

## Noise profiling for speech enhancement employing machine learning models

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This paper aims to propose a noise profiling method that can be performed in near real-time based on machine learning (ML). To address challenges related to noise profiling effectively, we start with a critical review of the literature background. Then, we outline the experiment performed consisting of two parts. The first part concerns the noise recognition model built upon several baseline classifiers and noise signal features derived from the Aurora noise dataset. This is to select the best-performing classifier in the context of noise profiling. Therefore, a comparison of all classifier outcomes is shown based on effectiveness metrics. Also, confusion matrices prepared for all tested models are presented. The second part of the experiment consists of selecting the algorithm that scored the best, i.e., Naïve Bayes, resulting in an accuracy of 96.76%, and using it in a noise-type recognition model to demonstrate that it can perform in a stable way. Classification results are derived from the real-life recordings performed in momentary and averaging modes. The key contribution is discussed regarding speech intelligibility improvements in the presence of noise, where identifying the type of noise is crucial. Finally, conclusions deliver the overall findings and future work directions.

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## 1 I. INTRODUCTION

2 Research in speech signal processing and enhancement has attracted considerable interest over the  
3 past decades. Major progress has been achieved in various applications, including automatic speech  
4 recognition (Li, 2021; Korvel et al., 2021; Michalopoulou et al., 2021), speaker recognition (Krcadinac  
5 et al., 2021), and emotion recognition from speech (Gosztolya, 2019; Liu et al., 2021; Morgan et al.,  
6 2021). However, when referring to robust speech processing, i.e., in noisy conditions, the progress in  
7 this field is below expectations (Li et al., 2015; Srinivasan et al., 2019). Environmental or ambient noise  
8 decreases the quality and intelligibility of the speech signal (Trujillo et al., 2021). Therefore, it is vital  
9 need to improve the assessment of speech intelligibility in the presence of interference noise. Various  
10 noise-robust approaches are adopted for this purpose. Typically, signal processing techniques are  
11 employed to reduce noise and enhance voice quality. There is a rich body of work focused on speech  
12 enhancement algorithms that use sparse Bayesian learning to solve the sound source localization  
13 problem of speech mixtures in noise (Xenaki et al., 2018) and improve speech enhancement by  
14 considering power spectral density (PSD) characteristics (Kavalekalam et al., 2018; Kim and Shin,  
15 2022), or aim to improve the quality and intelligibility of noise-corrupted speech through spectral or  
16 temporal modifications (Cooke et al., 2019; Kaçkol et al., 2020). The limitation of speech enhancement  
17 algorithms is that they are based on additive background noise or statistical properties of the speech  
18 and noise signal. However, the performance of speech enhancement in a real noisy environment, such  
19 as traffic, wind, or a cocktail-party effect when people talk simultaneously (i.e., babble speech), is often  
20 unsatisfactory. That is why the challenge of increasing real-world noise recognition robustness is still  
21 a significant problem, especially in cases where noise profiling is a necessary step for correct speech  
22 signal processing and quality and intelligibility enhancement is the primary goal.

23 In the literature, there exist several definitions of noise profiling that are related to the task needed,  
24 e.g., automatic annotation of noise data (Lin and Tsao, 2021) or attenuation of the noise to certain

25 predefined target levels (Zou et al., 2011). It may also be defined by the automatic threshold selection  
26 within lower and higher limit values (Dias et al., 2022), by clustering classification sound types (Kong  
27 et al., 2019), or by a noise profile observation in detected silent intervals (Xu et al., 2020).

28 The present study goes beyond the state-of-the-art methodology of speech enhancement as it  
29 incorporates noise inference profiling. In this work, noise profiling is understood similarly to noise  
30 type recognition but with a slightly different focus. While for the sound recognition models, it is crucial  
31 to obtain correct sound classification (e.g., whether it is a train sound or speech), for profiling task, it  
32 is critical to identify the sound characteristics (e.g., spectral features) which are specific to a given type  
33 of sound (i.e., noise in our case). In the latter case, precise noise identification is of less importance  
34 (Zou et al., 2011). Our previous research (Korvel et al., 2020) demonstrated that using the Lombard  
35 effect might improve speech intelligibility in the presence of noise. However, it is crucial to know the  
36 noise type to apply the best possible speech modifications. That is the context of our research.

37 To some extent, our research fits the paradigm of gathering experience based on interactions with the  
38 environment through some actions, as the process of noise recognition is sequential, and a decision  
39 on enhancing the speech signal should be taken based on satisfying the reward hypothesis (Mahmud  
40 et al., 2018).

41 This work aims to prepare the machine-based model recognizing the noise type and correctly  
42 classifying it in near real-time. Based on noise classification, it may then be possible to modify the  
43 speech signal appropriately to increase the probability of improving its quality and intelligibility. The  
44 study is conducted with a new perspective, focusing not on assigning a disturbance to a given class  
45 only but rather on investigating the stability of this assignment – understood as a classification  
46 consistency over a longer time, i.e., at least 5 seconds. This allows for stabilizing the decision rules,  
47 which might be placed in the system after the profiling block. This adds a new quality to noise profiling  
48 that is time-dependent. This research area requires a thorough analysis of speech and noise elements

49 based on a microscopic scale. Therefore, we left the large-scale deep learning analysis outside of this  
50 research, disregarding that noise recognition robustness is well served by deep learning methods (Roch  
51 et al., 2021; Watanabe et al., 2017). However, state-of-the-art baseline techniques that incorporate the  
52 extraction of features and machine learning, such as Naïve Bayes (Zhang, 2014; Barber, 2012), linear  
53 SVM (Cortes and Vapnik, 1995; Platt, 1999), SVM with the polynomial kernel (Wu et al., 2004),  
54 Gaussian process classifiers (Rasmussen and Williams, 2006; Byrd et al., 1995; Zhu et al., 1997),  
55 Decision tree (DT) (Kamiński et al., 2017), Random forest (RF) (Ho, 1995), Multilayer Perceptron  
56 (MLP) (Pedregosa et al., 2011), AdaBoost classifier (Rojas, 2009), and Quadratic Discriminant Analysis  
57 (Ghojogh and Crowley, (2019) that arose from different families and areas of knowledge (Fernández-  
58 Delgado, 2014) are used. It is worth noting that the methodology based on feature extraction and  
59 baseline classifiers shows its superiority in many speech signal processing tasks such as speech emotion  
60 recognition (Bhavan et al., 2019; Tuncer et al., 2021) or allophones classification (Piotrowska et al.,  
61 2019). These studies focused on preparing an optimized feature vector and utilizing this vector in the  
62 classification process. In the case of speech emotion recognition, the SVM classifier is used for  
63 classification in the mentioned above works. According to Bhavan et al. (2010), SVMs provide  
64 reasonably good estimates with lesser effort. In contrast, hidden Markov models and deep neural  
65 networks are more challenging to build and train and require higher computational power and time.  
66 In the work of Piotrowska et al. (2019), automatic classification methods, such as artificial neural  
67 networks (ANNs), the k-nearest neighbor (kNN), and self-organizing maps (SOMs), are applied to  
68 lateral allophone analysis and returned satisfactory results.

69 Also, we justify why the process of improving speech quality and intelligibility should be adaptive and  
70 specific modifications may depend on the noise characteristics and be reinforced by them. Based on  
71 the rate of change in intensity, noise can be classified into continuous, periodic, impulsive, and low-  
72 frequency noise (Tsalera et al., 2020). Therefore, a stable noise profiling method is needed – stable in

73 terms of being consistent over a longer period of time (Yang and Ritzwoller, 2008). Possible speech  
74 modifications must fit the disruption to provide the best results in terms of potential loss in  
75 intelligibility because of the noise presence. It is because every disturbance has different characteristics  
76 and impacts speech differently. However, it is more important to have the noise recognition process  
77 repetitive and stable rather than classify a given type of noise as a babble speech or airport noise. Also,  
78 noise signals with similar frequency characteristics should always be analogously classified to ensure  
79 that the speech signal modification is appropriate and durable.

## 80 **II. MATERIAL AND METHODS**

### 81 **A. Extraction of signal features**

82 In the learning process, the Aurora noise dataset was employed (Hirsch and Pearce, 2000). The Aurora  
83 database contains various speech recordings prepared mainly for speech recognition systems,  
84 especially for distributed speech recognition (Kshirsagar and Falk, 2021; Bandela et al., 2021). The  
85 noise database within the Aurora dataset has been prepared directly for speech processing, and it is,  
86 therefore, appropriate for our research. The noise signals contained in the Aurora dataset are as  
87 follows: airport, babble speech, car noise, exhibition, restaurant, street noise, subway, and train. Some  
88 noises are reasonably stationary, e.g., the car noise and the recording in the exhibition hall. Others  
89 contain non-stationary segments, e.g., recordings on the street and at the airport (Hirsch and Pearce,  
90 2000). In addition, pink noise was generated as this noise type was not present in the Aurora database.  
91 To be noted, pink noise is a signal with a frequency spectrum such that the power spectral density is  
92 inversely proportional to the signal's frequency, i.e., the power per Hertz in pink noise decreases as  
93 the frequency increases (<https://www.livescience.com/38464-what-is-pink-noise.html>). In pink  
94 noise, each octave interval carries an equal amount of noise energy, so the sound of pink noise is  
95 perceived as being even.



96 The following frequency characteristics were chosen and extracted to classify noise types (Klapuri and  
97 Davy, 2007; McFee et al., 2015; Das et al., 2021), i.e., spectral centroid, spectral bandwidth, spectral  
98 flatness.

99 The most important factor in evaluating the usefulness of the given feature is the separation of the  
100 calculated values in the context of the considered noise type. Three frequency characteristics,  
101 calculated in real-time, were considered to increase the separation of different types of noise. What is  
102 more, for each of the characteristics, the following short-term statistical parameters are calculated:  
103 maximum value, minimum value, average value, amplitude, standard deviation, variance, and median.  
104 The given statistic values should provide great noise parameters separation. The frequency  
105 characteristics are calculated from the Fourier spectrum computed with a Hamming window of 2048  
106 samples (25% overlap). Below the analyses performed have been described.

### 107 **1. Spectral centroid**

108 Spectral centroid is a metric used in digital signal processing that characterizes the spectrum of the  
109 signal. It allows calculating where the center of mass of the spectrum is located. This measure is related  
110 perceptually to the impression of the sound brightness. In this study, the spectral centroid is calculated  
111 as the weighted mean of the frequencies present in the signal with their magnitudes as the weights:

$$112 \quad SC = \frac{\sum_{n=0}^N f(n)X(n)}{\sum_{n=0}^N X(n)} \quad (1)$$

113 where  $X(n)$  is the weighted magnitude of the Fourier transform at frequency bin  $n$ , and  $f(n)$   
114 represents the center frequency of that bin.

### 115 **2. Spectral bandwidth**

116 The spectral bandwidth (SBW) is used to define the bandwidth of the signal spectrum. This measure  
117 shows the concentration of spectrum around the centroid and is computed by:

$$118 \quad SBW = (\sum_{n=0}^N X(n)(f(n) - SC)^p)^{1/p} \quad (2)$$

119 where  $X(n)$  is the weighted magnitude of the Fourier transform at bin  $n$ ,  $f(n)$  represents the center  
120 frequency of that bin,  $SC$  is the spectral centroid (see Eq. (1)). Variable  $p$  is equal to 2 – this  
121 corresponds to a weighted standard deviation around the centroid.

122 Spectral bandwidth values are calculated for all analyzed noise types and frames within the signal.

### 123 **3. Spectral flatness**

124 Spectral flatness is a measure of an audio sound spectrum that provides a way to quantify how tone-  
125 like a sound is, as opposed to being noise-like. High spectral flatness - approaching 1.0 for white noise  
126 - means that the spectrum has a similar amount of power in all spectral bands. Low spectral flatness  
127 values (approaching 0.0) convey that the power is concentrated in a small number of bands – typically,  
128 it is a mixture of sine waves.

129 The spectral flatness is calculated by dividing the geometric mean of the power spectrum by the  
130 arithmetic mean of the power spectrum, i.e.:

$$131 \quad SF = \frac{[\prod_{n=0}^{N-1} PX(n)]^{1/N}}{\frac{1}{N} \sum_{n=0}^{N-1} PX(n)} \quad (3)$$

132 The power spectrum  $PX(n)$  at bin number  $n$  is given by the following formula:

$$133 \quad PX(n) = \frac{1}{N} \sqrt{X(n)_{re}^2 + X(n)_{im}^2} \quad (4)$$

134 where  $X(n)$  is Fourier transform coefficient at bin  $n$ ,  $re$  means a real part, and  $im$  – an imaginary  
135 part.

### 136 **B. Noise type recognition model**

137 Based on the previously described frequency characteristics, the recognition models were built. For  
138 that purpose – as already mentioned – several baseline algorithms were employed, i.e., Naïve Bayes  
139 (NB), linear SVM (Support Vector Machines), SVM with the polynomial kernel, Gaussian process  
140 classifiers, Decision tree (DT), Random Forest (RF), Multilayer Perceptron (MLP), AdaBoost

141 classifier, and Quadratic Discriminant Analysis (QDA). For both learning and evaluation, the scikit-  
142 learn modules from the Python environment were used (<https://scikit-learn.org/stable/>).

143 Every recording containing noise was processed in the following way:

- 144 – each frame was 2 seconds in length - to retrieve the statistical features for the training
- 145 process,
- 146 – a 2-second window was moved by 0.1 seconds in each analysis step.

147 The classification models built use relatively long recording fragments because the measured  
148 parameter values change in time to a great extent. To clarify, the duration of the Aurora noise  
149 recordings is 10 seconds, and the generated pink noise recording is 5 seconds. Since the training is  
150 performed on the 2-second long frames, moved by 0.1 seconds, every Aurora noise recording resulted  
151 in 81 equally long 2-second frames, while the pink noise resulted in 31 frames of the same length. All  
152 frames were represented in the learning process by their calculated parameters – spectral centroid,  
153 spectral bandwidth, and spectral flatness. It means that in total, we had 679 samples (frames) – 81 for  
154 all 8 Aurora noise recordings and 31 for pink noise recordings.

155 The above dataset was divided into two almost equal parts: training (consisting of 339 samples) and  
156 testing used in generating predictions and calculating scores (composed of 340 samples). The training  
157 process was performed on the training set, while calculating scores and generating confusion matrices  
158 were performed on the testing set. In other words, the model evaluation process used data that were  
159 not seen by the learning process at all.

160 All classification models employed in the noise profiling task are briefly described below.

### 161 *Naive Bayes (NB) (sklearn.naive\_bayes.GaussianNB module)*

162 A posteriori probability was calculated using the following formula:

$$163 \quad P(C_k | \mathbf{X}) = \frac{P(C_k)P(\mathbf{X}|C_k)}{P(\mathbf{X})} \quad (5)$$



164 where  $\mathbf{X}$  represents the vector with  $n$  conditionally independent features  $X_1, X_2, \dots, X_n$ , and  $C_k$  is a  
165 possible outcome class.

### 166 ***Linear Support Vector Machines (SVM) (sklearn.svm.SVC module)***

167 A kernel used to train linear SVM takes the following form:

$$168 \quad K(\mathbf{x}_i, \mathbf{x}_j) = \phi(\mathbf{x}_i)^T \phi(\mathbf{x}_j) \quad (6)$$

169 where  $\phi$  is a function that maps training data into higher dimensional space,  $\mathbf{x}_i, \mathbf{x}_j \in R^n$ . The  
170 following parameters of linear SVM were implemented: regularization  $C=0.025$ , probability  
171 estimates have been enabled, and tolerance for stopping criterion is equal to 0.001.

### 172 ***SVM with polynomial kernel (sklearn.svm.SVC)***

173 The following parameters of the polynomial SVM were implemented: regularization parameter  $C =$   
174 1, gamma coefficient ( $\gamma$ ) set to auto (which means that it uses  $1/\text{number\_features}$ ), probability  
175 estimates were enabled, independent term in kernel function equals 0, tolerance for stopping criterion  
176 is equal to 0.001.

### 177 ***Gaussian process classifiers (GPCs) (sklearn.gaussian\_process.GaussianProcessClassifier*** 178 ***module)***

179 In our test, the exponential kernel was used – it takes one base kernel and a scalar parameter and  
180 combines them via:

$$181 \quad k_{exp}(\mathbf{X}, \mathbf{Y}) = k(\mathbf{X}, \mathbf{Y})^p \quad (8)$$

182 In this study, the exponent is equal to 2. As a source kernel, a Rational Quadratic kernel was used. It  
183 is parameterized by the length scale parameter and a scale mixture parameter. The kernel is given by:

$$184 \quad K(\mathbf{x}_i, \mathbf{x}_j) = \left(1 + \frac{(\mathbf{x}_i - \mathbf{x}_j)^2}{2\alpha l^2}\right)^{-\alpha} \quad (9)$$

185 where  $\mathbf{x}_i$  and  $\mathbf{x}_j$  are vectors of features computed from training or test samples,  $\alpha > 0$  is the scale  
186 mixture parameter,  $l > 0$  is the length scale of the kernel.

187 The L-BFGS-B (a limited memory Broyden–Fletcher–Goldfarb–Shanno) algorithm is used in the  
188 context of finding a (local) minimum of an objective function.

189 ***Decision Tree (DT) (sklearn.tree.DecisionTreeClassifier module)***

190 The parameters used in this test are as follows: the quality of the split is Gini impurity, maximum  
191 depth of the tree is 5.

192 ***Random Forest (sklearn.ensemble.RandomForestClassifier module)***

193 Parameters used in this research: the quality of the split is Gini impurity, the maximum depth of the  
194 tree is 5, number of estimators (trees in the forest) is set to 10.

195 ***Multilayer Perceptron (MLP) Classifier (sklearn.neural\_network.MLPClassifier module)***

196 The following parameters of the MLP classifier were used: L2 regularization parameter (alpha) is set  
197 to 1, and the maximum number of iterations equals 1000. The hidden layer contains 100 neurons, and  
198 the activation function is ReLU. The optimizer used for weight is Adam optimization, which refers to  
199 the stochastic gradient descent optimizer (Pedregosa et al., 2011).

200 ***AdaBoost classifier (sklearn.ensemble.AdaBoostClassifier module)***

201 In this study, the following parameters were used: the maximum number of estimates at which  
202 boosting is stopped equals 50, the learning rate equals 1, and SAMME.R is used as the boosting  
203 algorithm.

204 ***Quadratic Discriminant Analysis***

205 ***(sklearn.discriminant\_analysis.QuadraticDiscriminantAnalysis module)***

206 The quadratic Discriminant Analysis classifier is based on the Bayes rule presented above in the  
207 description of the Naïve Bayes classifier (see Eq. 5). If there is an assumption that the covariance  
208 matrices are diagonal, then the input features are assumed independent - the resulting classifier is then  
209 equivalent to Naïve Bayes. For our test, the regularization parameter is set to 0.

### 210 III. COMPARISON OF THE CLASSIFIER RESULTS

211 The classification results are provided in the form of overall accuracy and a confusion matrix, allowing  
212 for a straightforward interpretation of the results. For the multiclass classification problems, the  
213 following metrics have been used (Grandini et al., 2020):

- 214 – overall accuracy for the whole prediction process,
- 215 – precision, recall, and F1-score for every class.

216 The F1 metric was used because, in our classification procedure, both false positives and false  
217 negatives are equally undesirable, which is best reflected by F1 (Lipton et al., 2014). The dataset used  
218 in our study is well-balanced; therefore, AUC ROC has been chosen as it suits balanced datasets  
219 (Huang and Ling, 2005).

220 To calculate these metrics, the following prediction results need to be obtained:

- 221 –  $TP_n$  – the number of true positive recognitions for distortion type  $n$  (e.g., subway),
- 222 –  $TN_n$  – the number of true negative recognitions for distortion type  $n$ ,
- 223 –  $FP_n$  – the number of false positive recognitions for distortion type  $n$  (in other words – the number  
224 of samples recognized incorrectly as type  $n$ ),
- 225 –  $FN_n$  – the number of false negative recognitions for distortion type  $n$  (in other words – the number  
226 of  $n$  distortion samples recognized as something different than type  $n$ ).

227 The overall accuracy can be measured only using the full recognition results. For the multiclass  
228 classification problem, the formula is as follows:

$$229 \text{Acc} = \sum_n \frac{TP_n}{N} \quad (10)$$

230 In other words – it is a sum of true positives for all distortion types divided by the number of samples  
231 being recognized. The typical definition of two-class accuracy has the sum of true positives and true

negatives in the denominator of the equation. Still, it is the same as the sum of all true positives if both classes are treated as being detected.

Precision for type  $n$  is defined as follows:

$$Precision_n = \frac{TP_n}{TP_n + FP_n} \quad (11)$$

Recall for type  $n$  is defined as follows:

$$Recall_n = \frac{TP_n}{TP_n + FN_n} \quad (12)$$

F1-score for type  $n$  is defined as follows:

$$F1score_n = 2 \cdot \frac{Precision_n * Recall_n}{Precision_n + Recall_n} \quad (13)$$

Tables I-III show the comparison of the above-described classification models. Also, metrics such as  $P$  – precision,  $R$  – recall,  $F1$  – F1-score, and  $S$  – support are included. The pair of the best accuracy and ROC AUC (area under the receiver operating characteristic curve) achieved – is highlighted in bold. Moreover, recognition time for all models is included as well. Values of recognition time for all models are calculated as a time used for classifying all 340 testing samples.

TABLE I. Results of the classification using Naïve Bayes, Linear SVM, and SVM polynomial classification models.  $P$  – precision,  $R$  – recall,  $F1$  – F1-score,  $S$  – support.

	Naïve Bayes	Linear SVM	SVM polynomial
Accuracy	96.76%	96.17%	94.41%
ROC AUC	0.99	0.99	0.99
Recognition time	0.67 ms	1.56 ms	1.29 ms

Noise distortions	P	R	F1	S	P	R	F1	S	P	R	F1	S
Airport	1.00	0.96	0.98	45	0.86	0.96	0.91	45	0.84	0.91	0.87	45
Babble speech	0.90	0.90	0.90	39	1.00	0.95	0.97	39	0.90	0.95	0.93	39
Car	1.00	1.00	1.00	46	0.96	1.00	0.98	46	1.00	0.93	0.97	46
Exhibition	0.98	1.00	0.99	39	1.00	1.00	1.00	39	1.00	1.00	1.00	39
Pink noise	1.00	1.00	1.00	17	1.00	1.00	1.00	17	1.00	1.00	1.00	17
Restaurant	1.00	0.91	0.95	32	1.00	1.00	1.00	32	0.94	1.00	0.97	32
Street noise	0.92	0.98	0.95	48	0.91	0.81	0.86	48	0.88	0.79	0.84	48
Subway	1.00	0.97	0.98	32	1.00	1.00	1.00	32	1.00	1.00	1.00	32
Train	0.95	1.00	0.98	42	1.00	1.00	1.00	42	1.00	1.00	1.00	42

247  
248 TABLE II. Results of classification using Gaussian process, Decision tree, and Random forest  
249 classification models. All denotations are as shown in TABLE I.

	GPC	Decision tree	Random forest
Accuracy	85.88%	95.59%	92.94%
ROC AUC	0.98	0.98	0.99

Recognition time	45 ms				0.25 ms				1.66 ms			
Noise distortions	P	R	F1	S	P	R	F1	S	P	R	F1	S
Airport	0.83	0.89	0.86	45	0.94	0.98	0.96	45	0.98	0.98	0.98	45
Babble speech	0.78	0.97	0.86	39	0.97	0.77	0.86	39	0.87	1.00	0.93	39
Car	0.93	0.89	0.91	46	1.00	1.00	1.00	46	0.98	1.00	0.99	46
Exhibition	0.80	0.95	0.87	39	0.98	1.00	0.99	39	0.98	1.00	0.99	39
Pink noise	0.89	1.00	0.94	17	1.00	0.88	0.94	17	1.00	0.94	0.97	17
Restaurant	0.88	0.94	0.91	32	0.84	0.97	0.90	32	0.84	0.97	0.90	32
Street noise	0.94	0.63	0.75	48	0.92	0.98	0.95	48	1.00	0.58	0.74	48
Subway	0.92	0.72	0.81	32	1.00	0.97	0.98	32	1.00	0.97	0.98	32
Train	0.84	0.86	0.85	42	1.00	1.00	1.00	42	0.82	1.00	0.90	42

250

251 TABLE III. Results of the classification using MLP, AdaBoost, and QDA classification models. All

252 denotations are as shown in TABLE I.

	MLP	AdaBoost	QDA
Accuracy	67.05%	67.64%	93.52%

ROC AUC	0.95				0.95				0.94			
Recognition time	0.49 ms				15.66 ms				0.72 ms			
Noise distortions	P	R	F1	S	P	R	F1	S	P	R	F1	S
Airport	0.75	0.40	0.52	45	0.48	0.96	0.64	45	0.72	0.96	0.82	45
Babble speech	0.74	0.74	0.74	39	0.51	0.92	0.65	39	0.98	1.00	0.99	39
Car	0.85	0.85	0.85	46	1.00	0.98	0.99	46	0.94	1.00	0.97	46
Exhibition	1.00	0.33	0.50	39	1.00	1.00	1.00	39	1.00	1.00	1.00	39
Pink noise	0.55	0.94	0.70	17	0.00	0.00	0.00	17	0.00	0.00	0.00	17
Restaurant	0.59	1.00	0.74	32	0.00	0.00	0.00	32	1.00	1.00	1.00	32
Street noise	0.40	0.33	0.36	48	0.00	0.00	0.00	48	0.98	0.94	0.96	48
Subway	0.54	1.00	0.70	32	1.00	1.00	1.00	32	1.00	1.00	1.00	32
Train	0.92	0.79	0.85	42	0.56	0.83	0.67	42	1.00	1.00	1.00	42

253

254 One can notice that most tested algorithms give sufficiently good results with an accuracy of over  
 255 90%; however, only three have better accuracy than 96%, i.e., Naïve Bayer, Linear SVM, and Decision  
 256 Tree. For all three algorithms, all other metrics (averages of precision, recall, and F1 for all noise types)  
 257 are similar; however, Naïve Bayer is a little better than Linear SVM and Decision Tree. The

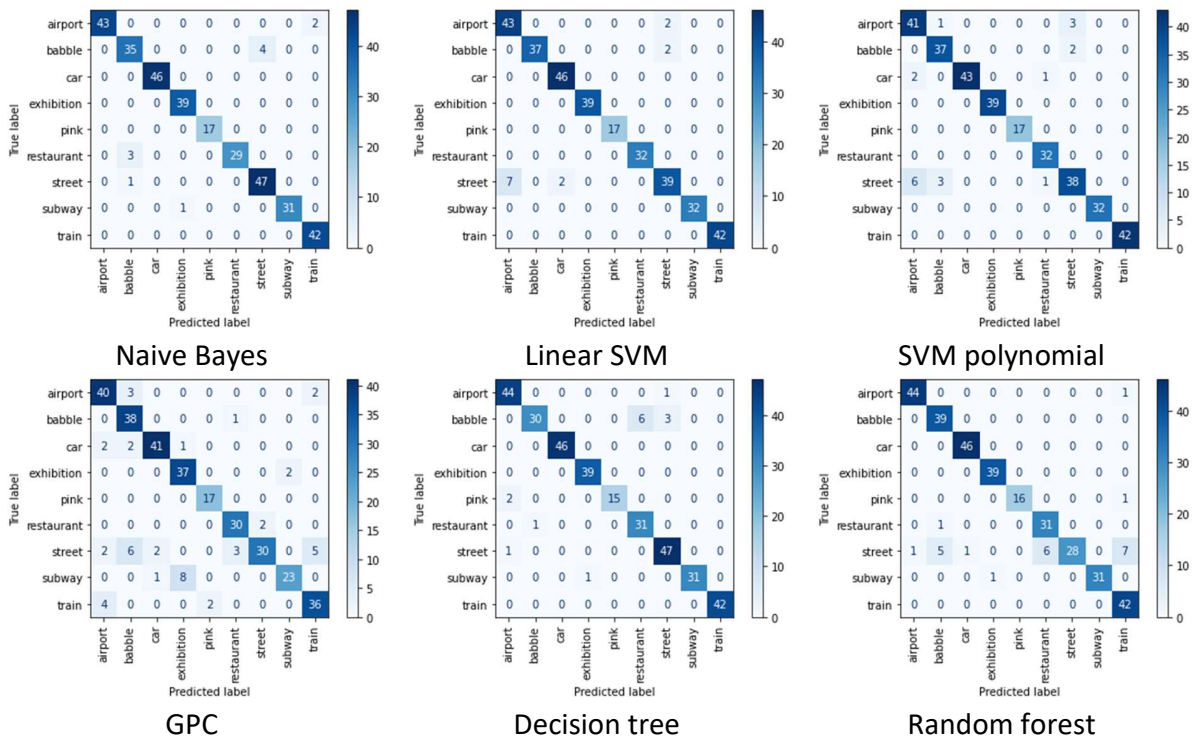
258 computational complexity for inference for all these methods is also similar and linearly dependent on  
 259 the number of dimensions (for Linear SVM and Naive Bayer) or the number of tree depths for the  
 260 Decision Tree.

261 The other algorithms are not as accurate as the three mentioned above. Some of them have no true  
 262 positives for some noise types, which results in zeroing the basic metrics for these types. This can be  
 263 observed in Figure 1 (e.g., pink noise recognition for the AdaBoost classifier). That is why these  
 264 algorithms have been disqualified, i.e., MLP, AdaBoost, and Quadratic Discriminant Analysis.  
 265 Moreover, since these times are of a millisecond level, we can assume that near-real-time recognition  
 266 is possible with the assumption that the initial 1-second recognition has already passed.

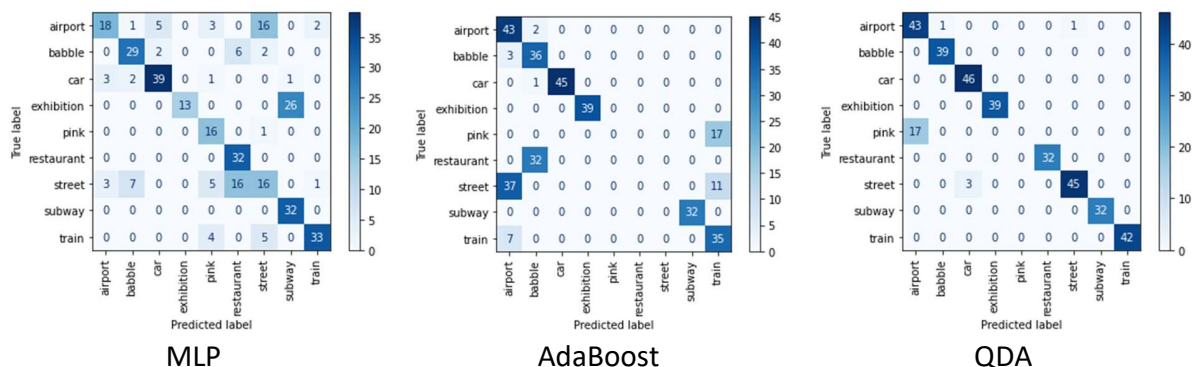
267 Considering the above results, we have selected the Naïve Bayes model as a source model for the  
 268 subsequent experiments.

269

270 In Figure 1, confusion matrices are presented that were prepared for all tested models.







271 FIG. 1. Confusion matrices for all tested models.

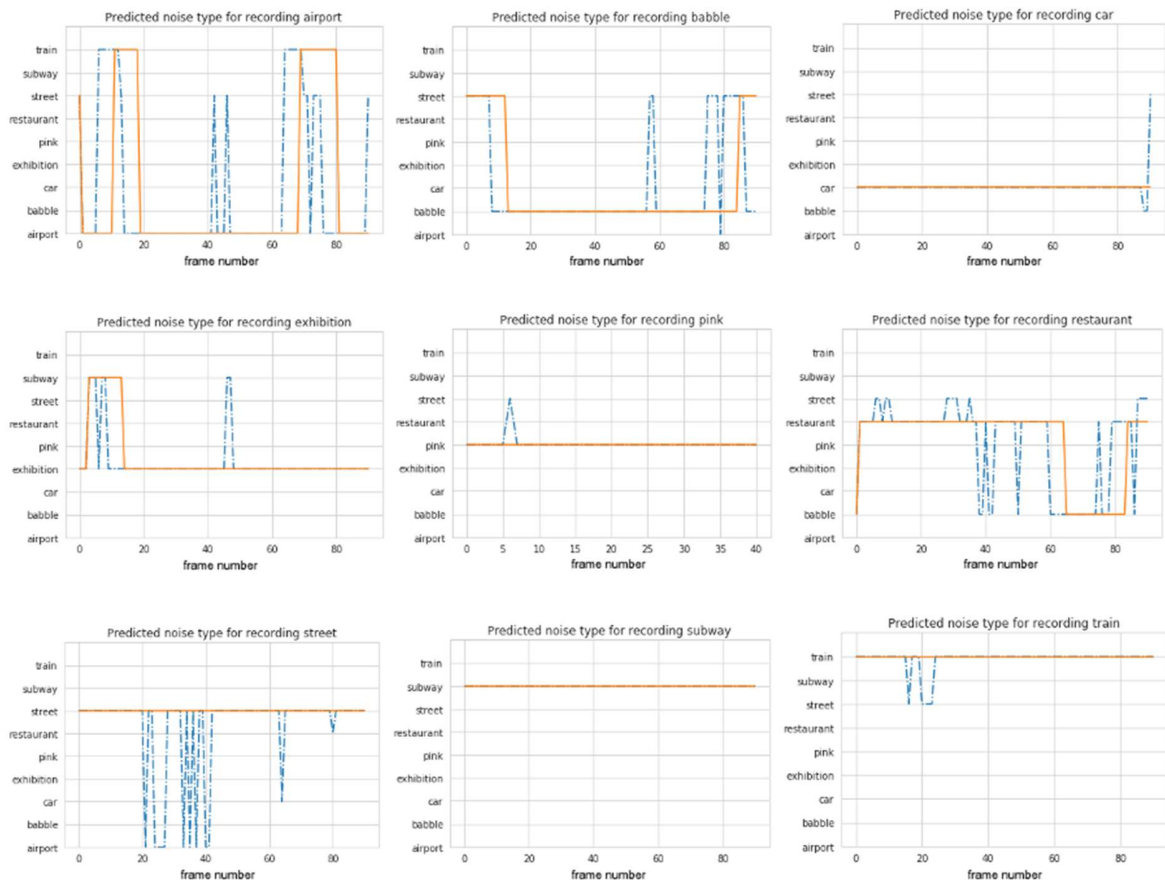
272 **IV. DISCUSSION**

273 The created model using the Naïve Bayes classification was tested on recordings that were used for  
 274 training (but different parts of these recordings) and on the additional recordings from the multimodal  
 275 corpus of English speech recordings called MODALITY (Czyzewski et al., 2017). As mentioned  
 276 before, in the context of noise profiling, the model's usefulness is measured by evaluating its stability,  
 277 understood as a classification consistency over a longer period of time, not correctness – presumed  
 278 as class-level accuracy. This is because the recording conditions might be very different - such as the  
 279 recording method and equipment, source of noise, and its characteristics. Therefore, for instance, the  
 280 airport recording might be identified as street noise. What matters here is that this recording is always  
 281 (or almost always) identified as street noise. That is why the correctness of classification is of less  
 282 importance in general. The value of this model is in recognizing the abstract type of distortion using  
 283 its frequency parameters – and this is the basis of improving speech intelligibility in the presence of  
 284 noise. The process of speech quality/intelligibility enhancement requires particular conditioning – and  
 285 the values of the parameters used should correspond to the type of noise. These values strongly impact  
 286 the efficiency of speech intelligibility improvement. So, it is crucial to effectively classify the particular  
 287 types of distortion to an assigned number of classes, enabling to modify the speech in the best way in  
 288 given noise conditions.

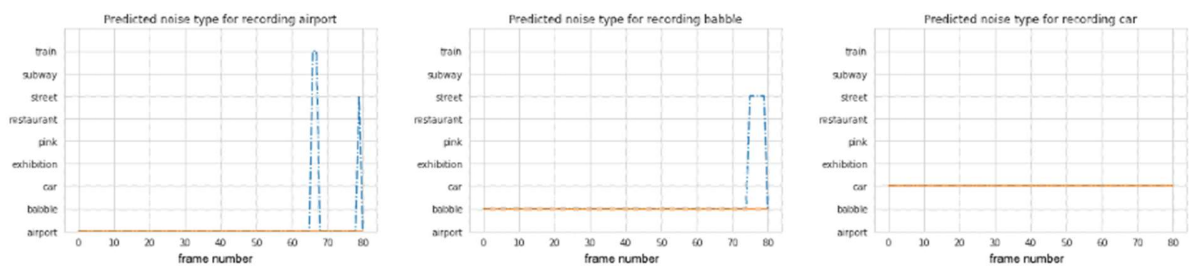
289 The recognition process was carried out in two modes: momentary and averaging. In both modes, the  
290 window/frame analyzed was 1 or 2 seconds, and the window was moved by 0.1 seconds with every  
291 step. In the momentary mode, classification was performed for every frame. In the averaging mode,  
292 the classification was made with delay - it means that the momentary classification should change  
293 across five consecutive frames to calculate the average classification. However, it does not mean that  
294 the results should be considered valid if and only if the five consecutive frames will occur. What is  
295 more, the 1-second frame does not necessarily have to be an uninterrupted fragment. It only means  
296 that the system should wait a little longer for the first recognition.

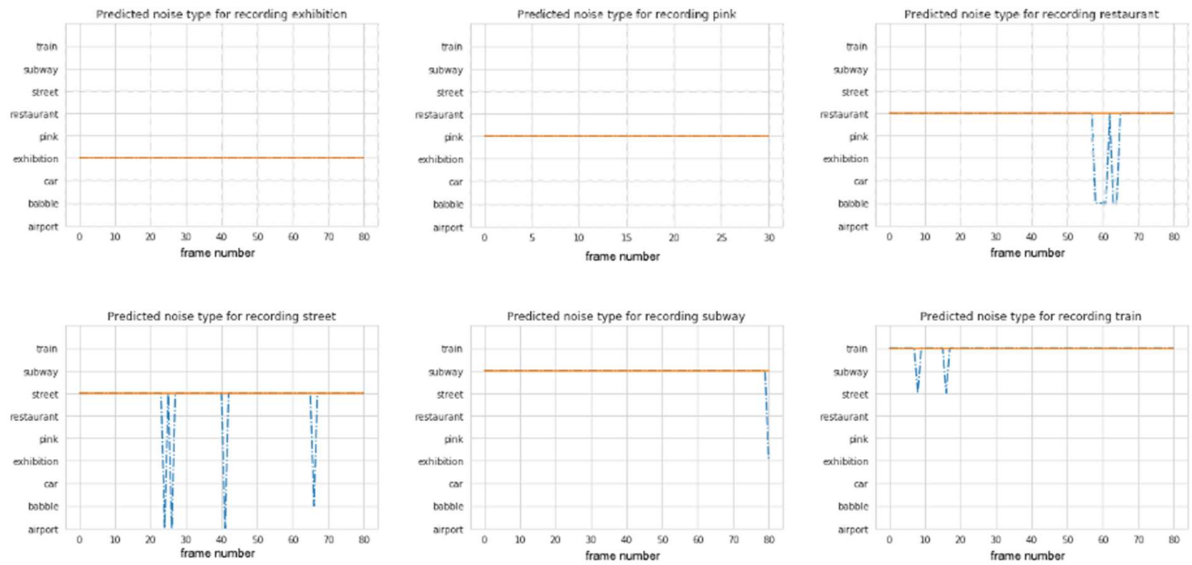
297 Thanks to this procedure, the recognition model avoids a temporary disturbance, usually caused by  
298 non-stationary noise.

299 Figures 2 and 3 present the outcomes of classification. The solid line represents the classification in  
300 the averaging mode, while the dashed line represents the momentary classification. The classification  
301 results for 1- and 2-second frames are different – first of all, it is because the learning process was  
302 performed using a 2-second frame; what is more, a longer window allows for better evaluation of the  
303 statistical features of the frequency characteristics. When using 2-second windows, the classification  
304 results are very good. For a 1-second window, the statistical characteristics might not be clearly visible,  
305 but the averaging mode provides satisfying results.



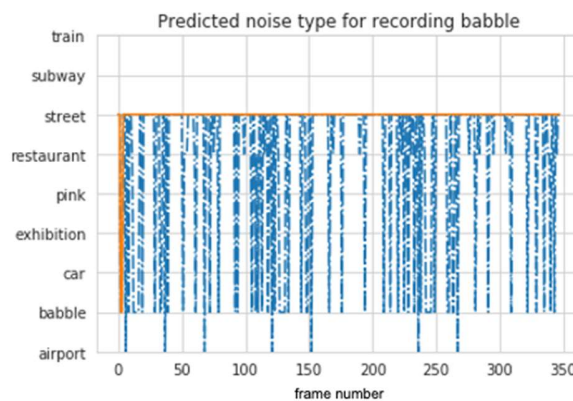
306 FIG. 2. Classification results on the real-life recordings using a 1-second-length frame (dashed line –  
 307 result from momentary mode, solid line – result from averaging model).





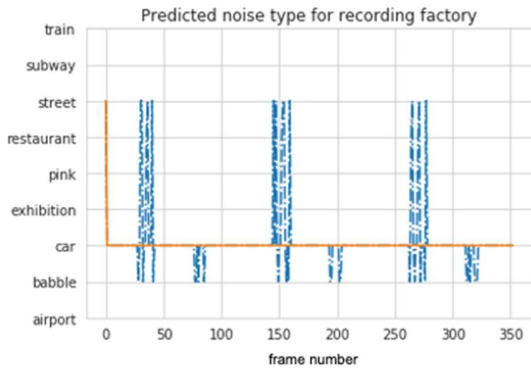
308 FIG. 3. Classification results on the real-life recordings using a 2-second-length frame (dashed line –  
 309 result from momentary mode, solid line – result from averaging model).

310 The recognition process was also performed on a completely different set of noise recordings  
 311 contained in the MODALITY multimodal corpus of English speech recordings (Rasmussen et al.,  
 312 2006). The recordings used in this test were very long (between 11 minutes 45 seconds and 14 minutes  
 313 54 seconds). The test was performed only for a 2-second frame, and the window was moved by 2  
 314 seconds (due to the overall recording length) with every step. The averaging was also used to remove  
 315 random fluctuations in the recognition results. Figures 4-6 present recognition results, where dashed  
 316 lines represent the single window classification and the solid line depicts the averaged result.



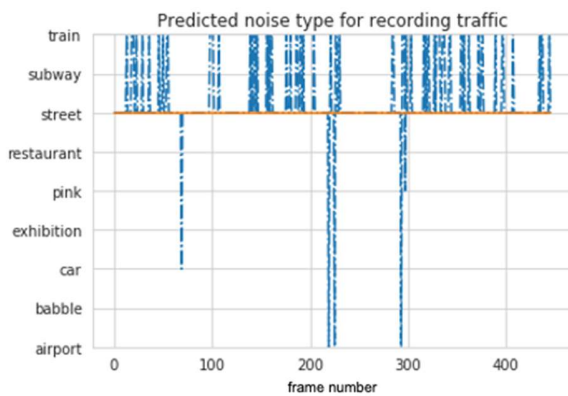
317

318 FIG. 4. An example where the classification model has selected both “street” and “babble speech,”  
319 but after averaging, the resulting classification was “street.”



320

321 FIG. 5. An example where the “factory” recording was classified as “car noise” (there was no such  
322 class as “factory” in the training set).



323

324 FIG. 6. An example where the recording “traffic” was classified as “street,” which is the correct  
325 classification.

326 As pointed out, it must be underlined that the classification quality is impacted by the stability of the  
327 classification, not correctness. That is why the results are generally satisfying, even if the noise  
328 recordings are not always correctly classified. As previously mentioned, the classification would  
329 strongly be impacted by the recording place, recording equipment, sampling frequency, etc.

## 330 V. CONCLUSIONS

331 In this study, an efficient method of noise profiling was presented, understood as critical to identify  
332 the sound characteristics specific to a given type of sound. It was demonstrated that stable and  
333 predictable noise profiling is possible using noise spectral characteristics. These characteristics can be  
334 calculated almost in real time so that noise profiling can be fast and efficient. The stability, however,  
335 depends on the length of the frame and the number of frames used in the averaging process. It may  
336 mean that the noise profiling process is delayed up to 2-3 seconds), but it can strongly be decreased  
337 after a couple of initial seconds of a signal. This means that the presented method can efficiently be  
338 used when trying to improve speech quality and intelligibility when the speech is played back in noisy  
339 conditions. The experiments, however, assumed that noise was separated from the speech signal. This  
340 can be extended to situations where speech is recorded with noise by separating both signals and  
341 processing them in separate flows, which could be the next step in improving the overall speech  
342 intelligibility improvement model.

343 Overall, the proposed method is fast and stable so that it can be used in near real-time systems. The  
344 algorithmic simplicity of the machine learning models employed results in low computational  
345 complexity while classifying the recorded noise, thus allowing for obtaining low inference times. Even  
346 though the classification is not binary, and the number of classes is quite large, a relatively simple  
347 model using spectral measures provides high accuracy. This allows for building applications on top of  
348 the model proposed.

349 In future research, we plan to use noise profiling along with the P.563 objective metric ITU-T  
350 Recommendation P. 563 (2004) as an input to the feedback system in classical reinforcement learning.  
351 We will follow the methodology in which predictors are trained on human quality ratings (Reddy et  
352 al., 2021) but use the reward derived from the Reinforcement Learning (RL) paradigm. This is because  
353 RL refers to learning by interacting with the environment (Sutton and Barto, 2018).

354 Indeed, our focus will be on the speed of stable recognition in our future research. Following our  
355 experiments, future research should also be directed to reducing the time needed for noise profiling  
356 and trying to use this approach in noise suppression systems.

357

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