2 Noise in Urban Forest

Noise in urban forest is produced by the sound field of different sources which can be detected in the surroundings. The acoustic intensity of this field is characterized by the following parameters: the amplitude of the disturbance, the excess pressure, the particle velocity, the density change or corresponding change in refractive index, the steady pressure on a surface due to the impact of sound waves, the thermal changes produced by alternating compression and rarefaction and the power which may be absorbed from the sound waves. From a theoretical point of view, three fundamental types of sources are recognized: the simple point source, the doublet (or dipole, equivalent to two simple and equal sound sources), and the quadrupole (the combination of two doublet sources, termed longitudinal and lateral quadrupole). A simple point sound source can be produced by a single-shot propane gun source.

To study impulse source scattering in forest, Rogers et al. (1992) used a propane gun, which contains a significant amount of low-frequency acoustic energy, and a microphone located in a stand, at 10 m from the source. It was observed that the received signal is composed of two main components: (a) a direct zone produced by sound wave direct propagation from the source to the microphone and (b) a scattered zone induced by the presence of woods. Scattering phenomena in a stand are very complex and rather difficult to estimate accurately. In order to make a detailed assessment of the influences of all factors producing scattering in a forest stand (biomass, density of trees/ha, tree height, tree diameter, crown shape and size, size and shape of leaves and needles, etc.), it has been accepted to study a global parameter expressed by the excess attenuation, which includes the absorption, dispersion, reflection and refraction of sound.

The specification of noise in physical terms depends upon its nature. One of the best representations is given by its spectrum. For noise measurement, three techniques are used: recording the wave-form to identify the disturbing frequency components, narrowband analysis and broadband analysis when determining the requirements for noise control. For most purposes, it is sufficiently accurate to use octave band analysis.

In the first part of this chapter, several acoustical notions necessary for the understanding of the theoretical and practical approaches are proposed. Factors affecting sound propagation and scattering phenomena are discussed.

The second part of this section is devoted to a presentation of the equipment for noise measurements.

2.1 Sound Propagation

2.1.1 Definitions and Theoretical Considerations

The sound is produced by a disturbance induced in air, causing alternative pressure and displacement of the air molecules. The dictionary of acoustics (Morfey 2001) and basic reference books (Stephens and Bate 1966; Beranek and Vèr 1992; Fahy and Walker 1998; Harris 1998; see also sources for noise level data in journals such as: Acta Acustica, J Acoust Soc Am, J Sound Vibr, Noise Control Eng J; and the US National Bureau of Standards and the ISO standards noted in Annex 4), in an acoustical context, define noise as an undesired and extraneous sound. A sound wave can be composed of a single frequency (pure tone), or a combination of this frequency harmonically related or not.

The measurable aspects of sound propagation in air can be described by many parameters. In this book, I selected only 12 parameters, as follows:

1. *Sound pressure* is the variation in pressure above and below atmospheric pressure and is expressed in Pascals (Pa). The normal audible frequency range is roughly between 15 Hz and 16 kHz. Frequencies between 3 kHz and 6 kHz are the most sensitive. A young person can detect pressure as low as 20 µPa, compared to normal atmospheric pressure, which is 101.3×10^3 Pa.
2. *Speed of sound* in air (noted c in m/s) is calculated as:

$$c = \sqrt{\frac{1.4P_s}{\varrho}} \tag{2.1}$$

where P_s is the ambient pressure (Pa) and ϱ is the air density (kg/m^3). The speed of sound in air is dependent on temperature. Some theoretical aspects related to this interaction are presented in Annex 3.

For practical purposes, the speed of sound is determined with the following approximate formula:

$$c = 331.4 + 0.607\theta \tag{2.2}$$

where θ is the ambient temperature in °C, or with the exact formula:

$$c = 331.4\sqrt{\frac{T}{273}} = 331.4 + \sqrt{1 + \frac{\theta}{273}} \tag{2.3}$$

where T is the absolute temperature (K). At the normal temperature of 20 °C, the speed of sound is 344.8 m/s.

3. *Sound intensity* (W/m^2) is the sound energy transmitted through a specific area and measured in a specific direction. In free space, the sound intensity is related to the total power radiated into the air by a sound source and to the sound pressure. Sound intensity at a point is a vector, having a minimum and a maximum. The maximum is obtained when its plane is perpendicular to the direction of travel; when parallel, the sound intensity is zero. The sound intensity is related to the sound pressure. In an environment without reflecting surfaces, at any point, the sound pressure of freely traveling waves (plane, cylindrical, spherical) is related to the maximum intensity $I_{\max}$, through the equation:

$$I_{\max} = \frac{p_{\mathrm{rms}}^2}{\varrho \cdot c} \tag{2.4}$$

where p_{rms} is the root-mean-square (rms) sound pressure (expressed in Pa or N/m^2), ϱ is the density of the air (kg/m^3), c is the speed of sound in air (m/s), $c\varrho$ is the characteristic impedance of the air $\left(\frac{\mathrm{m}}{\mathrm{s}} \cdot \frac{\mathrm{kg}}{\mathrm{m}^3}\right)$.

4. *Sound power level* is the measure of the total acoustic power radiated by a source and is expressed in dB *re* W_0, which is the reference sound power, standardized at 10^{-12} W, and is defined as:

$$L_W = 10\log_{10} W/W_0 (\mathrm{dB}\, re\, W_0) \tag{2.5}$$

where W is the sound power (W) and W_0 is the reference sound power, standardized at 10^{-12} W, corresponding to the reference pressure of 20 µPa (2×10^{-5} N/m^2).

The relationships between the sound power and sound power level are given in Table 2.1, from which it can be seen that power ratio < 1 lead to negative levels. Different international standards describe methods for determining the sound power levels of noise sources (see Annex 4).

5. *Sound intensity level* noted IL or L_I (dB) is the measure of the acoustical disturbance produced at a point removed from the source and is defined as the ratio of two sound sources intensities, I_1 and $I_2 = I_{\mathrm{ref}}$ expressed in logarithmic form as:

$$IL = L_\mathrm{I} = 10\log_{10} \frac{I_1}{I_{\mathrm{ref}}} \tag{2.6}$$

where I_{ref} is the reference intensity of 10^{-12} W/m^2 (if the reference is different, one must note explicitly the reference value). The sound intensity level depends on the distance from the source and the losses in the air path (ISO 3740, ISO 3744; see Annex 4).

Table 2.1. Sound power level (dB) and sound radiated power (W) in linear, exponential and dB-log scale (data from Beranek 1960, 1992)

Sound radiated power (W)		Sound power level (L_w, dB)	
Usual notation	Exponential notation	Relative to 1 W	Relative to 10^{-12} W
100 000	10^5	50	170
1,000	10^3	30	150
100	10^2	20	140
10	10^1	10	130
1	1	0	120
0.1	10^{-1}	−10	110
0.01	10^{-2}	−20	100
0.001	10^{-3}	−30	90
0.00001	10^{-5}	−50	70

6. *Sound pressure level* noted SPL or L_p (dB) is the ratio between the effective measured sound pressure and the sound pressure at a source reference:

$$L_\mathrm{p} = 10 \log_{10} \frac{\overline{p^2(t)}}{p_\mathrm{ref}^2} = 20 \log_{10} \frac{p_\mathrm{rms}}{p_\mathrm{reference}} \tag{2.7}$$

where $p_\mathrm{reference}$ is the reference pressure of 20 µPa (2×10^{-5} N/m^2), for sound propagation in air, since it corresponds to the rms pressure of a pure tone at 1 kHz, which is just audible by the human ear. The rms corresponds to the acoustic pressure fluctuations of the acoustic wave and is given by the equation:

$$\overline{p^2(t)} = \underline{\lim_{T\to\infty} \frac{1}{T} \int_{-T/2}^{T/2} p^2(t)\,\mathrm{d}t} \tag{2.8}$$

where T is the averaging time, very large compared to the period of pressure fluctuation and should extend to infinity for random fluctuations, whose statistical properties remain stationary with time. Since this parameter has the dimensions of pressure squared, the label "root mean square" was associated with this fluctuation. The parameter p_rms is given by the square root of the mean square pressure. In practice the range of variation of p_rms is very large, from 10^{-5} Pa to 10^3 Pa. For this reason, the logarithmic scale is always used.
Typical values of the rms pressure fluctuation and the corresponding sound pressure levels are given in Table 2.2.
The sound pressure level at different frequencies produced by different sources (wind, cars, train, etc.) is given in Table 2.3.

Table 2.2. Typical rms pressure fluctuations and their sound pressure levels (Fahy and Walker 1998, with permission)

Source	Pressure fluctuation p_{rms} (Pa)	Sound pressure level L_p (dB *re* 2×10^{-5} Pa)
Jet engine at 3 m	200	140
Pneumatic hammer at 2 m	2	100
Conversational speech	0.02	60
Residential area at night	0.002	40
Rustling of leaves	0.0002	20
Threshold of hearing	0.00002	0

Table 2.3. Noise data at octave-band center frequency for different noise sources (Egan 1988)

Source	Sound pressure level (dB) at various frequencies (Hz)								SPL
	63	125	250	500	1,000	2,000	4,000	8,000	dB
Birds at 33 m	–	–	–	–	–	50	52	54	57
Cicadas	–	–	–	–	35	51	54	48	57
Large dog at 17 m	–	50	58	68	70	64	52	48	72
Lawn mower at 1.7 m	85	87	86	84	81	74	70	72	86
Pistol shot at 82 m	–	–	–	83	91	99	102	106	106
Surf at 3 m, moderate sea	71	72	70	71	67	64	58	54	78
Wind in trees, 16 km/h	–	–	–	33	35	37	37	35	43
Large trucks	83	85	83	85	81	76	72	65	86
Passenger cars	72	70	67	66	67	66	59	54	71
Motorcycle	95	95	91	91	91	87	87	85	95
Snowmobile	65	82	84	75	78	77	79	69	85
Train at 33 m	95	102	94	90	86	87	83	79	94
Car horn at 5 m	–	–	–	92	95	90	80	60	97
Commercial turbofan airplane	77	82	82	78	70	56	–	–	79
Military helicopter	92	89	83	81	76	72	62	51	80

The relation between sound pressure in microPascals and sound pressure level in decibels (re 20 µPa) for various sources of noise is given in Fig. 2.1. All confusion between sound power level (often expressed in Bels) and sound pressure level (expressed in dB) must be avoided. The former corresponds to the measure of the acoustic power radiated by the source and the later depends on the power of the source, the distance from the source and the acoustical characteristics of the space surrounding the source.

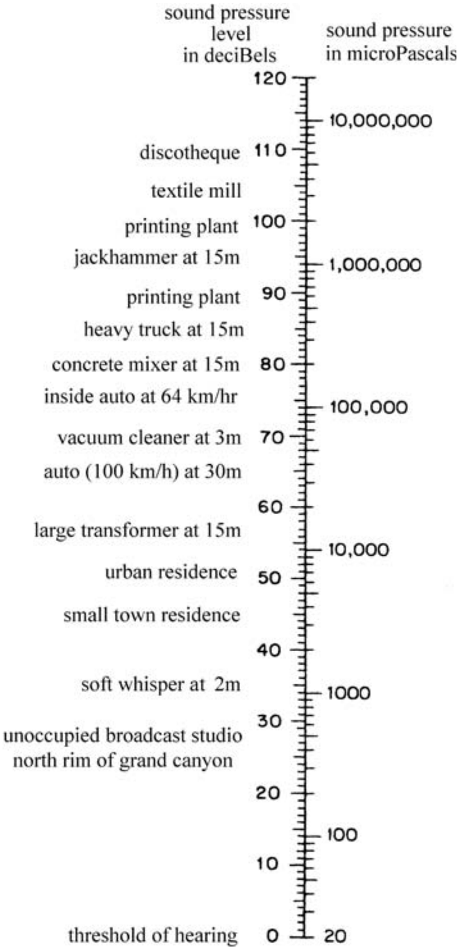

Fig. 2.1. Various sources of noise and the corresponding sound pressure level (dB) and sound pressure (μPa; Harris 1998). Reprinted with permission from the Acoustical Society of America, copyright 2005

7. *A-weighted sound pressure level* (L_A, in dB) is defined as:

$$L_A = 10 \log \left[\frac{p_A(t)}{p_{ref}} \right] \tag{2.9}$$

where $p_A(t)$ is the instantaneous sound pressure measured using the standard frequency-weighting A.

8. *Average sound level* ($L_{av,T}$) during time T, is expressed in decibels and is defined as:

$$L_{\mathrm{av},T} = 10\log\frac{\frac{1}{T}\int_0^T p^2(t)\,\mathrm{d}t}{p_{\mathrm{ref}}^2} \tag{2.10}$$

where T is the long time over which the averaging takes place (e.g. 8 h).

9. *Averaged A-weighted sound level* ($L_{\mathrm{eq}} = L_{\mathrm{A},T}$) is defined as:

$$L_{\mathrm{eq}} = L_{\mathrm{A},T} = 10\log\frac{\frac{1}{T}\int_0^T p_{\mathrm{A}}^2(t)\,\mathrm{d}t}{p_{\mathrm{ref}}^2} \tag{2.11}$$

where T is 8 h for a working day or 24 h for a full day.

10. *Day night noise level* ($L_{\mathrm{d,n}}$), between 07.00 and 22.00 hours (dB) is given by:

$$L_{\mathrm{d,n}} = 10\log\frac{1}{24}\left[\frac{\int_{7:00}^{22:00} p_{\mathrm{A}}^2(t)\,\mathrm{d}t}{p_{\mathrm{ref}}^2} + \frac{\int_{22:00}^{7:00} 10p_{\mathrm{A}}^2\,\mathrm{d}t}{p_{\mathrm{ref}}^2}\right] \tag{2.12}$$

11. *A-weighted sound exposure* ($E_{\mathrm{A}T}$; Pa^2 s) is proportional to the energy flow (intensity × time) in a sound wave in the time-period between t_1 and t_2 and is given by:

$$E_{\mathrm{A},T} = \int_{t_1}^{t_2} p_{\mathrm{A}}^2(t)\,\mathrm{d}t \tag{2.13}$$

12. *A-weighted noise exposure level* ($L_{\mathrm{EA},T}$) which is:

$$L_{\mathrm{EA},T} = 10\log\left(\frac{E_{\mathrm{A},T}}{E_0}\right) \tag{2.14}$$

where the reference E_0 at $(20\,\mu\mathrm{Pa})^2$ is equal to $(4\times10^{-10}\,\mathrm{Pa})^2$ s (Beranek 1992) or, following ISO 1996-1 (see Annex 4), E_0 is equal to $(1.15\times10^{-10}\,\mathrm{Pa})^2$ s.

The system used for noise control contains three major components: the source, the path and the receiver, associated with emission, transmission and immission. The sound energy emitted by a noise source is transmitted to the receiver where it is immitted. The immission is described by the sound pressure level (dB). The strength of the noise source is described by the sound power level and its directivity, which is a function of angular position around source and frequency.

To match the assumed frequency response of the ear, implied by equal loudness contours, A, B and C frequency weighting curves were standardized.

Table 2.4. Electrical weighting networks for sound-level meter, for several frequencies (Beranek 1992)

		Frequency (Hz)								
		10	20	50	100	200	400	1,000	10,000	20,000
A-weighting	dB	−70.4	−50.5	−30.2	−19.1	−10.9	−4.8	0	−2.5	−9.3
C-weighting	dB	−14.3	−6.2	−1.3	−0.3	0	0	0	−4.4	−11.2

The network specification of these curves is given in Annex 4. The A-frequency weighting was for 40 dB sound level, the B-frequency weighting for 70 dB sound level and C-frequency weighting for 100 dB level. For outdoor community noise measurements, A-frequency weighting is mostly used. This weighting reduces the sensitivity of the sound level meter to low and high frequency sounds, as compared with the mid-band frequency, between 1 kHz and 4 kHz. Another important practical advantage of A-weighting is the relative immunity against wind noise generated at the microphone (Fahy and Walker 1998). Today, the B and C frequency weightings are out of use. The time of weighting can be fast (F), 125 ms (corresponding approximately to the ear integration time), slow (S), with an exponential time constant of 1 s, and impulse (I), which has a fast rise (35 ms) and slow decay.

The A-weighted sound or noise power level is defined as:

$$L_{\mathrm{WA}} = \log \frac{W_{\mathrm{A}}}{W_0} \tag{2.15}$$

where W_{A} is the A-weighted sound power and W_0 is the reference sound power, 10^{-12} Watt. Table 2.4 gives the A-weighted sound pressure level for different frequencies. When A-weighting is used with the overall sound power level, the noise power emission level can be expressed in Bells or decibels (1 Bel = 10 dB).

The radiation source field varies with distance from source, which can be in the near field, far field or reverberant field. In the far field, the sound pressure decreases by 6 dB for each doubling of the distance from the source. In outdoor measurements, the sound power must be performed in the far field of the source (choosing an array of microphone positions over the surface).

2.1.2 Factors Affecting Sound Propagation

Sound or noise propagation in a forest stand is affected by the presence of trees, soil surface, ground vegetation, topography and meteorology of the site. The interaction between the sound field and the vegetation is complex and determines mainly the decrease in the sound level, but sometimes can induce a small increase in the sound level under the canopy. The presence of a solid barrier between the source and receiver introduces a much more complex mechanism of sound propagation, producing a decrease in sound level behind the canopy and barrier with another possible zone of unstable increased or decreased sound level, depending on particular in situ conditions.

Acoustic scattering in woodlands was a subject of interest for many authors (Eyring 1946; Embelton 1963; Aylor 1972a, b; Decourt 1975; Fricke 1984; Price 1988; Attenborough 1992; Barrière and Gabillet 1999), with the principal focus on the attenuation of sound between the source and receiver induced by geometric spreading and attenuation effects due to absorption where the ground effect plays an important role. Scattering effects will be largely covered in Chap. 4.

To study the influence of meteorological conditions, Heimann (2003) proposed a simulation of wind propagation through an idealized stand of trees with a tri-dimensional numerical fluid-dynamics model which allowed isolation of single influences (trees, wind, ground, etc.) in a virtual way.

In atmosphere (Piercy et al. 1977; Brown 1987; Naz et al. 1992), sound undergoes the following physical processes: reflection from surfaces, refraction by temperature and wind gradients, diffraction by edges and changes in surface impedance, scattering by turbulent fluctuations in temperature, wind, rain and snow, molecular absorption and attenuation induced by scattering-out of finite angular-width beams. Relative humidity variations produce negligible changes (<3%) on the speed of sound. Near the ground, the propagation of sound waves (Noble et al. 1992) is very complex, involving: geometric spread and molecular absorption, reflection with phase change due to finite impedance of the ground, refraction by the mean wind and temperature gradient and scattering by atmospheric turbulence, at small and large scale.

Very roughly, the meteorological effects (Ingård 1953) can be summarized as follows: for a windy day and wind speed of 6–11 m/s, the variation in SPL was 15–20 dB at 2 kHz frequency; and the average attenuation was 4–6 dB over 100 m, with a maximum of 20 dB. Noble et al. (1992) noted that "large scale atmospheric features have a large effect on phase but little effect on the amplitude of the signal".

2.2 Equipment for Noise Measurement

2.2.1 Instrumentation and Noise Sources

The techniques and instruments for the measurement of surrounding noise are determinants for noise control and abatement at any point in the acoustic field as a function of time and frequency. At any observation point in the acoustic field, the microphone (which is an electroacoustic transducer) transforms sound pressure into a corresponding electrical signal. Environmental noise fluctuates greatly and it is essential to be able to measure acoustic phenomena accurately, with a good repeatability of readings with a sound-level meter. Sound-level meters are more or less complex, depending on their practical utilization which can be for:

- laboratory reference intended for calibration of other apparatus,
- precision sound level meters for laboratory and field accurate measurements,
- general purpose sound level meter for noise recording level and later data analysis, and survey sound level meter for noise environment.

The environmental acoustical signal is processed by the adapted devices which compose a sound-level meter. As presented in the literature and many commercial pamphlets (Fig. 2.2), the sound-level meter is a portable apparatus, battery-operated and equipped with a microphone, a pre-amplifier, an amplifier, a weighted network (A, B, C and linear) or an external bandpass filter, an output amplifier and a read-out meter, giving sound pressure level (L_p) and other parameters. Example: L_{10} = sound level exceeded 10% of the time; L_{50} = sound level exceeded 50% of the time; L_{90} = ambient level; L_A = sound level on A scale; L_{eq}, $L_{d,n}$ = day/night equivalent noise level. The microphone converts the incident acoustical signal into an electrical signal. Commonly, the dynamic range of sound pressure is 10^7 and for this reason a output voltage must also be provided. Most frequently, the attenuator is arranged in 10-dB steps. Frequency analysis of the sound field can be performed in one-third band analysis. Because of the very big fluctuations in sound level, the apparatus is provided with three responses, a "fast" response having a time constant of 10 ms, a "slow" response for 1 s and an "impulse" for a time constant of 35 ms. More complex data about the noise signal can be obtained when parameters such as the peak or duration of a transient (50 µs), the cross-power spectra, the computation of correlation functions, etc., are computed.

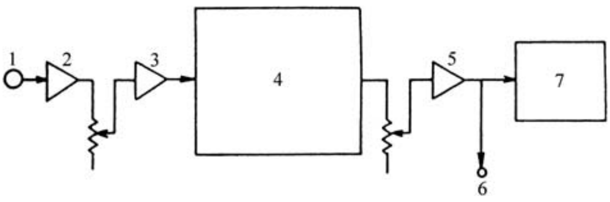

Fig. 2.2. Block diagram of a sound level meter. 1 Microphone, 2 pre-amplifier, 3 amplifier, 4 linear all-pass or weighted network (A, B, C) or external bandpass filter, 5 amplifier, 6 external output, 7 rectifier and read-out meter

It is very important to note that the sound-level meter must be checked regularly for acoustical calibration (accuracy 0.2 dB) and electrical calibration.

For scientific measurements and consistent data analysis, specific noise sources are used. Very popular is the single-shot propane gun source which contains a significant amount of low-frequency acoustic energy. A freon-powered horn was used by Rogers and Lee (1989) for in situ measurements. This source was located 60 m away from the forest and the microphone was located on a line from the source normal to the edge of the stand.

Also, as an acoustic source, a loudspeaker driven by a pre-recorded signal with traffic noise or white noise has been used (Fig. 2.3) for experimental measurements in a stand. For normal incidence, the source and the reference

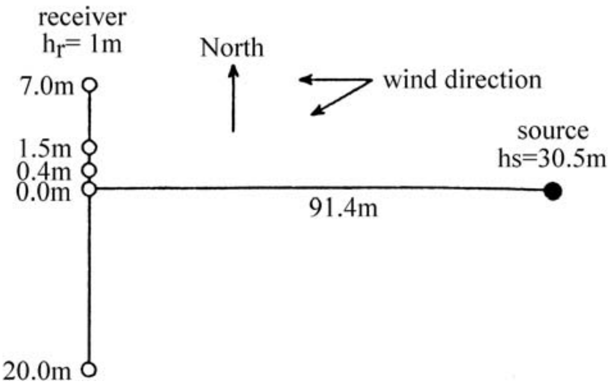

Fig. 2.3. Measurement geometry for in-field measurements on a flat surface (Noble et al. 1992). Reprinted with permission from the Acoustical Society of America, copyright 2005

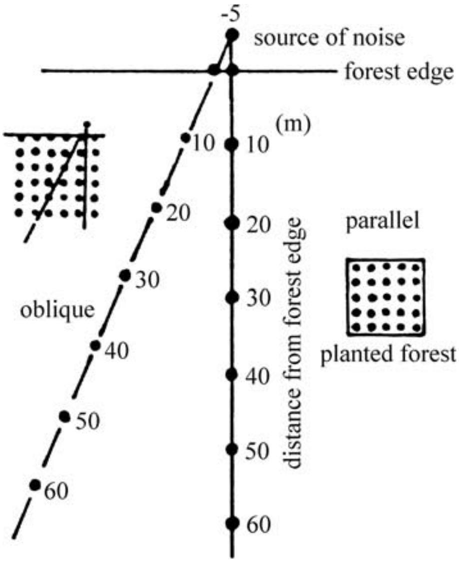

Fig. 2.4. Experimental arrangements for measurements in horizontal plane, in different stands (Tanaka et al. 1979)

microphone were positioned on a straight line (180°) at different distances from the ground (max 30.5 m). In order to control the experimental conditions, the sound level of the background noise must first be determined.

Measurements at an oblique incidence (Fig. 2.4) were performed by Tanaka et al. (1979) when the site configuration was very complex.

2.2.2 Measurement In Situ

Because of the diversity and complexity of experimental situations, in this section we propose to select only the aspects related to the amplitude and phase measured at the microphone, in situ, at 91 m from the source (Fig. 2.5).

The main difference between the amplitude variations and phase variations can be observed in a long-term excursion on the time-scale. The Fourier transform of phase variation represented in coordinates – amplitude and frequency – displays both short- and long-term variations. Several spectral peaks can be identified, probably related to the experimental conditions, such as length of sample analyzed and large atmospheric features related to turbulence. The peak at 0.0073 Hz shows that the dominant scale is on the order of hundreds of meters in size, corresponding to atmospheric or field features.

As a general remark, note that peaks must always be analyzed in terms of corresponding wave length, which can give an indication about the size and nature of the objects producing them. In practice, the understanding of those peaks requires fine equipment and very skilful operators having a good theoretical background.

2.2.2.1 Effects of Distance

The effects of scattering by woodlands related to the distance of measurement and frequency are addressed in the literature. Various empirical equations have been proposed to predict the influence of distance on noise level. One of these equations (Cook and van Haverbeke 1971) is given below:

$$S_d = S_0 - 20 \log \frac{d}{d_0} \tag{2.16}$$

where S_d is the sound level at distance d, S_0 is the sound level at the source and d_0 is the reference distance where the sound level is known. Figure 2.6 shows the influence of distance on decreasing sound level noise measurements in a tree belt (width of 30 m and in-row spacing of about 2 m). The belt was composed of deciduous trees of 25 m height. A very important decreasing effect was

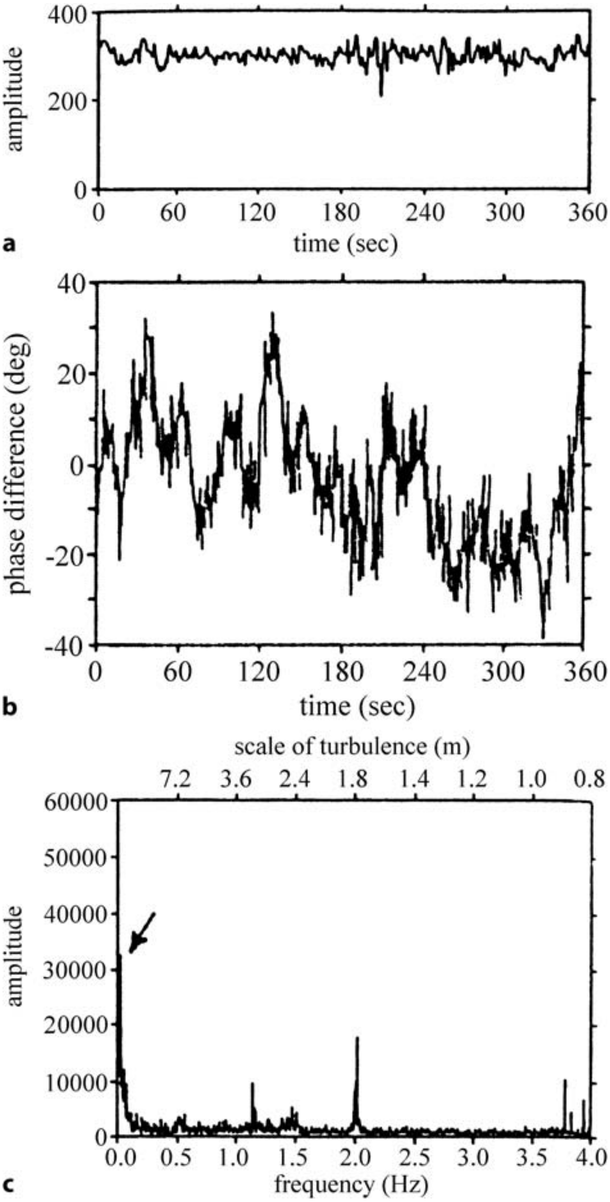

Fig. 2.5. Physical parameters measured at the microphone, situated 91 m from the source, on a grass field. The source is a loudspeaker (pre-registered traffic noise) located on top of a 30.5 m tower (Noble et al. 1992; reprinted with permission from the Acoustical Society of America, copyright 2005). **a** Amplitude versus time for 360 s, **b** phase difference vs time (360 s), at the receiver, **c** Fourier transform of the phase variation displayed in **b**, in the frequency range 0 – 4 Hz

observed for the distance between 33 m and 66 m, after which a quasi-constant level was observed.

To determine the optimum position of the tree belt as a noise screen, a series of experiments were performed, during which source and microphone position were varied simultaneously (Fig. 2.7). The vertical structure of the belt is

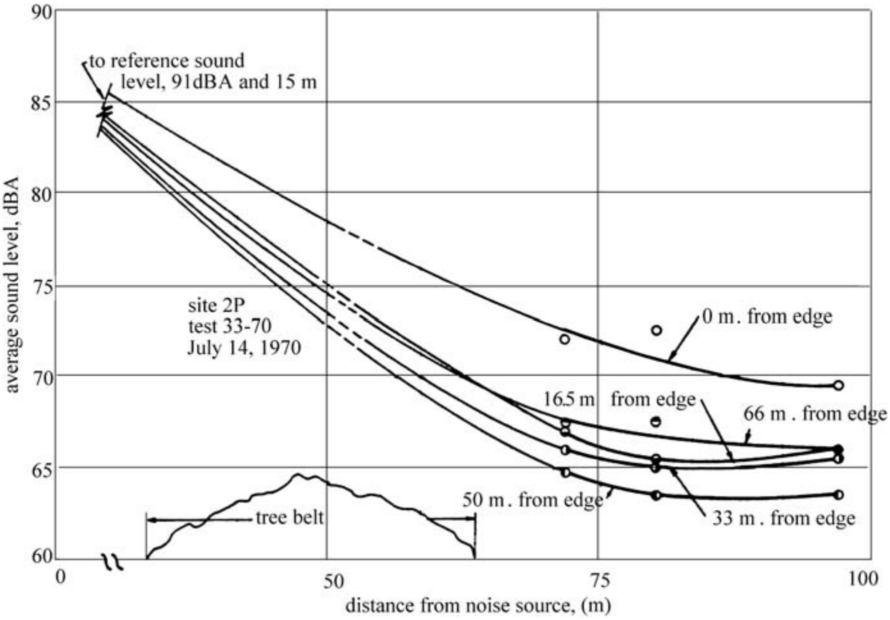

Fig. 2.6. Influence of distance on average sound level measurements (Cook and van Haverbeke 1971)

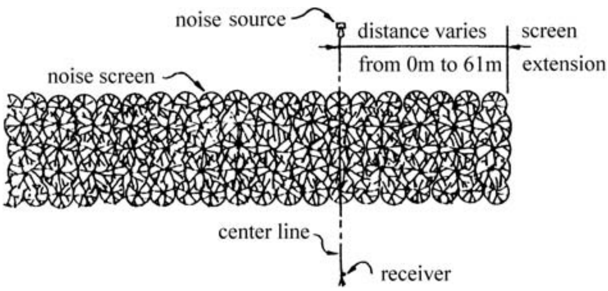

Fig. 2.7. Experimental arrangements for noise attenuation measurements through a tree belt, at different "screen extension" distances ranging from 0 m to 66 m (Cook and van Haverbeke 1971)

shown in Fig. 2.8. For quantifying the variation in relative attenuation, the reference variable was the ratio of the distance from receiver to source (R/S; Fig. 2.9). From this figure, one can see a minimum corresponding to R/S = 1, which corresponds to a tree belt situated midway between the source and the receiver. After this inflection point, the attenuation increases as the ratio R/S increases, indicating a more effective action of the tree belt. Cook and van Haverbeke (1971) noted: "it would seem that planting distances from 12 m

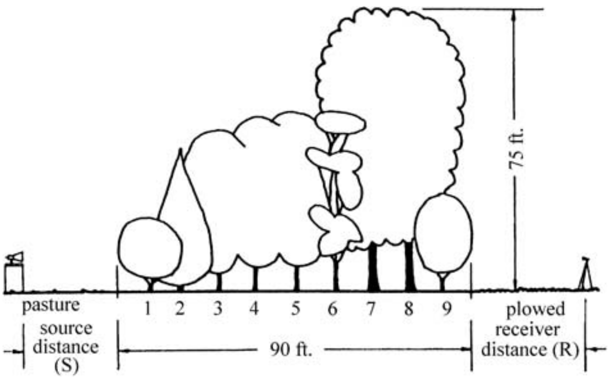

Fig. 2.8. Vertical structure of a belt of trees of width 30 m, of different species. Row spacing 3.3 m, in-row spacing 2.0 m (Cook and van Haverbeke 1971). 1 Russian olive, 2 pine and eastern red cedar, 3 catalpa, 4, 5 hackberry, 6 honey locust, 7, 8 cottonwood, 9 mulberry (90 ft = 27.45 m; 75 ft = 22.80 m)

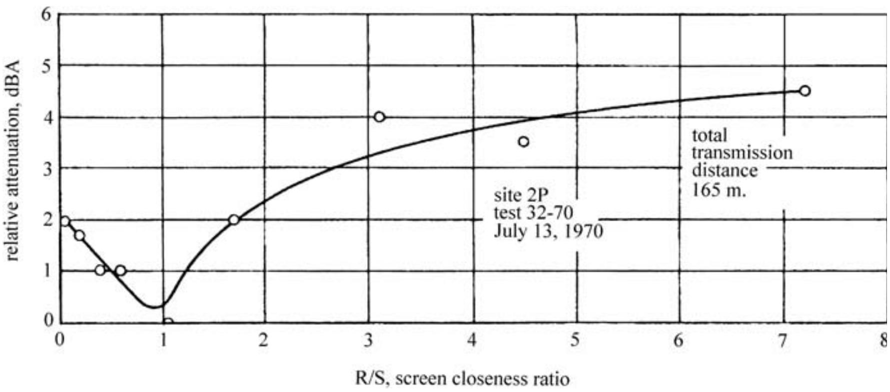

Fig. 2.9. Relative attenuation vs the ratio of receiver to source (R/S), for 165 m total transmission distance (Cook and van Haverbeke 1971)

to 22 m from the noise source would yield optimum results for tree belts of considerable height and depth in rural areas". Also: "placing trees and shrubs close to a noise source is recommended, a distance of 3 m to 8 m from a noise source to nearest shrub would seem to yield optimum results". The relative attenuation increases with the distance from the source (Fig. 2.10).

Beside the role of distance in outdoor sound transmission and therefore in noise reduction by tree belts, the roughness and acoustic impedance of the media interposed between the source and receiver play an important role (see also Chap. 4, the section related to ground effect).

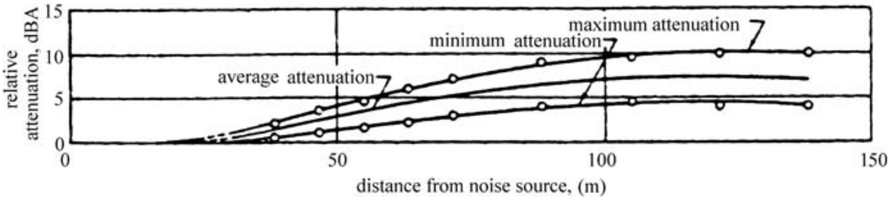

Fig. 2.10. Relationships between relative attenuation and distance from the noise source for different media in three characteristic situations presenting: *maximum attenuation* trees and corn, *average attenuation* gravel and *minimum attenuation* paving (Cook and van Haverbeke 1971)

2.2.2.2
Effect of Frequency

To identify and quantify acoustic features of interest for the studies related to noise control, a detailed analysis can be performed, using more or less sophisticated frequency analysis instruments which serve different purposes, such as: the assessment of the severity of an environment, the identification of system response properties and the identification of sources and transmission paths.

The acoustic signals can be steady-state, transient, stationary or nonstationary. Depending on signal complexity, the parameters calculated are different. For a steady-state stationary signal represented in amplitude-time coordinates, the mean value and the mean-square value, or the weighted average (which is a simple linear sum of values over a specific time interval), are calculated. The spectral functions provide a frequency decomposition of the signal values. The computation of rms values in one-third octave bands is widely used for frequency analysis of acoustical data. Much fine frequency analysis requires very complex calculation of frequency spectra, using fast Fourier transform algorithms.

The reader interested in more details related to data analysis is invited to study Piersol's (1992) chapter "Data analysis" in the book edited by Beranek and Vèr (1992). See also Goodfriend (1977), von Gierke et al. (1998) and Gygi et al. (2004).

2.2.2.3
Effect of Visibility

The effect of visibility in the forest on the attenuation of noise was and still is a very intriguing question from the beginnings of "forest acoustics". To estimate the density of tree belts, Eyring (1946) and Embleton (1963) proposed the parameter "visibility", defined as the distance at which an object is obscured by the vegetation. Eyring (1946) stated that attenuation is correlated with visibility inside the tropical rain forest and that it increases with increasing frequency.

As regards visibility, we note the results reported by Pal et al. (2000), performed in different forests that were intentionally planted or preserved around coal mines in India in order to protect the neighborhood from pollutants and noise. The resulting data set was used to derive a linear relationship of excess noise attenuation (dB) as a dependent variable, with the independent variables of number density, average height, canopy branch cover, trunk diameter and both vertical and horizontal light penetration. It was observed that light penetration (which depends reciprocally on leaf density and branches) is the most decisive parameter, while the average density (number of trees per unit area) has only a negligible effect. It was supposed that "sound waves propagate through the gaps between the trees even with maximum plantation density". It was stated also that meteorological effects were supposed to be negligible for this experiment.

More recently, Fang and Ling (2003) studied tree belts in Taiwan (*Ficus microcarpa, Podocarpus macrophyllus, Palaquium formosanum, Camelia japonica*, etc.) and noted a negative logarithmic relationship between the visibility, belt width and the relative attenuation determined "as the difference between the measurements on open ground and data from the tree belt which includes the effects of distance and vegetation" (Fig. 2.11). On the graph, four regions can be observed, noted A, B, C and D: region A – reducing noise in that the relative attenuation exceeded 10 dB, region B – between 10 dB and 6 dB, region C – between 6 dB and 3 dB and region D – less than 3 dB.

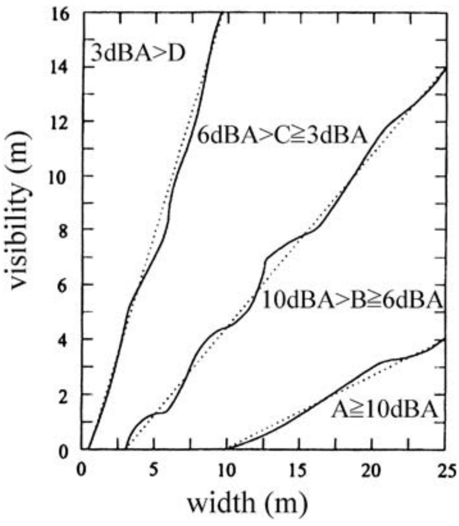

Fig. 2.11. Relative attenuation in a very large range (lower than 3 dB and higher than 10 dB) as a function of visibility and width of tree belt (Fang and Ling 2003). Reprinted with permission from Elsevier, copyright 2005

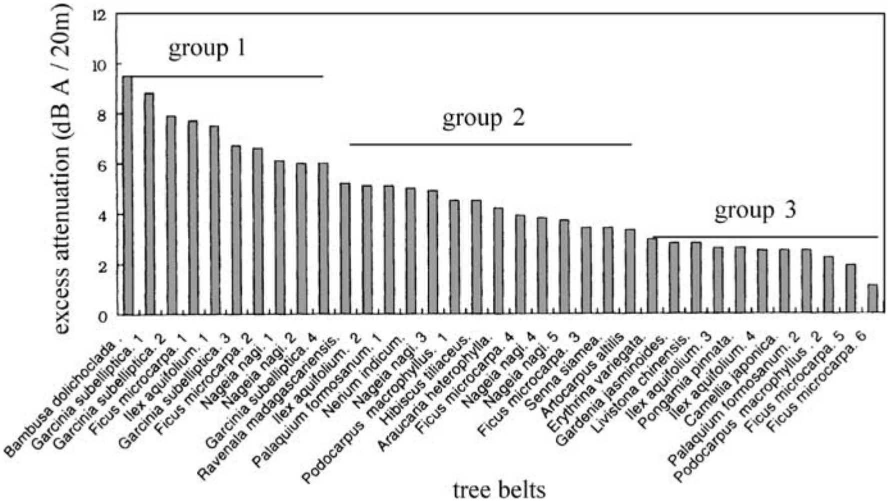

Fig. 2.12. Classification of tree belts following the excess attenuation value (Fang and Ling 2003). Reprinted with permission from Elsevier, copyright 2005

Figure 2.12 represents the excess attenuation of 35 tree belts classified in three groups:

- Group 1: "effective reduction" region for which the excess attenuation (dB/20 m) exceeded 6 dB, for a visibility less than 5 m.
- Group 2: "sub-reduction" region for which the excess attenuation is between 3.0 dB and 5.9 dB, for a visibility between 6 m and 19 m.
- Group 3: "invalid reduction" region, for which the excess attenuation is less than 2.9 dB, for a visibility exceeding 20 m.

From this study, it was recommended that a belt of trees and shrubs can reduce noise by 6 dB via suitable plantings and visibility (e.g. 1 m visibility and 5 m width, or 10 m visibility and 18 m width).

2.3 Summary

Noise in urban forest is produced by the sound field of different sources which can be detected in the surroundings. The acoustic intensity of this field is characterized by the following parameters: amplitude of disturbance, excess pressure, particle velocity, density change or corresponding change in refractive index, steady pressure on a surface due to the impact of sound waves, thermal changes produced by alternating compression and rarefaction, and the power which may be absorbed from the sound waves.

The specification of noise in physical terms depends upon its nature. One of the best representations is given by the spectrum. For noise measurement, three techniques are used: recording the wave-form to identify the disturbing frequency components, narrowband analysis and broadband analysis when determining the requirements for noise control. For most purposes, it is sufficiently accurate to use octave band analysis.

The measurable aspects of sound propagation in air can be described by the following parameters: sound pressure, speed of sound, sound power level, sound intensity level, sound pressure level, average sound level, averaged A-weighted sound level, day/night noise level and A-weighted noise exposure level. The system used for noise control contains three major components: the source, the path and the receiver, which are associated with emission, transmission and immission. The sound energy emitted by a noise source is transmitted to the receiver where it is immited.

The techniques and instruments for the measurement of the surrounding noise are determinants for noise control and abatement at any point in the acoustic field as a function of time and frequency. The environmental acoustical signal is processed with the sound-level meter. Outdoor measurements are mainly influenced by the distance effect, the frequency effect and the visibility, which is important mainly in tropical and subtropical areas.