peaks corresponding to the maxima of ***I x (p , w k) l .*** In between these peaks, the short-time spectrum values are strongly attenuated because they are close to or below the estimated average noise spectrum. **As** a result, the residual noise is composed of sinusoidal components with random frequencies that come and go in each short-time frame (l), (9). This artifact is known as the “musical noise phenomenon”; the term “musical” is a reference to the presence of pure tones in the residual noise. Some modifications of the basic suppression rules have been proposed in order to overcome this problem (l), (12), but these techniques ***only reduce*** the musical noise without completely eliminating it. The complete elimination of the musical noise phenomenon is generally only obtained by a crude overestimation of the noise average spectrum. An unwanted consequence is that the short-time spectrum is attenuated much more than would be necessary; this is a fact that can generate audible distortions in the audio signal (3). It has been reported that the noise suppression rule proposed by Ephraim and Malah (4), (5) (which will be referred to as the EMSR in the following) makes it possible to obtain a
Significant noise reduction while avoiding the musical noise phenomenon described above. This feature explains why this suppression rule is an excellent choice for the restoration of musical recordings where the musical noise artifact is to be strictly avoided (10). In the original papers by Ephraim and Malah, this aspect of the suppression rule was only mentioned as an experimental finding. In this paper, we investigate the mechanisms that counter the musical noise phenomenon.

2. DESCRIPTION OF THE EMSR

The EMSR was proposed by Ephraim and Malah in [4] and developed in [5], and two other suppression rules along the same principle were introduced later by the authors in [5] and [6]. Here, we will focus only on the EMSR, because the fundamental mechanism that counters the musical noise effect is basically the same in all these suppression rules. The EMSR can be expressed as a spectral gain ***G (p , w k)*** that is applied to each short-time spectrum value ***X(p, W k) ;*** this gain is given by [41, [5]

$$G = \frac{\sqrt{\pi}}{2}\sqrt{\left(\frac{1}{1+\mathcal{R}_{\text{post}}}\right)\left(\frac{\mathcal{R}_{\text{prio}}}{1+\mathcal{R}_{\text{prio}}}\right)} \times \mathbf{M}\left[(1+\mathcal{R}_{\text{post}})\left(\frac{\mathcal{R}_{\text{prio}}}{1+\mathcal{R}_{\text{prio}}}\right)\right] \quad (1)$$

where **M** stands for the function

$$\mathbf{M}[\theta] = \exp\left(-\frac{\theta}{2}\right)\left[(1+\theta)I_0\left(\frac{\theta}{2}\right) + \theta I_1\left(\frac{\theta}{2}\right)\right]$$

where ***I(O)*** and ***1(1)*** are the modified Bessel functions of zero and first order, respectively (5). In (l), the time and frequency indexes ***p*** and ***Wk*** have been omitted for reasons of compactness. The spectral gain depends on two parameters (Rpost(p***W***, ***k)*** and **Rprio(p*W, k))*** evaluated in each short-time frame and for all spectral bins. These two parameters are interpreted as follows: ***The a posteriori signalto- noise ratio (or a posteriori SNR)*** Rpost(p***W***, ***k)*** is given by

$$\mathcal{R}_{\text{post}}(p,\omega_k) = \frac{|X(p,\omega_k)|^2}{v(\omega_k)} - 1 \quad (2)$$

where ***'u(Wk)*** denotes the noise power at frequency ***Wk.*** Equation (2) indicates that ***R p o s t (p , W k)*** is a local estimate of the

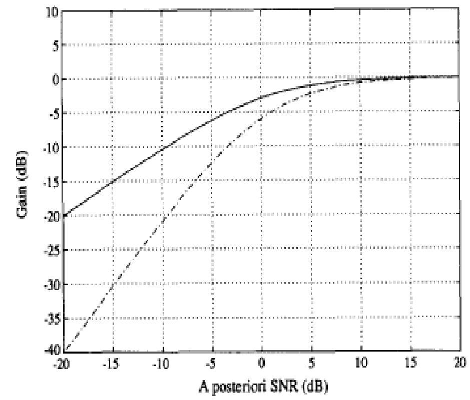

Fig. 1.Gain versus *a Posteriori* SNR solid line: power subtraction;
dashed line : Wiener

SNR computed from the data in the current short-time frame. Note that in the original papers by Ephraim and **Malah,** the definition of the ***a posteriori*** parameter is slightly different ***[5].*** The definition of (2) was preferred because it allows a simpler interpretation of **Rpos*(t* *p,***w k*).* ***The so-called a priorisignal-to-noise ratio (or a priori SNR)*** **Rprio(p**w, k*)* represents the information on the unknown spectrum magnitude gathered from previous frames and is evaluated in the "decisiondirected" approach (5) by

$$\mathcal{R}_{\text{prio}}(p,\omega_k) = (1-\alpha)P[\mathcal{R}_{\text{post}}(p,\omega_k)] + \alpha\frac{|G(p-1,\omega_k)X(p-1,\omega_k)|^2}{v(\omega_k)} \quad (3)$$

v(wk)
where *P[z]* = z if z *2* 0, and *P[z]* = 0 otherwise. As **RpOSt(***w***p***k,)* defined by (2) is not necessarily positive, the operator *P* guarantees that R**prio(p,wk**i**)**s always nonnegative
or, equivalently, that the expression of the gain given by (1) is valid. On the second line of **(3),** ***G(p - 1, w k) X (p - 1,*** *wk)* corresponds to the noiseless signal spectrum value as estimated in the previous frame. The term ***IG(p - 1, wk)X(p - 1,*** *wk)I2/v(wk)* thus corresponds to **an** estimation of the **SNR** in the

frame of index ***p- 1. Rprio(p****w, k)* ***is*** therefore **an** estimate of the **SNR** that takes into account the current short-time frame, with weight (1 - ***cy),*** and the result of the processing in the previous frame, with weight *a.* On the basis of simulations, the parameter ***cr*** was set by the authors to about 0.98. For standard suppression rules, the gain applied to each short-time spectral coefficient depends only on the signal level ***l X (p , ~ k)***m***(***e~as ured in the current frame. The gain can be expressed as a function of **Rpost(p***w, k).* Fig. 1 displays such suppression characteristics for the power subtraction and the so-called Wiener suppression rules (8), (11). The two curves of Fig. **1,** although they correspond to different strategies, illustrate the same intuitive principle that those points where the **SNR** is close to *-CO* dB are the ones that should be attenuated. These two curves are strongly related because the Wiener gain is the square of the power subtraction gain (**8).**

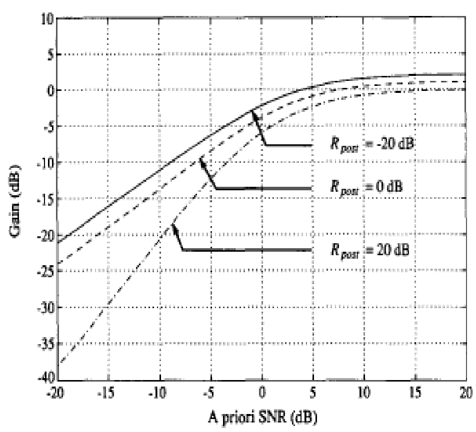

Fig 2 EMSR gain versus *a priori SNR* for different values of the ***a posteriori*** **SNR top-most curve:** *R,,,t(p,wk)* = **-20 dB; middle curve:** ***RpOst(p,wk)***= 0 **dB; bottom curve:** *RR,,.t(p,wk)* = **20 dB.**

The connection between the **EMSR** and more standard suppression rules is made clearer by plotting the gain of the **EMSR** versus the ***a priori*** **SNR** (in their original papers **[4],** ***[5],*** the authors used a reverse representation). The alternate representation of Fig. 2 highlights the respective influence of the two parameters of the **EMSR:** The ***a priori*** **SNR** is the dominant parameter. Strong attenuations are obtained only if **Rprio(p***W, k)* is low (left half of Fig. 2), and low attenuations are obtained only if **Rprio(p***w, k)* is high (right half of Fig. 2). Moreover, note that the overall shape of the gain is similar in Figs.
2 and 1 (although it must be stressed that the abscissa corresponds to **Rpost** in Fig. 1 and to **Rprio** in Fig. **2).**
The ***a posteriori SNR*** acts as a correction parameter whose influence is limited to the case where the ***a priori SNR*** is low (what is intuitively expected: The larger **Rpos*(t*****p *,****w k),* the stronger the attenuation. This over attenuation is a consequence of the disagreement between the ***a priori*** and the ***a posteriori*** **SNR's.** Why this counter-intuitive behavior is actually useful will be explained later. Comparison between Figs. 1 and 2 indicates that the **EMSR** is very close to the Wiener suppression rule ***evaluated as a function of*** **Rprio(p***w, k)* when **Rpost(p***w, k)* is 20 dB (bottom curves in the two figures). This remains true for values of **Rpost(p***w, k)* above 20 dB. Conversely, when **Rpost(p***w, k)* is - 20 dB, the **EMSR** gets very close to the power subtraction suppression rule evaluated as a function of **Rprio(p***w, k)* (top curves in the two figures). This is actually true for values of **RPost(p,wk**b**)** elow -5 dB. In practice, it can be considered that the **EMSR** corresponds to a smooth transition between the two uppression rules of Fig. 1; the ***a priori*** SNR **Rprio(p,wk**c**)**o ntrols the *x* coordinate along the suppression characteristics, whereas the ***a posteriori*** **SNR Rpos*(*t *p,*****w k)* controls the transition between the two asymptotic curves.

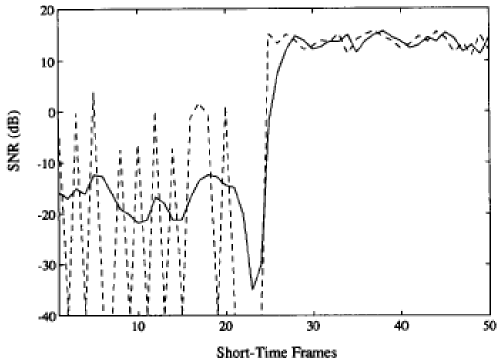

Fig. 3. ***SNR's*** **in successive short-time frames; dashed curve:** ***A posteriori***
SNR solid curve: ***A priori*** **SNR. For the first 25 short-time frames, the analyzed signal contains only noise at the displayed frequency; for the next 25 frames, a component with 15-dB SNR emerges at the displayed frequency. Parameter** ***a*** **is set to 0.98.**

3. ELIMINATION OF THE MUSICAL NOISE

A. The Smoothing Effect in the EMSR

The a ***priori SNR*** is evaluated by the nonlinear recursive relation of (3). ***An*** experimental study of (3) indicates two different behaviors for the a ***priori SNR:***

1) When **Rpost(p,wk)** stays below or is sufficiently close to 0 dB, the ***a priori SNR*** corresponds to a highly smoothed version of the ***a posteriori*** **SNR** over successive short-time frames. As a consequence, the variance of **Rprio(p,wk)** is much smaller than that of

2) ***On*** the contrary, when **Rpost(p***w,* *k)* is much larger than 0 dB, the a ***priori*** **SNR** follows the ***a posteriori*** **SNR** with a simple delay of one short-time frame. To see that, note that when the a ***priori*** **SNR** is high, the attenuation brought to the spectrum is negligible (right part of Fig. 2). Then, (3) reduces to

$$\mathcal{R}_{\text{prio}}(p,\omega_k) \approx (1-\alpha)\mathcal{R}_{\text{post}}(p,\omega_k) + \alpha\frac{|X(p-1,\omega_k)|^2}{v(\omega_k)}.$$

As $\mathcal{R}_{\text{post}}(p,\omega_k) \gg 1$, this can be written as

$$\mathcal{R}_{\text{prio}}(p,\omega_k) \approx (1-\alpha)\mathcal{R}_{\text{post}}(p,\omega_k) + \alpha\mathcal{R}_{\text{post}}(p-1,\omega_k).$$

Finally, because the parameter (I: is generally chosen very close to **1,** we can make the following approximation

$$\mathcal{R}_{\text{prio}}(p,\omega_k) \approx \alpha\mathcal{R}_{\text{post}}(p-1,\omega_k). \qquad (4)$$

These two different behaviors of **Rprio(p,wk)** are visible on the example of Fig. 3. Notice how in the left-hand part of the figure, the variance of **Rprio(p,wk)** is much lower than that of ***Rpost(p****w,* *k),* whereas on the right-hand part, **Rprio(p,** *wk)* follows **Rpost(p***w***r***k)* with a one frame delay. The smoothness of the **Q** ***priori*** **SNR** helps red ucing the musical noise effect. In the parts of the short-time spectrum corresponding to noise only, the ***a posteriori*** **SNR** is -*ca* dB in average, which corresponds to the case 1 above: Due to the smoothing behavior, the ***a priori*** **SNR** has **a** significantly
reduced variance. Because the attenuation of the **EMSR** depends mainly on the value of the ***a priori*** **SNR,** the attenuation itself does not exhibit large variations over successive frames. As a consequence, the musical noise (sinusoidal components appearing and disappearing rapidly over successive frames) is reduced. The idea of calculating the attenuation from the short-time spectrum averaged over successive frames was also exploited in **[l].** However, the superiority of the **EMSR** lies in the nonlinearity of the averaging procedure. When the signal level is well above the noise level, (3) becomes equivalent to a mere one-

frame delay, and **Rprio(p,q)** is no longer a smoothed **SNR** estimate, which is important in the case of non stationary signals.

B. Protection from Local Overtaking

The preceding results remain true if the **EMSR** gain function **G** in (3) is replaced by the Wiener suppression rule, ***evaluated as a function*** of **Rprio(p***w, k) [5].* However, simulations show that this is not the case when the power subtraction rule is used: Because the power subtraction attenuation is too small for values of the **SNR** around 0 dB (about -3 dB), the ***apriori*** **SNR** undergoes less smoothing and still exhibits important fluctuations. In the **EMSR,** another effect helps in eliminating the musical noise. In the frequency bands containing only noise, we have seen that the ***a priori*** **SNR** is about - 15 dB in average (see Fig. 3). In that case, improbable high values of the ***a posteriori*** **SNR** are assigned an increased attenuation. In the left half of Fig. 2, the attenuation increases for high values of the ***a posteriori*** **SNR** (values above 0 dB). This over attenuation is all the more important because **Rprio(p,wk)** is small. Thus, values of the spectrum higher than the average noise level are "pulled down."

This feature of the **EMSR** is particularly important for the recordings where the background noise is non stationary (e.g., recordings of old analog disks). The use of the **EMSR** avoids the appearance of local bursts of musical noise whenever the noise exceeds its average characteristics.

4. INFLUENCE OF THE PARAMETERS

A. Influence of (Y The choice of the value of parameter (Y is guided by a tradeoff between the degree of smoothing of parameter **Rprio(p***w, k)* in noisy areas and the acceptable level of transient distortion brought to the signal. Simulations show that when the analyzed signal contains only noise at a given frequency, both the average value and the standard deviation of the ***a priori*** **SNR** are proportional to (1 - (I:) when (I: is sufficiently close to one (above 0.9). As a result, in order to counter the musical noise e ffect, one will choose values of (I: as close to one as possible.

1

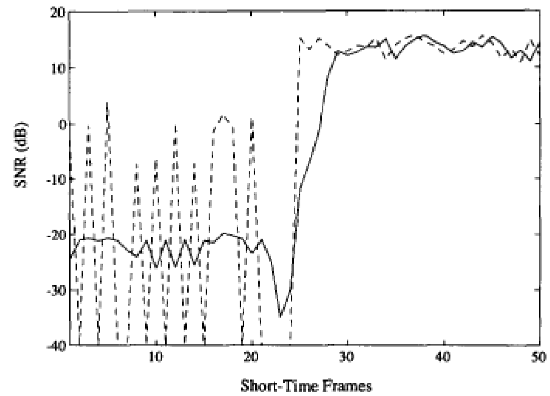

Fig. **4. SNR's in successive short-time frames; dashed curve: *A posteriori* SNR;** solid **curve: A *priori* SNR.** The **analyzed signal is the same as in Fig. 3. Parameter** *a* **is set to 0.998.**

On the other hand, when a signal component appears abruptly, the EMSR reacts immediately by raising the gain from a low value to a value close to 1 only if the SNR of the signal component is larger than 1/(1 - a). For signal components with lower SNR, simulations show that Rprio(pl *wk)* takes a longer time to reach its final value. This results in an unwanted attenuation of low-amplitude signal components during transient parts. The approximate limit of 1/(1 - a) is found by considering the study case where the ***a posteriori*** SNR is a deterministic quantity that equals zero before frame index ***po*** and has a fixed value of ***R*** for short-time frames with index ***p*** *2* ***po.* As** the gain of the EMSR is null before ***po,*** we have from (3)

$$\mathcal{R}_{\text{prio}}(p_0, \omega_k) = (1 - \alpha)\mathcal{R}.$$

If this first value satisfies **Rprio(p0,***wk)* >> 1, the gain of the EMSR evaluated at frame index ***po*** is already close to 1 (see Fig. 2). The condition that guarantees that there is no signal attenuation during the transient is thus (1 - a)R >> 1. The influence of parameter ***a*** appears clearly when comparing Figs. 3 and 4. In Fig. 4, the factor (1 - a) is divided by 10, compared with the case of Fig. 3. The average value of Rprio(p*w, k)* when noise is present drops from approximately -15 dB for the case of Fig. 3 to -25 dB for Fig. 4. The variance of Rprio(p,q) is also strongly reduced in Fig. 4, but there is now an important delay between the appearance of the transient component and the time when Rprio(p,wk) raises significantly above 0 dB. **As** a consequence, the signal component is incorrectly attenuated in the first short-time frames following the transient. In practice, the use of such a value of parameter a results in audible modifications of the signal transients. It should be noted that a more important overlap between successive windows reduces the transient distortion as the same number of short-time frame results in a shorter time delay. **As** a consequence, an overlap of 66% or more is sometimes preferred to the standard 50% setting [lo]. However, the variation of the overlap factor gives only slight perceptual differences because only the low-level transient components are distorted when reasonable values of a are used; for example, with ***CY*** = 0.98, the limit of 1/(1 - a) results in a SNR value of 15 dB.

B. Residual Noise Level

In the original paper by Ephraim and Malah, the gain function of (1) is tabulated for values ***of*** both SNR's between -15 and 15 dB [5]. The lower bound of this table is in fact a key parameter for the ***a priori SNR.*** Despite the smoothing performed by the procedure of **(3),** Rprio(p*w, k)*s till has some irregularities that can generate a perceptible low-level musical noise. **A** simple solution to this problem consists in constraining the ***a priori*** SNR to be larger to a threshold R(,in). In practice, the value of R(mini)s chosen to be larger than the average a ***priori*** SNR in the frequency bands containing noise only. **As** a consequence, in the frequency bands containing noise only, the average value of the constrained ***a priori SNR*** is close to R(,in). Furthermore, in the same frequency bands, most values of the a ***posteriori*** SNR are below 0 dB, and the gain function of the EMSR is close to the power subtraction whose squared gain can be shown to be equal to the SNR for low SNR values **[8]. As** a result, in the frequency bands containing noise only, the average squared gain is close to %+,in). l/R(min)c an therefore be interpreted as the average noise power reduction. When a equals 0.98, the average value of Rprio(prwk) is of **-15** dB, and a value of **R(min**a**)**r ound -15 dB is sufficient to eliminate the musical noise phenomenon, but R(minc)o uld be set to a larger value as well, with the effect of raising the level of the residual noise. The possibility to control the level of the residual noise is important for old recordings where the preservation of a certain amount of background noise is often judged as a positive aspect.

5. CONCLUSION

Speech signal is redundant not only in the time domain but also in the frequency domain. Perceptual restoration of missing information can occur also in the frequency domain (23). For example, a speech stimulus consisted of two widely separated narrow bands of speech is not very intelligible. However, when noise was introduced in the spectral gap separating the two speech bands, intelligibility increases significantly. This spectral restoration is analogous to the phonemic restoration in time domain: both represent mechanisms for minimizing interference when extraneous sounds replace portions of speech. These phenomena, as well as the phonemic restoration effect, demonstrate the sophisticated capabilities of the brain exploiting various types of redundancy in speech signal to realize stable recognition of linguistic messages even under noisy situations in the real world.

We have presented an analysis of the different mechanisms that counter the musical noise effect in the suppression rule proposed by Ephraim and Malah. The major factor was found to be the nonlinear smoothing procedure used to obtain a more consistent estimate of the SNR. With **an** appropriate choice of parameter a, the use of the smoothing procedure does not generate audible distortion in the signal. However, low-level signal components actually undergo a measurable over attenuation during abrupt transients. This transient distortion is hardly perceptible, and more precise listening tests would be necessary to decide whether it is useful or not to use an overlap factor larger than 50% Finally, it was shown that the attenuation function proposed by Ephraim and Malah avoids the appearance of the musical noise phenomenon even when the background noise is poorly stationary.

REFERENCES

1 S. F. Boll, "Suppression of acoustic noise in speech using spectral substraction," ***IEEE Trans. Acoust. Speech Signal Processing,*** vol. ASSP-27, no. 2, pp 113-120, 1979.
2 D. R. Brillinger, ***Time Series Data Analysis and Theory.*** San Francisco:
Holden-Day, 1981.
3 0. Cappi and J. Laroche, "Evaluation of short-time spectral attenuation
techniques for the restoration of musical recordings," to be published in
IEEE Trans. Speech Audio processing, 1994.
4 Y. Ephraim and D. Malah, "Speech enhancement using optimal nonlinear
spectral amplitude estimation," in ***Proc. IEEE In?. Conf. Acoust.Speech Signal Processing*** (Boston), 1983, pp. 11 18-1 5 **Y.** Ephraim and D. Malah , "Speech enhancement, using a minimum mean-square error log-spectral amplitude estimator," ***IEEE Trans. Acoust. Speech Signal Processing,*** vol. ASSP-32, no. 6, pp. 1109-1 121. 1984.
6 **Y.** Ephraim and D. Malah , "Speech enhancement, using a minimum mean-square error log-spectral amplitude estimator," ***IEEE Trans. Acoust.Speech Signal Processing,*** vol. ASSP-33, no. 2, pp. 44345, 1985.
7 J. **S.** Lim and A. V. Oppenheim, "Enhancement and bandwidth compression of noisy speech," ***Proc. IEEE,*** vol. 67, no. **12,** pp. 1586-1604,
Dec. 1979.
8 R. J. McAulay and M. L. Malpass, "Speech enhancement using a softdecision noise suppression filter," ***IEEE Trans. Acoust. Speech Signal Processing,*** vol. ASSP-28, no. 2, pp. 137-145, Apr. 1980.
9 **J.** A. Moorer and M. Berger, "Linear-phase bandsplitting: Theory and applications," ***J. Audio Eng. Soc.,*** vol. 34, no. 3, pp. 143-152, 1986.
10 J. C. Valiere, "Restoration of old recordings by means of digital
processing-contribution to the study of some recent techniques (text in French)," Ph.D. thesis, Universitc? du Maine, Le Mans, 1991.
11 P. Vary, "Noise suppression by spectral magnitude estimation- Mechanism and theoretical limits," ***Sign Processing,*** vol. 8, no. 4, pp. 387400, 1985.
12 S. Vaseghi and R. Frayling-Cork, "Restoration of old gramophone
recordings," ***J. Audio Eng.,*** vol. **40,** no. **10, pp.** 791-801, 1992.
13 R. M. Warren, Auditory Perception: A New Analysis and Synthesis (Cambridge University Press, Cambridge, 1999).
14 G. A. Miller and J. C. R. Licklider, ''The intelligibility of
interrupted speech,'' J. Acoust. Soc. Am., 22, 167–173 (1950).
15 R. M. Warren, ''Perceptual restoration of missing speech
sounds,'' Science, 167, 392–393 (1970).
(16) T. Sasaki, ''Sound restoration and temporal localization of noise in speech and music sounds,'' Tohoku Psychol. Folia, 39,79–88 (1980).
(17) M. Kashino and R. M. Warren, ''Binaural release from temporal induction,'' Percept. Psychophys., 58, 899–905 (1996).
(18) J. A. Bashford Jr., K. R. Riener and R. M. Warren, ''Increasing
the intelligibility of speech through multiple phonemic restorations,'' Percept Psychophys, 51, 211–217 (1992).
(19) M. Kashino, ''Perception of Japanese intervocalic stop consonants
based on the distributed cues in pre- and post-closure portions,'' J. Acoust. Soc. Jpn. (J), 48, 76–86 (1992).

ISSN 2229-5518

(20) R. M. Warren and G. L. Sherman, ‘‘Phonemic restorations based on subsequent context,’’ Percept. psychophys 16, 150–156 (1974).
(21) J. A. Bashford, Jr. and R. M. Warren, ‘‘Multiple phonemic
Restorations follow the rules for auditory induction,’’ Percept Psychophys 42, 114–121 (1987).
(22) K. Saberi and D. R. Perrott, ‘‘Cognitive restoration of reversed
speech,’’ Nature, 398, 760 (1999).
(23) R. M. Warren, K. Riener-Hainsworth, B. S. Brubaker, J. A.
Bashford, Jr. and E. W. Healy, ‘‘Spectral restoration of speech:
Intelligibility is increased by inserting noise in spectral gaps,’’ Percept. Psychophys 59, 275–283