Developments in Modelling and Measuring Ground Impedance

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Models for representing the acoustical properties of the ground surface are reviewed in the light of recent data. The distinct influences of surface roughness are described. A full description requires information about the mean height and spacing of surface roughness as well as porosity, tortuosity, connected pore geometry, flow resistivity and layering. There is a trade-off between simplicity and accuracy. Progress towards in situ methods for ground characterization is outlined. The various alternative approaches involve measurement of short range propagation, level difference spectra templates or direct deduction of complex surface impedance.

MODELS FOR GROUND IMPEDANCE

Atmospheric sound propagation close to the ground is sensitive to the acoustical properties of the ground surface as well as to meteorological conditions. Surface porosity allows sound to penetrate and hence to be both absorbed and undergo phase change through friction and thermal exchanges. There is interference between sound traveling directly between source and receiver and sound reflected from the ground. This interference is known as ground effect [1,2]. Over porous surfaces, enhancement tends to occur at low frequencies since the longer wavelengths are less able it is to penetrate the pores. The presence of vegetation tends to make the surface layer of ground including the root zone more porous. Outdoor surfaces are likely to be rough as well as porous. Roughness effects have been modelled numerically by the Boundary Element Method and analytically by boss theory [3]. The influence of roughness that is small compared with the incident wavelengths may be represented through effective impedance [3,4].

Various models and parameters have been used to calculate the impedance of ‘smooth’ ground surfaces. The most important characteristic of a ground surface that affects its acoustical character is its flow resistivity or air permeability. A widely used model for the acoustical properties of outdoor surfaces involves a single parameter, the ‘effective’ flow resistivity, to characterise the ground.

This single-parameter model [5], describes the propagation constant, \( k \) and normalised surface impedance, \( Z \) in terms of a single adjustable parameter known as the effective flow resistivity, \( \sigma_e \). The propagation constant and normalised surface impedance are given by

\[
\frac{k}{k_i} = 1 + 0.0978(f/\sigma)^{0.780} + 0.189(f/\sigma)^{0.505} \quad (1a)
\]

\[
Z = \frac{\rho c_i}{\rho e} = 1 + 0.0571(f/\sigma)^{0.771} + 0.087(f/\sigma)^{0.772} \quad (1b)
\]

Harmonic time dependence, \( e^{-i\omega t} \), is understood.

There is considerable evidence that (1a) tends to overestimate the attenuation within a porous material with high flow resistivity. More sophisticated theoretical models for the acoustical properties of rigid-porous materials introduce porosity, the tortuosity (or ‘twistiness’) of the pores, factors related to pore shapes and sizes, and multiple layering [6-10].

A model based on an exponential change in porosity with depth [7,9] has enabled better agreement with measured data for the acoustical properties of many outdoor ground surfaces than (1). The two adjustable parameters are the effective flow resistivity \( (R_e) \), which differs from \( \sigma_e \), and the effective rate of change of porosity with depth \( (\alpha_e) \). The impedance of the ground surface is predicted by

\[
Z = \frac{1 + i}{\sqrt{\rho \gamma}} \frac{R_e}{\sqrt{f}} + \frac{i \gamma \alpha_e}{8 \pi \gamma}, \quad (2)
\]

where \( \alpha_e = (n' + 2)\alpha \Omega \) and \( n' \) is a grain shape factor such that the tortuosity is given by \( \Omega^{n'} \).
A boss model approach has been used to deal with coherent (in phase) and incoherent scatter from a rough surface [3]. As long as the sound wavelength is significantly larger than mean height and spacing of the roughness, it predicts that the roughness of the surface of an acoustically-hard material produces a non-zero effective admittance. For grazing incidence on a hard surface containing randomly distributed 2-D roughness normal to the roughness axes, the effective admittance may be written

$$\beta = \frac{3V^{1/2}k^2b}{2} \left[ 1 + \frac{\delta^2}{2} \right] - i \nu \delta (\delta - 1)$$  \hspace{1cm} (3)

where \( V \) is the roughness volume per unit area of surface (equivalent to mean roughness height), \( b \) is the mean center-to-center spacing, \( \delta \) is an interaction factor depending on the roughness shape and packing density and \( k \) is the wave number.

The same approach can be extended to give the effective normalised surface admittance of a porous surface containing 2-D roughness [4,11]. In general, the effective admittance is predicted to be a function of the observer geometry. A tolerably successful prediction of the effective normalised impedance of a rough porous surface is that of the smooth surface but with a reduced real part.

MEASUREMENT OF GROUND IMPEDANCE

Measurements of the magnitude of the excess attenuation from a point source at non-grazing incidence may be inverted, by means of least-squares or template fitting to impedance models and yield impedance as a function of frequency.

As well as not requiring the assumption of plane waves, which is valid only at sufficient source height and near normal incidence, the short-range propagation method also includes effects of small-scale surface roughness.

Fig. 1 shows measurements obtained over uncultivated grassland, using a direct impedance-fitting method [12,13] Note that the impedance, deduced from measured complex excess attenuation spectra, tends to zero above 3 kHz.

Roughness i.e. incoherent scatter, may explain these measurements.

![Figure 1. Normalised impedance data (open circles) obtained over established grassland. Solid lines represent predictions including roughness. Dotted lines are predictions without roughness.](image)

REFERENCES

Sound propagation above an impedance discontinuity in the presence of meteorological effects, using a BEM formulation

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Using the same approach as in Meteo-BEM [1, 2], a new model is derived for describing outdoor sound propagation above an impedance discontinuity in refracting conditions. It is based on a Boundary Integral Equation formulation including ground and meteorological effects in the Green’s function. The methodology is presented. Results are given, showing that this approach can be used for complex outdoor sound propagation prediction.

1. INTRODUCTION

Complex traffic noise problems involve sound propagation from low height sources above impedance discontinuities in the presence of meteorological effects. A number of studies have been presented in the past in order to develop models for predicting the effect of mixed impedance on sound propagation over a flat ground [3]. Most of these works have dealt with a quiescent atmosphere, except for a few models based on the Parabolic Equation approach [4] that can take refraction effects into account. The aim of this paper is to use the Boundary Integral Equation theory (BIE) to investigate the problem of outdoor sound propagation in a stratified atmosphere above an impedance discontinuity. Following the same approach as in Meteo-BEM [1, 2] ground and meteorological effects are included in the Green’s function of the BIE. Section 2 recalls briefly the BIE theory used for describing sound propagation above an impedance discontinuity. Then section 3 presents the Green’s function accounting for meteorological effects. Section 4 gives results and finally we conclude and give perspectives.

2. THE BOUNDARY INTEGRAL EQUATION FOR SOUND PROPAGATION OVER AN IMPEDANCE DISCONTINUITY

The BIE theory represents a very powerful tool in order to assess for any kind of shape and absorption of the propagation domain boundaries (in particular uneven terrains, various shapes of sound barriers, impedance discontinuities).
good approximation, which gives correct results and is less computationally expensive, consists of assuming that the acoustic pressure in $\Gamma$ can be approximated by what it would be if the whole boundary had admittance $\beta_1$. This yields the following approximation to $p(S,M)$:

$$p(S,M) = G_{\beta_1}(S,M) + i(\beta_2 - \beta_1)K$$  \hspace{1cm} (2)

with:

$$K = k \int_{\Gamma} G_{\beta_1}(S,P) G_{\beta_1}(M,P) d\Gamma(P)$$  \hspace{1cm} (3)

The integral $K$ can be directly evaluated using for instance the composite midpoint rule or one can follow the improved calculation method proposed in [6]. This approach has already been derived for a quiescent atmosphere but if one uses the appropriate Green’s function presented in section 3, the case of sound propagation above an impedance discontinuity can then be studied in refracting conditions with this formulation.

3. THE GREEN’S FUNCTION ACCOUNTING FOR METEOROLOGICAL EFFECTS

We consider here the case of downward refraction. In the case of a positive constant sound speed profile, we use for the Green’s function of the BIE the Normal Modes solution (linear source), with the same notations as in [7]:

$$p_s(r,z) = \frac{i}{2l} \sum_n \exp(i k_n r) \text{Ai}(\tau_n + z/l) \text{Ai}(\tau_n + z/l)$$

$$- k_n \tau_n [\text{Ai}(\tau_n)]^2 - [\text{Ai}^2(\tau_n)]$$

where $\tau_n = (k_n^2 - k_0^2)^{1/2}$

are the zeros of $\text{Ai}'(\tau_n) + q \text{Ai}(\tau_n) = 0$  \hspace{1cm} (4)

$S(0,z_s)$ denotes the source, $M(r,z)$ is the receiver. $k_n$ represents the horizontal wave number of the $n$th mode.

The Normal Modes solution is particularly attractive for the BIE since the variables involved in the analytical formulation are uncoupled and the derivatives of the Green’s function are straightforward to derive.

4. RESULTS AND CONCLUSION

Figure 2 shows the influence of downward refraction (linear sound speed profile : $c(z) = c_0(1+az)$, with $a = 2.9 \times 10^3$ m$^{-1}$ compared to the case $a = 0$ m$^{-1}$ for a quiescent atmosphere) in the presence of an impedance discontinuity 20 m away from the source ($z_S = z = 1$ m, $f = 500$ Hz, $(\sigma_1,\sigma_2) = (300000 \text{ cgs}, 300 \text{ cgs})$). The results show that the Boundary Integral Equation theory can be adapted to complex atmospheric sound propagation problems. More realistic sound speed profiles are going to be studied and scale model measurements will be undertaken for validation of this new model.

REFERENCES

Meteorological Aspects of Sound Propagation Modeling over Irregular Terrain

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Examples of coupled 3D meteorological and acoustical simulations are presented. Numerical experiments illustrate how topographical and meteorological effects act together. It is shown that topographical effects on the atmosphere cannot be neglected. In particular, the validity of the ‘effective sound speed concept’ is tested, which is frequently used in PE-type propagation models.

INTRODUCTION

The propagation of sound waves is subject to ground effects and atmospheric influences. The atmosphere in turn is influenced by the ground through the exchange of energy, momentum and mass (of water). As a consequence, all parameters that act on the sound as it propagates over irregular terrain are more or less range-dependent. Acoustical simulations over long ranges have to account for these influences in an appropriate way.

In the following two examples are presented: the effect of a noise screen in the presence of wind and the sound propagation around a hill under different meteorological conditions. In either case the acoustical simulation is preceded by the simulation of the atmospheric response to the respective topographical feature.

Different model experiments are performed to demonstrate the atmospheric effects.

EXAMPLE 1: NOISE SCREEN

The 3D simulation deals with a 3 m high noise screen which ends inside a 40 m x 20 m wide model domain. A coherent line source of 250 Hz is placed 5 cm above hard ground, 6.5 m in front of the screen (see Figure 1).

The large-scale wind is assumed to flow parallel to the x-axis with 5 m/s at 10 m above ground. The atmospheric model simulates how the air flows around the obstacle. The Euler-type linearized numerical model [1] of acoustical wave propagation considers the screen and the airflow which is distorted by the screen in all it’s spatial components.

Figure 2 shows how the sound field is influenced by the screen. The 250 Hz tone gives rise to various interference patterns. In order to analyse the efficiency of the screen under different wind situations,
the sound pressure was averaged in y-direction between \(-10 \text{ m} < y < 0\) and \(0 < y < 10 \text{ m}\). The efficiency of the screen near it’s edge is shown in Figure 3. Although the sound level is generally highest (lowest) under the condition of downwind (upwind), the efficiency of the screen is highest under upwind condition between the screen and \(x = 15 \text{ m}\).

**EXAMPLE 2: HILL**

In the following example a meteorological mesoscale model is used to provide three-dimensional high-resolution fields of wind, temperature and humidity near a 50 m high hill for given large-scale meteorological situations. A point source (100-2000 Hz) is located 25 m above the hill top (Figure 4). The ground has a finite impedance. The meteorological fields are taken as input in a ray-based sound propagation model. The coupled models allow a consistent simulation of ground and air effects.

**FIGURE 4.** Schematic vertical cross section (top) and plane view (bottom) of the hill and the flow across it.

**FIGURE 5.** Horizontal distribution of sound level (dB) for a strong-wind situation.

A Lagrange-type sound particle model [2] was used to simulate the propagation of sound outside shadow zones in 3D (Figure 5). Different model experiments were performed in which the meteorological fields were fully considered or partly approximated. The results along the axes shown in Figure 5 reveal the differences (Figure 6).

**CONCLUSIONS**

The meteorological mesoscale simulations show that the wind field is distinctly affected by the topography leading to a strong acceleration above a screen or a hill. Moreover, the flow is diverted around the hill in the case of low Froude number, while the air flows over the hill with corresponding upward and downward motion in the case of high Froude number.

The acoustical simulations suggest that the state of the atmosphere has a significant effect on the sound level and may not be neglected. The ‘effective sound speed’ approach turned out to be an acceptable approximation provided the atmospheric inhomogeneities are considered.

**REFERENCES**

A Study of Range-Dependent Meteorological Conditions and their Influences on Outdoor Sound Propagation

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A series of field trials to enable simultaneous detailed range-dependent meteorological and acoustical measurements are under way. A high-power omni-directional electro-acoustic source is used to provide a broadband sound power of 130dB and the sites allow measurements over well-defined terrains of around 1km. To ensure that sufficient details of the sound propagation are resolved, ten independent computer-based measurement systems are installed at approximately 100m intervals to make time-synchronised measurements. Correlations between the comprehensive propagation and meteorological data obtained so far are presented in this paper.

INTRODUCTION

Salford has recently acquired a meteorological Doppler infrared LIDAR system that is capable of making fast scans of aerosol backscatter and aerosol radial wind velocity from which atmospheric turbulence parameters may be derived. This offers, for the first time, the possibility of simultaneous detailed measurement of meteorological and noise propagation data. The objective of present research is to establish the variability of meteorological conditions and their significance on outdoor noise propagation along a variety of propagation paths with a distance up to 1km. The project involves the development and verification of an outdoor noise prediction model that can take into account range-dependent meteorological conditions. Realization of these project objectives involves a series of field trials in which noise propagation measurements are made simultaneously with meteorological data under a range of weather conditions over well-defined terrain. Discussed here are the results of the first of these field trials.

FLAT TERRAIN FIELD TRIAL

The acoustical measurement systems were numbered #1 to #10 from the source. Each station was used as a stand-alone data logger and audio recorder logging \( L_{eq} \), \( L_{fast} \) and 1/3-octave band spectra each second. Synchronised digital recordings made for 10 minutes each hour sampled a maximum audio frequency of 10kHz. Measurements were made over a period of a week, with the source operating for eight hours a day. Automatic weather stations were set-up on 10m masts to provide spot checks of meteorological conditions. The measured noise data were synchronized with measurements from the meteorological Doppler infrared LIDAR system to enable direct correlation. The LIDAR is installed at a suitable position to scan the atmosphere for meteorological data during the noise measurement period. Local meteorological stations were used to provide wind profiles measurements to check against the LIDAR data.

DESCRIPTION OF MEASUREMENTS

The sound source used a series of equalised pink noise, silences and tones. Although meteorological data was recorded by the automatic weather stations and radiosonde, regrettably no measurements were obtained from the LIDAR during this trial. The winds throughout the trial period were strong cross winds to the line of the acoustic array. The consequences were high wind noise on the microphones and little refraction in the line of the receivers.

COMPARISON WITH PREDICTION

The parabolic equation method [1] provides accurate analytical predictions particularly useful for comparisons with the measurements of propagation from the tones used in the investigation. However more practical for environmental noise applications is the JASPEN [2] model using a heuristic ray tracing method [3]. Although assuming a linear velocity gradient to determine the sound propagation paths, for
distances less than 1km the model has been shown to have accuracy comparable with that of the PE method [4]. Figure 1 compares JASPEN predictions of $L_{Aeq}$ with background compensated measurements of pink noise at each receiver. Though the vector wind speed was close to zero, the wind velocity was typically around 10m/s.

![FIGURE 1. Comparison with JASPEN prediction (O)](image)

A correlation analysis between sound pressure level and vector wind speed and various wind and temperature gradients has shown that the highest correlation is with the vector wind speed [5]. The variation in the $L_{eq}$ (30s) measured in the 500Hz 1/3 octave band with vector wind speed determined at the source position is shown in Figure 2 for three of the ten receiver positions. A spread of measured sound level of around 6dB at 229m is seen not to significantly differ at receivers further from the source, and the least-squares fit to the data for each receiver distances does not show a correlation between the vector wind speed and measured sound level. While these data broadly support the relationships drawn from analysis of the measurement observations of the JOULE trials [5], where the dependence of the measured sound level on the vector wind speed was too weak and the spread too large to be conclusive, due in part to the restricted range of vector wind speeds encountered during this trial these results do not resolve the problems identified by this earlier work.

CONCLUSIONS

Useful investigations into the characterisation of meteorological conditions for outdoor sound propagation prediction purposes have in the past been limited by the ability to accurately determine the wind speed profile. The first of a series of trials designed to study range-dependent meteorological conditions and their influences on outdoor sound propagation has been performed. Whilst the analysis of this field trial data was limited by the absence of range-dependent meteorology, further trials over various terrains are under way to address these sound propagation topics.

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Source Location by Ground Effect Inversion

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A ground effect inversion and localization algorithm (GEILA) for source localization in an unknown outdoor environment has been investigated. GEILA matches the complex acoustic pressure received simultaneously at a distributed microphone array. Tests on simulated data have shown the possibility of deducing both range and height of broad-band sources in refracting and turbulent conditions. The method is robust to changing environments. Alternatively, for known source positions, the algorithm can be used to deduce acoustical properties of the ground surface and meteorological parameters.

INTRODUCTION

Considerable progress has been made in adapting numerical techniques for predictions of sound propagation in the presence of refraction and atmospheric turbulence above an impedance ground \cite{1-3}. However, localization of an acoustical source in an outdoor environment including the refraction and turbulence is still in progress. Li \textit{et al} developed an algorithm for determination of source height using a vertical array of two microphones by ground effect inversion \textit{from a priori} knowledge of a range between source and receivers in the presence of a linear sound velocity gradient \cite{4}. The purpose of this work is to investigate localization of sources in an unknown outdoor environment using a ground effect inversion and localization algorithm (GEILA) based on propagation models that include ground effect and atmospheric refraction. The effects of turbulence have been taken into account implicitly.

FLUCTUATION OF SPL

The instantaneous sound pressure level (SPL) fluctuates under different environmental conditions in the presence of atmospheric refraction and turbulence above an impedance ground. Figure 1 shows examples of SPL spectra obtained during 2 seconds of measurement at a range of 229 m, with source and receiver at 2 m and 1.5 m \cite{5}. The fluctuation of SPL results from variation in sound velocity gradient due to turbulence. Li \textit{et al} fitted the ground effect dip in instantaneous level difference spectra for ground impedance parameters and fitted the first two dips for sound velocity gradients \cite{2}. Their results demonstrate that the fluctuations in the best-fit sound velocity gradients and ground impedance parameters are related to the fluctuations of SPL.

FIGURE 1. Sound pressure level obtained during a 2-second measurement. \( z_s = 2 \text{ m}, \ z = 1.5 \text{ m} \) and \( r = 229 \text{ m} \). The lines are instantaneous SPL spectra (one spectrum every 0.16 s).

DESCRIPTION OF GEILA

GEILA makes use of a multi-element microphone array, equivalent to three single arrays each containing several microphones separated vertically and arranged in an arbitrarily-shaped triangle, see Figure 2. A matched field processing method, i.e. Bartlett processor, which was developed for the underwater environment \cite{6}, matches complex acoustic pressure spectra received simultaneously on each microphone of these arrays with those predicted by a propagation model for a grid of possible source positions \((r, z_s)\). The estimation includes the instantaneous effects of sound velocity gradient and ground impedance on the source localization. The values of sound velocity gradient \( a \) and ground impedance parameters \( \sigma_e \) and \( \alpha_e \) can be deduced by environmental inversion. However, the deduced values \( a', \ sigma_e' \) and \( alpha_e' \) represent \textit{effective} values corresponding to instantaneous sound pressure spectra rather than \textit{true} values average over time in a natural atmosphere.
NUMERICAL RESULTS

It is assumed that the atmosphere is vertically stratified and linear sound velocity profiles are used to describe the atmospheric refraction. In the presence of turbulence, sound velocity profiles fluctuate about their mean profile. Corresponding to each instantaneous sound velocity profile, there are instantaneous sound pressure spectra. GEILA matches each instantaneous complex sound pressure, which is simulated by using a GF-PE propagation model, with those predicted by a propagation model (either ray-trace or residue series) for a source location for trial values of \( r', z', a', \sigma'_s \) and \( e^\alpha' \).

Table 1 shows numerical results for \( r', z', a', \sigma'_s \) and \( e^\alpha' \) deduced from GEILA using frequencies 500-1500 Hz. The complex SPL, due to a broad band source at \( r = 1000 \text{ m} \) and \( z_s = 20 \text{ m} \) above a ground surface \((\sigma_s = 10^7 \text{ mks rayls m}^{-1} \text{ and } \alpha_s = 15 \text{ m}^{-1})\), is simulated in sound velocity gradient \( a \) that fluctuates randomly from \( 0.16 \times 10^{-6} \text{ m}^{-1} \) to \( 8.32 \times 10^{-6} \text{ m}^{-1} \) due to low turbulence. Three microphones of each vertical array are separated at heights of 0.1, 2 and 3 m above the ground. Using three such vertical arrays (see Figure 2), we should be able to determine the source azimuth \( \varphi \) \cite{7}. Table 2 shows the results for the same geometry but in conditions where the simulated sound velocity gradient \( a \) fluctuates randomly from \( -0.12 \times 10^{-6} \text{ m}^{-1} \) to \( -91.3 \times 10^{-6} \text{ m}^{-1} \).

SUMMARY

The numerical results of this study have shown the possibility of deducing range and height of a fixed source in refracting and turbulent conditions.

<p>| Table 1. Numerical results deduced by GEILA in simulated downward refraction. |</p>
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Calculation of Sound Propagation over Non Flat Terrain using Parabolic Equation

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In this paper we develop a method which aims to evaluate the propagation of an acoustic wave above a non flat ground and which could include realistic outdoor conditions. The acoustic waves are propagated in a parabolic approximation. The effect of topography is taken into account by rotating the coordinate system. Our method is validated in the case of propagation above a convex cylinder. Then an application of our method to the acoustic propagation above a wedge is presented.

\section*{INTRODUCTION}

Due to the traffic increase, roads become more and more noisy. Function of the traffic composition and the outdoor propagation conditions, populations can be submitted to substantial noise level variations. Their prediction implies to model the mixed influence of ground and atmospheric conditions. In most cases, the ground profiles and composition are complex and the meteorological situations unstable. Results reported here deal with a calculation method based on a numerical approach relevant to simulate various situations which can be met outdoors. For this, we used a well-adapted and accurate model based on the parabolic approximation. Solution are calculated through a Split-Step Padé method which has already been validated \cite{1} and which appeared to be reliable with respect to its obvious advantages in terms of angular aperture, CPU time and its capability to consider the main phenomena: homogeneous and mixed grounds, acoustic barriers. We propose here an extension of the code to the case of a non flat ground.

\section*{PARABOLIC EQUATION}

The problem of the propagation above a non flat ground has recently been studied by several acoustic researchers. A curved terrain version of the parabolic equation has been adapted for acoustic propagation in the atmosphere over fairly simple terrain profiles which can be reduced to a set of joined circular section pieces \cite{2}. In that approach, a separate conformal coordinate transformation was applied to each circular section piece of atmosphere. Sack and West \cite{3} chose to use a transformation which follows the terrain profile. Their method can be used for any smooth terrain but seems difficult to apply for a parabolic equation including wind terms. We chose to develop another model which can be used above any terrain and with a parabolic equation including wind terms \cite{4}. The non flat ground is treated as a succession of flat domains (see figure 1). After each flat domain the coordinate system \((r,z)\) is rotated so that the \(r\) axis stays parallel to the ground. The calculation above each domain needs an initial solution. The values of the initial solution for the domain \(n + 1\) are obtained from the values of the pressure field of the domain \(n\), except for the first domain where a gaussian starter is used. The propagation code is based on the parabolic equation and a Split-Step Padé method \cite{5}. Introducing the envelope \(u\), \(u(r) = \exp(-ik_0r)p(r)\), a marching algorithm is obtained:

\begin{align}
[1 + q(\eta + \xi)]u(r + \Delta r, z) = [1 + p(\eta + \xi)]u(r, z) \quad (1)
\end{align}

where \(\eta = n^2 - 1\), \(\xi = \frac{k_0}{k}z\) and \(\Delta r\) is the wave number and \(n\) is the index of refraction. Equation 1 is then discretized by a finite difference technique. Reflexions at the top of the numerical grid are controlled by introducing a thin artificial absorption layer in the upper part of the computation domain. The code has been tested in realistic outdoor configurations. The case of an impedance discontinuity
(infinite/finite) in a stratified atmosphere has been validated.

The propagation above a cylinder is chosen as a benchmark case for our code. The calculation can be treated by a method using conformal mapping [2]. In the transformed domain where the ground is flat, the effect of topography is accounted for by an effective sound speed, which is exponentially increasing with height. For a convex cylinder, the sound speed profile used in the transformed domain is given by 

\[ c(z) = c_0 \exp(-z/R_0) \]

\( R_0 \) is the radius of curvature and \( c_0 \) is a reference sound speed. For the benchmark case the parameters of the calculation are a radius of curvature of 100 m, a source height of 5 m and a frequency of 1000 Hz. The sound level is evaluated on a curved line at a height of 5 m above the cylinder. In our simulation the curved surface is split in 8 flat domains; the angle between two flat domains is \( \pi/32 \) radians. On figure 2 we plot the relative sound pressure level \( 20 \log(p/p_{ref}) \), where \( p_{ref} \) is the sound pressure in front of the source at a distance of 1 m. The agreement between the two methods is very good. Other comparisons have been carried out. The agreement is good up to an angle of \( \pi/16 \) radians between the flat domains. Beyond this value, the approximation of the cylinder by flat domains gives rise to errors.

**PROPAGATION ABOVE A WEDGE**

An outdoor measurements campaign above a wedge is planned as another benchmark case for our code. We use it to simulate the acoustic propagation above the wedge (figure 3). The height of the source is 2 m, the frequency is 400 Hz. We use a finite impedance value to represent a grassy ground. This value is calculated with a Delany and Bazley model from an air flow resistivity value \( 200 \, kN\, sec^{-1} \). The wedge has a slope of 10 degrees. The deformation of the acoustic field induced by the wedge and more particularly diffraction phenomena at the junctions between the flat domain are illustrated on figure 3.

**CONCLUSION**

We have presented a new model for sound propagation above non flat ground. This model has been validated through a comparison with results obtained from a method using a conformal mapping. The next step of our work is to integrate wind effects in the code by using a new parabolic equation including wind terms [4]. An outdoor site near Saint Berthevin (France) has been selected to study the influence of meteorological conditions on noise traffic. Acoustical and meteorological measurements will be performed simultaneously. This survey will provide a database of noise level variations in a complex environment (non flat terrain, mixed ground) and thus allow us to validate our numerical simulation of outdoor sound propagation.

**REFERENCES**

Calculation of noise barrier performance in a three-dimensional turbulent atmosphere using the substitute-sources method

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Substitute sources between a noise barrier and a receiver are used to calculate the effect of atmospheric turbulence on barrier sound reduction. The method is extended for application to three-dimensional situations with both high and low barriers. Calculations are made for a thin, hard screen, without the influence of a ground surface. The Kirchhoff approximation is applied for the low screens and a more accurate diffraction model is used for the higher screens. The calculated results are compared with corresponding ones for two-dimensional situations, also by using the substitute-sources method (SSM). The two and three-dimensional calculations give very similar results, which indicates that only two-dimensional models are needed. The results are also compared with those obtained using a scattering cross-section method which, although it predicts a much weaker influence of turbulence than the SSM, shows the same trend, namely that the turbulence influence is large only within a range of lower screen heights.

INTRODUCTION

This paper describes an extension of the substitute-sources method [1]. The problem under study is the increase in sound level behind barriers due to the influence of atmospheric turbulence on the sound propagation.

In terms of physical modelling, the problem can be seen as arising from two interacting processes: diffraction (due to the barrier) and sound propagation in an inhomogeneous medium. A direct numerical solution to the whole problem would generally be very expensive computationally. The approach here is to describe the field of a receiver, reached by sound from an original source, as a superposition of fields from a distribution of sources on a surface located between the original source and the receiver. The surface is denoted the substitute surface, and the sources on it are substitute sources. (See Figure 1.) When the substitute surface is located between the barrier and the receiver, there is a free path from all of the substitute sources to the receiver, and the calculation of the sound propagation along the free path is possible for a variety of situations with an inhomogeneous atmosphere. A mutual coherence function for a turbulent atmosphere has been applied here (with the structure parameter $C_n^2$ describing the strength of velocity fluctuations). Another possibility is to take into account the refraction due to a sound speed profile.

In this model the turbulent atmosphere is assumed to increase the noise level behind the barrier by a decorrelation of the contributions from the substitute sources. This implies that, in the absence of turbulence, the substitute sources must be interfering negatively.

In a previous study [1], the Kirchhoff approximation was used, which gives accurate results only for flat geometries, i.e. when the barrier is low in comparison to its distance to the source and the receiver. The results were compared with those from PE calculations. Here, calculations are made for 2-D and 3-D situations, both with and without the Kirchhoff approximation. The results from using the different approaches are compared; a comparison with a scattering cross-section method is also made. The situations studied here are without the influence of a ground surface, for a thin hard screen with edge parallel to the $z$ axis, and both the source and the receiver at $z = 0$.

The main parts of the theory are described in [1], and a full description is in a submitted paper [2], where also more numerical results are shown.

RESULTS

Calculations with the diffraction accurately modelled, i.e. without the Kirchhoff approximation, were made for screen heights $H = 2.5, 5, 11, 20, 35,$ and $50$ m.
The maximum height needed for the substitute sources to give a good approximation of the field at the receiver positions was obtained from test calculations. It was found that the height needed is much lower for calculations with turbulence than without. This means that when the surface \( S \) is enlarged, the convergence is faster with turbulence than without, which is an interesting result and also leads to much shorter computation times. In the calculations for the homogeneous atmosphere, the maximum height needed to be approximately doubled. A discretisation distance of five points per wavelength was used for all of the calculations.

For the lower screen heights, the results with and without the Kirchhoff approximation show small differences, as expected. Above \( H = 5 \) m, however, they deviate significantly; using the Kirchhoff approximation is shown to lead to an underestimation of the sound level for the homogeneous examples.

Both the 3-D and the 2-D results, when not using the Kirchhoff approximation, show that the influence of turbulence is weaker for the highest screens. Moreover, the 3-D and 2-D results are very similar, which indicates that the sound level increase behind barriers, caused by atmospheric turbulence, can be predicted by using 2-D models.

Although the scattering cross-section method predicts a much weaker influence of turbulence than the substitute-sources method (SSM), the results show the same trend, namely that there is a range of lower screen heights for which the sound reduction is the most sensitive to turbulence. For the higher screens, where the turbulence influence is weak, the scattering cross-section results tend to those for the SSM.

For the higher screens, using the Kirchhoff approximation (outside its range of validity) shows an influence of turbulence that is very weakly linked to the screen height. This will not be the case if the Kolmogorov turbulence model used here is exchanged for a Gaussian model, resulting in a significant turbulence scattering only within a range of lower screen heights [3]. Probably, this contrast is caused by the fast decay with increasing wave number that the Gaussian model describes, since the smaller scales of the turbulence cause the large angle scattering.

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Recommendations for improvement of aircraft noise propagation assessment

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Huge demands for more accurate and reliable methods of aircraft noise calculations provide the necessity to investigate particular elements of existing methods for aircraft noise calculations. Few national methods (USA, UK, Ukraine, Germany, Netherlands) and some international recommendations (ISO, ICAO, CAEC) are analysed and their poor elements are outlined. It was found that noise propagation assessment of the methods needs for improvements more necessarily. Thus theoretical and measurement analysis of aircraft noise propagation are performed first of all. On their basis the improvements are proposed and the main idea is to implement the approach of routine generators – for “noise-power-distance”-relationships (as a most important acoustic characteristic of the aircraft to be calculated), for “lateral attenuation” and for “screen effect” assessment of the noise during propagation. Routine generators are the subprograms in a main aircraft noise calculation program and they must be supplied by necessary current input data – meteorological, flight and topographical.

INTRODUCTION

Environmental noise, caused by traffic, industrial and recreational activities is one of the main local environmental problems in Europe and the source of an increasing number of complaints from the public. For this reason recent proposal on the review of the Fifth Action Programme announces the development of a noise abatement programme for action to meet new targets that is outlined in Green Paper and working groups were established for solving of the particular tasks. For example, harmonised calculation methods and associated measurement methods shall be better than the present ISO or national models, shall be elaborated for noise from road, rail, and air traffic as well as outdoor machinery and industry, for a variety of geometrical and weather conditions, and they should be valid for propagation over given distances (to be agreed) and have an agreed accuracy.

The approach for total aircraft noise assessment described elsewhere [1]. Here the details on noise propagation effects are outlined, as they were investigated during last time.

GROUND EFFECT ASSESSMENT - “LATER”-GENERATOR

All relationships for the extra ground attenuation of noise $\Delta L_{\text{int}}$ are based on approximate solutions for the reflection for spherical sound waves from locally reacting plane surfaces. ISO 9613/2 proposes appropriate formulas for the ground effect calculations from for two kinds of ground conditions without reference to source parameters. However it has been found that the differences between the predicted attenuation effects on overall A-weighted levels $L_A$ can as much as 12 dB as a result of spectrum variation, either for various types of aircraft (engines) under the same conditions. The magnitude of the variation is the same as for variance of Noise-Power-Distance (NPD)-relationships $L_{\text{npd}}$ due to atmosphere conditions observed in operation. This means that not only the type and mode of the engine, but also the influence of ambient conditions must be taken into account with equal accuracy and reliability. Differences between the magnitudes of lateral attenuation are considerable for different types of reflecting surfaces also [2]. Data obtained for grass surfaces have been used as the most appropriate for calculations of noise levels around the airports (for environment impact assessment, for example). Impedance characteristics can vary greatly. Sometimes mixed types of reflecting surfaces (inside the aerodrome area) must be considered. It was found that the boundary between the coverings of various types (for example inside aerodrome concrete-soil or concrete-grass types of mixing are usual) behaves as a line of sound wave diffraction and the size of diffraction zone from discontinuity line is comparable with length of sound wave [3]. All corrections have been obtained numerically with Chien and Soroka approach to interference effects and with a semi-empirical model for impedance characteristics of
reflecting surfaces. For routine aircraft noise assessment a “ground effect”-generator has been proposed (LATER-generator in Isobell’a soft tool, designed in Ukraine).

SCREEN EFFECTS ASSESSMENT - SCREEN-GENERATOR

Any type of screens may be used for noise abatement for ground modes of aircraft operations and maintenance (engine run-ups) around the airports. The effects of screens are assessed by means of a model which takes into account the effects of sound diffraction at screen edges, interference of direct and reflected waves from various kinds of impedance surfaces, each type of noise spectra generated by the aircraft, etc [2]. Spectral efficiencies of screens must be calculated for different kinds of noise propagation. Predictions of inserting loss in terms of $OASPL (\Delta L_{in})$ and $L_{max} (\Delta L_{o})$ for different types of noise sources under identical conditions are shown in [2].

The variation in the influence of the same type of screen on the results of noise abatement indicates the need for screen affect assessment software for aircraft noise calculations – such as SCREEN-generator.

RECOMMENDATIONS

Recommended procedures for ground attenuation assessment in aircraft noise calculations have been produced for several kinds of calculation schemes. The stages in the procedure are outlined below:

1. Calculation of acoustic spectrum at point of noise control:
   1.1. Calculation of the sound wave reflection from finite impedance surfaces
   1.1.1. General case (Chien/Soroka solution)
   1.1.2. Particular case: influence of the type of noise source (monopole, dipole, quadrupole)
   1.1.3. Particular case: influence of discontinuity
   1.2. Assessment of the impedance characteristics of the surface
   2. Calculation of aircraft noise indexes performed by means of “generators” for real noise radiation spectrum:
   2.1. NPD-relationships are defined in a routine mode taking into account real ambient conditions, type of the airplane with its known basic acoustic model (its real spectral characteristics) by use of RADIUS-generator approach
   2.2. Generalized relationships may be used for assessment of the lateral attenuation for jet, fanjet or propeller engines, but the approach of LATER-generator is the most valid (for noise source with real noise radiation spectrum)

2.3. Generalized data may be used for assessment of screen effects for jet, fanjet or propeller engines, but the approach of SCREEN-generator is the most valid (for noise source with real noise radiation spectrum)

CONCLUSIONS

For best results of calculation of aircraft noise propagation under the reflecting surfaces the following features must be taken into account:

- Assessment must be provided in a spectrum domain (1/3-octave is preferable) and then the results can be recalculated in a form of noise indexes of any kind;
- Assessment provided for real noise source spectrum in accordance with engine type, mode and directivity pattern;
- Assessment accounts for dominant physical model of elementary acoustic sources (monopole, dipole, etc) for the noise source under consideration (jet, fan, turbine, propeller, etc.);
- Assessment accounts for impedance characteristics of the reflecting surfaces under consideration (homogeneous covering);
- Assessment accounts for discontinuity effect only in a case of near location of reflection point to discontinuity line. At distances larger than one wave length from discontinuity line the reflection effect is like to homogeneous covering only.

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Application of two impedance discontinuity models to real road traffic noise cases in inhomogeneous air conditions

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Regulations for road traffic noise are becoming more and more restrictive requiring the noise prediction to be achieved at ranges where meteorological effects have to be taken into account. It is then of interest to evaluate the effect of the impedance discontinuity at the edge of a road platform on noise propagation in inhomogeneous air conditions. This paper compares results obtained by two different models: a Boundary Element Method with an adapted Green’s function and a modified Rasmussen geometrical method. Agreement between results from both of them is good. It appears from calculations that for a traffic noise situation, the effect of the impedance discontinuity is sensible only when the celerity gradient becomes relatively high.

PRESENTATION OF THE MODELS

A first model presented in a twin paper [1] is a Boundary Element Method whose Green’s function is calculated analytically for the case of a positive linear celerity gradient using the Normal Modes solution [2,3]. This method (called Meteo-BEM) shown to be powerful is taken as a reference here.

The second model used for impedance discontinuity problems is based on the well-known Rasmussen ray formulation [4] modified here in order to introduce the celerity gradient effect [5]. The geometry of the such a problem with a constant positive celerity gradient is given in Fig. 1 where the geometrical rays relative to one of the secondary sources Si above the discontinuity are drawn in the case of single reflections.

The excess attenuation can be written as [4,5,6]:

$$\text{Att} = -20 \log \left[ \frac{d_1 + d_2}{d_s} \sqrt{\frac{8 \pi k_0}{\pi}} \int \left[ \frac{e^{i 2 \pi f (\tau_1 + \tau_2)}}{s_{i} \sqrt{s_{i} + s_{j}}} \right. \right. $$

$$\left. \left. + \hat{Q}_{1} e^{i 2 \pi f (\tau_1 + \tau_2)} + \hat{Q}_{2} e^{i 2 \pi f (\tau_1 + \tau_2)} \right] dt \right] \quad (1)$$

where \( f \) is the frequency, \( k_0 \) is the wave number at \( z=0 \), \( s_i \) and \( \tau_i \) are the curvilinear length and travel time relative to the \( i_{th} \) path (shown in Fig. 1). \( \hat{Q}_1 \) and \( \hat{Q}_2 \) are the spherical wave reflection coefficients calculated for paths 2 and 4 respectively, and corrected by an amplitude coefficient. This correction takes into account the fact that the ray-tube area at \( Si \) (and \( R \)) associated to a reflected path on the curved surface is lower than in the case of a flat ground with straight rays [6] inducing an increase of acoustical energy.

Instead of calculating the exact values of \( s_i \) and \( \tau_i \), we consider that the two reflections shown in Fig. 2 occur at points \( I_1 \) and \( I_2 \) on tangential flat planes (see Fig. 3).
The problem is then similar to the Rasmussen one [4] with the following corrected heights [5]:

\[
\begin{align*}
\tilde{z}_s &= z_s + \delta z_s, \quad \tilde{z}_R = z_R + \delta z_R, \\
\bar{z}_i &= z_i + \delta z_i, \\
\bar{z}_{i1} &= z_{i1} + \delta z_{i1}, \\
\bar{z}_{i2} &= z_{i2} + \delta z_{i2}
\end{align*}
\]  

(2)

\[
\begin{align*}
\delta z_s &= a_0 \left[ \frac{z_s}{z_s + z_R} \right] \frac{d_1}{2}, \\
\delta z_R &= a_0 \left[ \frac{z_R}{z_s + z_R} \right] \frac{d_2}{2}, \\
\delta z_i &= a_0 \left[ \frac{z_i}{z_s + z_i} \right] \frac{d_1}{2}, \\
\delta z_{i1} &= a_0 \left[ \frac{z_{i1}}{z_s + z_{i1}} \right] \frac{d_2}{2}, \\
\delta z_{i2} &= a_0 \left[ \frac{z_{i2}}{z_s + z_{i2}} \right] \frac{d_2}{2}
\end{align*}
\]  

(3)

RESULTS AND CONCLUSIONS

Simulations have been carried out at 500 Hz with a celerity gradient of \(a_0 = 2.9 \times 10^{-3} \text{ m}^2\text{s}^{-1}\) making source-receiver distance vary. Geometry values are \(z_S = z_R = 1 \text{ m}\) and \(d_1 = 20 \text{ m}\). Ground impedance is calculated through the Delany and Bazley’s one parameter model with \(\sigma_1 = 300,000 \text{ cgs}\) and \(\sigma_2 = 300 \text{ cgs}\) [1]. Results compared with Meteo-BEM simulations are shown in Fig. 4a. Agreement is good.

Another configuration representative of road traffic noise has been studied at 500 Hz with a celerity gradient of \(a_0 = 1 \times 10^{-3} \text{ m}^2\text{s}^{-1}\) making source-receiver distance vary. Geometry values are \(z_S = z_R = 1 \text{ m}\) and \(d_1 = 20 \text{ m}\). Ground impedance is calculated through the Delany and Bazley’s one parameter model with \(\sigma_1 = 300,000 \text{ cgs}\) and \(\sigma_2 = 300 \text{ cgs}\) [1]. Results compared with Meteo-BEM simulations are shown in Fig. 4a. Agreement is good.

Same calculations could be carried out with negative gradients since this geometrical model is also applicable to this situation, if out of the shadow zone.

This method shall be applied to more complex celerity profiles by considering equivalent linear gradients depending on frequency and geometry. It is also possible to adapt it to multi-reflections phenomenon occurring in the case of a positive gradient situation. The main limitation concerning the presence of caustics is being investigated as well as turbulence.

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Variation of the Characteristic Impedance in Air with Environmental Conditions

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With the demand for more precision in sound intensity and sound power measurements, it is necessary to calculate physical parameters such as the Characteristic Impedance \( Z \) (the product of density and sound speed) in air with less uncertainty. With the latest knowledge on the density and the velocity of sound in air, a method is described on the development of an empirical equation for the computation of the variation of the characteristic impedance in air with temperature, humidity and barometric pressure.

**INTRODUCTION**

The variation of \( Z \) with humid air at sea level had been investigated \[1\] with data usable from 0 to 30 °C. At a wider temperature range the computation of \( Z \) can be improved by dividing the process into two parts: sound speed and air density.

**SOUND SPEED IN AIR**

When the barometric pressure changes from 90 to 110 kPa, the sound speed in air, \( c \) (m/s), varies from the International Electrotechnical Commission (IEC) standard conditions of 23 °C, 101.325 kPa and 50 % RH, by approximately ± 50 ppm. This relatively small variation in sound speed due to barometric pressure changes can be ignored in most acoustical measurements. However, sound speed is also influenced by temperature, humidity and carbon dioxide content. For example, for an increase of 0.1 °C, the sound speed increases by approximately 0.058 m/s or 0.017 %; and an increase of 10 % RH at 23 °C increases the sound speed by approximately 0.04 %. Similarly, an increase of CO₂ concentration \[2\] by 1 %, the sound speed decreases by approximately 0.32 %. With the aim to have a low uncertainty in mind a simplified computation method \[3\] with known uncertainty for sound speed as functions of temperature \( t \), humidity \( h \) and carbon dioxide content \( C \) in sea level standard air is used. The sound speed ratio \( c/c_0 \) is:

\[
\frac{c}{c_0} = a_0 + a_1 t + a_2 t^2 + a_3 C + a_4 C t
+ a_5 C t^2 + a_6 h + a_7 h t + a_8 h t^2
+ a_9 h t^3 + a_{10} C^2 + a_{11} h^2 + a_{12} h C t
\]

where \( c_0 \) is the sound speed for dry air, (331.29 m/s), at 0 °C and at sea level, \( t \) is the temperature in Celsius; \( C \) is the percentage carbon dioxide content, (assumed to be 0.04 percent for standard air); \( h \) is the humidity with 0 to 1 to represent RH from 0 to 100 %; and the numerical constants \[3\] are:

\[
\begin{align*}
a_0 &= 1.000100 \\
a_1 &= 1.8286 \times 10^{-3} \\
a_2 &= -1.6925 \times 10^{-6} \\
a_3 &= -3.1066 \times 10^{-3} \\
a_4 &= -7.9762 \times 10^{-6} \\
a_5 &= 3.4000 \times 10^{-3} \\
a_6 &= 8.9180 \times 10^{-4} \\
a_7 &= 7.7893 \times 10^{-4} \\
a_8 &= 1.3795 \times 10^{-6} \\
a_9 &= 9.5330 \times 10^{-5} \\
a_{10} &= 1.2990 \times 10^{-4} \\
a_{11} &= 4.8016 \times 10^{-5} \\
a_{12} &= -1.4660 \times 10^{-6}
\end{align*}
\]

The sound speed at the above IEC reference condition is 345.67 m/s. The uncertainty of the sound speed computed with (1) is estimated at approximately 450 ppm. from 0 to 50 °C.

**THE DENSITY OF AIR**

The basic equation for \( \rho \) is given by Giacomo \[3\]:

\[
\rho = \frac{(pM_a)}{(ZRT)} \left[ 1 - x_c \left( 1 - \frac{M_v}{M_a} \right) \right] 
\]

(2)
where $p$, $M_a$, $M_v$, $Z$, $p_{sv}$, $R$, $T$, $x_v$, $x_{sv} (= h. x_{sv})$, $x_v (= h. f_e p_{sv} / p)$, and $f_e$ are the barometric pressure, molar mass of dry air, molar mass of water vapour, compressibility factor, saturated vapour pressure, universal gas constant, kelvin temperature, mole fraction of water vapour in moist air, and in saturated moist air, and an enhancement factor to compensate for gas imperfection, respectively.

By applying the constants, such as $R$, $M_a$ and coefficients for the calculation of the compressibility factor $Z$, etc. given by Davis [5]; and calculate $f_e$ from the coefficients given by Greenspan [6] for water for temperature from 0 to 100 °C; and compute $p_{sv}$ by curve fitting the data given by Wexler [7] for the same temperature range, one can compute the density of moist air.

By combining (1) and (2) above to give $\Delta c$ and apply curve fitting $\Delta c$ as functions of $t$, $h$, and $p$ an empirical equation is obtained:

$$\Delta c(t,p,h) = \left[a_0 + (a_1 + a_2.h)p + (a_3 + a_4.h^2)p^2\right] +$$

$$[b_0 + (b_1 + b_2.h)p + (b_3 + b_4.h^2)p^2].1 +$$

$$[b_0 + (c_1 + c_2.h)p + (c_3 + c_4.h^2)p^2].t^2$$

(3)

where the coefficients $a$, $b$ and $c$ are constants.

The uncertainty of (3) depends heavily on the uncertainty in the measurement of the parameters $t$, $h$, and $p$. With modern measuring instruments, one may assume the uncertainties of the measurement of the above parameters as 0.1 °C, 5 % RH and 100 Pa, respectively. Over the temperature range from 0 to 50 °C, barometric pressure from 90 to 110 kPa, and humidity from 0 to 90 % ($h$ from 0 to 0.9), the standard uncertainty for $\Delta c$ calculated with the above procedure is estimated to be less than 0.2 % with a confidence level of approximately 95 %.

CONCLUSIONS

The above is only an outline for the computation procedure. With the aim for a lower uncertainty in the computation of the variation of $\Delta c$ with the environment, work is continuing to arrive at the optimum values for the constants shown in (3).

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An Efficient Method for the Prediction of Sound Propagation in a Canyon

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An efficient method is proposed to calculate the acoustic field in a canyon with two rigid walls and an impedance ground. In this method the infinite sum in the integral representation of the fundamental solution of the Helmholtz equation is reduced to a number of terms and the remainder, which are treated explicitly. A pole subtraction technique is used to ensure smooth numerical integration of the derived functions. The developed expressions can be useful for efficient, 2-D boundary element modelling of sound propagation in city streets with high-rise building facades of irregular shape.

INTRODUCTION

In city street environments multiple reflections occur at the building facades and the ground and can contribute substantially to the overall sound levels. In outdoor acoustics a two-dimensional version of the boundary element method (BEM) has been exploited to predict the efficiency of road noise barriers [1] and the effect of building facades [2]. In principle, the BEM is not limited by the extent of the acoustic region of interest, although specific programming implementation can be restricted by the size of the accessible computer memory and processor speed. In this paper it is shown that the contributions from multiple reflections from the plane, rigid canyon walls can be incorporated analytically in the expression for the Green’s function, which can be adopted in efficient BEM implementations.

THEORETICAL FORMULATION

The acoustic field at an arbitrary observation point \( \mathbf{r} = (x, y) \) from a line source at \( \mathbf{r}_0 = (x_0, y_0) \) above an impedance boundary can be written as

\[
G_\beta (\mathbf{r}, \mathbf{r}_0) = G_0 (\mathbf{r}, \mathbf{r}_0) + P_\beta (\mathbf{r}, \mathbf{r}_0),
\]

where \( G_0 (\mathbf{r}, \mathbf{r}_0) \) is the 2-D Green’s function for sound propagation above a rigid boundary and \( P_\beta (\mathbf{r}, \mathbf{r}_0) \) is the perturbation term. The Green’s function \( G_0 (\mathbf{r}, \mathbf{r}_0) \) and the perturbation term \( P_\beta (\mathbf{r}, \mathbf{r}_0) \) can be expressed as Laplace-type integrals as [3]

\[
G_0 (\mathbf{r}, \mathbf{r}_0) = -\frac{e^{i\eta}}{\pi} \int_0^\infty \frac{\cos(\eta \cdot \mathbf{v}) + \cos(\eta_\beta \cdot \mathbf{w})}{\sqrt{\mathbf{v}^2 - 2i}} e^{-\mathbf{v} \cdot \mathbf{r}} \, dv
\]

and

\[
P_\beta (\mathbf{r}, \mathbf{r}_0) = \frac{2\beta e^{i\eta}}{\pi} \int_0^\infty \frac{\beta \cos(\eta \cdot \mathbf{w}) - i\omega \sin(\eta_\beta \cdot \mathbf{w})}{(\beta^2 - w^2)\sqrt{\mathbf{v}^2 - 2i}} e^{-\mathbf{v} \cdot \mathbf{r}} \, dv + P_s
\]

where \( \beta \) is the surface impedance of the boundary, \( \eta = kx_0 \), \( \eta_\beta = k(y + y_0) \), \( \eta_w = k(y - y_0) \), \( w = v\sqrt{v^2 - 2i} \), \( P_s \) is the surface wave term defined in [3] and \( k \) is the wavenumber in air. Integrals (2) and (3) can be combined so that the Green’s function for sound propagation above an impedance boundary can be expressed as

\[
G_\beta (\mathbf{r}, \mathbf{r}_0) = \frac{e^{i\eta}}{\pi} \int_0^\infty \frac{f(v^2)e^{-iv \cdot \mathbf{r}}}{v} \, dv + P_s
\]

where

\[
f(v^2) = \frac{w^2 \{ \cos(\eta \cdot \mathbf{w}) + \cos(\eta_\beta \cdot \mathbf{w}) \} - 2i\beta w \sin(\eta \cdot \mathbf{w})}{(\beta^2 - w^2)\sqrt{v^2 - 2i}} + \frac{\beta^2 \{ \cos(\eta \cdot \mathbf{w}) - \cos(\eta_\beta \cdot \mathbf{w}) \}}{(\beta^2 - w^2)\sqrt{v^2 - 2i}}
\]

The problem of multiple reflections from two parallel rigid walls can be reduced to the problem of sound propagation from an infinite number of periodic sources elevated at the same height above the impedance ground. If the equivalent sources are at positions \( \mathbf{r}_l = (x_l, y_0) \) spaced with period \( 2h \), then the resultant field at an arbitrary observation point \( \mathbf{r} = (x, y) \) is written as

\[
G_\beta^p (\mathbf{r}, \mathbf{r}_0) = \sum_{l=0}^{\infty} G_\beta (\mathbf{r}, \mathbf{r}_l)
\]

where \( l = 0, 1, \ldots \) and \( x_l = x_0 + 2lh \). The first \( N \) terms can be extracted from sum (6) to be evaluated explicitly, using the efficient calculation method proposed in [4]. Provided that \( x_N > x \), the remaining sum, from \( N+1 \) to infinity, can be expressed as a single infinite integral using the geometric series rule so that

\[
G_\beta^p (\mathbf{r}, \mathbf{r}_0) = \sum_{l=0}^{N-1} G_\beta (\mathbf{r}, \mathbf{r}_l) + \frac{e^{i\eta_0}}{\pi} \int_0^\infty g(v^2)e^{-iv \cdot \mathbf{r}} \, dv + P_s
\]

where

\[
g(v^2) = \frac{f(v^2)}{1 - e^{-2ih(v^2)}}
\]

and \( P_s \) is the surface wave term and \( \eta_0 = k(x - x_0 + 2Nhl) \). Integral (7) can be calculated numerically using the Gauss-
Can have two poles, $z_n = i(1 - \sqrt{1 - \beta^2})$ and $z_n = \frac{\arg e^{2ikb}}{2kh}$ which can lie close enough to the integration path to affect the accuracy of the numerical integration. It is suggested to subtract these poles to overcome this problem. To subtract the poles, the integral in exp. (7) can be presented in the form

$$
\int_0^\infty g(t) e^{-\alpha t} e^{-i \omega t} dt = \int_0^\infty p(t) e^{-\alpha t} e^{-i \omega t} dt + \frac{1}{\pi} \int_{\gamma_n b}^\infty \frac{1}{t | \zeta - z_n |} e^{-i \omega t} dt
$$

where $\nu = \frac{v}{2}, \ p(z) = g(z) - \frac{e}{z - z_n}, e_{z_n} = \text{Res} \ g(z)$. The last term in eqn (8) is a table integral [5]. The pole subtraction ensures that the integrand $p(t)$ is bounded and analytic, as a function of $t$, in a neighbourhood of the positive real axis. Exp. (7) can now be used to construct the acoustic field due to an array of sources extending to infinity in both directions, i.e. $G_{\beta}^{DP}(\mathbf{r}, \mathbf{r}_0) = \sum_{t=-\infty}^{\infty} G_{\beta}(\mathbf{r}, \mathbf{r}_t)$. If we let $\tilde{x}_r = 2x - x_0$ so that $\tilde{r}_r = (\tilde{x}_r, y_0)$, then the periodic impedance Green’s function is written as

$$
G_{\beta}^{DP}(\mathbf{r}, \mathbf{r}_0) = -G_{\beta}(\mathbf{r}, \mathbf{r}_0) + G_{\beta}^{e}(\mathbf{r}, \mathbf{r}_0) + G_{\beta}^{b}(\mathbf{r}, \mathbf{r}_0).
$$

The solution to the problem of sound propagation from a point source at $\mathbf{r}_0$ in the canyon formed by two parallel vertical walls emerging from an impedance boundary can be given in terms of the periodic Green’s function (9). The acoustic field in the canyon which occupies the strip $0 < x < h, y > 0$ in the $Oxy$ plane, has rigid walls at $x = 0$ and $x = h$ and an impedance boundary condition on $y = 0$ is given by

$$
G_{\beta}^{can}(\mathbf{r}, \mathbf{r}_0) = G_{\beta}^{DP}(\mathbf{r}, \mathbf{r}_0) + G_{\beta}^{DP}(\mathbf{r}, \mathbf{r}_0^*),
$$

where $\mathbf{r}_0^*$ are the images of $\mathbf{r}_0$ in the wall at $x = 0$. Exp. (10) was used to calculate the sound pressure level

$$
L = 20 \log_{10} \left( \sqrt{k} G_{\beta}^{can}(\mathbf{r}, \mathbf{r}_0) \right)
$$

in a city street canyon with a porous asphalt road surface. The positions of the source and receiver were chosen to be at $x_0 = 5.75$ m, $y_0 = 2.0$ m, $x = 1.5$ m and $y = 1.5$ m. The width of the canyon was set to 17 m. The Attenborough model [6] was used to predict the surface acoustic impedance of the porous asphalt road surface. The following non-acoustic parameters were adopted in the model: flow resistivity $R = 3500$ Pa s m$^{-2}$, porosity $\Omega = 0.335$, tortuosity $\eta = 1.91$, shape factor $s_p = 0.21$, thickness $d = 0.1$ m. The result is shown in Figure 1, where it is compared to that predicted by the method of normal mode decomposition [7]. Excellent agreement is demonstrated between the results of the two methods through the spectral range considered.

**CONCLUSIONS**

The proposed calculation procedure is efficient as long as $k(y - y_0^*)^2 / h$ is not too large. Unlike the method of normal mode decomposition the proposed method is robust when $k(y - y_0)$ is small. It is expected that the developed Green’s function will prove useful in boundary element modelling of sound propagation in city streets and waveguides and for problems of scattering by periodic structures.

**REFERENCES**


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**Figure 1. Comparison of predictions using the proposed method and the method of normal mode decomposition for sound propagation over a porous road surface.**
Acoustic Pulse Measurements over Mixed Impedance Ground Produced by a Snow Cover


U.S. Army ERDC-ERREL, 72 Lyme Road, Hanover, NH 03755 USA

By selectively removing sectors of a natural snow cover, acoustic propagation paths with different lateral ground impedance properties were produced. Measurements were then conducted over these paths to investigate the effect of mixed ground impedance on acoustic waves. Blank pistol shots were used to produce the acoustic pulses that were digitally recorded after propagating horizontal distances of 30 to 150 m. The first measurement was conducted over an undisturbed snow cover, a highly porous material with low acoustic impedance. Then, portions of the snow cover were removed and the measurements were repeated. Where the snow was removed, a less porous, higher impedance frozen ground surface was introduced into the propagation path. The snow cover was removed in stages so that several different inhomogeneous ground impedance conditions were sampled, and a final measurement was made with the snow cover entirely removed. Changes in the pulse waveforms were observed when only 10% of the propagation path was modified by plowing. Differences were also observed depending on whether the source was over snow or plowed ground.

Introduction

The interaction of sound energy with the ground is an important factor in outdoor sound propagation. Understanding this phenomenon is needed for accurate noise propagation predictions and for improved performance of sensor systems. In realistic situations, and especially in urban terrain, sound often encounters laterally varying ground, for example grass, paved, or snow-covered surfaces. Current understanding of the effects caused by changing ground properties along the propagation path is limited by a lack of experimental data, and only single-frequency computational methods are available for predictions [1–5].

Approach

We designed an experiment to investigate the effect of lateral variations in ground impedance on acoustic pulse propagation. By measuring propagation across snow-covered ground, and systematically removing areas of the snow cover by plowing, we were able to introduce large variations in the ground impedance in a controlled manner. Microphones were installed on the snow surface at three locations spaced 30 m apart. Blank pistol shots were used as the source of the waves. The pistol was held by hand 1 m above the snow or ground surface, pointed toward the sensors, and fired. Three different source locations were used giving propagation ranges between 30 and 150 m in length. After shots were recorded from the three locations, a swath of snow was removed by plowing and the measurements were repeated. This procedure was followed until all of the snow was removed. The signals from the microphones were recorded using a digital seismograph.

Results

Figure 1 shows a few waveforms recorded over the same 90-m path as sections of snow were removed. Visible changes in the recorded acoustic pulse are evident when only 10 m of the snow was removed (11% of the path length). Generally, we measured different pulse waveforms each time the ground conditions were changed by plowing.

Table 1. Waveform parameters from uniform impedance model.

<table>
<thead>
<tr>
<th>Label in Fig. 1</th>
<th>Snow–ground propagation distance (m)</th>
<th>Peak pressure (Pa)</th>
<th>Effective uniform flow resistivity (kRayls/m)</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>90–0</td>
<td>1</td>
<td>7</td>
</tr>
<tr>
<td>B</td>
<td>80–10</td>
<td>2</td>
<td>13</td>
</tr>
<tr>
<td>C</td>
<td>60–30</td>
<td>3</td>
<td>19</td>
</tr>
<tr>
<td>D</td>
<td>0–90</td>
<td>21</td>
<td>400</td>
</tr>
</tbody>
</table>

In our first attempt to model these measurements, we tested whether a uniform ground impedance could be found to match the observed data [6]. This approach was surprisingly very successful, as can be seen by the agreement with the dotted lines in the Figure.
The ground impedance parameters determined from the model are listed in Table 1.

As the proportion of plowed ground increased, the effective air flow resistivity ($\sigma$) needed to match the data increased from 7 kPa m$^{-2}$ (a typical snow value) to 400 kPa m$^{-2}$ (a typical bare ground value). We found that the equation

$$\ln(\sigma) \approx 6 - 4 \frac{R_{\text{snow}}}{R_{\text{snow}} + R_{\text{ground}}}$$

approximately described the relationship between the effective flow resistivity and the proportion of snow-covered ground in the propagation path.

Additional analysis of these data will be presented in a future paper. These data provide a robust test set for the development of models predicting sound propagation above mixed impedance ground.

Acknowledgements

We thank Mr. Leon Stetson of North Pomfret, Vermont, for providing the field site. This work was funded by the U.S. Army Corps of Engineers.

References


Figure 1. Measured waveforms (solid lines) over a 90-m-long path as the snow was removed. (A) 90 m snow (undisturbed). (B) 80 m snow, 10 m ground. (C) 60 m snow, 30 m ground. (D) 90 m ground. Dotted lines are theoretical predictions using a homogeneous ground impedance model. The waveform amplitudes are normalized.
The Background Noise Level Analysis of Cultural Centers In Taiwan

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The cultural center in Taiwan contains library, exhibition room and performance hall. While, the background noise level in these three different areas are measured. According to the acceptable noise criteria. The unqualified rates are 86% for libraries, 80% for exhibition rooms and 79% for performing halls. The major sources of noise all come from air-conditioning system. In addition, the managers of these areas and the performance groups are subjectively surveyed by questionnaires and it is found out that the dissatisfied rate of the cultural center’s background noise is around 17%-26%. It’s very interesting and valuable for further investigation.

INTRODUCTION

Cultural centers in Taiwan, which are open to general public and relevant staff, serve as major places to promote cultural activities. It contain library, exhibition room and performance hall multipurpose hall. This study is intended to discuss the background noise level in cultural centers. The aims of this study are to objectively examine the real situation of background noise level in cultural centers while air-conditioning system are turned on, and to have a subjectively questionnaire to center managers and performing groups in the aspect of background noise level.

ACCEPTABLE BACKGROUND NOISE LEVEL CRITERIA IN CULTURAL CENTERS

Refer to C. M. Harris’s 1979, [1] suggestion of acceptable background noise level in concert hall and large auditorium is 30-35 dB (A), in library is 40-45 dB (A). So we make a temporary acceptable background noise level criteria for performing hall is 35 dB (A), for library is 45 dB (A). As to exhibition room, we consider as library, guilt chatter is permissible, the acceptable background noise level is 45 dB (A).

METHODOLOGY

There are 23 cultural centers in Taiwan. The amounts of this study measured are 62 reading areas of libraries, 55 exhibition halls and 19 auditorium of performing halls. We measured each room’s background noise level with turned on air-conditioning system and noted noise sources. Then we evaluated the results by acceptable temporary background noise level criteria. We also took a questionnaire to ask 23 cultural center managers and 507 performing groups include music dance and drama groups about subjective noise impression. There were 23 valid questionnaires received from center managers and 87 valid questionnaires received from performing groups.

RESULTS

The results of background noise level with turned-on air-conditioning system in cultural centers are shown in Figure 1. The background noise impression of cultural center managers and performing groups is as present in Figure 2 and Figure 3.

FIGURE 1. The average of background noise level with turned-on air-conditioning system in Taiwan’s cultural centers.
centers.

FIGURE 2. The impression of cultural center managers about background noise with turned-on air-conditioning system in cultural centers.

FIGURE 3. The impression of background noise with turned-on air-conditioning system in the auditorium of performing hall between managers and performing groups.

CONCLUSION

1. In general, background noise levels in the reading area of libraries and exhibition rooms with turned-on air-conditioning system which are almost equal is about 51dB (A). However, it is higher than that in the auditorium of performing halls, which is about 42 dB (A). The qualified rate of background noise level been evaluated by temporary criteria is 12.3% in reading area of libraries; 17.3% in exhibition rooms, 21.2% in auditorium of performing halls.

2. According to study results, there are 95% spaces, regardless of reading area of libraries or exhibition rooms or auditorium of performing halls, which major background noise source came from air-conditioning systems; the second was traffic noises.

3. According to questionnaire results the rate of noisy annoyance answered by center managers in reading areas of libraries is 26% include 21.7% noisy and 4.3% very noisy, 17.4% in exhibition rooms, and 18.2% in the auditorium of performing halls. In the case of performing groups that the rate of noisy annoyance in the auditorium of performing halls is 24.5% include 23.3% noisy and 1.2% very noisy. Based on the objective measure, we found that the background noise level of reading area of libraries and the exhibition rooms are almost equal. But center managers felt that the noise in the reading area of libraries is greater than that in the other one.

4. Regarding to the rate of noisy annoyance in auditorium of performing halls there is difference about 6% between center managers and performing groups. The latter felt noisier than the former.

5. The cross analysis between measured results of background noise level and questionnaire results shows that there is an extremely difference. The disqualified rate of objective evaluation is 79-86%, the subjective questionnaire results that thought the background noisy is 17-26%. In this study, the results of using temporary criteria to evaluate background noise level in cultural centers in Taiwan can't match the results of subjective questionnaire. For this reason we suggested that more than two evaluation degrees (10dB(A)) could be down than the temporary background noise level criteria.

ACKNOWLEDGEMENTS

Gratitude must go to the Council for Cultural Affairs and Cultural Centers for their help.

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This paper develops two different aspects related to outdoor sound propagation. The first part of the paper deals with temperature influence, and the second part with sound attenuation due to abundant vegetation (trees/forests). Concerning temperature effects, some results will be shown for different source/receiver distances and for three different temperature conditions. Then the influence of trees density on sound propagation will be analyzed.

INTRODUCTION

Sound propagation outdoors is affected by three basic factors: ground characteristics, meteorological conditions and the presence of obstacles. Concerning the meteorological parameters, it is necessary to discriminate the effect of each of them separately (temperature, wind speed and direction, atmospheric pressure, relative humidity…) as well as the effect of turbulences and different distribution profiles for those parameters.

The purpose of our paper is to study how sound propagation outdoors is affected by temperature and by the presence of arboreal mass (woods) of different densities but over the same kind of ground, with a well known flow resistivity $\sigma$. We have performed measurements of the sound pressure level in third octave bands, at different distances from the source, in a rather flat area, and simultaneously several meteorological variables were measured.

The sampling place is the Centro de Investigación de la Baja ATMósfera –CIBA- (Low Atmosphere research Centre) located 30 km. away from Valladolid (Spain). From the experimental data it can be seen that temperature is significant parameter considering outdoor sound propagation, and that the presence of woods results in an unequal attenuation for the different frequencies.

MEASUREMENT SET UP AND INSTRUMENTATION

Measurements were carried out in a pine forest. The source and the receiver were placed at 0.8 and 1.2 m height respectively. The parameter measured was the one minute Leq for the following source-receiver distances: 2,10,20,40,60 and 80m. The B&K 4224 sound source generated a wide band pink noise. We measured in third octave bands, with lineal average and fast constant time using B&K 2260 modular analyzer. We also used an anemometer to verify wind speed.

TEMPERATURE INFLUENCE

It is our intention to try to identify the potential attenuation as a function of temperature. In order to evaluate this effect we have measured the sound pressure level at different source/receiver distances and under three different temperature conditions. In all cases the measurement positions were the same and registered wind speed was below 1m/s. Turbulent effects were not taken into account since the distances between all measurement positions were relatively small. Ground conditions can be considered basically identical in all three sets of measurements. Since we are interested in making a comparative study under different temperature conditions, we have not included the effect associated to geometric divergence since this effect should be equal for the three measurement sets. Sound attenuation for each receiver position was calculated as the measured level in that position minus the sound level measured at a distance of 2 m from the source. Measurement were made in March, May and July in order to ensure quite different temperatures, and early in the morning in order to keep the wind speed at low levels. For each measurement it was checked that the temperature remained approximately constant over all the measurement, being its value 7 ºC, 15 ºC and 20 ºC respectively.

Figure 1 sows sound attenuation in dBA versus the source/receiver distance, in meters, and for the three different measuring days. Similarly, Table 1 shows the maxima and minima attenuation differences in dBA between all cases and for all measurement distances. The maximum attenuation difference has been calculated taking the maximum attenuation values obtained (May) as reference and subtracting the minimum values obtained any other day. We have found the highest divergence at 40 and 60 m from the source. The minimum attenuation difference was calculated using the data obtained on July and March were the temperature difference was 15 ºC. It can be seen that the differences remain below 0.7 dBA shown.
In spite the fact that it is really temperature gradient and not temperature itself which is relevant, we can already observe significant attenuation differences just considering temperature variation, although the tendency cannot be clearly established. In order to explain the origin of these differences it is necessary to undertake a deeper study including the effect of wind and temperature profiles, as well as potential turbulent effects. We are developing such a study and although we do not have concluding results available yet, we have found rather good agreement between quite complete existing outdoor propagation prediction models and experimental results.

**REFERENCES**

Validation of an Efficient Outdoor Sound Propagation Model Using BEM

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An approximate, simple and practical model for prediction of outdoor sound propagation exists based on ray theory, diffraction theory and Fresnel-zone considerations [1]. This model, which can predict sound propagation over non-flat terrain, has been validated for combinations of flat ground, hills and barriers, but it still needs to be validated for configurations that involve combinations of valleys and barriers. In order to do this a boundary element model has been implemented in MATLAB to serve as a reliable reference.

INTRODUCTION

There are several schemes for prediction of outdoor sound propagation over non-flat terrain. One particularly efficient approach is based on a combination of ray theory, diffraction theory and Fresnel-zone considerations [1]. However, the reliability of this model still needs to be established for some shapes of the terrain, in particular combinations with a valley between the source and the receiving point.

The purpose of this paper is to examine the validity of the model for the case in which a valley and a barrier separate the source and the receiver.

THE BEM MODEL

As shown in figure 1 the source is separated from the receiver by a hollow of variable depth and a barrier. All surfaces are assumed to have a locally reacting impedance. To test the performance of the approximate sound propagation model for these configurations, a two-dimensional boundary element model has been developed.

The BEM model, which is implemented in MATLAB, uses a line source and solves numerically a coupled system of equations. The two-domain coupled problem consists of an exterior domain in which the barrier is placed, and an interior domain in which the hollow is placed. The two domains are coupled by a fictitious boundary over the hollow. The fictitious boundary has to be bent in order to fulfill a requirement of the boundary element method according to which bodies in standard BEM must have a volume inside, otherwise the problem is ill-conditioned.

Since an infinite impedance plane is considered, the exterior problem is solved using the modified Green’s function derived by Chandler-Wilde and Hothersall for such a case [2]. A two-dimensional boundary element model for sound propagation over a barrier on a homogeneous impedance plane was developed two years ago making use of this function [3]. The barrier is taken into account as a term in the boundary integral equation of the general exterior problem. The interior problem uses the free field Green’s function. The resulting coupled system of equations accounts for the two domains and their common boundary. The approach has been inspired by a paper by Peplow and Chandler-Wilde [4].

Initially the BEM model was validated with a cylinder (for which there is an analytical solution), and the coupled version was validated by comparison with a flat cutting (i.e. no hollow), and by comparing with an uncoupled BEM model in which the entire geometry was raised corresponding to the depth of the hollow. Therefore, the results of the model may be regarded as reliable. The coupled BEM model is much less time-consuming than the uncoupled model.

However, the boundary element method still requires a long calculation time at high frequencies since there must be at least five elements per wavelength. Therefore the BEM calculations end at 2 kHz.

RESULTS

To examine the reliability of the approximate model some examples have been calculated using the two methods.

The geometry is a valley consisting of two segments, with a depth of 0, 0.1, 0.25 and 0.5 m. There is
also a wedge barrier of fixed shape (1 m high with a width of 1 m at the base), 0.2 m away from the limit of the hollow. The barrier is considered to be rigid, whereas the cutting and the impedance plane have a non-zero admittance given by a flow resistivity of 200 kNs/m²⁴ [5].

The source is placed 0.5 m over the impedance plane, right above the starting edge of the hollow. The receiver height is 0.5 m, and it is 10 m away from the source, on the other side of the barrier.

The coupled BEM problem is solved for the one-third octave centre frequencies in the frequency range from 31.5 Hz to 2 kHz.

The two models are compared for the combinations of barriers and hollows described above. As can be seen by comparing figures 2 and 3 the predictions of the approximate model agree quite well with the BEM results, although small deviations occur at low frequencies and at the interference dip.

**CONCLUSIONS**

An approximate sound propagation model based on ray and diffraction theory and Fresnel zone considerations has been found to be in acceptable agreement with a boundary element model for various geometries involving combinations of hollows and barriers.

**ACKNOWLEDGMENTS**

The authors would like to thank Peter Møller Juhl (Odense University, Denmark) for his good advice and assistance with the boundary element method, and would also like to acknowledge Birger Plovsing (Delta Acoustics, Denmark), whose model is the reason for this work to be.

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**REFERENCES**


Improvement of reverberation time estimation for sound propagation study in a street scale model

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This work is part of a project concerning the realisation of a software for prediction of sound propagation in cities. This paper presents the application of a non linear regression method from the energy decay for time reverberation estimation in a scale model. Results are given for impulse responses simulated and measured in a real street and in a street scale model. This method improves efficiently the time reverberation estimation. Characteristics of the sound source are finally exposed.

INTRODUCTION

A research project in transport industries concerning sound environment in cities was recently carried out. Models based on scattering phenomenon were first developed to describe sound propagation in streets \cite{1}. To validate these models, a street scale model (1:25) has been built \cite{2}. Reverberation time (RT) and sound attenuation were measured and compared to measurements in a real street and to modeling predictions. This work allowed to validate the scale model and the scattering approach but not in the whole road noise bandwidth ([0.4-5] kHz). This limitation is mainly due to weak signal to noise ratio (SNR) in high frequency domain. Consequently, our work consist in the increase of the impulse source power and in the use of the non linear regression method (NLM) developped by Xiang \cite{3} for RT estimation. The first part of this paper presents the application of this method to impulse responses simulated \cite{4} and measured. The last part finally presents requirements for the impulse source.

REVERBERATION TIME ESTIMATION

Principle of the method

The reverberation time (RT) is determined classically from the energy decay versus time with a linear regression method (LM), in a scale where the SNR is large enough. This method is highly dependent on the SNR, and yield bad results for low SNR. Therefore, it is advantageous to take into account the whole energy decay and not only the part free of noise. Considering that the impulse response (IR) is the sum of an ideal IR free of noise $h_i(t)$ with an additive noise $n(t)$, which are assumed uncorrelated, the discrete-time sound decay, after integration by the Schroeder method \cite{5}, can be written as following \cite{3}:

$$d(t_k) = \sum_{m=k}^{L} h_i^2(m) + \bar{n}^2(L - t_k), \quad (1)$$

where $t_k$ is the discrete time, $L$ is the length of the IR and $\bar{n}^2$ is the mean square value of noise. The first term of the right side of eq. (1) corresponds to the Schroeder integration for an ideal IR which is assumed exponentially decaying. The second term traduces the additive noise which is linearly decreasing after integration. Consequently, the continuous-time energy decay curve model can be established as:

$$d(t) = x_1 e^{-x_2 t} + x_3 (L - t), \quad (2)$$

where $x_1$ and $x_3$ are respectively the amplitudes of the direct sound energy and of the noise energy. The RT is obtained from the estimation of $x_2$. The first step of the procedure consists in estimating roughly these three parameters to get an initial vector $(x_1, x_2, x_3)$. Then, a non linear regression method yields a more precise value of $x_2$.  

Simulated impulse responses

In order to validate this method, it was applied to different simulated IR. The first one is a random noise damped exponentially (constant RT). The second one is, for each time, the sum of exponentials decaying with squared frequency (decaying RT) \cite{4}. Both IR are also contaminated by an additive noise. Figure 1 shows results of RT estimation for third octave bands with these simulated IR and for low SNR (40 and 30 dB). These results indicate that standard deviation is lower with the NLM and that the classical LM method diverges for SNR lower than 40 dB, while the Xiang method yield good results down to 30 dB in both cases.
FIGURE 1. Estimation of reverberation time by the linear (LM) and the non linear (NLM) methods from impulse responses simulated with a constant reference RT$_{ref}$ of 1 s (left) or decreasing from 1.5 to 0.5 s (right).

FIGURE 2. Estimation of reverberation time by the linear (LM) and the non linear (NLM) methods from impulse responses measured in a real street (left) and in a street scale model (right); $d_{SR}$ is the full scale distance source-receiver.

### Measured impulse responses

Figure 2 shows results obtained for IR measured in a real street and in a scale model. Firstly, standard deviation is always lower with the NLM. Secondly, the LM yields an increase of RT from 1 kHz, which seems to be not justified physically (RT should decrease with frequency due to the increasing atmospheric attenuation). This phenomenon does not appear any more with the NLM. Finally, the validity of the LM, limited at 1 kHz is extended with the NLM to 2.5 kHz in a real street and 2 kHz in the scale model.

### Characteristics of the sound source

We can now estimate RT with a SNR of 30 dB minimum. Background noise measured by a 1/8” microphone is around 75 dB (SPL). Consequently, the source power has to reach at least 105 dB in the scale model up to 125 kHz. Note that the source has also to be of small size (lower than 2 cm) and omnidirectional.

### CONCLUSION

The non linear regression method of Xiang allows to treat impulse responses more efficiently than the classical linear regression method. It is therefore possible to achieve this extraction down to 30 dB of signal to noise ratio. Works deals now with the impulse source which is an electrostatic spark. Theoretical and experimental studies of the influence of the geometrical and electrical parameters of the source are in progress to shift the spectrum towards high frequencies and to get more power.

### REFERENCES

Diffraction of a Cylindrical Wave by a Two-Impedance Ground

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The problem of sound propagation over a flat plane with two impedances is considered. The problem is two-dimensional as the excitation sound source is a line parallel to the line of impedance discontinuity. The solution to this problem is elaborated from considering the plane wave spectrum decomposition of a cylindrical source and then to incorporating in it the solution to the problem of plane wave diffraction by a wedge with different face impedances. The propagation over a plane surface is taken as the special case of a wedge whose angle is equal to 180°. The present solution is however a high frequency asymptotic approximation, and comparisons are thus made with two other available approximate models using instead a point-like sound source.

Noise resulting from road traffic activity is considered as an environmental problem and its control still constitutes a serious challenge to engineers. Artificial barriers or elevated road side banks are thus often used in residential areas for reducing this noise. However, researchers are often in need of theoretical prediction schemes for foreseeing the performance of newly designed noise reducing devices before testing their prototypes in real full-scale experiments. In the case of analytical models, calculations require the combination of a model accounting for sound wave reflections on the ground, and a model for the diffraction of waves by a straight-edged barrier. Furthermore, for assessing the insertion loss of a noise barrier it is also needed to calculate the sound level prior to erecting the barrier. As the ground in the case of traffic noise is often a combination of at least two ground types with different acoustical properties, models are then necessary to deal with such situations. It is therefore the purpose of the present work to present the results of a study made on a theory that has recently been developed for the problem of wave diffraction by a two-impedance wedge, the sound source being considered as linear.

DIFFRACTION OF A CYLINDRICAL WAVE BY AN IMPEDANCE WEDGE

The problem of diffraction by an impedance wedge has been solved exactly by Maliuzhinets for a plane incident wave [1], and the solution for a line source was made through considering the plane wave spectrum decomposition of a cylindrical wave [2]. Figure 1 illustrates the geometry of the wedge, its faces being at the angles 0 and π, and having the impedances \( Z_0 \) and \( Z_\infty \), \( \sin \theta_0 = Z_0/Z_\infty \), \( \theta_\infty \) being the Brewster angles of the wedge faces and \( Z_c \) the impedance of air, i.e. \( Z_c = \rho c \). The incident wave is assumed being \( e^{-i\omega t}/\sqrt{kr} \), \( k \) being the wave-number, and a time dependence factor \( e^{i\omega t} \) is understood and omitted throughout. The total pressure field is the sum of the geometrical components, the direct field \( u_d \), and the reflected field \( u_r \), and a contribution due to the edge diffraction. This component, \( u_d \), may be written as the sum of four terms, i.e. \( u_d = \sum_{i,j=1}^n u_{i,j} \) with:

\[
 u_{i,j} = \frac{e^{-ikr}}{\sqrt{kr}} P_{i,j}(\varphi,\varphi') \bar{P}_{i,j} \left[ \frac{k r' r}{r + r' a_{i,j}} \right] e^{i\omega t} \quad (1)
\]

where:

\[
 P_{i,j}(\varphi,\varphi') = -\frac{e^{-i\pi/4}}{2n\sqrt{2\pi}} \Omega_{\varphi} (\varphi,\varphi') \cdot (-1)^{i-j+1}.
\]

\[
 \cdot A_n \left[ (-1)^i \varphi, (-1)^j \varphi' \right] \cot \frac{\pi + (-1)^i \varphi + (-1)^j \varphi'}{2n} \quad (2)
\]

This expression is a high frequency asymptotic form which further is uniform and in full agreement with the UTD expression for the hard wedge [3]. For details on the expression of \( A_n, \Omega_{\varphi} \) reference is made to the original work or to a more recent application [4].
APPROXIMATE MODELS FOR A SPHERICAL SOURCE

Rasmussen’s Model
This model is for a spherical sound source, and is developed by means of a Green’s function concept. In the original work [5], the pressure field is evaluated by means of a surface integration along the entire vertical plane over the impedance discontinuity. In a subsequent paper [6], the integration was instead replaced by a more practical line integration in the same plane. As compared to other approximations, this model has shown high accuracy of calculations, as well as good agreement with experimental measurements, and for interesting discussions reference is made to [7].

DeJong et. al’s Model
This model is semi-empirical and as for the model above the source-receiver line is supposed to be normal to the line of impedance discontinuity. For a source not too near the ground, and for not too large distances, this model gives also satisfactory results [8].

In the models above, the reflection coefficients on the impedance planes are needed. For the line source use is made of Chandler-Wilde’s model [9], while Thomasson’s model is used for the spherical wave incidence [10].

EXAMPLE OF CALCULATION
As an example, calculation is made on the excess attenuation over a plane ground composed of asphalt-grass. This example may be used for instance to evaluate the efficiency of a noise barrier prior to erecting it on such a ground. A theoretical comparison between the three models is represented in the curves of Figure 2.

The face impedances were evaluated according to Delany-Bazley’s single-parameter model [11] with the resistivities as determined experimentally by Embleton et al. [12]. The model of the present study gives a further contribution to calculations on multi-impedance grounds, and despite its two-dimensional character, it shows fairly reasonable agreement with the three-dimensional models. It may be noted that the two dimensional predictions are somehow underestimated at the higher frequencies and at the interference dips. To achieve a satisfactory convergence in Rasmussen’s model, the upper limit of integration in [6] was taken as \( 80 \lambda \), \( \lambda \) being the wavelength. Unfortunately, the calculations using this model take exceedingly larger amounts of time as compared to the two other models. At low frequencies, it is hard to speculate on the validity of the models as the two-dimensional one is rather a high frequency asymptotic, and Rasmussen’s has shown poor convergence in this frequency range [7]. The position of the sound source near the ground may give rise to the propagation of surface waves. Hence, an improvement of the present model may be achieved through incorporating the effect of this type of waves [13]. Lastly, used in combination with Chandler-Wilde’s model on two-dimensional reflection by an impedance plane [9], the present model can be taken as reliable in calculations on noise barriers as recent studies have confirmed the relative similarity between using either two- or three-dimensional models [14].

REFERENCES
Comparison of measurements and calculations of sound attenuation for real urban situation

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In the paper the results of measurements and calculations of the excess attenuation during sound propagation in built-up area will be presented. The influence of input data and the simplification of the prediction procedures on the calculated results and their agreement with experimental data will be under discussion. The calculation were carried out using geometrical model of sound field in urban areas and a simplified calculation methods acc. to ISO 9613.

INTRODUCTION

The basic documents illustrating the state of the acoustic environment are acoustic maps that are being drawn up mainly with analytical methods. One of the key issues is the problem of calculation accuracy of sound propagation in areas of varied topographic profiles and in urbanised areas. Another issue is the problem of errors resulting from the inaccuracy of input data. In the event of a complex urban situation and a heterogeneous area surface, the result of model calculations is burdened with an error ensuing from a simplified description of the space and its acoustic properties. For real situations in the area, the acoustic parameters of the space, such as: coefficients of reflection from building facades or absorbing properties of the heterogeneous ground surface can be determined only approximately, assuming possible values of these parameters.

The purpose of this work was a comparative analysis of measurement and calculation results gained from calculation methods of a different degree of simplification, and the analysis of the influence of input data on the conformity of measurement and calculation results.

RESEARCH

A relatively simple urban situation was chosen for comparative research – an urban interior formed by a U-shape configuration of three buildings. The distance between two parallel buildings \((63 \times 12 \times 15 \text{ m})\) was 30 m, the distance between those buildings and the perpendicular one \((55 \times 12 \times 15 \text{ m})\) was 25 m and 40 m respectively. The area surrounding the buildings was heterogeneous: internal asphalt and concrete roads and pavements, macadam external roads; the remaining area covered by grass and low vegetation.

Measurements and calculations were done for various combinations of source and observer configurations, which were selected so as to take into account the basic propagation effects or to have a dominant influence of one of the acoustic phenomena for a particular configuration.

The loudspeaker set generating pseudo-random noise and an impulse source (a magneto) were used as the sound sources. In both cases, the geometrical centre of the real sound source was at the height of \(h_s = 0.5 \text{ m}\). The measurement microphones were at the height of \(1.5 \text{ m}\) and \(6 \text{ m}\). The octave-band frequency characteristics were measured using FFT techniques. For impulse source time history of measurement signals were also registered, enabling a later analysis of their structure. For each position of the magneto, four shots were registered in order to check the repeatability of the results.

Simultaneous, the measurements of meteorological conditions were carried out. The air temperature and wind speed at heights of \(2 \text{ m}\) and \(10 \text{ m}\) were measured as well as the atmospheric pressure, relative humidity and wind direction.

The calculations were done by means of: 1) computer programme Envira, based on the geometrical model of acoustic wave propagation, in which all basic elementary phenomena were taken into consideration [1], 2) a general calculation method according to the ISO 9613, which is based on simplified calculation rules. Calculation methods taken for comparison differ not only in calculation algorithms, but also in the way of describing the properties of the real urban situation as an acoustic wave propagation space.

In the theoretical model, properties of the ground surface is characterised by flow resistance \((\sigma)\). In the model a plane wave reflection coefficient \((R_p)\) was assumed. The simplified model is valid for an incident angles \(\varphi > 5^\circ\); for smaller angles it may be a source of...
calculation errors. The space description defines the border of different ground surfaces. For describing the diffraction, the relationships given by Pierce on the grounds of Keller’s theory were used. In the ISO method the ground surface properties are defined by the G parameter for three distinct regions: the source region, the receiver region and the middle region, whose range depends on the source and the receiver heights. Screening correction is determined on the grounds of a semi-empirical relationship.

For the same configurations variant calculations were done in order to determine:

A) influence of input parameters:
- accuracy of the observer and the source localisation – a change of the position within ± 1...5 m, a change of the source height h_s = 0.5 m ± 0.1 ... 0.2 m.
- the absorbing properties of the ground surface – the following variants were assumed:
  - roads and pavements (asphalt, concrete) – hard ground: σ = 2000 [mks units] or G = 0.2...0,
  - macadam road – a moderately reflecting surface: σ = 400...800 [mks units], G = 0.2 ... 0.4 ,
  - grass-covered area – porous ground: σ = 30...400 [mks units], G = 0.2 ... 0.4.

B) simplifications of calculation algorithms:
- with and without taking into account the correlation between signals arriving along different sound paths,
- with and without taking into account the scattering influence of the diffracting edges.

The calculations using Envira programme were done for the 3-th order of image source. In the ISO 9613 method, the following sound paths were considered: direct, reflected (for a single reflection) and diffracted at the side or top edges of buildings. The air absorption coefficient was assumed for temperature T and humidity H occurring during the measurements.

RESULTS

1. The good repeatability of impulse measurements, for d_s < 50 m, was found. For longer distances (d_s > 100 m). The dispersion of measured octave band levels increases – it amounts to 2...10 dB; at the same time, the shape of the measured frequency characteristics is maintained. Average values after rejecting extreme results were taken for comparison.

2. Better conformity of the results of measurements and calculations according to both methods occurs for observer at h_o = 6 m. The greatest differences occur in high frequency range (f_0 > 1000 Hz), which indicates on the influence of scattering on small architectural forms which had not been taken into account in the calculation models.

3. For configurations with a dominant direct sound, the values of excess attenuation (Δ_L_{e}) in octave bands of f_0 = 250 Hz and 500 Hz amounted to 2...15 dB, depending on the distance d_s, sound source location and the calculation method applied. For sound source located near a border of different ground surface (e.g. concrete – grass) Envira calculations render much better evaluation of ground effect than ISO 9613, which points to an error resulting from averaging the values of the G parameter for the source region.

4. For observers located in the shadow zone, both calculation methods well render the character of changes Δ_L_{e} as a function of frequency. However there are considerable differences in estimated values of Δ_L_{e}.

5. For configurations with a dominant direct sound, the values of excess attenuation (Δ_L_{e}) calculated using Envira programme, amounts to:
- change of height and location of the substitute sound source: -3 dB ÷ +7 dB, f_0 = 500 ÷ 4000 Hz,
- change of the ground properties: ±(1...3) dB
- influence of the averaging method: ±(0...3) dB.

CONCLUSIONS

The results of acoustic measurements made in a real situation are burdened with a random error due to temporary changes in the state of atmosphere, and also from many factors whose influence is disregarded in model calculations, e.g. the influence of scattering on small decorative architectural forms. Even for a relatively simple urban situation, the errors of input parameters of calculation models may be greater than the error resulting from simplified calculation algorithms.

REFERENCES

We present statistical properties of the variations in sound pressure level from a point source, induced by meteorological factors. The data set used for the analysis is generated by a simulation model driven by standard climatic data recorded over a 29-year time period. The method is firstly described. The set of hourly sound levels obtained over the 29-year period is then used to perform several statistical analysis and estimate the "long-term" noise level.

INTRODUCTION

The statistical properties of meteorological effects on noise level are not well known. Long-term reconstitution of wind and temperature gradients allows a set of $L_{Aeq}(1h)$ to be obtained over a long period for given location, distance and source/receiver orientation [1]. Statistical properties of noise level can then be estimated from this data, for the specific case considered.

We present results from such a study, focusing on some statistical properties of $L_{Aeq}(6h-22h)$ and $L_{Aeq}(22h-6h)$ for a point source. These variables were chosen because they correspond to the periods used in the French road noise regulation.

REMINDER OF THE METHOD

A first version of the methodology was presented and used in [1]. It was later improved by using an acoustic model taking into account a logarithmic vertical sound profile. The data input are standard climatic data (as provided by local meteorological offices), and the acoustic and geometrical characteristics of the ground and source/receiver position.

The climatic data is fed into a micrometeorological model [2] calculating the hourly vertical gradients of wind speed and air temperature. These gradients, together with wind direction, enable us to reconstitute the time values of sound speed across the vertical source/receiver plane.

The resulting hourly profiles are then used in a numerical acoustic model based on the resolution of the Helmholtz equation by parabolic approximation [3]. We finally obtain a set of $L_{Aeq}(1h)$ allowing us to carry out various types of analysis.

LONG-TERM RESULTS

Propagation assumptions

The results presented here relate to a point source with a road-noise type spectrum of following relative values: (125 Hz, -10 dB), (250 Hz, -6 dB), (500 Hz, -3 dB), (1000 Hz, 0 dB), (2000 Hz, -3 dB), (4000 Hz, -8 dB). This source is located at 0.05 m above the ground (with $\sigma=300$, as used in the empirical model of Delany-Bazley), and at 300 m away from a receiver at a height of 1.2 m. The receiver/source direction is 225 degrees from the North. The meteorological data was measured near Angers (western France). It covers 29 years, from January 1, 1962 to December 31, 1990.

Results of the reconstitution

Figure 1 gives an example of the time variation in $L_{Aeq}(1h)$, obtained by simulation for 4 periods of 48 h. On top of the daily cycle, random fluctuations in noise level, also due to meteorological effects, can be seen.
Statistical characterisation

By grouping the results in equivalent 6h-22h and 22h-6h noise levels, we obtain for each period 10592 values that can be analysed statistically. Figure 2 represents the corresponding cumulative distribution functions. The overall range is as large as 30 dB(A). The night-time levels are higher than their day-time counterparts because of the nocturnal temperature inversion. This representation makes it possible to determine the percentage of time during which a noise level is reached or exceeded. Using real data instead of relative values, these functions can be compared, for example, with regulation levels.

Moreover, this error is further reduced if the considered 6-month period is the period between the winter solstice and the summer solstice. The results are roughly similar for day-time conditions.

In case 2 we studied the possibility of estimating $L_{A_{eq}}(T)_{LT}$ from a measurement over only one month, but with the help of a statistical model. The model was established over 20 years and the validations performed over the remaining 9 years. The best regression form was found as:

$$L_{A_{eq}}(T)_{LT} = a_0 + a_1 L_{A_{eq}}(month) + a_2 L_{A_{eq}}(month)^2$$

where $L_{A_{eq}}(month)$ is the equivalent level measured over the considered month and $a_0$, $a_1$ and $a_2$ are empirical regression coefficients.

As an example, Table 1 gives the regression coefficients and the mean absolute error (M.A.E.) in dB(A) obtained for each month over the period 6h-22h.

**Table 1.** Estimation of $L_{A_{eq}}(6h-22h)_{LT}$ from a measurement period of only one month.

<table>
<thead>
<tr>
<th>Month</th>
<th>$a_0$</th>
<th>$a_1$</th>
<th>$a_2$</th>
<th>M.A.E.</th>
</tr>
</thead>
<tbody>
<tr>
<td>January</td>
<td>-180.1</td>
<td>+7.59</td>
<td>-0.059</td>
<td>0.08</td>
</tr>
<tr>
<td>February</td>
<td>-71.3</td>
<td>+4.02</td>
<td>-0.030</td>
<td>0.33</td>
</tr>
<tr>
<td>March</td>
<td>+241.5</td>
<td>-5.78</td>
<td>+0.047</td>
<td>0.14</td>
</tr>
<tr>
<td>April</td>
<td>+217.6</td>
<td>-5.01</td>
<td>+0.041</td>
<td>0.03</td>
</tr>
<tr>
<td>May</td>
<td>+87.9</td>
<td>-0.76</td>
<td>+0.006</td>
<td>0.09</td>
</tr>
<tr>
<td>June</td>
<td>-154.5</td>
<td>+7.23</td>
<td>-0.060</td>
<td>0.08</td>
</tr>
<tr>
<td>July</td>
<td>+306.2</td>
<td>-8.21</td>
<td>+0.069</td>
<td>0.07</td>
</tr>
<tr>
<td>August</td>
<td>-478.2</td>
<td>+17.73</td>
<td>-0.145</td>
<td>0.11</td>
</tr>
<tr>
<td>September</td>
<td>+238.5</td>
<td>-5.81</td>
<td>-0.048</td>
<td>0.11</td>
</tr>
<tr>
<td>October</td>
<td>+168.5</td>
<td>-3.45</td>
<td>-0.028</td>
<td>0.06</td>
</tr>
<tr>
<td>November</td>
<td>-94.7</td>
<td>+4.78</td>
<td>-0.036</td>
<td>0.16</td>
</tr>
<tr>
<td>December</td>
<td>+78.2</td>
<td>-0.66</td>
<td>+0.006</td>
<td>0.20</td>
</tr>
</tbody>
</table>

It is thus found that one can estimate a "long-term" noise level with a satisfactory accuracy, from measurements performed over one month only.

**REFERENCES**

Calculating Sound Impulse Responses in City Streets.

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A hybrid method has been used to calculate the impulse response to sources in city streets. The TLM method, which takes wave effects into account, has been used for the low part of the spectrum, while ray tracing has been used for the higher.

INTRODUCTION

In order to assess the efficiency of noise reduction measures in city streets, auralisation based on numerical simulations, can be an efficient technique. With this technique, we have to construct a numerical model taking the important part of the geometry into account, and study the acoustic propagation in such a model. Binaural listening can be achieved by convolution of calculated impulse response functions with appropriate source signals, for instance the sounds of cars and trucks recorded under special conditions.

NUMERICAL MODEL

In our project we have used two different numerical techniques for the calculations, both implemented in the time domain. The high frequency part of the impulse response function has been calculated under the assumptions of geometrical acoustics by a ray-tracing program [1]. As wave effects like diffraction have important effects at low frequencies, this part of the spectrum had to be treated by a different technique. The Transmission Line Modelling (TLM) [2,3] method was chosen. This method requires a full 3-dimensional mesh of the calculation domain, but has good stability characteristics and is easy to implement.

Details of the TLM model

3-dimensional modelling up to some hundred Hertz in a typical urban canyon (see figure 1), requires a large computer memory and calculation time. One is therefore obliged to do the calculations within a limited domain, as indicated by the planes ABCDEF, EDJK, and GHIJKL. In some cases one can profit from symmetry by making the first plane perfectly reflecting, while the other two should be transparent. A TLM model node is shown in figure 2.

A point source can be implemented by specifying pulses leaving such a node in the 6 directions: e, w, n, s, u, d. The TLM model regards a source to be of the velocity type. Therefore, to obtain a pressure impulse response, (the response to a delta function) the system has to be excited by a step function. Updating each node at discrete time steps defines the sound propagation. New values at the node i, j, k are defined by relating them to pulses leaving the neighbouring nodes at the previous time step by simple transmission line rules:

\[
\begin{align*}
    p_{i,j,k}^e &= 1 & p_{i+1,j,k}^e &= 1 \\
    p_{i,j,k}^w &= 1 & p_{i,j+1,k}^w &= 1 \\
    p_{i,j,k}^n &= 1 & p_{i,j,k-1}^n &= 1 \\
    p_{i,j,k}^s &= 1 & p_{i,j,k}^s &= 1 \\
    p_{i,j,k}^u &= 1 & p_{i,j,k}^u &= 1 \\
    p_{i,j,k}^d &= 1 & p_{i,j,k}^d &= 1 \\
\end{align*}
\]

The transparent planes EDJK and GHIJKL must have as good a sound absorption as possible. The boundaries are defined at the middle of a transmission line, thus half a cell...
away on the outside of the last nodes of the grid. In the TLM formulation, some local reflection is necessary from the boundary to achieve maximum absorption. This reflection might be estimated by Taylor expansion of the reflections from neighbouring nodes at previous time steps [4].

**IMPULSE RESPONSES**

The impulse responses at chosen microphone positions were calculated by the two methods. The combination of the two responses were done in the frequency domain by low pass filtering of the TLM signal’s Fourier transform and high pass filtering of the one given by the geometrical acoustics model (see figure 4). Adjustment of signal levels was also required. This was done by calculating free space sound transmission and matching the levels obtained by the two methods in the low frequency part of the spectrum. It is in this region the TLM model has its smallest cell length to acoustic wave-length ratio and therefore will give results with little numerical dispersion. The final impulse response could then be obtained by the inverse Fourier transform of the combined spectrum. Figure 5a shows the total impulse response at a microphone position not visible from the source (behind a screen). The diffracted sound obtained by the TLM model is seen as the first part of the signal. The impulse response based only on geometrical acoustics is seen in figure 5b.

![Figure 3](image3.png)

**Figure 3** shows a snapshot of the direct and reflected wave pulses in an urban canyon with a low barrier calculated by the TLM method. The source is a point source situated underneath the rectangular box. The axes are numbers of cells.

![Figure 4](image4.png)

**Figure 4.** The combined Fourier transform of the low frequency (TLM) part, and the high frequency geometrical acoustics part. The cut-off frequency is 250Hz.

![Figure 5a](image5a.png)

**Figure 5a** Total impulse response at a microphone position not visible from the source.

![Figure 5b](image5b.png)

**Figure 5b** Geometrical acoustics part of the impulse response seen in figure 5a.

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Localization of Infrasonic Signals in an Urban Environment

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The US Army Research Laboratory has operated two arrays of microphones designed to resolve frequencies less than 10 Hz, with continuous signal processing for nearly three years. A first array (operated since 1998) has been combined with a second array located about 1 km away. Signal processing for these arrays includes adaptive minimum variance distortionless response (MVDR) beamforming and a localization algorithm. The first array is a centered triangle of four microphones, while the second array is a cross of five microphones. Both acoustic arrays consist of Chaparral Physics microphones spaced 20 m apart. Each microphone has six 20-foot lengths of porous garden hose attached radially for wind noise suppression. Data from these arrays has been localized to determine source locations. Source signals include the local utility infrastructure, local explosive testing, over-flight of private, commercial, and military aircraft. Discussion will include our ability to locate known sources. We will illustrate the ability of the large array's and the localization algorithm to accurately locate signal sources.

BACKGROUND

By selecting an array geometry of 20m spacing we have directivity in the frequency range of 3 - 8 Hz. The study of the infrasonic energy available at these frequencies includes signatures of many natural events and man made machines such as; thunder storms, power stations, aircraft and gas supply lines\cite{1}. We have chosen the Chaparral Physics Model 2 microphone as the sensor for the arrays. We have implemented a 24-bit analog to digital converter. Consultation with Dr. Al Bedard of NOAA and Dr. Rod Whitaker of Los Alamos National Laboratory has suggested the application of a radial arrangement of porous garden hose for the reduction of wind created background noise.

SIGNAL PROCESSING

Data was collected at 100 Hz and filtered to 25 Hz, however, we only processed data to 8 Hz to avoid grating lobes. The Average Spectral Coherence (ASC) across the array was employed as a signal detector\cite{2}. An adaptive Minimum Variance Distortionless Response (MVDR) beamformer was used. The relationship used for the MVDR beamformer (Equation 1) can be found in many references\cite{2}.

\[
P_{\text{MVDR}} \left( f, k \right) = \frac{1}{A^T R^{-1} A}
\]

\(A\) in Equation 1 is the steering vector and \(R\) is the correlation matrix. Once ASC indicates that a signal is present the beamformer creates 120 beams. Beamforming over the wavenumber range based on the measured ambient temperature and the frequency range of interest (3 to 8 Hz) the direction of arrival is selected by determining the maximum amplitude reaching the array. The direction of the maximum beam power is thus determined to be the direction of signal propagation. Once direction is established the beamforming process is extended to consider the full wavenumber range and a range of apparent propagation speeds or slowness \(\left( \frac{\text{angle}}{\text{frequency}} \right) \). This slowness value is the apparent propagation speed, which infers the angle of elevation of the incoming signal\cite{3}. The current integration period on the infrasonic baseline is longer than the propagation delay so we can assume that transient arrivals will appear simultaneously at both sites. For each given time slice, the localization algorithm forms all combinations of LOB pairings from the available sites.

DETECTIONS

Figure 1 shows the reported arrival angle in degrees of the LOBs versus time in seconds. The cluster of LOB's at 60 degrees comes from a power station over 38 km away, while the slightly tilted LOB's from 180 to above 270 degrees have been generated by overflying aircraft. The solid LOB's after mid-graph were created by the testing of an automatic gun about 13 km away.
LOCALIZATIONS

We have attempted to take the LOB's shown in Figure 1 above and create localized results to define source locations. The two arrays are spaced 1 km apart, consequently we do not expect to be able to localize much beyond 10 km away. It was found that, in order to localize signals coming from much outside 10 km from the arrays, we would need to beamform to 1 degree increments. Results showed that the localization needs further work. The algorithm seems to display the proper angle to the source, but not the correct location.

CONCLUSIONS

Our goal was to localize out to perhaps 20 km. We must have 1 degree accuracy from the arrays to accomplish this and clearly we are not achieving that. From a first look at localization performance, we would have suggest that for two arrays to accurately determine the source of such a signal and track it, the arrays would need have a separation distance of at least 40 km. As time progresses and more data is process these requirements will become more accurately defined.

REFERENCES

A New Model of Ground Effect

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The ground effect is an essential part of noise propagation in the atmosphere. In many cases, it’s very difficult to describe the interaction between the noise and the ground surface. Sometimes the accurate source’s height isn’t known. Therefore simple models of the ground effect are still in use [1, 2, 3]. On the basis of exact theory, a new model of the ground effect for a point source was developed, which can be applied for a train noise. Free parameters were estimated for “typical railway traffic power spectrum”.

INTRODUCTION

Railway noise is one of the most annoying in the environment. Annoyance of railway noise can be assessed in terms of the equivalent continuos time-average sound level,

\[ L_{Aeq} = 10 \cdot \log \left( \frac{1}{T} \sum_{i=1}^{n} N_i \cdot 10^{0.1 L_i} \right), \quad t_o = 1 \text{ s}, \]

where \( L_{Aeq} \) represents the sound exposure level for the \( i \)-th category of trains (i.e. intercity, passengers, goods), and \( N_i \) express the number of noise events during the time period \( T \). The sound exposure level depends on following wave phenomena: geometrical spreading, ground effect, turbulence, air absorption and refraction. Close to the source the ground effect is the most important factor and the other phenomena can be neglected.

THEORY

The moving train can be modeled by the continuous line of incoherent point sources with the dipole directivity [4]. Sound exposure for the line source of length, \( l \), which moves with steady velocity, \( V \), may be written as [5],

\[ E_s = \frac{l}{V l_o} \int_{-\frac{\pi}{2}}^{\frac{\pi}{2}} \frac{3W_s \rho c \cos^2 \Phi}{4\pi D} \cdot G_s d\Phi, \quad l_o = 1 \text{ m}, \]

where \( D \) is the horizontal distance between the track and the receiver and \( W_s \) is the A-frequency weighted sound power of the unit length of the line source.

Sound exposure (eqn (2)) takes into consideration the exact theory of the ground reflection, which is given by

\[ G_s = \sum_{n}^{} \frac{W_n}{W_s} G_n (D, H_s, H_o, f, Z_n), \]

where \( W_n \) is the sound power in the \( n \)-th frequency band and \( G_n \) is the exact form of the ground effect determined by the Weyl-Van der Pol solution [6]. Function \( G_n \) depends on the source’s height, \( H_s \), the receiver’s height, \( H_o \), and the ground impedance for the \( n \)-th frequency band, \( Z_n \). Sometimes the ground impedance is unknown. Therefore, the simple form of the ground effect function must be used. The exact form of \( G_n \) (eqn (3)) may be approximated by the following expression

\[ \tilde{G}_n = \frac{K}{1 + a \cdot e^{(b \cdot H)}} \cdot D^2, \]

where \( K, a, b \) are free parameters and

\[ H = (H_s + H_o)/2, \]

is the mean height propagation. Using expression (3) and (4), one can estimate free parameters of the model.

RESULTS OF CALCULATIONS

The calculations of free parameters were performed for “typical railway traffic power spectrum” [7] and various ground surfaces (characterized by the effective flow resistivity, \( \sigma \) [8]). Parameters of propagation: \( \sigma = 0.2\pm0.2 \text{ m}, \quad H_s = 7.2\pm10.2 \text{ m} \quad \text{and} \quad d = 10\pm1000 \text{ m}. \) The variations of \( G_s \) (eqn. (3)) and \( \tilde{G}_s \) (eqn. (4)) are shown on Figure 1.
The obtained values of $a$, $b$ and $K$ are listed in Table 1.

Table 1. The values of the free model’s parameters (eqn. (4))

<table>
<thead>
<tr>
<th>$\sigma$ [kPa·s/m²]</th>
<th>$a$</th>
<th>$b$</th>
<th>$K$</th>
</tr>
</thead>
<tbody>
<tr>
<td>20</td>
<td>15.0·10⁻¹</td>
<td>79.0·10⁻¹</td>
<td>1.5</td>
</tr>
<tr>
<td>35</td>
<td>49.9·10⁻³</td>
<td>65.9·10⁻¹</td>
<td>1.4</td>
</tr>
<tr>
<td>50</td>
<td>18.5·10⁻²</td>
<td>54.8·10⁻¹</td>
<td>1.4</td>
</tr>
<tr>
<td>100</td>
<td>21.8·10⁻¹</td>
<td>34.7·10⁻¹</td>
<td>1.4</td>
</tr>
<tr>
<td>300</td>
<td>99.3·10⁻⁵</td>
<td>15.8·10⁻¹</td>
<td>1.3</td>
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<tr>
<td>500</td>
<td>21.8·10⁻²</td>
<td>93.0·10⁻²</td>
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</tr>
<tr>
<td>1000</td>
<td>39.0·10⁻⁶</td>
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<td>2000</td>
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<tr>
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<tr>
<td>5000</td>
<td>30.4·10⁻⁷</td>
<td>28.8·10⁻²</td>
<td>1.7</td>
</tr>
</tbody>
</table>

**EXPERIMENT**

To validate the model of the ground effect presented here, measurements and calculations of the railway noise were performed. The trains were moving with steady speed, $V$, on an embankment of height $H_s = 1$ m. The terrain between source and the receiver was open, without buildings or any other reflecting objects nearby. Three simultaneous measurements of the sound exposure level, $L_{AE}$, were performed at the distance $D_1 = 25$ m, $D_2 = 100$ m and $D_3 = 104.5$ m, with microphones at the heights: $H_{\omega}^{(1)} = 1.2$ m, $H_{\omega}^{(2)} = 1.3$ m and $H_{\omega}^{(3)} = 1.5$ m, respectively. From the measurement of $L_{AE}(D_1, H_{\omega}^{(1)})$ we calculate the sound power level, $L_{WA}$ [9]. Then, using the exact theory [6], the ISO model [1], and new theory model presented here, we calculated the sound exposure level, $L_{AE}$. Figure 2 presents comparison of the calculated and measured values $L_{AE}(D=100$ m, $H_{\omega}^{(1)} = 1.3$ m). The line indicates the ideal relationship between the levels for which the prediction corresponds exactly to the measurements. The best agreement between calculated and measured values of $L_{AE}$ yields the new model.

**CONCLUSIONS**

In the present study, a new model of the ground effect has been developed. Free parameters were estimated on the basis of the exact theory, for “typical railway traffic power spectrum”. The field measurements indicate a good agreement between the new model and measured values of sound exposure level, $L_{AE}$.

**REFERENCES**