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Modeling the Acoustic Channel of Voice Information Leakage

Anatoliy Oleynikov
ORCID 0000-0002-4458-8833
dept. of Computer Radio Engineering and Technical Information Security Systems
Kharkiv National University of Radio Electronics
Kharkiv, Ukraine
anatoly.oleynikov@nure.ua

Oleksii Bilotserkivets
ORCID 0000-0002-8514-9650
dept. Microprocessor Technologies and Systems
Kharkiv National University of Radio Electronics
Kharkiv, Ukraine
oleksii.bilotserkivets@nure.ua

Oleksandr Shirokyi
dept. of Computer Radio Engineering and Technical Information Security Systems
Kharkiv National University of Radio Electronics
Kharkiv, Ukraine
d_ref@nure.ua

***Abstract*—The article analyses the modelling of an acoustic channel for voice information leakage. The model consists of 5 parts that are coordinated with each other by mathematical expressions. The results of experimental studies are presented. The proposed model can be used as a basis for software for modelling the acoustic channel of voice information leakage.**

***Keywords*—modeling, channel, voice, information, leakage**

I. Introduction

The method of instrumental control doesn't allow to estimate a priori information security from leakage through the acoustic channel. It may be required for example at the stage of designing of a complex protection system. This possibility can be provided by creating a mathematical model of acoustic information leakage channel.

The model is a consistent set of mathematical expressions that allows calculating the intelligibility of the speech signal at the output of the technical means of intelligence (TMI), depending on a given set of spatial and energy characteristics of the acoustic information leakage channel. It can be decomposed into 5 models: speech signal source (SSS), acoustic interference sources (AIS), propagation environment (PE), TMI, and speech intelligibility assessment (SIA).

II. Mathematical Expression of Models

The formant method, also known as Articulation Index (AI), is used for intelligibility estimation, in which the articulation (formant) intelligibility is the sum of the intelligibility of individual bands into which the spectrum of the speech signal is divided (1).

$$R = \sum_{i=1}^{n} k_i \cdot r_i(Q_i) \tag{1}$$

where i - number of frequency bands, n - their number, k_i - probability coefficients of formants presence in bands, $r_i(Q_i)$ - coefficient of formants perception, depending on signal/noise level, corrected for energy redundancy of speech spectrum $Q_i = q_i - \Delta A_i$. Syllabic and verbal intelligibility are related to formant by known relationships.

Thus, the input data for the model is an array of signal to noise ratios to be derived from the TMI model output.

The partitioning of the spectrum into bands is arbitrary, but in practice 20 equal-articulation bands with equal k_i, or 21 third-octave bands, and in a simplified version 7 or 5 octave bands are used [1].

The input parameters for the TMI are the speech and noise signal levels at the point of reconnaissance contact for the frequency bands (FB), taking into account the arrival directions. The latter point is important if the TMI has directional properties. In such a case, the directional characteristic (DC) will be the mathematical relationship describing it. For example, for a reflector microphone it is approximated by the expression (2).

$$R(\Theta) = \frac{2 * J_1(\psi)}{\psi}, \text{where } \psi = \frac{2 * \pi}{\lambda} \rho_0 * sin\Theta \tag{2}$$

and $J_1(\psi)$ is a Bessel function of kind 1.

In an analogy to antennas, the concept of the coefficient of protective action (CPO) is introduced, which is the ratio of the sensitivity in the direction of interference arrival to the axial sensitivity of a narrow-field microphone. In the case of isotropic interference, an integral index, the directivity index, is applied.

The propagation environment model must convert the arrays of speech and noise signal levels in the FB from their sources to the corresponding levels at the point of reconnaissance contact, taking into account the spatial conditions.

According to the interstate standard, sound propagation attenuation in the terrain (3):

$$A = A_{div} + A_{atm} + A_{gr} + A_{bar} + A_{misc} \tag{3}$$

where A_{div} is attenuation due to geometric divergence, A_{atm} is attenuation due to atmospheric sound absorption (neglected at small distances), A_{gr} is attenuation due to ground effect, A_{bar} is attenuation due to shielding, A_{misc} is attenuation due to other effects (e.g., foliage spread). In the simplest case, only the first summand is retained (4):

$$A_{div} = \left[20\lg(\frac{d}{d_0} + 11)\right] \quad (4)$$

where d is the distance from the noise source to the receiver, m; d_0 is the reference distance (RD) (d0=1m). Constant 11 relates the sound power level of a non-directional point source to the sound pressure level at the RD.

If the source and receiver are located in adjacent rooms and the acoustic signal penetrates the building envelope (BE), the signal levels at the design point are determined by the formula (5):

$$L = L2м - R + 10lg\,S - 10lg\,B - 10lg\,k, \quad (5)$$

where R - coefficient of sound insulation of BE, dB; S - area of BE, m2; B - acoustic constant of the room with TMI; k - coefficient taking into account the sound field diffusivity disturbance; $L2м$ - sound pressure level in the room with the source at 2 m from BE, dB, determined by the formula (6):

$$L = L_w + 10lg\left(\frac{\chi\Phi}{\Omega r^2} + \frac{4}{kB}\right) \quad (6)$$

where L_w - source power level, dB; r - distance from acoustic centre of source, m; χ - coefficient taking into account influence of near field; Φ - directivity factor; Ω - spatial angle of source radiation, rad.

In addition to the above relationships, the PE model must account for other possible reconnaissance contact options to maximize its completeness.

The acoustic source (AS) model must convert a given power level or sound pressure level at a reference distance into an array of sound levels in the FB.

For a speech signal source, the spectral levels are calculated as follows (7):

$$Lsi = Ls + Vi\ [дБ], \quad (7)$$

where Ls is integral level of acoustic signal at distance of 1 m; Vi is weight energy coefficient of the i-th band, in dB it has negative sign; $fagi$ is average geometric frequency of the i-th band.

To model acoustic noise sources (ANS), some types of coloured noise and noise with a spectrum close to speech should be considered [2]. In order to mask speech, a "speech chorus" or near-spectrum pink noise, whose levels in the FB are often used (8):

$$L_{pn} = 10 \cdot log\left[g_{pn} \cdot (ln(f_{ui}) - \ln(f_{di}))\right], [dB] \quad (8)$$

where $g_{pn} = \frac{10^{0,1 \cdot L_{П}}}{\ln\left(\frac{1}{f_{d_1}}\right) - \left(\frac{1}{f_{u_n}}\right)}$,

where $L_{П}$ is the integral level of the noise source.

The directional coefficient can also be taken into account, showing how much the equivalent sound pressure level in a given direction differs from the sound pressure level of a non-directional source with the same sound power level [3].

III. Results of a Partial Model Implementation

The fig. 1 and fig. 2 shows the results of a partial model implementation for free space in a single plane, representing the verbal intelligibility coefficients for each point in 1 m steps, converted for clarity into greyscale luminance levels.

As TMI was chosen narrow-directional microphone of "linear group" type with the following characteristics: number of microphones n=20, step d=0.05 m, DC of a single microphone - cardioid, directional index of TMI at frequency 1000 Hz - 20 dB. The level of the speech signal at the RD is 60 dB, the level of isotropic noise is 40 dB, the noise levels at the RD from point sources is 80 dB.

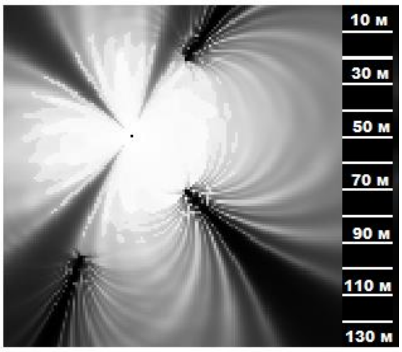

Fig. 1. 3 point noise sources.

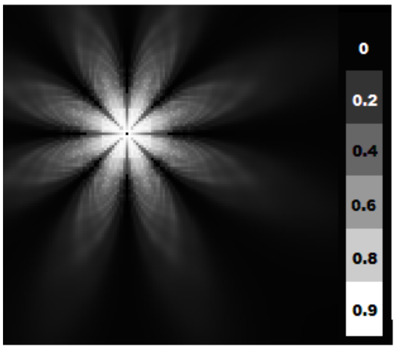

Fig. 2. 8 noise sources surrounding the speech source.

IV. Conclusion

The mathematical model considered can form the basis of software that will greatly simplify and speed up the procedure for modelling the acoustic leakage channel.

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