Time-domain modeling for impulse source localization in urban environments

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Summary
The localization of acoustic sources with a network of distributed microphones is difficult in urban environments, due to the complex propagation effects such as reflections and diffractions on buildings. The present study discusses our efforts toward localizing impulse sources using a Finite Difference Time-Domain (FDTD) model. Our model is evaluated against theory and observations in presence of urban-type obstacles. The computational burden is made tractable on the basis of comprehensive physical considerations. Two approaches for localization are introduced. The first approach essentially relies on the time reversal of the microphones measurements with the FDTD model. In the second approach, the source is localized by matching the observed characteristics to a look-up table obtained from time-domain simulations for various source positions. The two approaches are tested and compared with impulse measurements in an urban area, and their respective sensitivities are discussed.

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1. Introduction

The localization of sound sources is a topic of wide interest, with civilian needs e.g. in building engineering as well as defense applications, e.g. to localize shots or explosions. The hardware and software of available acoustic sensing systems are adapted to the characteristic properties of the target sounds (frequency range of interest, loudness etc). They must also account for the physical processes related to outdoor propagation from the sound source to the system sensors. Cheinet and Broglin [1] recently demonstrated that the localization performance of shot sensing systems critically depends on propagation in open environments, due to atmospheric refraction and reflections on trees.

The modulations caused by urban environments combine reflections and diffractions on buildings. They alter the times and angles of arrival of the acoustic waves, and deter the use of the standard straight-line propagation assumption of many shot localization systems. Extended antennas may be developed with distributed sensors in the urban environment, either communicating to a central unit, or forming a network. However, even when the hardware can be adapted, the data processing and accounting for urban propagation effects is still a challenge.

In the last decade, the Finite Difference Time Domain (FDTD) numerical modeling approach has been used to numerically solve the linearized Euler equations in three dimensions and time. It is capable of reliably capturing the urban propagation effects. It can ingest arbitrarily complex urban geometries, with impulse or continuous noises. This versatility however comes with a large computational time. The promises of using FDTD to localize sound sources largely meets with the challenge of developing simulation strategies which decrease the computational cost and/or circumvent its real-time application.

The present study aims at presenting some recent progresses in that direction. It is composed as follows. Section 2 briefly introduces the FDTD model used in this study, and evaluates it against theory and observations. In section 3, two approaches for localization from FDTD simulations are introduced, tested and compared. The tests use some impulse sound measurements in an urban area. Section 4 summarizes the results.

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2. The FDTD model

The model used in this study is hereafter referred to as the ISL Time-domain Model (ITM), as detailed e.g. in Ehrhardt [2]. It numerically resolves the linearized Euler equations in the atmosphere, and the counterpart equations in porous grounds. The prognostic, three-dimensional variables are the acoustic pressure and the velocity of the acoustic particle. They are calculated over half-staggered meshes. The simulation forcings include:

- The sound sources, specified by the time series of the acoustic pressure,
- The boundary conditions: perfectly matched layers absorb radiating waves, reflection and periodicity are also implemented,
- The three-dimensional atmospheric parameters: wind, density, sound celerity,
- The three-dimensional ground parameters.

The temporal integration uses a Runge-Kutta scheme of order 4. The spatial derivatives and interpolations are evaluated with fourth-order central differences. The temporal and spatial steps of the model are calculated from the smallest acoustic wavelength to be resolved. The model is hereafter run with the MPI parallelization standard, using 128 processors of a computational cluster.

The ITM model has already been evaluated in various configurations: geometrical dispersion, reflection over a perfectly reflecting ground and over a finite-impedance ground, refraction by the wind, propagation through turbulence [2]. We here discuss some evaluation tests specific to urban propagation.

The first test was detailed by Ehrhardt [2]. It addresses the diffraction of a monochromatic sound over a thin barrier with a perfectly reflective ground, as simulated with a 2D version of the ITM. An excellent agreement was obtained between the ITM simulation and an analytical solution for this scenario. We have found a comparable agreement with the 3D version of the ITM.

The second test is more complex; it follows from a FDTD model evaluation originally proposed by Liu and Albert [3]. Specifically, it addresses the propagation of an impulse sound around a corner wall (two perpendicular barriers). Figure 1a shows the acoustic pressure as the wave front passes around the corner wall, as obtained in the ITM 3D simulation. The simulation has been compared to the microphone measurements of Albert and Liu. A good agreement is obtained between the simulated and recorded acoustic pressure time series (Fig. 1b,c), both in terms of timing and in terms of wave shape.

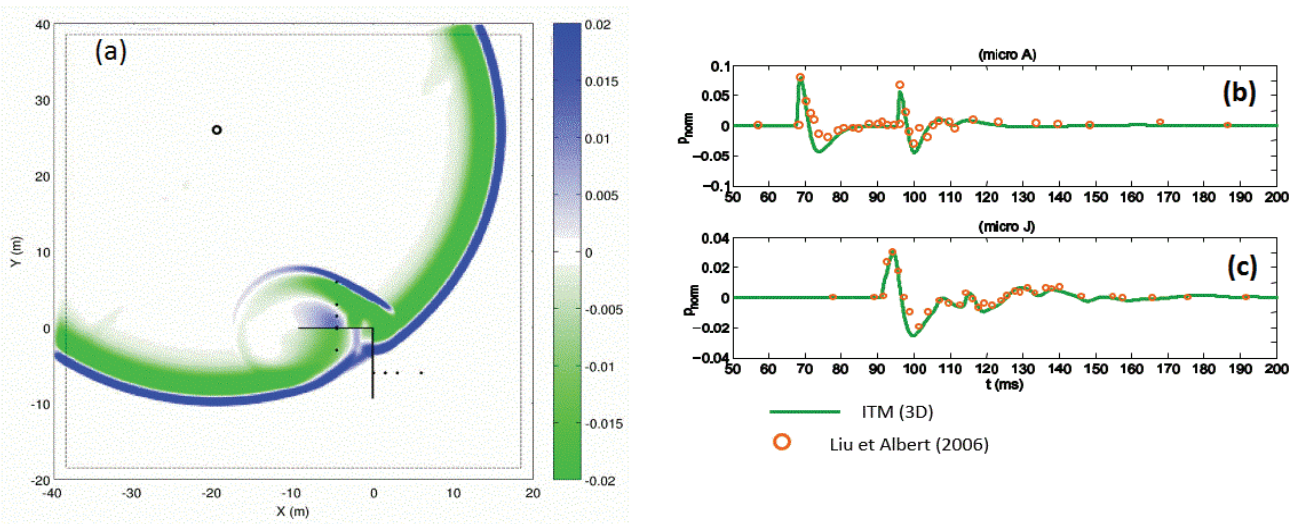


Figure 1: Sound levels (Pa) simulated with ITM (3D version), in the case of an impulse sound propagation near a corner wall. (a) At 1m above ground, the black circle gives the source position. Also shown are comparisons between the ITM simulation and the measurements of Liu and Albert at two points, (b) in front of the corner wall (the micro A of Liu and Albert, with a reflection) and (c) behind the corner wall (micro J, in a shadow zone).

The third evaluation test is based on an experiment made in June 2012. The observational area includes 15 buildings of heights from 4 to 10m (Fig. 2). The acoustic source is a propane cannon. Its impulse emission is horizontally isotropic with a characteristic frequency of 1000Hz. Two source positions are tested. Synchronized acoustic recordings are made at 16 positions, including in complex configurations (several buildings between the source and microphone).

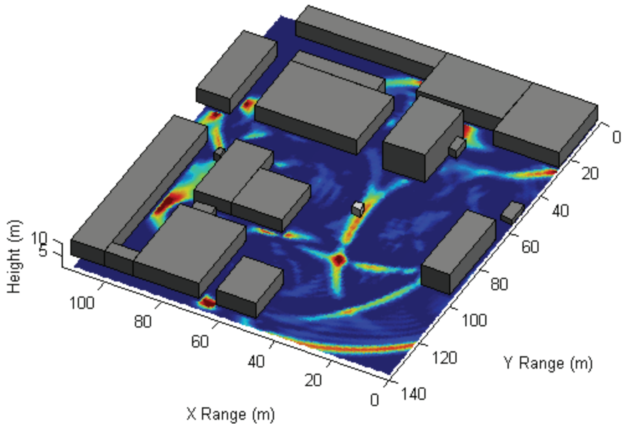


Figure 2: Acoustic pressure at 1m above ground, as simulated by ITM in 3D, 0.2s after emission from the source (located at the white cube). The grey areas indicate the buildings as modelled in the simulation.

Ehrhardt [2] evaluates the 2D version of the ITM in this scenario. Here we reach a 3D simulation thanks to frequency truncation. If the considered highest frequency in the simulation is divided by a factor d, the temporal and spatial steps can be multiplied by d so the computational time for a 3D simulation is divided by d^4. Figure 2 illustrates the acoustic pressure propagating from an idealized impulse signal of characteristic frequency 20Hz, i.e. with $d \approx \frac{1000}{20} = 50$ compared to the real cannon signature. This 3D simulation uses a spatial mesh of 1m and lasts some minutes. The frequency truncation receives a physical support in urban environments, since the diffraction due to urban obstacles tends to act as a low-pass filter. The propagation of low frequency signals is also less sensitive to geometrical details of the urban obstacles, which may be difficult to document.

The simulated signal is different from the observation due to the low-pass filtering. Still, the Times Of Arrival (TOA) should match, as the wave celerity does not depend on frequency. Figure 3 shows that the ITM proves reliable in reproducing the TOAs. The uncertainties could be decreased by refining the modeled building geometries and the TOA quantification – in the observations (e.g. multiple arrivals) and in the simulation (the wave front spans over 0.02s).

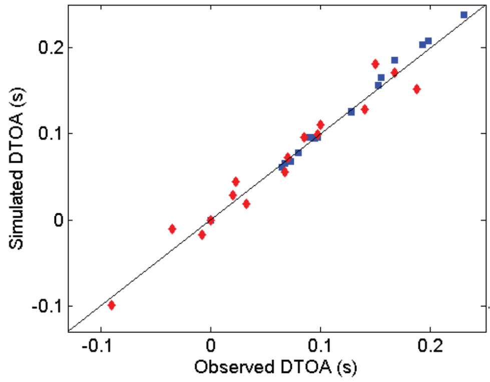


Figure 3: Comparison between observed and simulated TOA of the first arrival at the 16 microphones (times are relative to the TOA at one microphone). The two colours stand for the considered source positions.

The acoustic signatures of the propane cannon and of the pulse used in our simulation both contain frequencies of the order of 20Hz. It is therefore possible to compare the microphone-to-microphone transfer functions observed and simulated at frequencies sufficiently low to enter the ITM resolution, say, below 30Hz. These transfer functions vary by 20dB in the frequency band 10-30Hz (Fig. 4). This illustrates the impact of propagation processes in the considered urban environment. The ITM model captures these large variations, which provides a further evaluation of it.

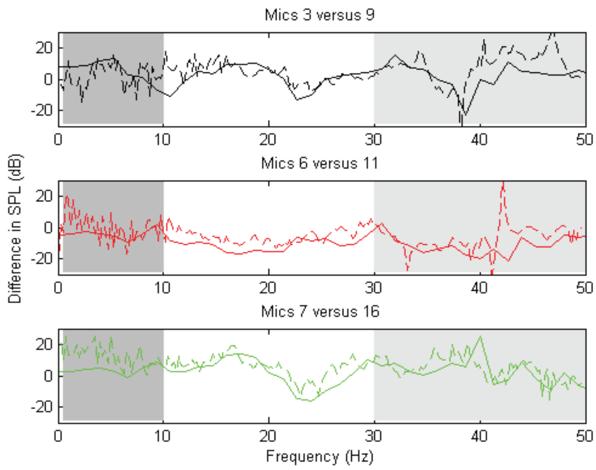


Figure 4: Difference (in dB) between the transfer functions at two microphones, with the simulation (full lines) and the observations (dashes). The non-grey area gives the frequency range of validity of the comparison. Below 10Hz, the observations are affected by the ambient noise and the bandwidth of the microphones.

3. Source localization

The complex acoustic field of fig. 2 hints the challenge of localizing an impulse sound source in presence of urban propagation effects. This section addresses the issue on the basis of the FDTD simulations, and process the observations of June 2012. In a search for representative configurations toward applications, the ITM will be configured so that localization (at receipt of acoustic recordings) lasts less than 10 seconds.

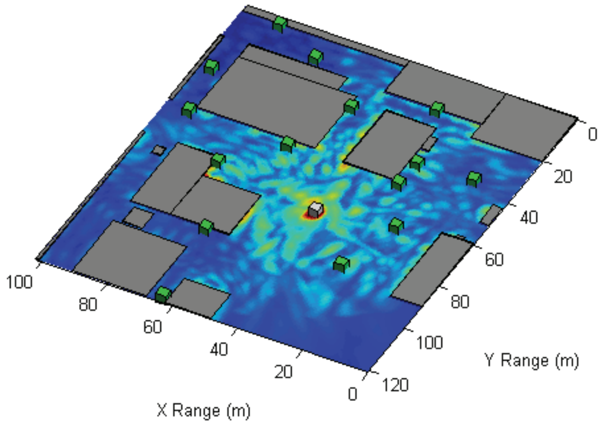
3.1. Time reversal localization

Following Albert et al. [4], time reversal can be used to localize the source. The principle is as follows. At occurrence of an acoustic event of interest, the acoustic pressure is synchronously recorded at various locations in the considered environment. The recordings are communicated to a processing unit. Using microphones as sound sources, the unit numerically simulates the propagation of the time-reversed recordings. A criterion to detect the constructive interference of the various sound contributions supports the localization of the original sound source.

To reach a tractable computational time, we use the 2D version of ITM with a spatial mesh of 0.4m and a time-step of 0.8ms to time reverse the observed signals, implicitly neglecting all propagation paths over roofs. The source signal is modeled from a near-source observation resampled at the model time-step. We use the maximum absolute acoustic pressure after the peak emission of the closest microphone to the source as a proxy to the localization criterion.

Figure 5 is obtained with a time reversal of 15 microphones - we ignore the microphone located at the source position. With the 2D set-up, low frequency truncation and MPI parallelization, the simulation lasts some seconds. After all microphones have emitted their time-reversed sequence, the acoustic pressure field clearly shows a large magnitude peak at the time and location of the cannon explosion. The source can thus be localized in this configuration.

The above localization is for a source centered in the microphone network, not too close to buildings, with 15 microphones. The time reversal based on 5 microphones (not shown) is far more uncertain, with localization peaks within a 20m-radius disk around the real source. Besides, as shown on fig. 5, a localization error of 30m is obtained based on signals recorded with the source on the second position, 1m from a building and more masked from the majority of microphones. In this case, the signal emitted at one microphone (the one in 60,120) dominates over the others, its reflections over buildings create constructive interferences, the magnitude of which are larger than the interference among microphones.



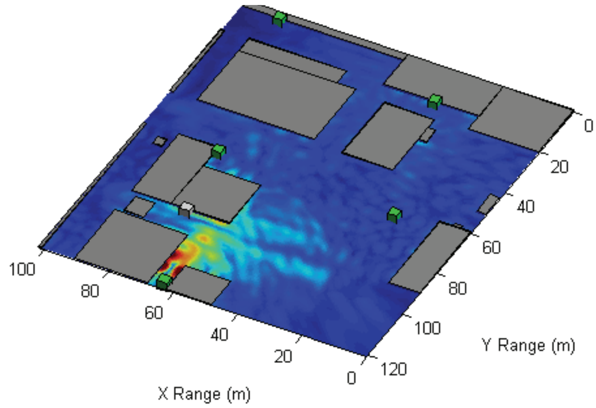


Figure 5: Maximum absolute acoustic pressure as simulated by the ITM time reversal of observed recordings. (Top) with 15 microphones, (bottom) with 5 microphones. The green (respectively, white) cubes denote the positions of the receiving / emitting microphones (resp., of the source).

Our results suggest that our implementation of time reversal can be used to localize impulse sound sources in urban environments, to the extent that the sensing network is relatively dense. This limitation is not obtained by Albert and Liu [4]. They use FDTD to generate the forward-propagated signals. In comparison, we time reverse some observed signals. Our processing is thus impacted by the ambient noise in the signals and by the signal low-pass filtering. Besides, the forward propagation is with the real, 3D urban environment (i.e. with trees, cars, finite impedance ground and facades etc), whereas our backward propagation is with the modeled, 2D environment. These limitations, inherent to the processing of real data, are not met in the model approach of Albert and Liu [4].

3.2. Matched fingerprint processing

An alternative approach has been investigated, in lines with the so-called matched field processing, and also sometimes referred to as a fingerprint approach. The rationale is to form a database of simulated signals characteristics at the microphone positions, for all possible source positions. The simulations are all compared to the measurements, the best match simulation gives the source position. Our implementation has 3 key steps:

(A) As a preparatory step, we define a grid of all possible source locations, with index i. For each source position, we perform a simulation of an impulse sound propagation [i.e. we approximate the impulse response of the medium]. The TOA of the first arrival is calculated at all microphone positions (index j, j=1..M, with M the number of microphones). The overall procedure fills the TOA matrix $t_{i,j}$. Numerical simulation is the method of choice to constitute this database. We use the 3D version of the ITM model with the set-up used in Sec. 2.

(B) At occurrence of an event, the acoustic pressure is recorded at the microphones. A microphone-level processing derives the TOA of the first arrival $\widehat{t_j}$, the peak pressure of the first arrival $\widehat{p_j}$, the background noise $\widehat{n_j}$, and the signal-to-noise ratio $\widehat{s_j} = \widehat{p_j}/\widehat{n_j}$ at the considered microphone. The hat stands for observations. The quantities $\widehat{t_j}$ and $\widehat{s_j}$ are communicated to a processing unit.

(C) The microphone j1 of maximum $\widehat{s_j}$ is used as reference. Let us define the differences in TOA at microphones j vs. j1:

$$\Delta t_{i,j} = t_{i,j} - t_{i,j1}, \quad (1)$$

$$\widehat{\Delta t_j} = \widehat{t_j} - \widehat{t_{j1}}. \quad (2)$$

For the simulation with the source at the correct position, the simulated and observed differences are approximately equal for all j. Let us define:

$$f_i = \sqrt{\frac{\sum_{j\neq j1}\left(\Delta t_{i,j} - \widehat{\Delta t_j}\right)^2}{M-1}}. \quad (3)$$

This function scales the timing error made when comparing the TOAs simulated for source at position i with those observed. The TOAs of signals with lowest signal-to-noise ratio are less reliable in Eq. (3). Therefore we introduce:

$$\tau_i = \sqrt{\frac{\sum_{j\neq j1}\widehat{s_j}\left(\Delta t_{i,j} - \widehat{\Delta t_j}\right)^2}{\sum_{j\neq j1}\widehat{s_j}}}. \quad (4)$$

The index of minimum τ gives the source position.

In practice, the possible sources grid is taken equidistant with a 10m mesh. This yields a hundred ITM simulations in step (A). For each simulation, we save the TOA over a grid of all possible microphone positions. The database can thus be used with an arbitrary network of microphones (number, positions). The microphones grid is equidistant with a 5m mesh. The above meshes pose a limit to the localization performance of the method. They are selected in lines with the positioning accuracy obtainable with standard techniques and desirable for applicative purposes. The localization (steps B and C) takes 0.05s, i.e. it is real time.

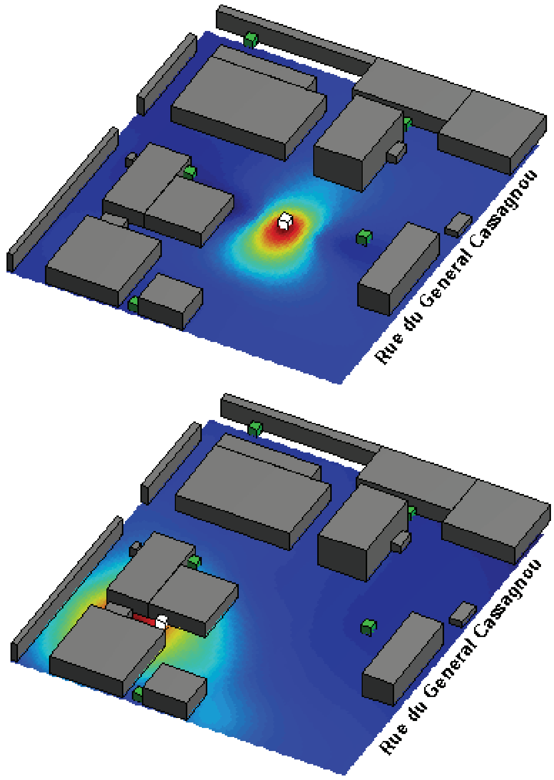


Fig. 6: Localization criterion (m^{-1}) within the matched TOA approach, with 5 microphones measurements with (top) source in position 1, (bottom) source in position 2. The panels are interpolated from the 10x10m source grid. The 3D appearance of buildings hints that the approach is based on 3D simulations.

Figure 6 shows the localization criterion $1/c\tau_i$ for all source positions - c denotes the sound speed. The localization is shown from measurements from 5 microphones, with the source in the two considered positions. In both cases, a well-behaved maximum of the localization criterion is obtained (compare with fig. 5). The matched TOA approach provides an excellent localization for source in position 1 – as is obtained with 15 microphones. The localization for source in position 2 has an error of the order of 10 meters, i.e. one mesh of the source grid.

4. Summary and discussion

Urban environments alter the characteristics of the signals emitted from an acoustic source, and hamper source localization. The FDTD modeling approach of the linearized Euler equations has proven reliable in capturing these urban propagation effects. This study presents some progresses in order to use this propagation model to support the development of source localization algorithms in urban environments.

We introduce our FDTD model, and find that it reproduces diffractions and reflections around urban obstacles, in idealized as well as real conditions. The 3D simulation of urban propagation is made computationally tractable by limiting the resolved frequencies. This approach proves still reliable in reproducing the times of arrival and propagation effects at low frequencies. It ignores fine geometrical details of the urban obstacles, which may be difficult to document.

The simulations are used to investigate localization processing. In order to obtain meaningful results for applicative purposes, the model configurations are chosen so that localization is obtained in less than ten seconds. Two localization approaches are tested. The first uses the FDTD model to time reverse the observed signals, with a further localization criterion. The second is based on matching the observed TOAs in a database of pre-formed FDTD simulations with known source position.

The approaches are evaluated against acoustic measurements in an urban area. They both allow source localization with a large number of microphones. The matched fingerprint approach is found to be more robust to the number of microphones and/or masking of the source, with errors limited to 10m in unfavorable configurations. The notably larger uncertainties obtained here with time reversal are also reported by Ehrhardt [2] without frequency truncation and with various localization criteria.

The time reversed pressure signals include measurement noise and are subject to a second round of propagation loss. Conversely, the match fingerprint approach analyzes the direct pressure fields. Besides, time reversal implies an accurate and on-line propagation calculation, with a trade-off between computational time and realism of the simulation - e.g. the 2D assumption ignores all contributions diffracted at building tops. In comparison, in our match fingerprint approach, the TOA database is calculated in advance (off-line), with a 3D sound propagation model. Efficient models could accelerate these calculations. The localization step is comprehensive and real-time.

The time reversal approach requires that the full pressure time series at each microphone is transmitted to the central unit. The matched fingerprint approach requires transmitting two values (TOA, signal-to-noise ratio). This difference stems from the derivation of the useful acoustic metrics at the sensor unit level. The fingerprint approach can integrate match on additional metrics, e.g. the relative weights of frequency bands, the TOA of late arrivals or the angles of arrival (as measured with antennas, e.g. Polprasert et al., [5]). With such extensions, this approach could in principle operate with one microphone – as could the time reversal approach.

Last, in terms of hardware, both methods use sensors distributed in the environment. The sensor positions are known, the sensors communicate to a central unit, and they are synchronized. The first features can be accessed with GPS and XBee-ZigBee radio links. To address synchronization, the acoustic recorder can be taken stereo, with one input taken from a GPS-PPS module. The PPS (Pulse-Per-Second) provides synchronized counts; the initial timing is documented on start of recordings. This solution has been tested and found to provide excellent (below micro-second) synchronization among the sensing units.

Acknowledgement

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