1. **Models for Noise Control and Sound Design**
   1. **Airborne Sound Insulation**

The ability to predict sound and vibration transmissions in built-up structures such as buildings, trains and automobiles is important for human comfort, health and safety. There is a concern about steadily growing annoyance due to the noise in private dwellings and the background speech in commercial work sites that leads towards reduced power of concentration during physical or mental work. Many surveys reveal that in multifamily apartments people are annoyed by noises mainly caused by indoor activities that are responsible for this annoyance. The studies show that people are also exposed to the noises from neighbours, which causes consequences of disturbances in sleep, physical or mental work imparities, and the disturbances in conversation in private dwellings and working performance in office premises. In the residential and worksite premises, especially in the urban areas, the international standards provided by ISO have reflected the trends in growing annoyance due to the indoor as well as outdoor noise. These trends are increasing both in number and covering broader aspects. However, there are certain aspects of noise to be taken into account to provide an optimal acoustic satisfaction and an accurate evaluation of building performances. These aspects, generally, include the specific sound such as a conversation varying in intelligibility originate from the adjacent office and a transient noise from outdoor moving sound source which are the main cause for the disturbances in daily life’s physical and mental work.

In architectural structure, such as dwellings, the sound is transmitted mostly through walls, floors and ceilings by setting the entire structure into vibration. When the sound energy (i.e. a sound wave) incident on a surface of a wall or any other element of the buildings, it is partly reflected back to the source room and partly dissipated as heat energy within the material of the wall. Some sound energy propagates to other connecting structures and some energy is partly transmitted into the building (i.e. rooms). Sound insulation has to do with the reduction of sound energy when sound is transmitted through a barrier or a wall elements. There are typically two types of sound transmissions, which are termed as airborne sound transmission and structure borne transmissions. In Section 5.17 and Section 5.24, we discussed the quantities used to characterize the airborne sound transmission and the quantities that are found in common building regulations and requirements for the sound insulation properties of building elements and constructions. The detailed description about sound transmission (airborne and structure borne) is given in Chapter 5, however, the objectives in this section are to understand the airborne sound insulation models and design of corresponding noise control techniques based on available prediction methods and tools in the perspective of auralization of different indoor and outdoor environments. We discuss the fundamental concepts of sound insulation prediction techniques from the filter design and auralization perspectives. An important aspect of air sound insulation is the frequency which, generally, is considered in the range defined by one-third octave bands from 50 to 5000 Hz. Airborne sound insulation tends to be lowest in the low frequency range and highest in the high-frequency range (i.e. acting as low pass filter). Hence, significant transmission of airborne sound above 5000 Hz is not usually an issue as far as the predictions of sound insulation metrics are concerned. However, when noise control design models and auralization of sound insulation comes into play, it very important to consider sound insulation filter design at frequencies outside the building acoustics frequency range in filters and signals domain. Therefore, interpolation and extrapolation techniques are applied very carefully to cover a full audible frequency range which is typically from 20 to 20,000 Hz. In this way, it is obvious that the measurement procedures applied in the laboratory or in the field are just one aspect of the sound insulation. There are several standards and methods, as discussed in Chapter **5**, which describe the performance of buildings elements in terms of noise reduction and level reduction indices in the form of a single number value and/or frequency dependent curves. Nevertheless, it can be assumed that these quantities are insufficient to describe the real situation for the perceptual evaluation of noise and deigning the airborne sound insulation models for auralization.

As discussed above, it is now desirable to introduce airborne sound insulation auralization methods that simulates the sound field at listener’s end from predicted or measured data with better subjective impressions, and psychoacoustic and psychological factors to move toward noise control and design methods. The basic principle of sound insulation auralization is to simulate the alteration of a sound signal from its source to the receiver end via transmission through the building elements involved in transmission of sound energy. The auralization of a situation where either the speech spoken in one office or the noise produced by an outdoor source is transmitted through building structures, requires the sound source modelling, sound propagation (e.g. from an adjacent room or from a façade etc.) and its transmission through building wall elements (for example direct or flanking wall elements which include separating walls between rooms, doors, portals, floors, ceilings and external walls etc.), and the insulation characteristics of the direct and flanking parts of the dwellings. In order to auralize these situations whether these are office-to-office situations or these are because of the outdoor sounds, both the level and the spectral characteristics of sources are highly dependent on the sound insulation curves of the building elements which separate source and the receiver (e.g. listener). In the proceed sections of this chapter we explain the design procedures for airborne sound insulation filters based on the knowledge of performance of building elements and characteristics of sound transmission through these elements (i.e. from sound insulation metrics either predicted or in-situ measured) which are discussed in Chapter 5 with greater details. The design and development techniques of the corresponding filters are also discussed in this section, which comprehends an office to office sound transmission (which is the adjacent-rooms case) and sound transmission from outdoor environment (e.g. façade sound insulation). The prerequisite for sound insulation filters design is an accurate sound insulation model which is supposed to be based on the sound insulation predictions theories as well as on available international standards and/or in-situ measurements. Most of the input data for filters design and auralization comes from standardized data formats of sound insulation (for example, ISO 12354-1, ISO 12354-3 and ISO 3382-2) from where point to point transfer functions from source to the listener are possible.

* + 1. **Sound Insulation Models for Adjacent Rooms**

The most commonly used methods for sound insulation prediction are based on ISO 12354 (Part I/Part III), SEA (Statistical Energy Analysis), ASEA (Advanced Statistical Energy Analysis) and FEM (Finite Element Methods). It is very challenging to assume that which one of the available methods is accurate and precise for the purpose of sound design and control, and corresponding insulation filters and auralization. In fact each method has certain limitations depending on the assumptions made during their development. ISO 12354 models are widely used in most of the European countries which are based mainly on publications of Gerretsen and on the updates from Vigran, Rindel, Fahy, Davy and Cramer [**REF**]. As an established method, we will make use of the ISO 12354: 1-3 as the foundation for sound insulation model for filter designs and to build up the auralization chain. The first application of airborne sound insulation auralization was introduced by Vorländer and Thaden, who implemented Gerretsen's [**REF**] prediction method in the filters domain. Based on ISO 12354-1, they presented an auralization of airborne sound insulation using binaural technology with headphones. As discussed in Chapter **5**, the transmission coefficients for airborne sound insulation can be expressed by of the transmission path between two spaces (rooms) and hence the standardised sound level difference terms of transmission coefficients is expressed in Equation **3.1**. Here and denote the source and receiving room wall elements respectively for the transmission path , with receiving room volume and the separating (direct) element area between the two rooms. In Figure **3.1**, a typical adjacent source and receiver rooms, separated by wall element are shown.

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|  | **(3.1)** |

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| **Figure 3.1:** *Typical adjacent source and receiving rooms with receiving room walls represented as secondary point sources radiating elements* |

The resulting average sound pressure level in the receiver room is calculated for all transmission paths by Equation **3.2**. By introducing the sound energy signals and as mean squared energies in the source and receiving room respectively, Equation **3.2** is expressed in energetic form and is given in Equation **3.3**.

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|  | **(3.2)** |
|  | **(3.3)** |

In these equations the sound energy radiating elements i.e. the receiving room walls (denoted as secondary sources (SS)) are approximated as point sources located at the centres of the wall and representing whole bending wave patterns as a single point source. The balance between direct and reverberant part of sound field is very important in perception of the spatial characteristics of the rooms. This energy balance is obtained by using definition of reverberation distance , and is computed through the ratio of energies given by relationship, . The quantities, and are the energies of direct and reverb sound field at a distance from the secondary source and is the equivalent absorption area of the receiving room. For uncorrelated direct and reverberant sound fields, the contribution of the transmission path to the mean squared pressure in terms of the reverberant and the direct field can be written as . The temporal effects of the receiving rooms in terms of reverberation are included by measured or simulated impulse response, of this room. Here, at first, the direct sound is removed from the impulse response as it is already included in the transmission path calculation in its binaural form . Subsequently, it is approximately equalised to white spectrum and normalised in energy. The time domain representation of the binaural signal from source to receiver of the transmission path is given in Equation **3.4**. All binaural contributions from the radiating wall elements (i.e. secondary sources) are summed up to get final signal which is shown in Figure **3.2** (left) and Figure **3.2** (right) in frequency and time domain respectively.

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|  | **(3.4)** |

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| **Figure 3.2:** *Final binaural signal at the listener: Frequency domain* ***(****left****)*** *and time domain* ***(****right****)*** *representation* |

This model laydowns the foundations of sound insulation auralization of typical rectangular adjacent rooms. However, this model is based on several simplifications as can easily be observed in Equation **3.4**, which we discuss in next section along with extended approach for this this model.

* + 1. **Improved Sound Insulation Models (Adjacent Rooms)**

In Section **3.13.1**,we discussed sound insulation filters for simple adjacent rooms which is the starting point of sound design models and auralization. Certain simplifications were made in the simple model presented above, such as, point to point transmission from source to receiver room, excluding portals and the doors between the adjacent rooms, radiation of transmitted sound energy from the centre of the receiver walls and diffuse field assumptions in both source and receiver rooms etc. Another important aspect is source directivity that is likewise neglected, which, might reasonably contribute toward the distribution of sound pressure on the surfaces of the source room walls. In non-diffuse fields the amount of transmitted energy would be different for different paths, in particular if sources are placed close to walls (such as loudspeakers or TV sets). In this section, we discuss the improvements in this model by taking into account the source room acoustics whereas considering a more complex sound field incident on the source room walls consisting of a direct and a diffuse field components as introduced, which includes the temporal decay of the room responses. The sound energy transmitted via direct and flanking paths to the adjacent receiving room is now specific for all surface elements, depending on the sound pressure hitting the corresponding wall elements in the source room due to source position and its directivity. Secondly, the influence of reverberation of source and receiving rooms and the balance between direct and reverberant energies inside the receiving room are incorporated into sound insulation transfer functions.

* + - 1. Sound Source Directivity and Room Impulse Response

Generally, the diffuse sound field conditions are assumed while predicting the sound insulation metrics. Which means that the sound sources are assumed to be omnidirectional and casting sound energies all over direct direction homogeneously. However, the sound source directivity in the source room might have a significant influence on calculate the sound field in the sending room (i.e. source room) and in turn on the energy transmitted to the receiver room. As a matter of fact, in normal residential apartments and in commercial work sites the sound fields are normally are not diffuse. We need to include source directivity which plays an important role in calculating the incident sound energies on the boundaries of the source room (i.e. corresponding direct as well as flanking wall elements of the source room). This, in turn, considerably influence the transmitted energies to receiver room through specific transmission paths and hence the radiated energy from the receiving room wall elements. On the other hand the room impulse response (RIR), denoted by plays an important role in non-diffuse fields. The RIR which is, generally, simulated based on the reverberation time of the source room and artificial noise representing the sum of the room reflections. The synthesis of source room impulse responses to the surface elements of the walls is necessary to include the effects of absorption of room boundaries as well as to simulate the cases where an equivalent real room is not present. It means that we require room impulse response at any point in the source room (especially near the surface of the each wall element) from where we can estimate the sound energy at a particular wall surface. The approximated is obtained through a linear combination of filtered exponential decay signals.

* + - 1. Sound Field in the Source Room

In closed spaces, it is assumed that the direct sound field propagates and decay with time as in free-field conditions, whereas, the reverberant sound field is evenly distributed [**REF**] throughout the closed spaces. This phenomenon is described in classical sound field theory for sound propagation in rooms given by Equation **3.5** in its simple form. Let us consider two simple rectangular adjacent rooms, with given dimension and separated by main partition, similar to one shown in the Figure **3.1**. A sound source with specific directivity patterns is placed at an arbitrary position the source room. The directivity of this source is represented by and its sound power level by . The source produces a sound pressure level at distance inside a source room with an equivalent absorption area , and is given by Equation **3.5**.

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|  | **(3.5)** |

Equation **3.5**,includes source room reverberation, the directivity of the source, and the same balance between direct and reverberant energy as considered in simple model discussed in Section **3.13.1**. Let the acoustics power in the source room is (source acoustic power in Watts). The mean squared sound pressure at any point inside the source room in energetic notation can be calculated by Equation **3.6**,

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|  | **(3.6)** |

To calculate the sound pressure, in signal and filters domain, at any point in the source room a loudspeaker with typical “HiFi system” directivity might be selected as an example sound source to analyse the influence of the source directivity on the transmitted energy to the receiving room walls for the direct as well as flanking paths. Let is source signal normalized in power and is energetically normalised impulse response of the source room calculated at a distance form it. Let the directivity of the source in the direction of receiver point on wall is , the time domain representation of Equation **3.6** is rewritten in the form of Equation **3.7**.

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| --- | --- |
|  | **(3.7)** |

From Equation **3.7**, it is possible to calculate the sound pressure inside the source room on any point on the surface of wall elements. Which means that the incident sound pressure and hence sound energy on each segment (i.e. “patch”) can be calculated. In other words, if the building wall elements consist of an assembly of components such as doors and portals, the sound pressure at each component is feasible to calculate.

In reverberant rooms, as a general case, the diffuse sound field is a good approximation while dealing with the sound field propagation, and the results for stationary conditions and sound decays might be applied to measure the sound power of a source. However, in the ordinary rooms the diffuse sound field is usually not the case. Therefore, a rather simple modification to the stationary sound field is to separate the direct sound from the reverberant part of the impulse responses (i.e. reverberation tail). Hence, the incident sound power on any wall element in the source room with surface area is taken as a combination of direct and the diffuse sound fields. In real dwellings the walls (specially the outer wall elements) are usually consists of an assembly of two or more components or surfaces; such as doors, portals and windows, which are known as composite walls. For this reason, we can divide the walls elements into components (termed in this thesis as “patches”). Let’s consider a patch on the wall element , with surface area . Under diffuse sound field conditions the reverberant part of the incident sound power on any patch of any wall element is given by Equation **3.8**.

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|  | **(3.8)** |

On the other hand, under free-field conditions, the incident direct sound power is calculated on this patch and is given in Equation **3.9**.

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|  | **(3.9)** |

In Equation **3.9**, is representing the source directivity in the direction of this patch, and and are the distance and the incidence angle from the source point to the infinitesimal patch area . These quantities ( and ) depend on the room geometries. If the integral in Equation **3.9** is represented by Equation **3.10**,then by combining Equation **3.8** and Equation **3.9** the incident sound power on a single patch of wall element results in the form of Equation **3.10**.

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|  | **(3.10)** |
|  | **(3.11)** |

The integral is approximated numerically for not very large patches and in not very close positions to the walls as introduced by [**REF**] and it is even more appropriate after the wall has been subdivided into small segments (patches), thus relaxing the condition of constant conditions on the surface. This integral is obtained by assuming that , and do not vary significantly along,, therefore, these factors are taken out of the integral and approximate solution of Equation **3.10** is now given as . The vector is the distance from the source to the centre of patch with an incidence angle and denotes mean directivity value in the direction . The integral is calculated by the adaptive Simpson’s integration method. The incident power on patch given in Equation **3.11**, is now represented by its corresponding instantaneous incident sound power in time domain as follows.

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|  | **(3.12)** |

In Equation **3.12**, is the source signal normalized in power and is energetically normalised impulse response of the source room for patch of wall element , from which the direct sound is removed. The synthesis of source room impulse responses at the surfaces of the patches is necessary to include the temporal effects of the source room, the effects of absorption of room boundaries as well as to simulate the cases where an equivalent real room is not present.

* + - 1. Sound Transmission

ISO standard, provides comprehensive formulation for calculating transmission coefﬁcient which are based on spatially averaged sound insulation factors, such as; vibration velocities on the surface of elements, radiation efficiencies and bending wave transmission across the junctions, hence represents point to point transmission. Furthermore, it assumes the diffuse field theory and average the transmission coefﬁcient as . However, for an isotropic, uniform thickness plane wall the sound transmission coefﬁcient of a plane wave depends on the angle between the direction of propagation of the incident plane wave and the normal to the plane of the wall (see Chapter **5**). This is made possible by setting the coincidence angle equal to 90°, which does not exist below the critical frequency. A major advantage of this approach is that there is only a very slight discontinuity at the critical frequency. The angle dependent transmission coefficient is denoted by and is given in Equation **3.13**.

|  |  |
| --- | --- |
|  | (**3.13**) |

is the bending wave impedance of the wall. The single sided radiation efficiency is explained in Chapter **5**, is given in the following from as (for ). Therefore, Equation **3.13** can be rewritten as,

|  |  |
| --- | --- |
|  | (**3.14**) |

By inserting impedance and sigma values from Chapter **5** the final transmission factor is given by,

|  |  |
| --- | --- |
|  | (**3.15**) |

However, due to the assumption of infinite size plates in the derivation of Equation **3.15**, important discrepancies may be found between predicted and experimental data [**REF**]. An approach proposed by Vigran [**REF**] to obtain the transmission coefficient for finite plate is to calculate the transmission coefficient by using Equation **3.16**.

|  |  |
| --- | --- |
|  | (**3.16**) |

On the other hand, flanking transmission is more complex phenomenon than the direct sound transmission as it involves building elements connected to each other through junctions. It is very important to take into account the bending waves while considering the flanking transmission. The bending wave travels through one element in the source room hits at the junction and travels to other element in the receiving room. The detailed theoretical concept for the calculation of sound power from flanking transman is discussed in Chapter **5**. Once the transmission coefficients for individual patches are computed, it can be proceeded towards calculating transmission coefficients for each path from source room to the receive room defined in ISO 12354-1 and is given in Equation **3.17**. Here and are the surface areas, and and are the transmission coefficients the flanking element and of the source and receiving rooms respectively. The surface area of the partition between the source and receiver rooms is denoted by and the vibration transmission over junction between the elements and element is represented by , which can be measured in accordance with ISO 10848-3 or ISO 10848-4 [**REF**].

|  |  |
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|  | (**3.17**) |

* + - 1. Sound Field in the Receiver Room

Once the sound is transmitted form source room through building elements via direct as well as flanking elements, it is radiated from the receiver room walls to the receiver end. From the ISO 2354-1, the sound power transmitted from element of the source room to element of the receiver room for direct and flanking paths is defined by Equation **3.18**, which is the final sound power of any radiating element in the receiver room.

|  |  |
| --- | --- |
|  | **(3.18)** |

Using Equation **3.17** in Equation **3.18**, we get the expression of radiated sound power in the following form.

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|  | **(3.19)** |

In Equation **4.19**, the sound power is obtained by taking sum of incident sound power of all the single patch from Equation **3.12** and using in Equation **4.19**, the expression of sound power for the receiver room can be written in the following form.

|  |  |
| --- | --- |
|  | **(3.20)** |

In simplified approach (Section **3.1**), it assumed that the sound is apparently radiated from a single point. Here we take each radiating element , of the receiver room, are represent it by a set of evenly distributed point sources on its surface as shown in the Figure **3.3**.

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|  |
| **Figure 3.3**: *Segmenting receiver room wall as secondary source* |

At this point, we can distribute the transmitted acoustic power radiated by element , among these secondary sources homogeneously by a factor where are the total number of secondary sources on element . The sound energy , radiated by a single secondary source of wall element , with as its directivity is then calculated from Equation **3.21**.

|  |  |
| --- | --- |
|  | **(3.21)** |

The mean squared sound pressure of a secondary source for path in the receiving room can be derived from Equation **3.22**.

|  |  |
| --- | --- |
|  | **(3.22)** |

Using from Equation **3.21** into Equation **3.22** we get the final sound pressure for transmission path given in Equation **3.23**.

|  |  |
| --- | --- |
|  | **(3.23)** |

In Equation **3.23**, represents the distance between the acoustic centres of the radiating secondary source of the wall element of receiver room to evaluation point (i.e. position of the receiver). Finally, the time domain representation of the binaural signal at receiver point is obtained by introducing room impulse response of the receiver room and the HRIR filters for each secondary source to the receiver depending on its position and orientation relative to the secondary sources.

|  |  |
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|  | **(3.24)** |

The term in Equation **3.24** is the phase of the patch radiations, which depends on the incidence angle of the plan wave in air and bending wave number. All are statistically valid for all points inside both the source and the receiving rooms that is why can be synthesized before implementing the auralization filter chain. However, as it can be assumed that does not vary significantly for different source/receiver positions, hence, and can be computed independently to avoid coherent interferences in the reverberant field coming from different radiating elements.

* + 1. **Façade Sound Insulation Filters**

Façade sound insulation model is based on the filter design techniques described in the previous sections, however, the procedures of filter design for sound levels should cover sound transmission loss of exterior walls, roof constructions, windows, and some doors. Therefore, we keep the idea of segmenting building elements into finite size patches know as secondary sound sources (SS), because generally the exterior walls of common buildings are consisted of an assembly of two or more parts or surfaces (e.g. windows etc.). ISO 12354-3 provides basic guide lines for airborne sound insulation against outdoor sound source which are based on diffuse field theory. The diffused sound field approximations are very advantageous for the indoor cases where the adjacent rooms are assumed to be diffusive, however, this is not the case for outdoor environments as there are no reverberant or diffused sound fields outside the buildings. Hence, we will take into account the direct part of the sound field at surfaces of the building elements exposed. We might consider direct sound transmission paths through each small part of the element (secondary sources) because it can be assumed that the transmission for each secondary source is independent from the transmission from the others. Therefore, for a direct sound field (i.e. non-diffuse) the sound transmission coefﬁcient of a plane wave depends on the angle of incidence between the direction of propagation of the incident plane wave and the normal to the plane of the exterior elements (i.e. walls). We shall take the angle dependent radiation efficiency to calculate the . The detailed model to calculate this transmission factor is described in Section **3.13.2**, therefore, in this section we will modify the formulation for sound insulation against outdoor sources.

To proceed for the filters design, let us take a sound source with directivity, and the acoustics power (in Watts). The mean squared sound pressure at any point at a distance , from the source in energetic notations is given by Equation **3.25**.

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|  | (**3.25**) |

The sound power on the external surfaces of the building elements can be calculated with a simple modification to the stationary sound fields of the ordinary room, which mean that to separate the direct sound field and the reverberant part. Therefore, under free field conditions the direct incident sound power denoted by is given by Equation **3.26** on a secondary sound source (a component of the walls such as door, window etc.) of a surface area of . Where, is the distance from the source acoustic centre to the infinitesimal element (i.e. secondary source SS on the surface on the wall) and is the incidence angle. The quantities , and depend on the element geometry.

|  |  |
| --- | --- |
|  | **(3.26)** |

Thus the incident power on the each secondary source of an element can be calculated by,

|  |  |
| --- | --- |
|  | **(3.27)** |

Using Equation **3.26** and definition of the transmission coefficient, the sound power transmitted from source to receiver room by one secondary source is given by,

|  |  |
| --- | --- |
|  | **(3.28)** |

Using Equation **3.18** into Equation **3.27**, we can calculate , and finally the contribution of the path for the finite size patch to the mean squared pressure in the receiving room is derived as follows by using the expression given by,

|  |  |
| --- | --- |
|  | (**3.29**) |

The time domain representation of the sound pressure for single is given by,

|  |  |
| --- | --- |
|  | (**3.30**) |

For an input time signal, , and transfer function, , the final sound pressure in temporal domain form source to receiver for one SS is given by,

|  |  |
| --- | --- |
|  | (**3.31**) |