ICASSP2021 TORONTO Canada June 6-11, 2021

Adaptive Reverberation Absorption using Non-Stationary Masking Components Detection for Intelligibility Improvement

G. Zucatelli and R. Coelho

Laboratory of Acoustic Signal Processing (LASP)
Military Institute of Engineering (IME), Rio de Janeiro, Brazil
zucatelli@ime.eb.br, coelho@ime.eb.br
lasp.ime.eb.br

Laboratory of Acoustic Signal Processing

Abstract

This letter introduces a new time domain absorption approach designed to reduce masking components of speech signals under noisy-reverberant conditions. In this method, the non-stationarity of corrupted signal segments is used to detect masking distortions based on a defined threshold. The non-stationarity is objectively measured and is also adopted to determine the absorption procedure. Additionally, no prior knowledge of speech statistics or room information is required for this technique. The results show that the proposed scheme leads to a higher intelligibility improvement when compared to competing methods for three objective measures. A perceptual listening test is also considered and corroborates these results.

Reverberation and Non-Stationarity

The natural non-stationary behavior of speech signals is closely related to speech intelligibility. The main goal of adaptive methods is to preserve transient speech regions in order to improve speech intelligiblity. The reverberation effect is usually described by a Room Impulse Response *h(n)*. In real environments, acoustic noises *w(n)* are also a common distortion, which means that both effects can occur simultaneously. Therefore, the resultant noisy-reverberant speech signal *s(n)* can be obtained by:

$$s(t) = h(t) \circledast x(t) + w(t)$$

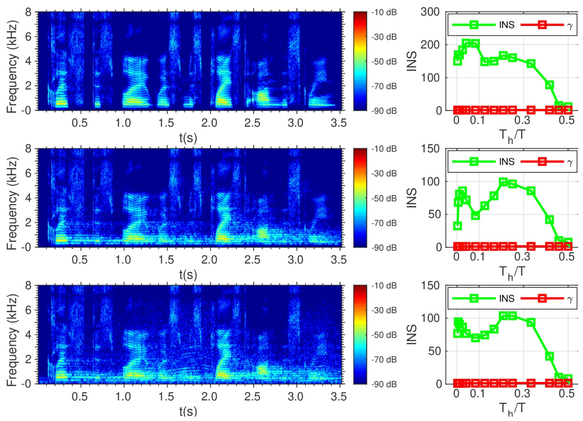


Fig. 1. Spectrogram and related INS for direct signal (top), reverberated signal with T_{60}= *4.9s* and SRR=*7.1* dB (middle), and reverberated signal with Chainsaw noise at *-3* dB (bottom)

The Index of Non-Stationarity (INS): In this work, the INS is adopted to objectively examine the non-stationarity of speech signals under noisy-reverberant environments.

This measure is a time-frequency approach that compares the target signal with stationarity references called surrogates for different time scales T_h/T, where T_h is the short-time spectral analysis length and T is the total signal duration. For each length T_h, a threshold γ is defined to keep the stationarity assumption considering a *95%* confidence degree:

$$\text{INS} \begin{cases} \leq \gamma : & \text{stationary,} \\ > \gamma : & \text{non-stationary.} \end{cases}$$

***Important* Notes:**

- Reverberation and acoustic noise significantly change the temporal and spectral structure of speech signal.
- Natural non-stationary behavior of speech signal can be considerably attenuated (INS: 200→100).
- As INS alters on noisy-reverberant scenarios, it can be a useful instrument for detection and reduction of such effects.

ARA_{NSD}

The Adaptive Reverberation Absorption with Non-Stationary Detection (ARA_{NSD}): The approach works similar to a physical element, changing the low absorption characteristic of materials that compose a room. The proposed technique can be described in two main phases: reverberation detection and acoustic absorption.

Reverb Detection

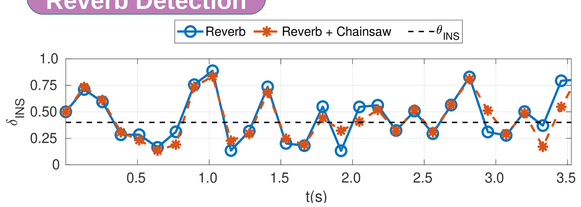


Fig. 2. Non-stationarity variation $\delta_{INS}(m)$ for the reverberated and noisy-reverberated signals.

Reverberation Groups (RG): The m-th segment composed of eight consecutive frames of the reverberant speech $s_{RG}(m,n)$.
This window duration is selected to enable a long-term temporal observation of the reverberation effect using the INS.

INS vector (v_{INS}): For each $s_{RG}(m,n)$, the INS values are grouped into v_{INS} vectors. which characterizes the non-stationary behavior of the RG.
Consecutive vectors are then used to compute a normalized variation $\delta_{INS}(m)$ of the non-stationary property as:

$$\delta_{INS}(m) = \frac{||\mathbf{v_{INS}}(m) - \mathbf{v_{INS}}(m-1)||}{||\mathbf{v_{INS}}(m)|| + ||\mathbf{v_{INS}}(m-1)||}$$

Absorption

Implemented on a frame-by-frame basis and established depending on the value of θ_{INS}, the theshould of non-stationarity. For each frame *l*, a INS vector $v_{frm}(l)$ is extracted and a short-time distance *d(l)* is computed, similarly as in the RG case.

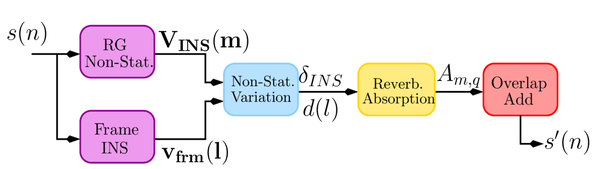


Fig. 3. The ARA_{NSD} scheme for adaptive absorption of acoustic masking effects .

Absorption: Sigmoid functions are selected to assign each value of *d(l)* to a corresponding absorption *A(m,l)* because of their smoothness and monotonic property. The proposed adaptive absorption *A(m,l)* is therefore defined in every frame *l* by

$$A(m,l) = \begin{cases} F(l) \cdot \frac{L(m) - S}{1 + \exp(-k.(d(l) - d_0))} + S, & \delta_{INS} \leq \theta_{INS}; \\ \frac{L'}{1 + \exp(-k'.(d(l) - d_0'))}, & \delta_{INS} > \theta_{INS}, \end{cases}$$

The resulting signal is reconstructed by overlap-add steps of processed frames.

Experiments and Results

Experimental Setup:

- 240 speech signals from TIMIT database
- Reverberation database AIR for rooms Stairway (T_{60}=1.1 s) and Aula Carolina (T_{60}=4.9 s)
- Four SNR conditions (-3dB, -2dB, -1dB, 0dB, 1dB) for Babble, Cafeteria, Chainsaw and SSN

Non-Stat. Restored

The ARA_{NSD} is able to absorb masking components of the corrupted signal, e.g. near *0.7* s and *1.5* s, which makes the resulting signal more similar to its anechoic version.
Moreover, the **proposed method restores the natural non-stationarity behavior** raising the INS value from *100* up to *150*, which is closer to the direct signal (INS=*200*).

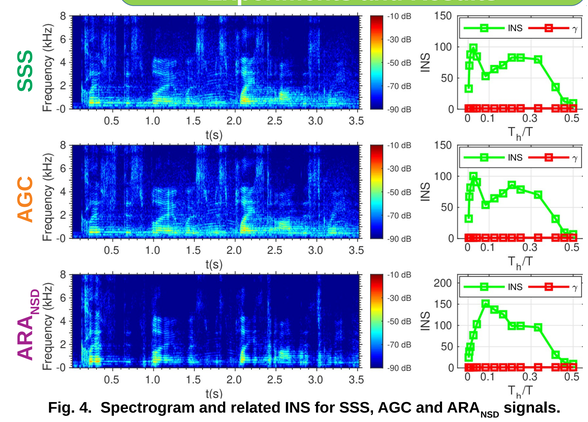


Fig. 4. Spectrogram and related INS for SSS, AGC and ARA_{NSD} signals.

Objetive Evaluation

The ESII and $ASII_{ST}$: These measures are adopted for the intelligibility prediction because they are explicitly designed to deal with the non-stationarity of speech and its distortions.

		ESII (%)					
SNR (dB)		-3	-2	-1	0	1	Avg.
Babble INS=40	UNP	63.8	64.8	65.7	66.6	67.3	65.6
	SSS	65.7	66.7	67.4	68.3	68.9	67.4
	AGC	66.9	67.9	68.6	69.3	**69.9**	68.5
	ARA_{NSD}	**68.8**	**69.2**	**69.5**	**69.7**	**69.9**	**69.4**
Cafeteria INS=15	UNP	65.1	66.0	66.7	67.3	68.0	66.6
	SSS	67.2	67.9	68.5	69.0	69.6	68.4
	AGC	68.2	68.9	69.6	70.2	**70.6**	69.5
	ARA_{NSD}	**69.6**	**69.8**	**70.1**	**70.3**	**70.6**	**70.1**
Chainsaw INS=80	UNP	56.9	58.2	59.5	60.6	61.8	59.4
	SSS	58.9	60.2	61.5	62.5	63.7	61.3
	AGC	60.3	61.6	62.8	64.0	65.1	62.8
	ARA_{NSD}	**67.2**	**67.6**	**68.1**	**68.4**	**68.8**	**68.0**
SSN INS=2	UNP	62.2	63.4	64.5	65.4	66.3	64.4
	SSS	64.4	65.6	66.6	67.4	68.2	66.4
	AGC	65.7	66.7	67.7	68.6	69.3	67.6
	ARA_{NSD}	**68.6**	**68.9**	**69.3**	**69.6**	**69.9**	**69.3**

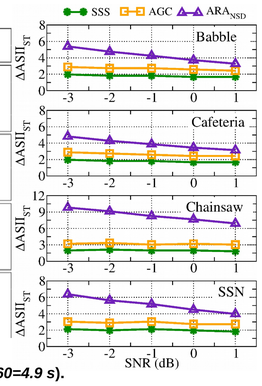


Fig. 5. ESII and $ASII_{ST}$ for Aula Carolina room (*T60=4.9 s*).

The $SRMR_{norm}$: metric estimates the human perceived reverberation effect on speech signals.

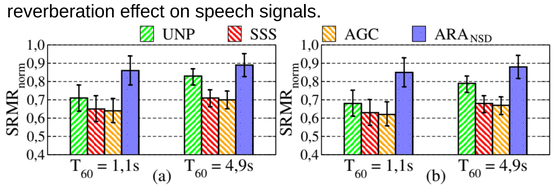


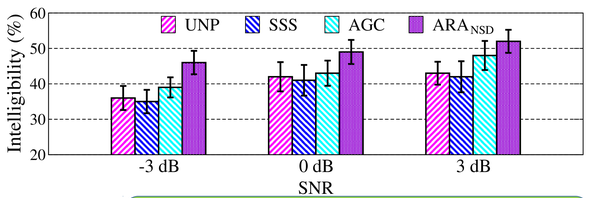
Fig. 6. $SRMR_{norm}$ for (a) Noise-free reverb and (b) reverb with SSN at *0dB*.

The ARA_{NSD} attains the best objective results for all measures regarding multiple noise-reverberant conditions. The highest gain is observed for *-3dB* Chainsaw, the most non-stationary acoustic noise.

Perceptual Evaluation

- Listening test with *10* male Brazilian volunteers
- Simulated room with **Image Source Method (ISM)** - T_{60}=*1.0 s*
- **SSN noise** at *-3dB, 0dB, 3dB*.

Fig. 7. Perceptual intelligibility evaluation for ISM room (T_{60}=*1.0 s)* and SSN additive acoustic noise.



Conclusion

This letter proposed a new time domain absorption approach designed to reduce masking components of speech signals under noisy-reverberant conditions. In this method, the non-stationarity of corrupted signal segments are assessed via INS and used to detect masking distortions based on a defined threshold. The results showed that the proposed scheme leads to a higher intelligibility improvement when compared to competing methods. A perceptual listening test was also developed and corroborate the objective results.

References: Highlights

- R. Bolt and A. MacDonald, “Theory of speech masking by reverberation,” J. Acoust. Soc. Amer., vol. 21, no. 6, pp. 577–580, 1949.
- T. Arai, N. Hodoshima, and K. Yasu, “Using steady-state suppression to improve speech intelligibility in reverberant environments for elderly listeners,” IEEE Trans. Audio, Speech, Lang. Process., vol. 18, no. 7, pp. 1775–1780, Sep. 2010.
- P. N. Petkov and Y. Stylianou, “Adaptive gain control for enhanced speech intelligibility under reverberation,” IEEE Signal Process. Lett., vol. 23, no. 10, pp. 1434–1438, Oct. 2016.
- P. Borgnat, P. Flandrin, P. Honeine, C. Richard, and J. Xiao, “Testing stationarity with surrogates: A time-frequency approach,” IEEE Trans. Signal Process., vol. 58, no. 7, pp. 3459–3470, Jul. 2010.
- L. Zão, R. Coelho, and P. Flandrin, “Speech enhancement with EMD and hurst-based mode selection,” IEEE/ACM Trans. Audio, Speech, Lang. Process., vol. 22, no. 5, pp. 899–911, May 2014.
- R. Coelho and L. Zão, “Empirical mode decomposition theory applied to speech enhancement,” in Signals and Images: Advances and Results in Speech, Estimation, Compression, Recognition, Filtering and Processing, R. Coelho, V. Nascimento, R. Queiroz, J. Romano, and C. Cavalcante, Eds. Boca Raton, FL, USA: CRC Press, 2015.
- R. Tavares and R. Coelho, “Speech enhancement with nonstationary acoustic noise detection in time domain,” IEEE Signal Process. Lett., vol. 23, no. 1, pp. 6–10, Jan. 2016.
- S. Ghimire, “Speech intelligibility measurement on the basis of ITU-T Recommendation P.863,” 2012.