16 DESIGN AND CONSTRUCTION OF OFFICE BUILDINGS

Commercial buildings present a set of design issues that differ somewhat from those encountered in residential structures. In critical spaces such as classrooms, theaters, and studios, much of the information presented in Chapt. 15 is still applicable. In commercial office buildings the uses can vary, and may include less sensitive spaces. Structures are likely to be multistory, with an air handling unit on each floor located in a central core, along with other services such as elevator shafts, stairwells, and bathrooms. Mechanical equipment is also located on the roof, sometimes directly above the most prestigious and expensive floor space. The main air ducts are sized, based on the clearance afforded by the ceiling heights, and velocities may be relatively high. Air is likely to be returned through a common plenum, which complicates the room-to-room noise transmission problem.

In high-rise office towers exterior walls can be continuously glazed curtain walls, framed outside the flooring system, with narrow mullions, leaving little opportunity for closure of gaps at the interior walls and floors. Floor coverings may not be selected based on considerations of footfall noise. Conflicting uses may abut one another. These and many more details, which are critical to a satisfactory work environment, may not be totally under the control of the acoustical engineer, but nevertheless should be addressed.

16.1 SPEECH PRIVACY IN OPEN OFFICES

Privacy

When the intelligibility of speech is low, not surprisingly it follows that the privacy is high. Table 16.1 (ANSI S3.5) shows the relationship between Articulation Index, signal-to-noise ratio, and privacy. Written characterizations of these degrees of privacy were given in Table. 3.6.

In an office environment, where speech privacy is the goal, we can strive to achieve the required signal-to-noise ratio in several ways. The general form of the intelligibility equation is

$$\begin{bmatrix}\text{Source}\\ \text{Sound}\\ \text{Level}\end{bmatrix} - \begin{bmatrix}\text{Sound}\\ \text{Attenuation}\\ \text{(Reduction)}\end{bmatrix} - \begin{bmatrix}\text{Masking}\\ \text{Sound}\\ \text{Level}\end{bmatrix} = \begin{bmatrix}\text{Signal to}\\ \text{Noise Ratio}\\ \text{(Privacy)}\end{bmatrix} \tag{16.1}$$

TABLE 16.1 The Relationship between Intelligibility and Privacy

Articulation Index	Signal to Noise	% Sentences Understood	Intelligibility	Privacy
> 0.4	> 0 dB	> 90	Very Good	None
0.3	−3 dB	80	Good	Poor
0.2	−6 dB	50	Fair	Transitional
0.1	−9 dB	20	Poor	Normal
< 0.05	−12 dB	0	Very Poor	Confidential

Clearly we have three possible ways of influencing the signal-to-noise ratio: 1) control the sound source, 2) increase the path attenuation, and 3) raise the masking sound level. The source can be oriented to take advantage of the natural directivity of the human voice. The direct sound path can be controlled by using full or partial height barriers. Reflections can be attenuated with absorptive materials, and we can electronically generate background noise to mask intrusive speech. The details of each of these features depend on the office configuration and other design considerations.

Privacy Calculations

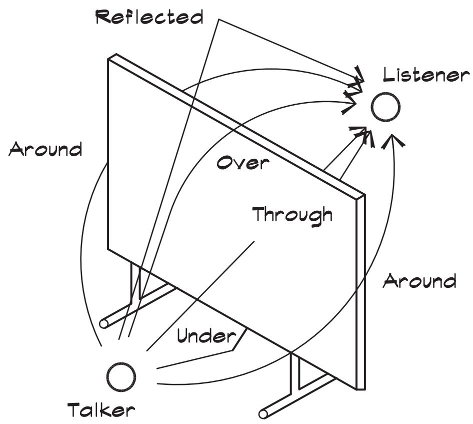
Calculations based on Eq. 16.1 can be carried out in individual third-octave or full-octave bands. The most accurate method is to use third octaves and the Articulation Index to determine intelligibility; however, there are also useful composite methods. Articulation calculations begin with the layout of the working environment.

Offices are configured either as separate rooms with full or partial height walls or as an open plan layout. Open-office plan refers to a system of workstations, distributed about an open floor, separated by partial height barriers. This design approach yields a flexible and relatively inexpensive work space, which if properly designed can furnish a degree of privacy for telephone and other conversations. Originally developed in the 1960s, its effectiveness unfortunately was oversold at first and its reputation subsequently suffered from unfulfilled expectations. Problems also arose when the system was only partially implemented.

Privacy in an open-office work environment can be achieved only when all the critical components are present and properly implemented. These include: 1) careful arrangement of the furniture and occupants, including the orientation of both talkers and listeners; 2) partial height barriers of the correct type, height, and location; 3) highly absorbent ceiling and wall panels; and 4) masking sound having the proper spectral content and level.

Chanaud (1983) and others have developed detailed methodologies for evaluating speech privacy in open offices by calculating the speech intelligibility between workstations. All calculations are based on Eq. 16.1; in the office environment, however, there is a large number of potential sound paths to be considered. Figure 16.1 shows several, each of which must be evaluated to arrive at the final signal-to-noise ratio at a receiver. Figure 16.2 presents a diagram of the separate calculations broken down into individual steps. The final result is a composite signal-to-noise ratio, which leads to the Articulation Index at the receiver. Note that each of these calculations is based on direct-field transmission, where the sound wave has undergone at most one or two reflections. In open-office environments a classic

FIGURE 16.1 Panels and Screens as Speech Barriers



reverberant field does not exist, particularly with intervening barriers and highly absorptive ceilings.

The Articulation Index (AI) is calculated from partial articulation indices (PAI), which are signal-to-noise ratios in each third-octave band, weighted according to the importance of the band for the understanding of speech. The Articulation Index is the sum of these individual weighted signal-to-noise ratios

$$AI = \sum_{i=200}^{5000} PAI_i \tag{16.2}$$

The signal-to-noise ratio in a particular third-octave band, between 200 Hz and 5000 Hz, is calculated from the source strength, the particular path attenuation (which is called the speech reduction), and the masking spectrum present at the receiver location. Each partial Articulation Index is determined from the signal-to-noise ratio multiplied by the weighting factor particular to a given band

$$PAI_i = (VS_i - SR_i - MS_i)WF_i \tag{16.3}$$

where PAI = partial Articulation Index in a given third-octave band

VS = average peak voice spectrum, in a given third-octave band, of the male voice at one meter, on axis (dB)

SR = speech reduction—difference between the VS level and the level at the receiver in a given third-octave band (dB)

MS = third-octave masking sound spectrum (dB)

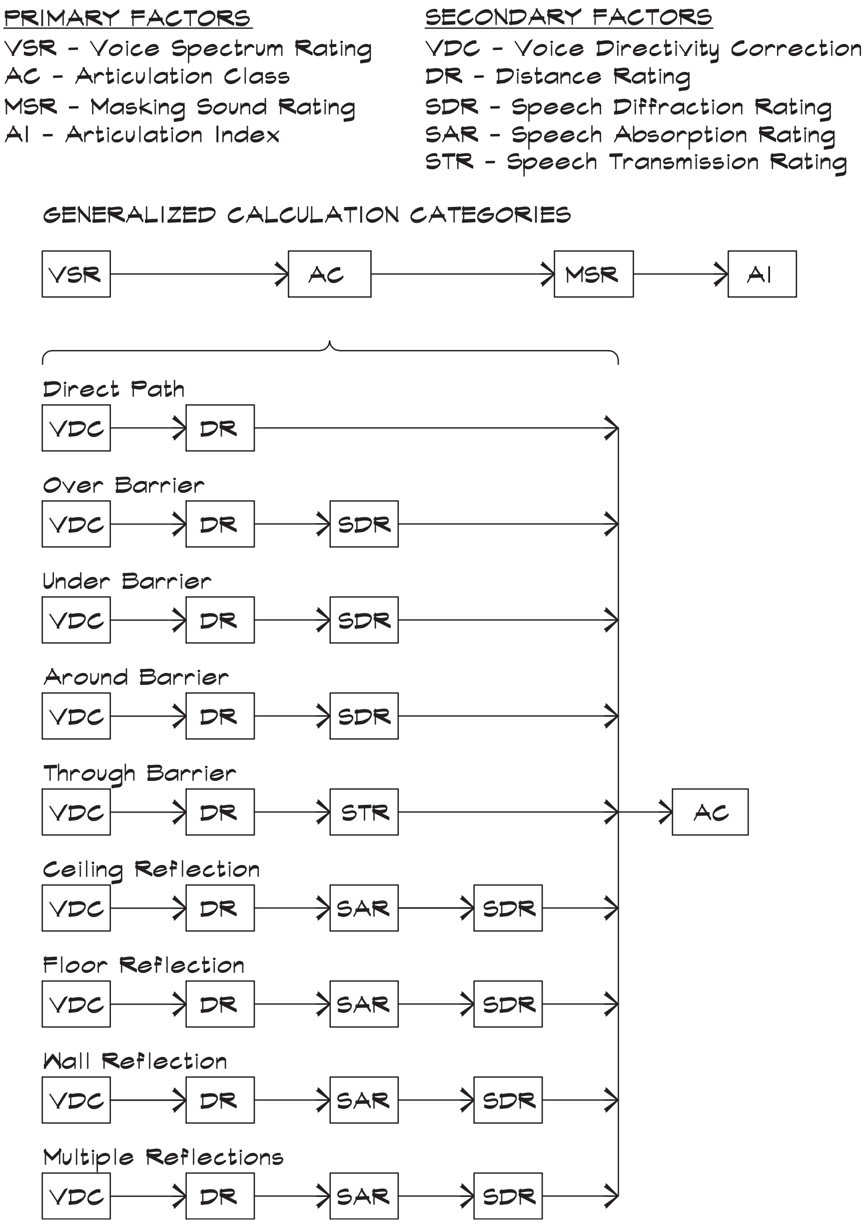
WF = third-octave Articulation Index weighting factor

All calculations are limited by conditions on the partial AI components such that

$$\text{If } PAI_i < 0 \quad \text{then set} \quad PAI_i = 0 \tag{16.4}$$

$$PAI_i > 30\ WF_i \quad \text{then} \quad PAI_i = 30\ WF_i$$

FIGURE 16.2 Open Plan Speech Rating Diagram (Chanaud, 1983)



We begin with the sound pressure level and spectrum generated by a male voice adjusted for voice effort and weight it according to the way the ear hears. The voice spectrum (VS) in Fig. 16.3 is the third-octave sound pressure level of the male voice peaks, which differs somewhat from the energy average levels cited previously. The levels in each band are weighted with the AI speech weighting factors, WF, given in Table 16.2. For simplicity we can use the Speech Rating Factor (SRF), which is 30 times the AI weighting factor (WF) and sums to one.

FIGURE 16.3 Average Peak Male Speech Spectra (ANSI S3.5, 1997)

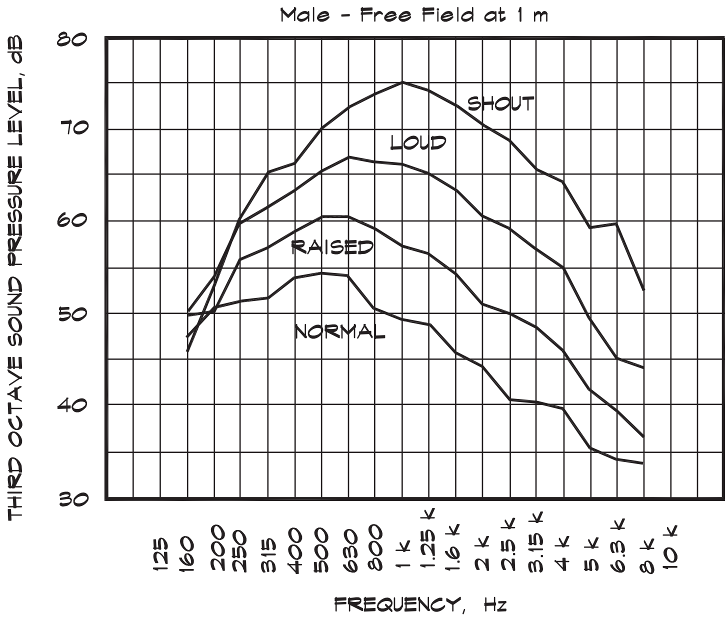


TABLE 16.2 Speech Weighting Factors (ASTM E1110-86)

	One Third Octave		One Octave		
Freq.	**SRF**	**WF**	**SRF**	**WF**	**A-Weighting**
200	.012	.0004			−11
250	.030	.0010	.072	.0024	−9
315	.030	.0010			−7
400	.042	.0014			−5
500	.042	.0014	.144	.0048	−3
630	.060	.0020			−2
800	.060	.0020			−1
1000	.072	.0024	.222	.0074	0
1250	.090	.0030			1
1600	.111	.0037			1
2000	.114	.0038	.328	.0109	1
2500	.103	.0034			1
3150	.102	.0034			1
4000	.072	.0024	.234	.0078	1
5000	.060	.0020			1

The speech reduction (SR) is calculated from the directivity relative to the on-axis level associated with the listener direction, the loss due to distance, and the attenuation due to the interaction with objects associated with the particular sound path, including diffraction over barriers, transmission through barriers, or absorption due to reflections. When each path level is computed the overall level in each third-octave band is determined and we compare the composite sound pressure level with the masking sound spectrum $(MS)_i$ to calculate the Articulation Index and the degree of privacy.

To make the AI easier to understand, a metric called the Privacy Index (Chanaud, 1983) was introduced, which increases as the isolation increases. The Privacy Index (PI) is defined as the percentage of privacy in terms of the Articulation Index as

$$\mathrm{PI} = 100\,(1 - \mathrm{AI}) \tag{16.5}$$

The Privacy Index is easier to understand for many users. For example, a 95% Privacy Index is excellent privacy, and 60% is poor.

Articulation Weighted Ratings

The Articulation Index can be estimated (Chanaud, 1983) from three ratings instead of being calculated individually in each third-octave band.

$$\mathrm{AI} \cong \frac{1}{30}\,[\mathrm{VSR} - \mathrm{AC} - \mathrm{MSR}] \tag{16.6}$$

where the AI weightings have been included in the individual rating terms. The ratings are the voice spectrum rating (VSR), which is based on voice level; the speech reduction rating (SRR), originally developed by Chanaud (1983), which has been standardized (ASTM E1110-86) and renamed the Articulation Class (AC); and the masking spectrum rating (MSR) defined in Eq. 16.9.

$$\mathrm{VSR} = \sum_{i=200}^{5000} (\mathrm{VS}_i)\,(\mathrm{SRF}_i) \tag{16.7}$$

$$\mathrm{AC} = \sum_{i=200}^{5000} (\mathrm{SR}_i)\,(\mathrm{SRF}_i) \tag{16.8}$$

$$\mathrm{MSR} = \sum_{i=200}^{5000} (\mathrm{MS}_i)\,(\mathrm{SRF}_i) \tag{16.9}$$

The speech reduction calculation varies depending on the attenuation mechanism associated with a particular path. Each path attenuation can also be discussed in terms of a composite rating. The Articulation Class (AC), as contrasted to the STC rating, is a weighted measure of the noise reduction between two given positions. It can be determined by calculating the losses associated with each sound path.

For example, if the sound is diffracted over a barrier, we can calculate the barrier loss and the associated speech diffraction rating (SDR).

$$\mathrm{SDR} = \sum_{i=200}^{5000} (\Delta L_B)_i \, \mathrm{SRF}_i \tag{16.10}$$

For transmission through a barrier, the sound attenuation is the point-to-point direct-field noise reduction, based on the direct-field transmission loss at a given angle and frequency. The speech transmission rating (STR) is written as

$$\mathrm{STR} = \sum_{i=200}^{5000} \left\{ \left[\Delta L_{TL}(\theta) \right]_i \mathrm{SRF}_i \right\} \tag{16.11}$$

For surface reflections the speech absorption rating (SAR) is

$$\mathrm{SAR} = \sum_{i=200}^{5000} \left\{ \left[\Delta L_s \right]_i \mathrm{SRF}_i \right\} \tag{16.12}$$

where ΔL_s is the attenuation due to a reflection, based on the specular absorption coefficient given in Eq. 16.13.

These rating simplifications have the advantage of allowing a general discussion of the components, while retaining most of the original accuracy. The differences between the results using the separate calculation method and the one-number ratings are shown in Fig. 16.4, where Eq. 16.5 has been inserted into Eq. 16.6 for the estimation of the Privacy Index. The only thing that is dropped in this generalization is the limitation imposed under Eq. 16.4, which leads to a slight over-design at high isolation values.

Experience has shown that there is no single masking spectrum, which provides the most privacy at the lowest overall sound level, and is best for every situation. The shape of the spectrum depends critically on the details of the sound attenuation. Chanaud (2002)

FIGURE 16.4 Privacy Index vs Articulation Class Rating (Chanaud, 2000)

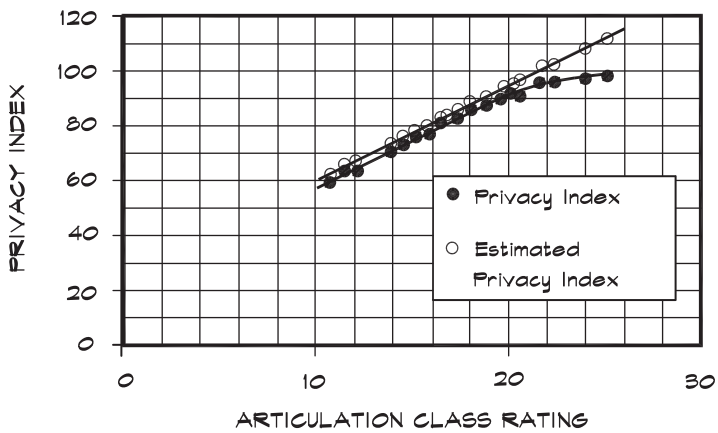


TABLE 16.3 Masking Sound Spectra (Chanaud, 2002)

	One-Third Octave		One Octave	
Freq.	**Open Office**	**Closed Office**	**Open Office**	**Closed Office**
200	43	43		
250	43	41	47	46
315	42	40		
400	41	39		
500	41	37	45	42
630	40	35		
800	39	34		
1000	38	32	43	37
1250	36	30		
1600	34	29		
2000	32	27	37	32
2500	30	25		
3150	28	23		
4000	26	21	31	26
5000	24	19		
dBA	47	43		
MSR	41	37		

suggests two good compromise spectra, one for open offices and another in closed offices, in Table 16.3.

Speech Reduction Rating and Privacy

In a calculation of the Articulation Index, based on the geometry sketched in Fig. 16.5, we begin with the source level and associated spectrum, and calculate the attenuation due to the directivity of the talker, the intervening distance, and barriers or absorbers, finally comparing the level in third octaves to the background level. The masking spectrum (MS) is the actual background noise in the receiving space.

For purposes of this analysis it has been assumed to be the open-office spectrum from Table 16.3. The worst case (least private) condition, at a given distance, would be represented by a face-to-face orientation with no intervening barriers. Table 16.4 shows the results of such a calculation, giving a compilation of the privacy expected from various speech reduction ratings, for a person talking at a normal voice level based on face-to-face orientation and an open-office masking sound level.

What we learn from this figure is that the entire span of privacy ratings falls within a 10 dB range of levels, and that a change of 1 dB in a rating results in a change of .03 in the Articulation Index. Thus even a one-point difference can have a noticeable effect on the degree of privacy. Given these relationships, we can refer to this chart when discussing speech privacy in terms of calculated AC values in more complicated configurations.

FIGURE 16.5 Noise Reduction Ratings by Path

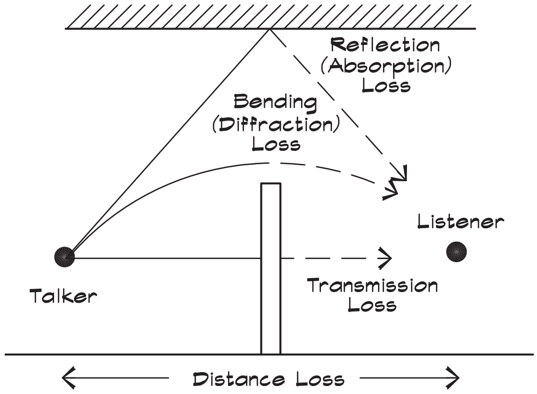


TABLE 16.4 Influence of Sound Attenuation on Speech Privacy (Chanaud, 1983)

Articulation Class	Articulation Index	Privacy Index	Degree of Privacy
16	.37	63	None
17	.33	67	None
18	.30	70	Poor
19	.27	73	Poor
20	.23	77	Poor
21	.20	80	Transitional
22	.17	83	Transitional
23	.13	87	Normal
24	.10	90	Normal
25	.07	93	Normal
26	.05	95	Confidential

Source Control

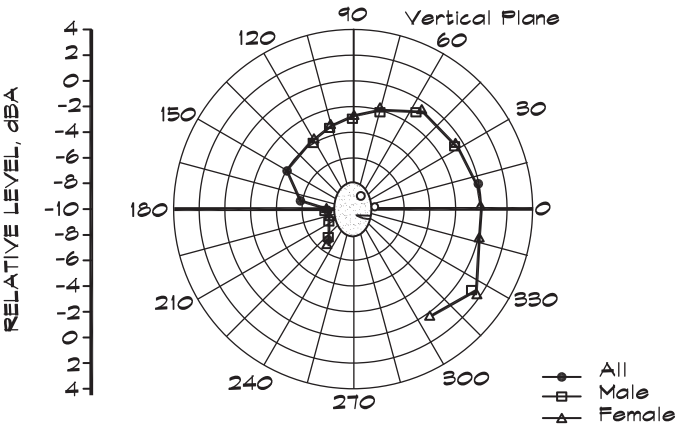
In open plan analyses the source model is the male voice, which has a characteristic spectrum, level, and directivity. In the analysis of speech, average voice peaks from Fig. 16.3 are used. The overall changes can be approximated by the vocal effort table in Table 16.5.

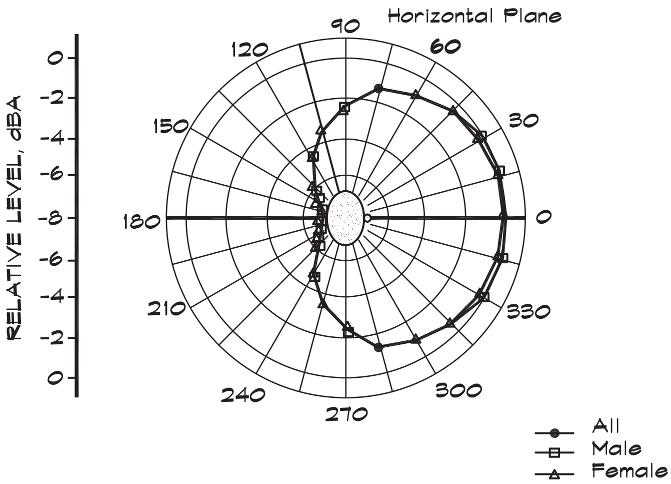
There are characteristic directivities associated with the human voice, which are shown in Fig. 16.6, relative to on-axis. It is assumed that the receiver has no directivity. From this table we see that a degree of natural attenuation may be achieved by taking advantage of the directivity of the human voice and the orientation of the talker and listener. If workstations are arranged so that the line between two conversing individuals in one workstation is at right angles to the direction of the listener in the next workstation, the directivity correction is maximized.

TABLE 16.5 Voice Level Corrections

Vocal Effort	Correction to Voice Spectrum
Normal	0 dB
Raised	+ 6
Loud	+ 10
Shouting	+ 20

FIGURE 16.6 Comparison of the Relative A-Weighted Levels in the Frontal Vertical and Horizontal Planes for Male and Female Talkers (Chu and Warnock, 2002)





Outside noise sources should also be controlled in open-plan offices. HVAC noise should be limited to NC 35 in open-office areas and to NC 30 in private offices. HVAC noise is not helpful as a masking source, since its spectrum is rarely the proper shape and, in any case, it is not adjustable or uniformly distributed throughout the space. Noisy office equipment should be located in separate rooms. Offices should be carpeted to reduce walking and furniture noise. The use of intercoms, personal radios, pagers, speaker phones, and other extraneous noise sources should be discouraged. Telephone rings should be set to their minimum volume.

Partial Height Panels

Although source orientation and control are helpful, barriers are always necessary. Partial height barriers include walls, which extend up to or even beyond the acoustical tile ceiling, but not to the slab or roof above. Prefabricated furniture panels can be used as speech barriers, but their inherent transmission loss limits their ability to perform this function. All the potential transmission paths must be considered, namely over, under, around, and through the barrier.

The total direct-field attenuation of a panel is the composite of the paths in Fig. 16.2. First, the barrier must block the direct path between talker and listener. If the listener is within 12 m (40 ft) of the talker, as in Fig. 16.7, the barrier must overlap the line of sight by at least 0.3 m (1 ft) to be effective. At distances of less than 4.5 m (15 ft) the sound path should have to bend at least 90° or more, as in Fig. 16.8, to be sufficiently attenuated.

Panel height is also an important factor. Barrier attenuation for both over and around paths may be calculated using Maekawa's relationship from Eqs. 5.10 and 5.11. The attenuation depends on the relative heights of the talker, barrier, and listener, as well as the frequency, and approximate results are given in Fig. 16.9.

FIGURE 16.7 Determining the Line of Sight for Distant People (Chanaud, 1983)

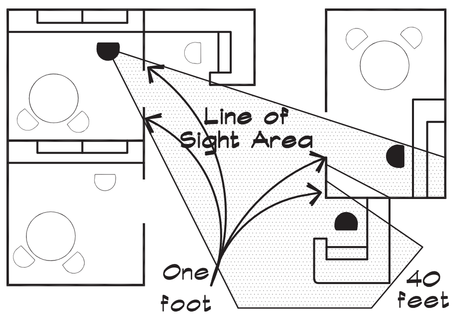


FIGURE 16.8 Determining the Line of Sight for Near People (Chanaud, 1983)

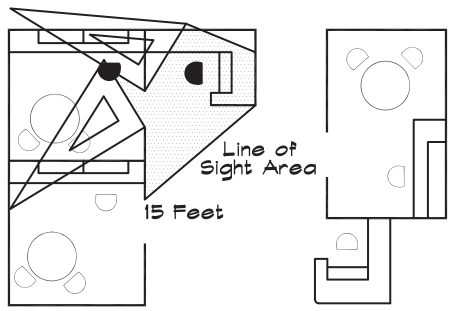


FIGURE 16.9 Influence of Panel Height on Speech Diffraction (Chanaud, 1983)

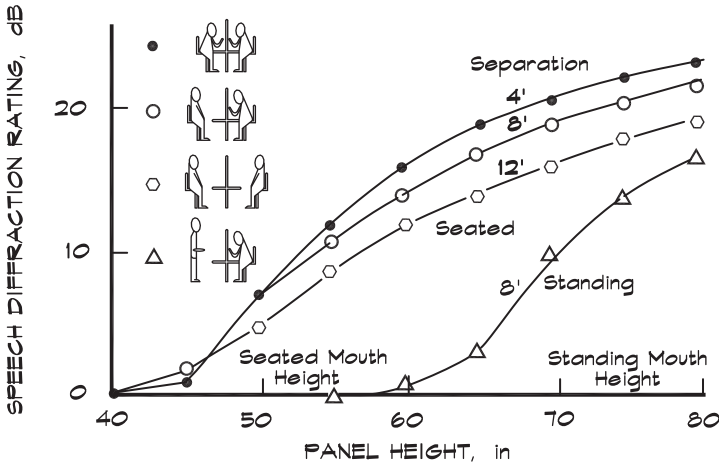
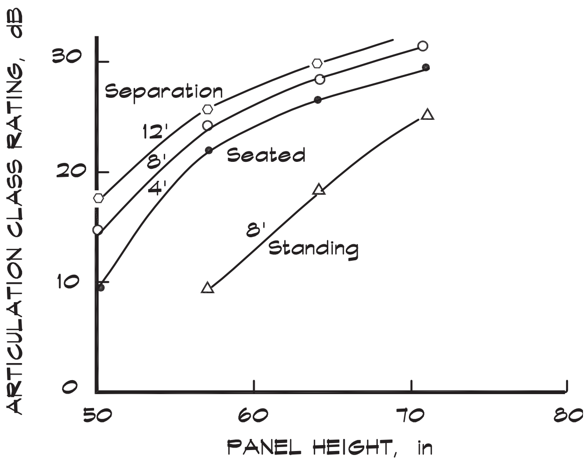


FIGURE 16.10 Influence of Panel Height and Distance on Speech Reduction (Chanaud, 1983)



When the distance and directivity attenuations are taken into account, the results are the speech reductions shown in Fig. 16.10. For seated occupants, at least a 1.8 m (70 in) panel height should be used, and if there are a significant number of standing conversations, a 2 m (80 in) panel height is recommended. In some situations it is desirable to be able to look around the room or over individual panels. In these cases panels may incorporate sections of glass without significant degradation.

In all cases it is also important that partial height barriers extend down to the floor to seal off the transmission path under the panel. A well-designed system includes blockage of this reflection. Panels should leave no more than a 25 mm (1 in) gap at the floor since carpet yields relatively low absorption values in the speech frequencies.

FIGURE 16.11 Influence of Panel Width on Speech Reduction (Chanaud, 1983)

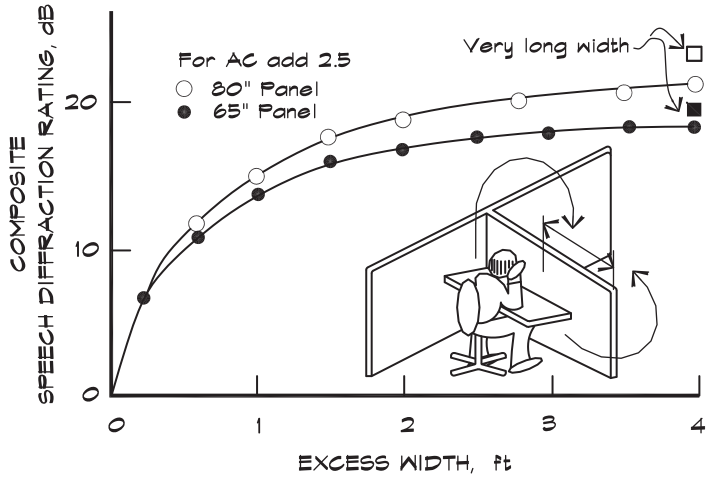
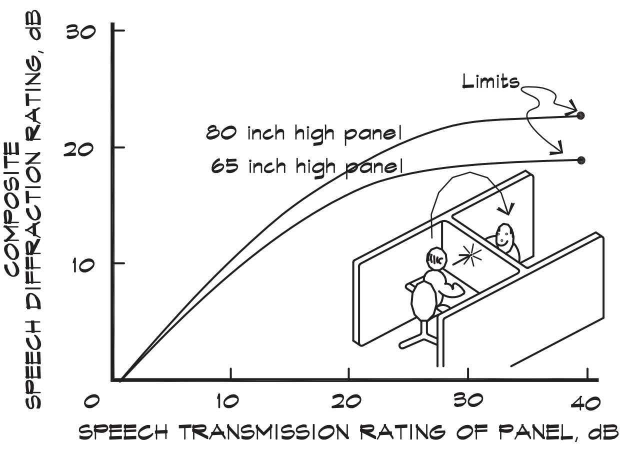


FIGURE 16.12 Influence of Transmission Loss on Speech Reduction (Chanaud, 1983)



Diffraction around the end of a panel must also be controlled. Figure 16.11 shows the effect of excess width on the speech diffraction rating of a side panel. Free-ended panel runs should be avoided and the ends capped with a right-angle piece. For 1.8 m (70 in) high panels the free end should extend at least 1.2 m (4 ft) beyond a seated worker. For 2 m (80 in) high barriers, no free-end conditions should be allowed. Free-ended panels are a problem, not only because of the diffraction, but also because they allow easy transmission of reflected sound.

Transmission of sound directly through a panel is also a concern. Since there is a natural limit to the effectiveness of a panel for diffraction losses, this sets a maximum on the necessary transmission loss. Figure 16.12 gives the diffraction limits for two heights of panel. The corresponding STC limits are 20 for 1.65 m (65”) panels and 24 for 2 m (80”) panels.

To achieve an STC rating of 20 to 24 a surface mass density of about 5 kg/sq m (1 lb/sq ft) is required. This can be accomplished using 3/8" plywood, 1/4" gypboard, or a 22 Ga steel sheet sandwiched between two absorbent fiberglass boards. Manufacturers of office furniture sometimes refer to this type of panel as "acoustical" or "high performance" in their literature, although the presence of these descriptors does not guarantee this rating. Manufacturers must be contacted for the actual STC ratings of a prospective material.

Absorptive and Reflective Surfaces

Absorptive surfaces should be used in locations where they will prevent reflected sound from flanking the main barrier, including ceilings, rear, and sometimes side walls. The sound reflecting from a surface is attenuated by an amount given by

$$\Delta L_s = 10 \log (1 - \alpha_s) \tag{16.13}$$

where α_s is the specular absorption coefficient.

Although the NRC ratings are useful for a general discussion of the properties of a given panel, if calculations are to be undertaken more detailed data are necessary. Random-incidence absorption coefficients are measured in third-octave bands, although they are published in octaves. Due to the details of the testing process, values of these coefficients sometimes exceed 1. In the case of ceilings and other reflecting surfaces the specular absorption coefficients are of more interest but are rarely available. Chanaud (2000) has developed an empirical relationship relating the diffuse (α) and specular (α_s) absorption coefficients for open-office calculations. He suggests

$$\begin{aligned} &\alpha_s \cong \alpha \quad \text{for } \alpha < 0.5 \\ &\alpha_s \cong .092 + .82\,\alpha \quad \text{for } 0.5 < \alpha \leq 1.1 \\ &\alpha_s \cong 0.999 \quad \text{for } \alpha > 1.1 \end{aligned} \tag{16.14}$$

Since most reflections between workstations are within 20° to 30° of the normal, there is little change in absorption with angle, as we saw in Fig. 7.19. When losses from reflections are combined with the directivity and the distance attenuation, we obtain the speech reduction for a reflected path

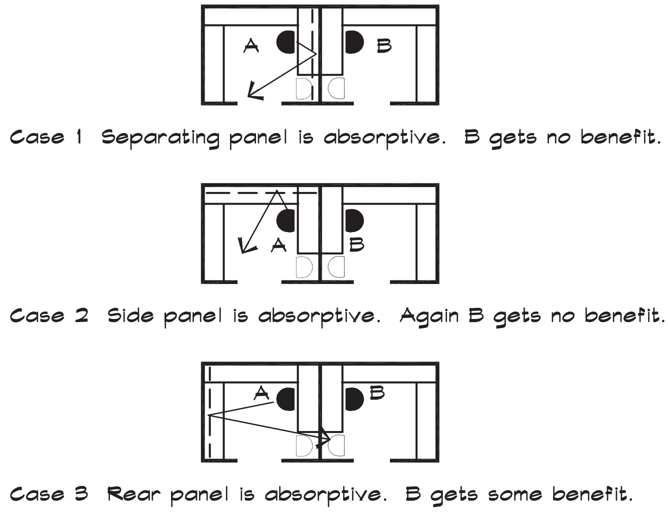
$$SR = \Delta L_\theta + \Delta L_s + 20 \log (D_s) \tag{16.15}$$

where D_s is the total reflected path length in meters.

Figure 16.13 shows several examples of absorptive panel placement. If the separating panel, located directly along the line of sight between the source and receiver, is absorptive, its first reflection does not go toward the receiver, so the absorption is not in the most beneficial position. A side-panel reflection may not go directly to the receiver and so here, too, absorption may not be effective. An absorbing ceiling and rear panels decrease the sound transmission reflecting to the receiver and are effective. We therefore arrive at the conclusion that we place absorptive panels overhead and in the direction opposite the person to be given the privacy. Where there are people in all directions the application of this rule will result in several panels being covered with soft material.

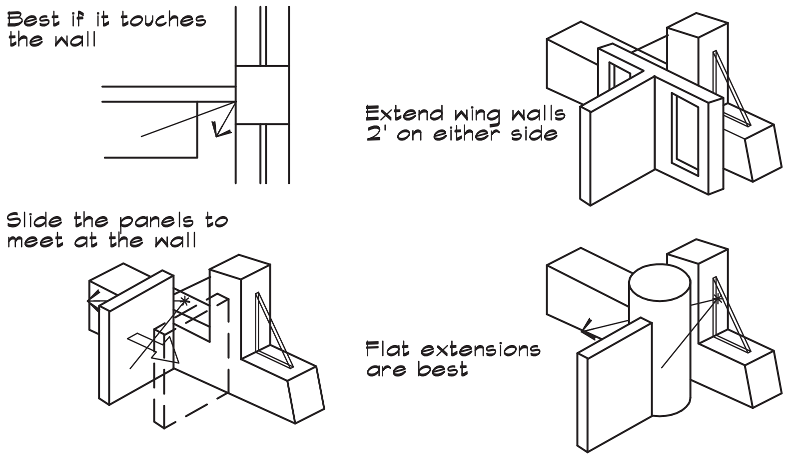
Absorption should also be placed where a reflection would create a flanking path around the barrier panel. Reflections can occur from other panels, or from the walls and windows of

FIGURE 16.13 Absorbing Panels in Workstations (Chanaud, 1983)



Panel Type	Typical NRC	Reflection Attenuation
Hard Surface	0.05	0 dB
Acoustical	0.80	13 dB
Sound Screens	0.90	15 dB

FIGURE 16.14 Open Plan Panel to Window Connections



the building. If the reflecting surface is within 2 m (6.6 ft) of the talker the noise reduction coefficient (NRC) of the panel must be at least 0.9. For each additional 0.3 m (1 foot) of separation the NRC of the panel may be reduced by 0.05. Absorbent panels placed on potential reflecting surfaces must be long enough to cover all possible talker and listener mirror locations within their respective workstations. This is illustrated in Fig. 16.14.

Carpeted floors do little to improve speech privacy when panels extend all the way down to the floor. They do help control extraneous noise from sliding furniture, walking, and general reverberant noise at high frequencies. Their use is recommended in open-office areas for these reasons; however, they do not have to be chosen based on their NRC ratings.

Open-Plan Ceilings

The ceiling is a potential reflecting surface, which must be treated in open offices. The choice of ceiling materials was once limited to plaster, gypsum board, and mineral or fiberglass tile. Recently these traditional materials have been supplemented with many additional products including perforated and linear metal shapes, wire mesh, and sintered metal products. The traditional acoustical tile products have become more interesting visually and may be wrapped with cloth, perforated vinyl, or other porous materials.

An absorptive ceiling tile consists of three parts: backing, core, and facing. The most commonly available combinations are listed in Table 16.6. For open-plan offices, ceilings must be highly absorptive. Fiberglass tiles with a cloth facing have the highest ratings. A fiberglass tile having a 19 mm (3/4”) thickness will have an NRC rating of about 0.90 and at a 38 mm (1.5”) thickness its rating can sometimes exceed 1.0. Mineral-tile ceilings have NRC ratings, which range from 0.55 to 0.65 depending on thickness and surface treatment, whereas a gypboard ceiling has an NRC of about 0.05. Metal-foil backings should be used where the ceiling plenum is also a return air conduit. Table 16.7 (Chanaud, 1983) gives the speech reduction rating or Articulation Class for various ceiling absorption coefficients at an 8-foot talker-to-listener separation distance.

A comparison with the privacy ratings in Table 16.4 reveals that Normal Privacy will not be achieved if an acoustically hard ceiling is present. Transitional Privacy can be achieved with a mineral tile ceiling, but Normal Privacy can only be achieved by using a fiberglass ceiling tile in conjunction with 2 m (80 inch) high panels with masking. Figure 16.15 shows the variation of Articulation Index with ceiling material in the presence of masking sound.

The ceiling height can also have an effect on the degree of privacy. As the ceiling height increases the distance loss for reflected sounds becomes greater. Since speech reduction is a combination of distance attenuation and the absorption of the ceiling panels, a higher ceiling may allow the selection of a less expensive ceiling material. Table 16.8 (Chanaud, 1983) shows a comparison of the speech reduction rating for various ceiling heights.

A mineral tile ceiling must be 4.6 m (15 ft) high if it is to result in the same attenuation as a fiberglass tile ceiling at 2.6 m (8.5 ft). There is little advantage to be gained from raising

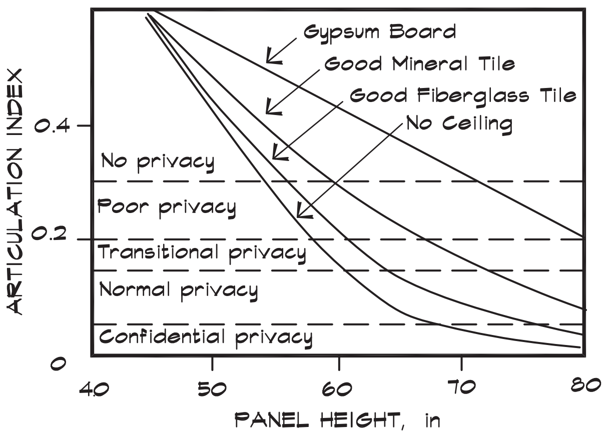
TABLE 16.6 Ceiling Tile Combinations

Backing	Core	Facing
None	Mineral Tile	Glass Cloth
Metal Foil	Fiberglass	Vinyl
		Perforated Vinyl
		Painted Cloth
		Metal Pan

TABLE 16.7 Speech Reduction Caused by Ceiling Absorption (Chanaud, 1983) 2.5 m (8 Foot) Separation

Ceiling Type	NRC	AC (dB)
1.65 m (65 Inch) Panels		
None		27
Ideal Material	1.15	26.5
Good Fiberglass Tile	1.00	25
Standard Mineral Tile	0.65	21.5
Gypsum Board	0.05	17
2 m (80 Inch) Panels		
None		33
Ideal Material	1.15	31.5
Good Fiberglass Tile	1.00	28
Standard Mineral Tile	0.65	22.5
Gypsum Board	0.05	17

FIGURE 16.15 Factors Influencing Speech Privacy with Masking Sound (Chanaud, 1983)



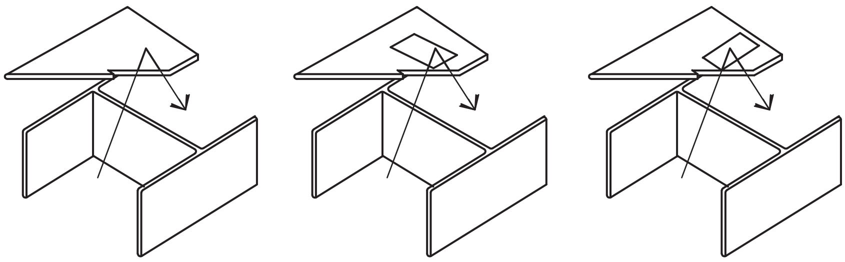
a ceiling, if it is already highly absorbent. Conversely, if a ceiling is quite high, there is little privacy improvement to be gained by changing to a more absorbent material.

Reflective light fixtures and metallic diffusers should not be located where a reflection will cause a degradation of the ceiling absorption. A flat reflective light fixture, such as those in Fig. 16.16, located above a separating partition, can produce a reflected level equal to the effect of changing the entire ceiling from fiberglass to mineral tile.

TABLE 16.8 Influence of Ceiling Height on Speech Reduction (Chanaud, 1983) 1.3 m (4 foot) Separation—1.35 m (65 Inch) Panels

Ceiling Height (m/ft)	AC (dB) Fiberglass	Mineral Tile
2.4 (8)	23	19
2.6 (8.5)	23	19.5
2.7 (9)	23.5	20
3.0 (10)	23.5	21
3.7 (12)	24	22
4.6 (15)	24	23
6.1 (20)	24.5	23.5
30 (100)	24.5	24.5

FIGURE 16.16 Influence of Light Fixtures on Ceiling Reflections (Chanaud, 1983)



Privacy for 65 inch panels on three sides,
fiberglass ceiling, and masking sound

Condition	Privacy Index	Degree of Privacy
No light fixture	84	Normal
Fixture in short direction	69	Poor
Fixture in long direction	67	Poor

Masking Sound

The addition of background or masking sound to an open-office environment is a critical component to obtaining overall privacy in the workplace. When properly done, masking sound raises the noise level in an unobtrusive way that increases privacy without being noticed by the occupants. It can be used in open-office plans with partial-height barriers and in private offices with walls that are either full or partial height. Even when a private office with a full-height wall is adjacent to an open-office area, it is still a good idea to introduce masking sound into the private office. This smoothes the transition between the spaces and makes the noise in the open area much less noticeable. Masking sound is produced by using an electronic pink noise generator and a filter that results in a spectrum that falls off at about

FIGURE 16.17 Influence of Masking Sound on Speech Privacy (Chanaud, 1983)

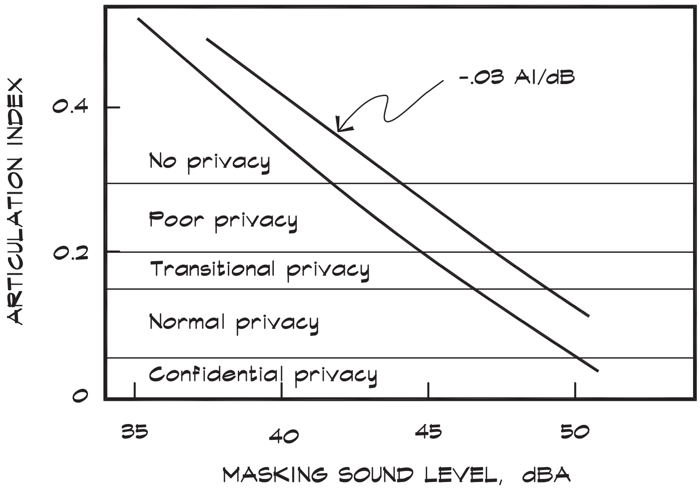
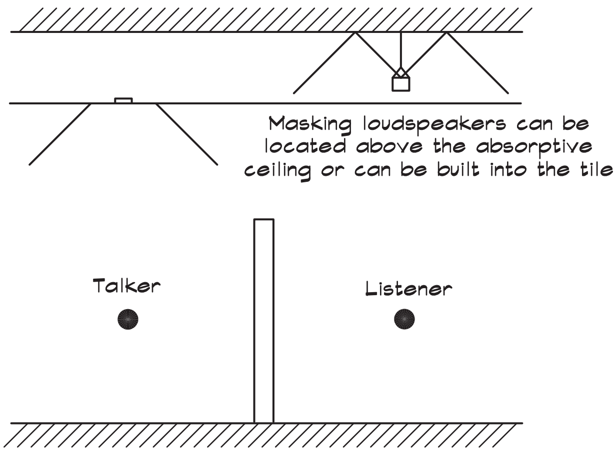


FIGURE 16.18 Masking Sound Loudspeaker Locations



5 to 6 dB per octave at the receiver. A possible spectrum is shown in Table 16.3; however, there is no single ideal curve. The overall level is set between 43 and 49 dBA. Though this seems like a relatively narrow range, below 43 dBA masking sound has little effect, and above 49 dBA it is an annoyance. Most systems end up being adjusted to somewhere around 47 dBA in open offices and 44 dBA in closed offices. The effect of masking sound on privacy for a typical workstation is shown in Fig. 16.17.

Loudspeakers can be located above the ceiling tiles, out of sight, and pointed up to increase their effective distance and widen their coverage pattern. They may also be built into the tiles themselves, which decreases the installation cost. Figure 16.18 gives two examples. The introduction of masking sound above a workstation provides privacy for a listener located within the sound field of the loudspeaker. It does not provide masking of the sound made by a talker located there. Hence, loudspeakers must be distributed through all areas where there are potential receivers. As an employee walks around the workspace, the sound level should not change drastically. He should not be aware of the presence of masking, so any changes in level or spectrum should be gradual. Work areas should be zoned so that levels may be adjusted according to the needs of a particular area. Masking systems can be controlled using a timer that varies the level in a given zone during the workday; however, the system should not be switched off during working hours.

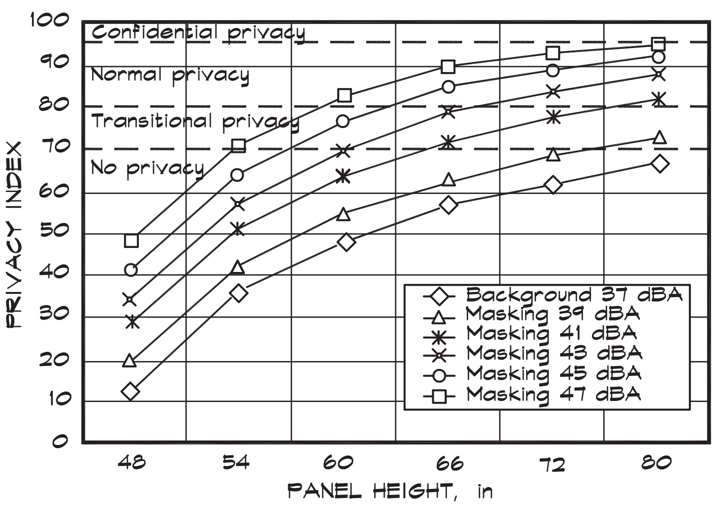
Music and paging may be mixed into the system, but the masking sound should only be muted for emergency pages. Music is not a good source of masking sound by itself, but it can provide a pleasant background environment for certain tasks. Music is not appropriate for areas where difficult tasks are performed. When music is introduced the masking sound must remain on.

Degrees of Privacy

We now have in place the tools to provide various degrees of privacy in an open (or private) office environment. As was emphasized previously, all component parts must be in place to achieve privacy. Figure 16.19 shows the influence of panel height and masking sound on levels of speech privacy. Clearly neither high panels nor masking sound alone can yield good privacy.

A successful open-office design must include four key elements and fails if any of the four is missing. The elements are: 1) partial height barriers at least 1.65 m (65 in) high, having sufficient transmission loss, provided by a 3/8" plywood or other interior panel; 2) absorptive material typically 25 mm (1 in) fiberglass panels on the reflecting walls or additional barriers to prevent flanking; 3) a highly absorbent ceiling (NRC > .85); and 4) an electronic sound-masking system with loudspeakers located above the acoustical tile ceiling set to emit a particular spectrum in the range of 45 to 49 dBA.

FIGURE 16.19 Privacy Index with Ceiling NRC = 0.9 and a 6-Foot Separation (Chanaud, 1983)



16.2 SPEECH PRIVACY IN CLOSED OFFICES

Private Offices

In a closed office the assessment of privacy is expressed in terms of STC values between adjoining spaces. Closed offices have the advantage of providing privacy throughout the enclosed space for a standing or seated occupant. The disadvantage is that the normal background level is lower, even when masking is included, and conversations may take place at raised voice levels even when confidential matters are being discussed. Specialized areas such as psychologists' offices, spaces used for conflict resolution, rooms with audio visual systems, lecture rooms, and classrooms, are all likely to have a need for extra isolation and should be identified as part of the initial planning process.

The analysis of sound transmission for private offices is similar to that for open offices. In critical locations detailed calculations using voice spectra, transmission loss, and background noise levels should be done. Figure 16.20 shows the calculated Articulation Index for various FSTC values at three voice levels, and Table 16.9 includes the corresponding sound attenuation requirements for various degrees of privacy. Chanaud (1983) recommends that the FSTC of a given structural component be 6 dB greater than the desired composite FSTC.

Full-Height Walls

The traditional approach to closed-office privacy is to use full-height walls and weather-stripped solid core doors. When the walls are full height, the total FSTC value is obtained. Sketches of several wall configurations are given in Fig. 16.21, along with estimates of their composite FSTC values. When the wall does not extend full height on both sides various plenum barrier materials can be used to make up the difference.

For Confidential Privacy with a normal voice talker, a good separation wall between rooms is a single 3 5/8" (90 mm) metal stud with 5/8" (16 mm) drywall each side and R-11 batt in the airspace. This wall rates an STC 44 (FSTC 39), and for voice provides a calculated noise reduction of approximately 42 dB. In terms of a rough calculation, a 60 dBA

FIGURE 16.20 Required FSTC for Speech Privacy for Various Voice Levels (Chanaud, 1983)

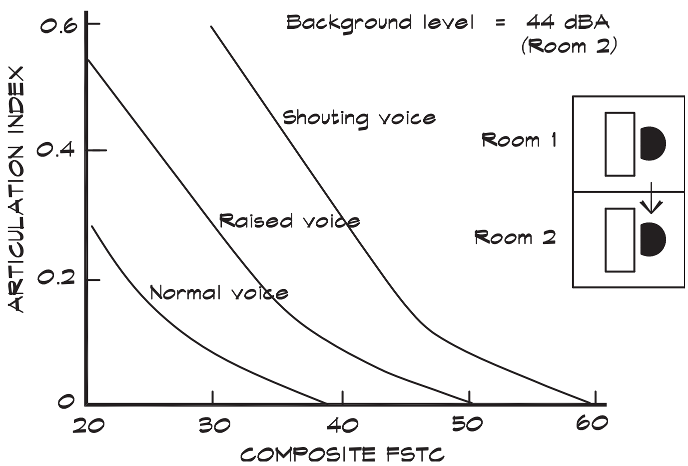


TABLE 16.9 Required Composite and Component FSTC for Various Voice Levels (Chanaud, 1983)

Background Noise = 44 dBA

Minimum Composite FSTC for

Voice Level	Normal Privacy	Confidential Privacy
Normal	26	32
Raised	36	42
Shouting	46	52

Recommended Component FSTC for

Voice Level	Normal Privacy	Confidential Privacy
Normal	32	38
Raised	42	48
Shouting	52	58

sound pressure level due to a normal voice in one room would generate about 18 dBA sound pressure level in the adjacent room. If the masking level in the receiving space were 35 dBA, we could achieve Confidential Privacy since the signal-to-noise ratio is less than –12 dB. This describes a typical private office with a background level of about NC 30 although the NC spectrum is not identical with the masking spectrum.

Plenum Flanking

The choice of an appropriate wall type is not the end of the design exercise, since there are many other routes that the sound can take from one room to another. If the wall does not extend from slab to slab, then the sound can travel over the top of the wall, passing through the plenum above the T-bar ceiling on either side. Since the mass of acoustical tile is quite low, it does little to attenuate noise from passing through it. Blazier (1981) has published measured values of the noise reduction of an acoustical tile ceiling, which are shown in Fig. 13.15. Even if these are doubled and increased by 6 dB, following Eq. 9.50 they still represent a significant degradation in transmission loss relative to the wall performance. Grille openings for the return air plenum can further degrade these ratings.

Figure 16.22 gives several examples of FSTC values for partial-height walls flanked by plenums. In this figure there are substantial differences in FSTC values depending on the ceiling material. Manufacturers of acoustical ceilings have sought to develop so-called high transmission loss tiles, which combine both absorption and transmission loss into a single product. These tiles have a modest absorption coefficient, between 0.5 to 0.7, and a relatively low effective STC rating (as measured with a wall in the STC 30 to 35 range). Even though they are better than lighter acoustical tiles, they can be compromised by openings for return air grilles.

FIGURE 16.21 Full Height Separation Wall Configurations

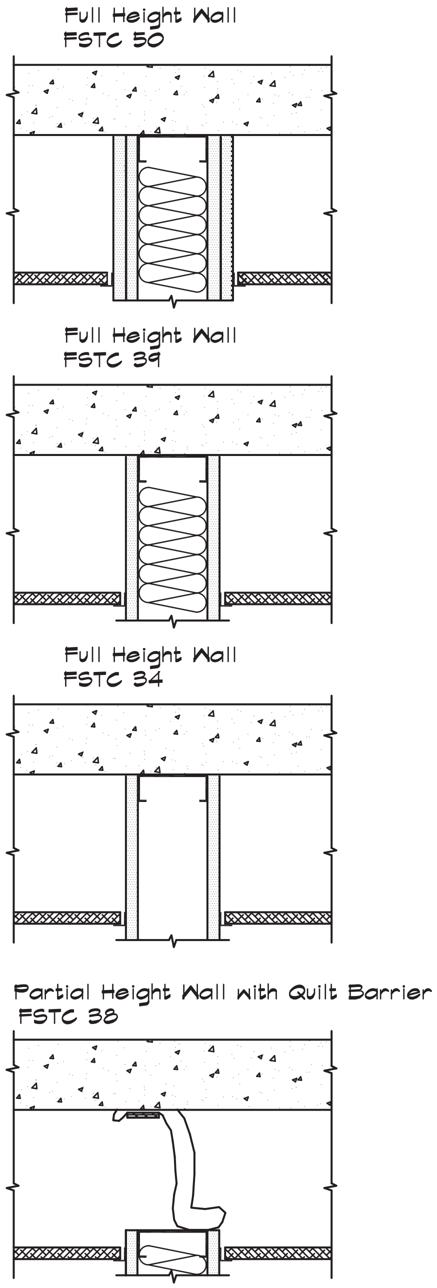


FIGURE 16.22 Wall Plenums with Estimated FSTC Values

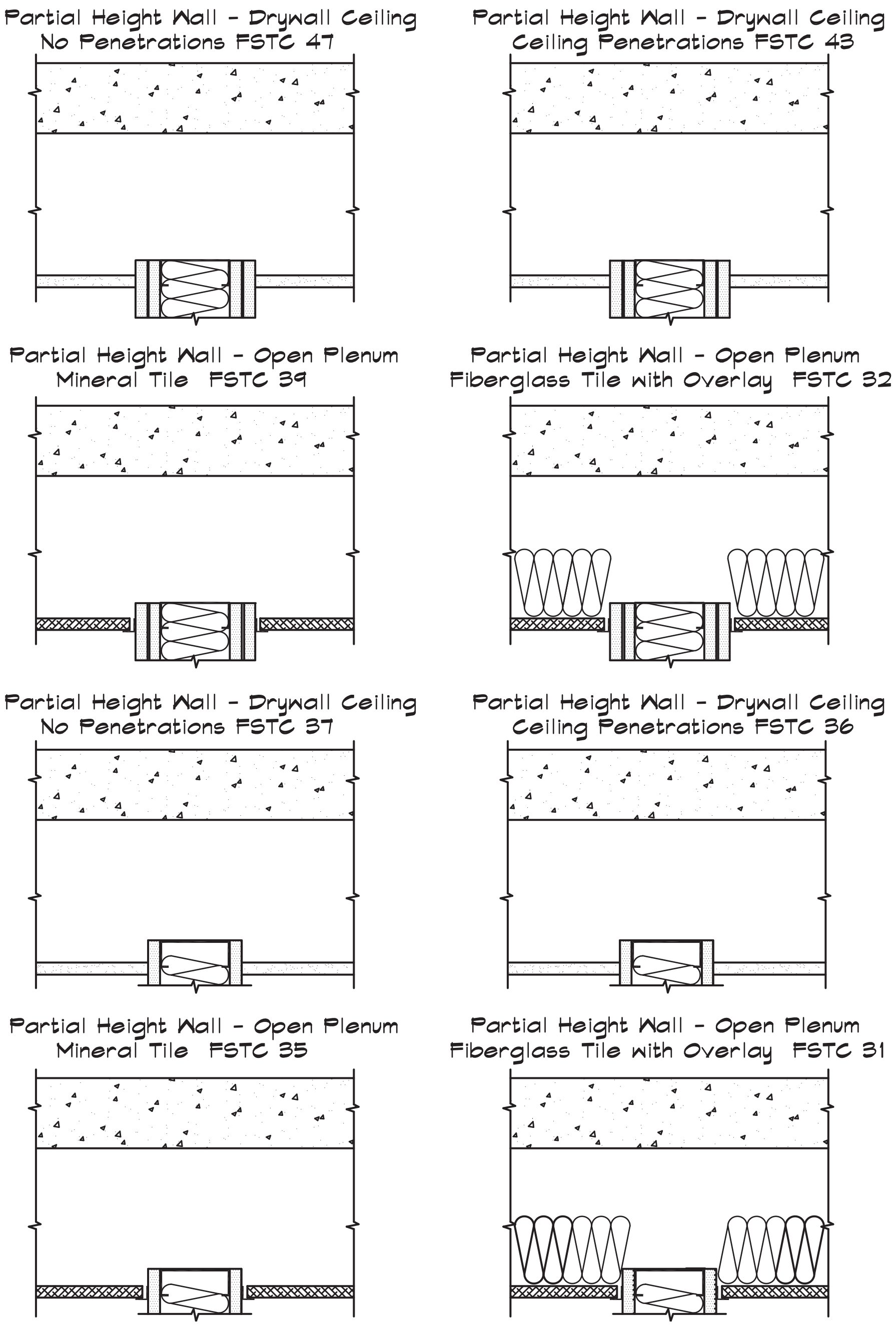
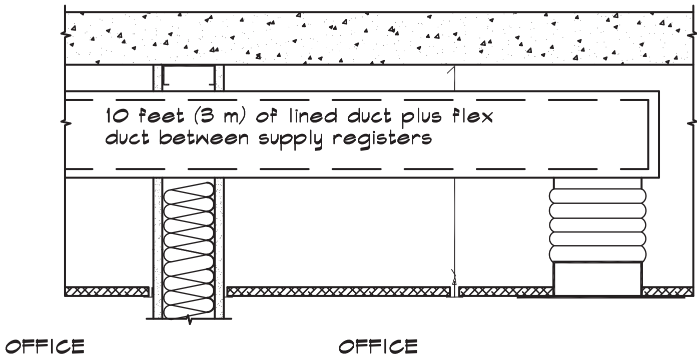


FIGURE 16.23 Flanking through Ductwork



The addition of batt insulation on top of the T-bar ceiling can provide some improvement for fiberglass tile ceilings, which have relatively low transmission loss values. Batt is not appropriate in return air plenums and instead a duct liner should be used.

Duct Flanking

HVAC ducts can also serve as a conduit for sound transmission between adjacent rooms, particularly if they are unlined. If an unlined duct directly connects diffusers in adjacent offices, sound can propagate along the duct and be heard clearly in the receiving room. To calculate this effect, the duct attenuation becomes the transmission loss and the cross sectional area of the duct becomes the transmitting area in Eq. 10.5. Where the supply or return air is ducted directly between sensitive spaces, a 3 foot (1 m) silencer or a total of 10 feet (3 m) of duct, lined with 1” (25 mm) duct liner, is required. If the ceilings are drywall, two 4-foot (1.2 m) lengths of flex duct can be substituted for the lined duct. A detail is shown in Fig. 16.23.

Where walls are constructed full height to the floor or roof above, they cut off the free circulation of air from the plenum to the rest to the return-air system. An opening in one of the walls, with 5 ft (1.5 m) of lined duct penetrating the plenum wall, is normally sufficient to isolate the two spaces and provide a low-pressure path for the air to return. Where the return air can be transported in the space above the corridors in an office complex, it provides for additional separation by forcing the noise to traverse two segments of lined duct. Figure 16.24 shows a typical arrangement.

Exterior Curtain Walls

Interior partitions separating adjacent offices will eventually meet an exterior wall. Ideally, the interior wall should join the exterior surface at an area of solid construction, so that acoustical isolation is maintained. When the walls intersect at a window mullion or at the glazing there can be a flanking path at the junction, particularly if the window is double glazed. Exterior window mullions are constructed from thin (typically 0.090” or 2 mm), hollow, rectangular aluminum extrusions with very little mass. Where the junction with an interior wall falls on the mullion the wall can be attached as shown in Fig. 16.25. For additional isolation an extra piece of drywall or heavy sheet metal can be attached to each side.

FIGURE 16.24 Return Air Plenum at a Full Height Wall

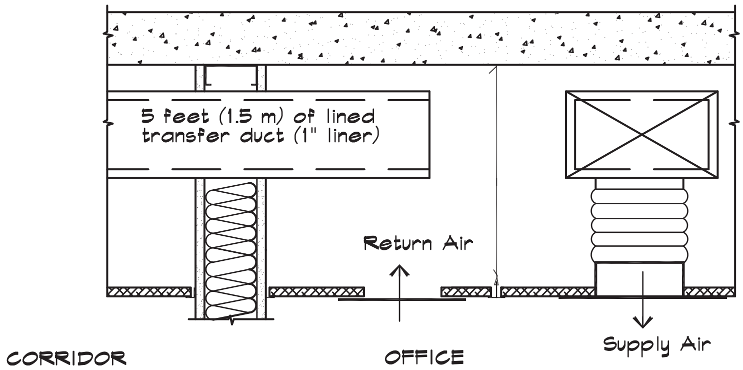
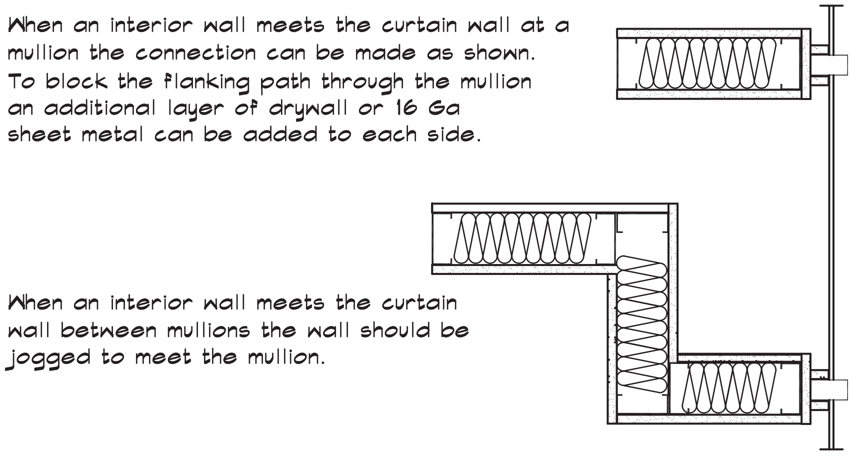


FIGURE 16.25 Flanking around Walls at Mullions

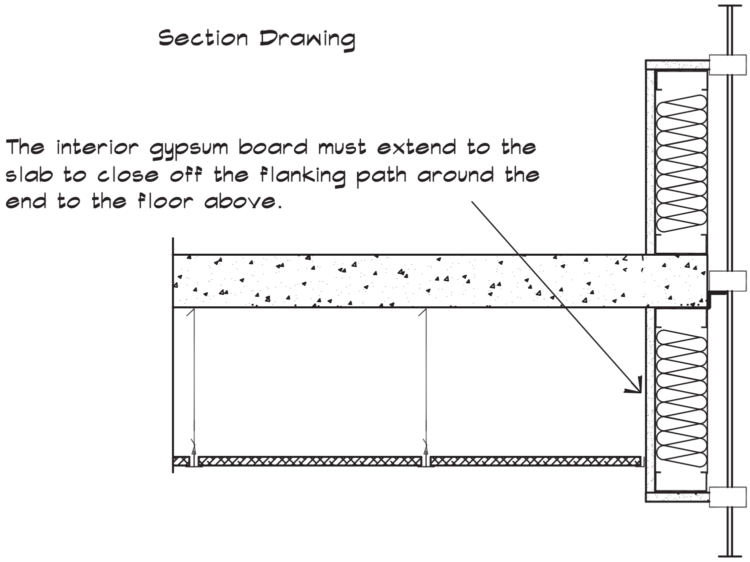


This thickens the appearance of the mullion as seen from the outside but provides much better isolation than stopping the wall short of the inner surface of the mullion.

Where the end of the wall falls between two mullions the wall should be jogged over until it falls directly on the mullion. Occasionally there is a continuous air bar around the outside of the building that penetrates the dividing wall. This can severely compromise acoustical isolation between spaces and must be closed off at the wall. Lined sections of duct can carry the air through the wall.

Not infrequently in high-rise construction the exterior curtain walls are supported from the edge of the floor slab and a gap is left between the slab and the glazing. Where there is spandrel glass and an interior wall, it is a good idea to continue the wall up past the ceiling as shown in Fig. 16.26. In other cases sheet-metal plates can be installed to bridge the gap between the slab and horizontal mullions both above and below the slab. The airspace should be filled with safing.

FIGURE 16.26 Flanking around Slabs at Curtain Walls



Flanking paths sometimes occur at column penetrations of slabs, particularly where the lower floor space has an open ceiling. In each of these conditions the openings around the column must be closed off with a material that has a sound transmission loss equivalent to the floor system.

Divisible Rooms

Moveable walls, used as dividers between meeting rooms, are commercially available with STC ratings as high as 50 or more. To achieve these ratings the panels weigh as much as 10 lbs/sq ft (49 kg/sq m). They are supported from a structural framework above that must be sized to carry the load without undue deflection. Unless this is taken into consideration the moveable walls can drag on the floor when they are moved into place. Some deflection is inevitable, and the panels are positioned and jacked up so that they support their own weight from the floor below. Moveable walls are the same as permanent walls in that the plenum flanking path must be blocked off. Figure 16.27 shows a design. Any stem wall must also accommodate structural deflection due to the weight of the panel.

Masking in Closed Offices

Where walls do not extend to the structural deck above the T-bar ceiling, it is not possible to achieve Confidential Privacy without the addition of masking sound. An office separation might consist of a single metal-stud wall with single layers of 1/2” gypsum board extending up to or slightly above the T-bar and a mineral-tile ceiling. Return-air grilles should be baffled with a lined sheet-metal enclosure illustrated in Fig. 16.28, with the open sides facing away from the receiver room. Table 16.10 shows the resulting degrees of privacy with this configuration for several levels of masking sound.

FIGURE 16.27 Installation of Moveable Wall Partitions

A wall of equivalent transmission loss should be built above the moveable wall. Studs and batt not shown for clarity.

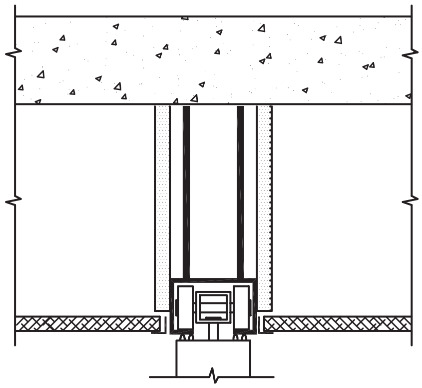


FIGURE 16.28 Return Air Boots

A sheet metal return air boot built over the return air grille. Orient the opening away from the receiver. Nominal dimensions 24" x 24" x 8" high.

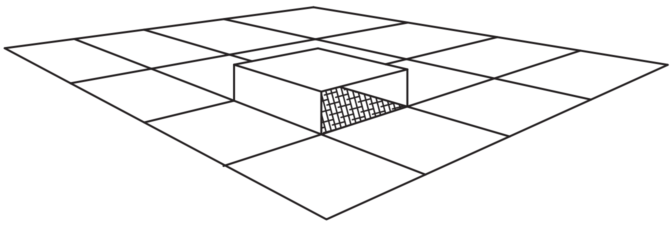


TABLE 16.10 Closed Office Speech Privacy for Mineral Tile Ceilings Normal Voice Levels (Chanaud, 1983)

	No Masking	Masking
Attenuation (NIC)	33	33
Masking Level (dBA)	35	39
Privacy Index	89	95
Degree of Privacy	Normal	Confidential

TABLE 16.11 Closed Office Speech Privacy for Fiberglass Tile Ceilings: Normal Voice Levels (Chanaud, 1983)

	Fiberglass Tile	Fiberglass with Overlay
Attenuation (NIC)	27	33
PI with 47 dBA	> 95	> 95
PI with 42 dBA	91	> 95
PI with 37 dBA	83	91

Note that in closed offices lower levels of masking sound are required to achieve Confidential Privacy than were necessary in open offices.

When fiberglass tile ceilings are present the low transmission loss values limit the attenuation between spaces. An overlay of 1/2" drywall on top of the tile can be helpful for improving the ceiling transmission loss. Table 16.11 summarizes the results of measurements for these conditions with booted return air grilles.

This solution is helpful in cases where the designer prefers the same style of ceiling tile in the open- and closed-office areas. Even with the drywall overlay the masking sound penetrates the ceiling, although the level must be increased. Loudspeakers should not be located near the return air grilles.

16.3 MECHANICAL EQUIPMENT

Mechanical systems in commercial spaces tend to be larger and more centralized than those in residential buildings. With larger units it becomes critical to address the vibration isolation aspects of the transmission problem and to enclose the mechanical spaces with heavy walls or buffer zones to protect the adjacent occupancies. Given the additional space, oft times these units are easier to treat, since there is room for silencers, lined plenums, and other conventional treatments.

System Layout

Architects and engineers can reduce the amount of treatment necessary to control mechanical noise in office spaces by shielding mechanical equipment rooms from sensitive spaces with intervening rooms such as bathrooms, storage rooms, and corridors. Bathrooms are particularly useful in this regard, since they include drywall or plaster ceilings, which can serve as return air plenums and provide additional space for the location of silencers.

Simply providing adequate space for mechanical equipment in a location that is isolated from potential receivers can be of great benefit. Many noise problems occur when this planning is not done. All too frequently air handlers or heat pumps are shoehorned into the available space in ceiling plenums above sensitive areas. Even when space is provided for the equipment, space is not left for a silencer, plenum, or other attenuating devices. This is often the case with down-shot air handlers, where the ducts emerge straight down from the underside of the unit, leaving little room for silencers and gradual turns. Where these devices are located above sensitive receptors it is very difficult to correct noise problems after installation.

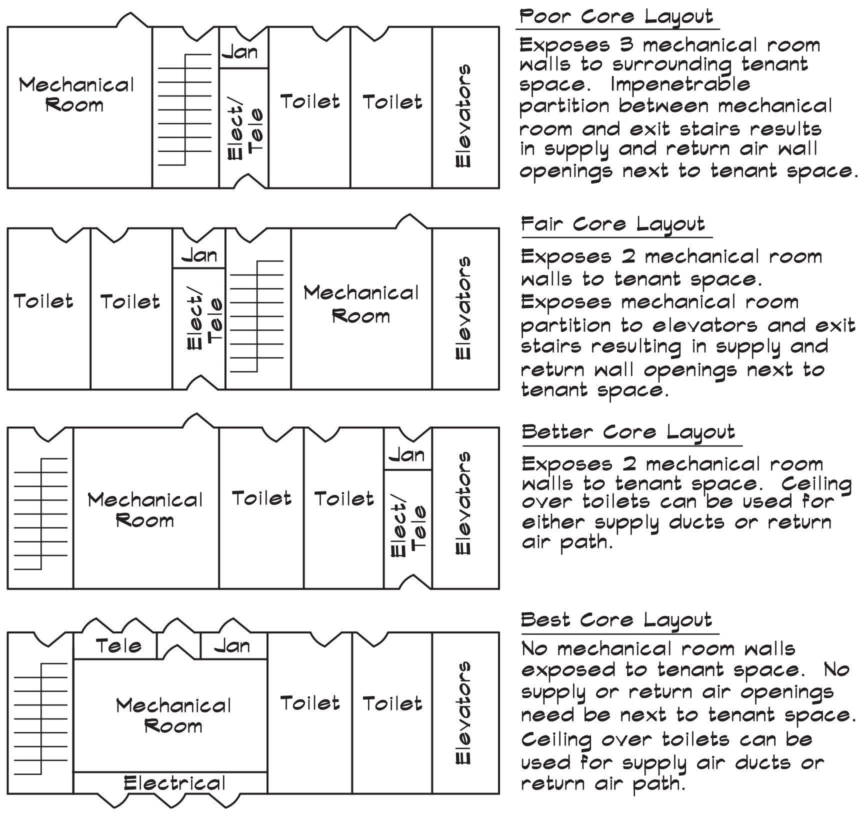
Mechanical Equipment Rooms

In a typical multistory office complex there is a central core where a mechanical equipment room is located. This is an excellent design configuration since it allows the necessary space for buffers, silencers, lined duct, or the other treatments. Several possible core layouts are shown in Fig. 16.29. The designs illustrate increasingly isolated arrangements with storage and acoustically benign equipment rooms acting as buffers.

From the central mechanical equipment room the supply air is fed into a duct loop serving the tenant spaces. The return air can be ducted or drawn from the ceiling plenum or from a plenum located above the central corridor. When the space is leased by a single tenant, there is often no reason for the corridor, and the lessor builds out the whole space to his liking. The full built-out condition is more difficult to treat acoustically than the subdivided condition since there is no buffering provided by the corridor.

In a central equipment room the air handler is located (Fig. 16.30) so that the supply ducts rise vertically to an elbow and pass horizontally through the walls of the mechanical equipment room to the duct loop. The air is returned through an opening in the mechanical equipment room wall and into the side of the air handler. Makeup air is supplied to the mechanical equipment room by means of a separate fan located on the roof.

FIGURE 16.29 Comparison of Various Building Core Layouts (after Schaffer, 1991)



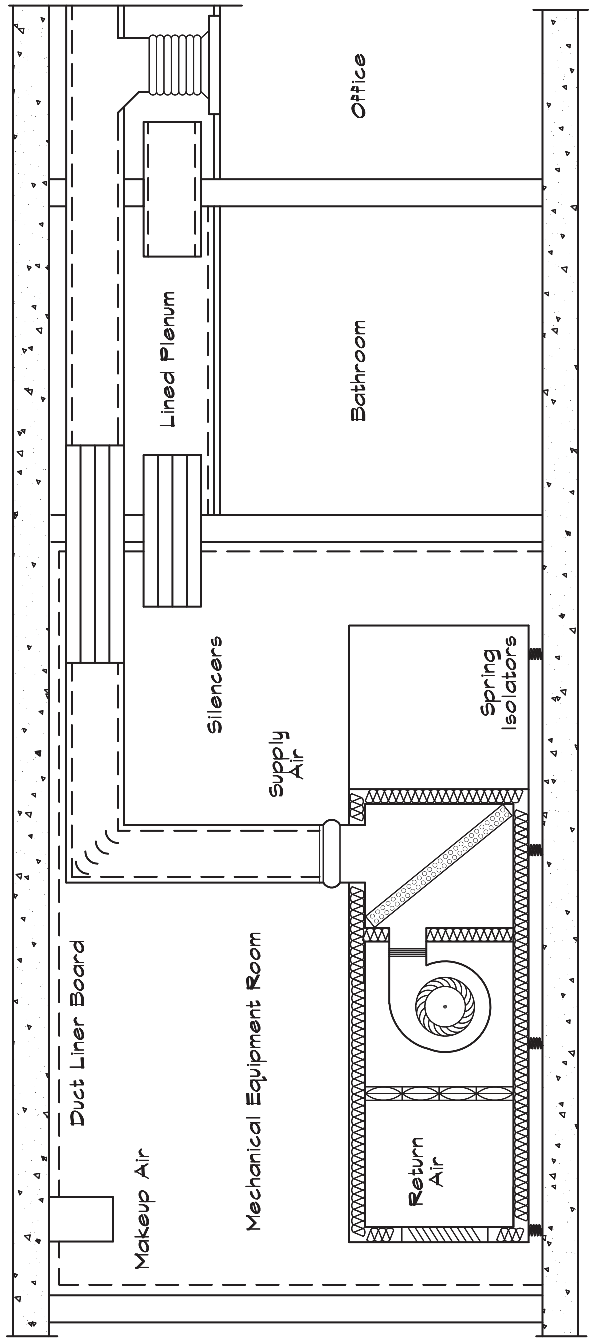


FIGURE 16.30 Central Core Air Handling Unit

Given the sound power level data on HVAC units at the specified operating point, a calculation can be carried out on both the supply side and return sides of the equipment as outlined in Chapt. 14. The final receiver sound pressure levels are compared with the interior noise level criteria to determine if any remedial steps need to be taken. In this case a silencer is added to the supply duct to provide the necessary attenuation.

On the return side the mechanical equipment room acts as an acoustical plenum with the return air opening on the air handler being the plenum entrance. If the radiating area of the air handler is not included as the entrance area, it is common to overestimate the attenuation due to the plenum effect. When the ceiling space above the bathroom is available it is useful to create an additional plenum and to use return-air silencers as well. Assuming that the exit from the plenum opens into the area above the acoustical tile ceiling we calculate the loss through the ceiling as an insertion loss to be subtracted from the sound power level before converting it to a sound pressure level using the appropriate room constant. An additional calculation of the sound radiated through the walls of the mechanical equipment room should also be done.

Roof-Mounted Air Handlers

In single story and low multistory projects packaged air handling equipment is frequently located on the roof. A packaged unit contains its own refrigerating condensing section along with the fan coil heat exchangers, filters, and one or more circulating fans. These units should be isolated with external spring isolation, on a series of open springs or curb rails, because internal isolation provided by the manufacturers may be inadequate and does not isolate casing vibration. The unit should be located over a stiff region of the roof having a deflection under the load of the equipment of no more than 1/6 to 1/8 of the spring isolator deflection. A housekeeping pad of at least 3” (75 mm) thickness should be provided for units up to 10 tons and 6” (150 mm) thickness above that. Pads must extend at least twice their thickness beyond the isolated equipment. Pads are strongly recommended, even when units are rail mounted on grade beams.

Side inlet and discharge are much preferred over down-shot units since they afford the space to install silencers. Down-shot units are often specified by mechanical engineers and contractors since they are self flashing, less prone to leaking, and less unsightly. Schaffer (1991) has published a series of figures showing various configurations of rooftop units, which are reproduced in Figs. 16.31 through 16.34.

Figure 16.31 shows a down-shot air handling unit with a large opening in the lightweight roof beneath. The opening allows the passage of noise through this space. To correct this condition, the unit must be picked up and mounted on rails to stiffen the structure and the opening must be boxed in from below with a metal stud wall with 2 layers of 5/8” (16 mm) drywall. Silencers are then installed at the points where the ducts penetrate the box. The details of the enclosure design must be determined by doing calculations using the sound power level emanating from the underside of the unit.

Figure 16.32 illustrates a configuration where some steps have been taken to control the noise. This arrangement is not sufficient above occupied office areas; however, with the addition of return-air silencers and a drywall box around the ductwork out to the silencers, it can be made to work in nonsensitive areas. Some manufacturers can provide high transmission loss ductwork of heavy (14–16 gauge) metal, which can help reduce breakout problems.

FIGURE 16.31 A Very Noisy Rooftop Air Handling Unit (Schaffer, 1991)

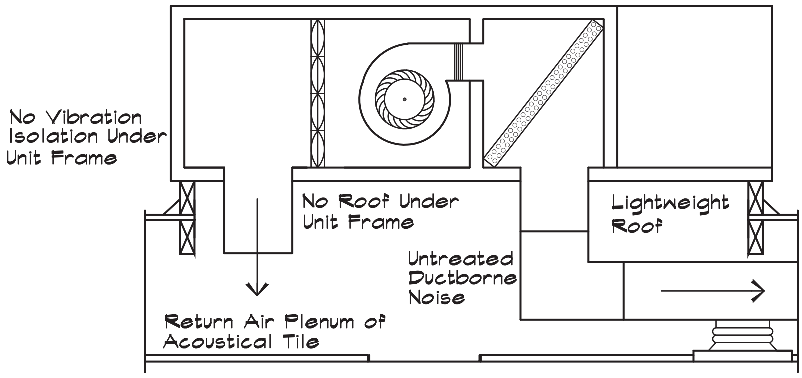


FIGURE 16.32 A Noisy Rooftop Air Handling Unit (Schaffer, 1991)

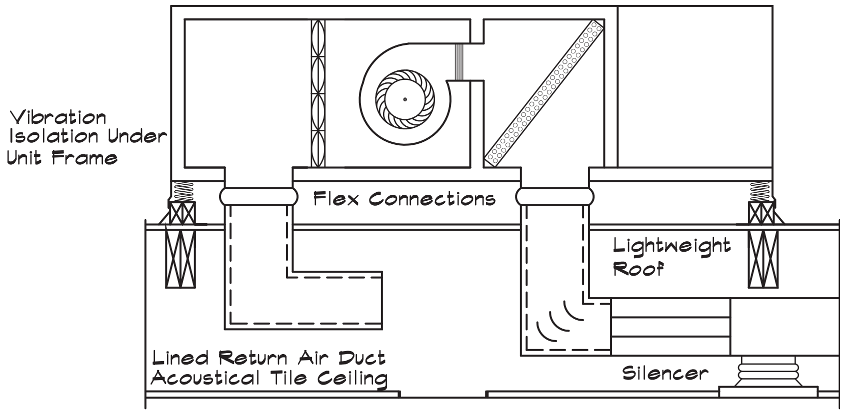


FIGURE 16.33 A Moderately Quiet Side Discharge Rooftop Air Handling Unit (Schaffer, 1991)

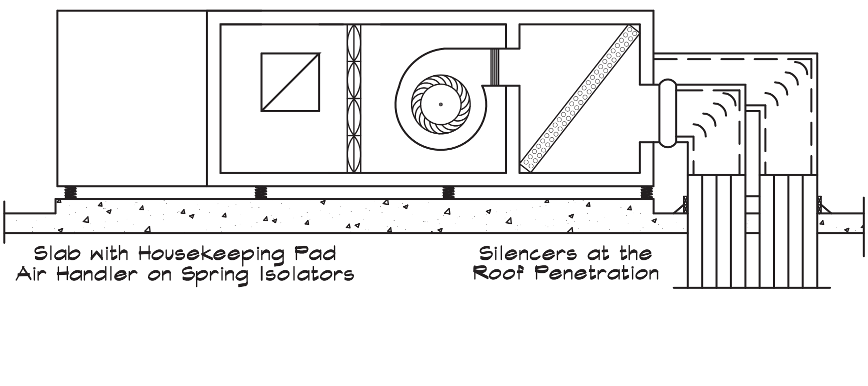


FIGURE 16.34 Quiet Rooftop Air Handling Unit (Schaffer, 1991)

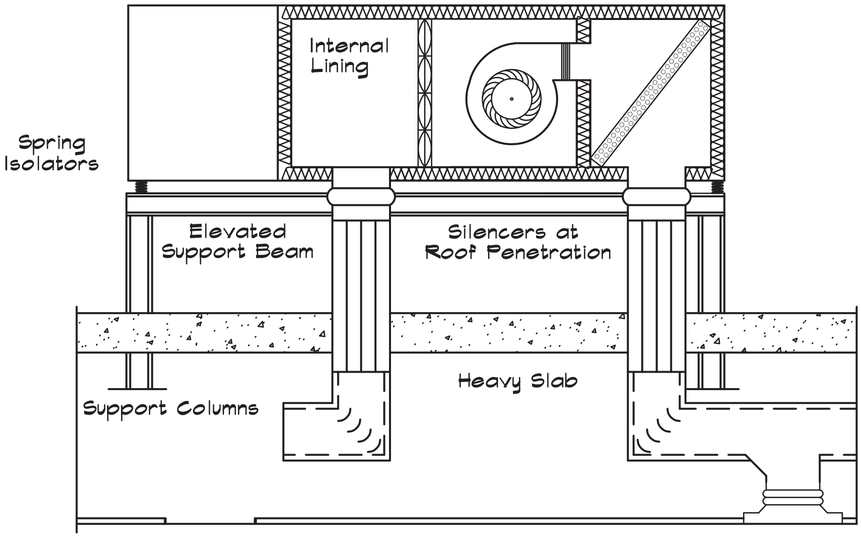


Figure 16.33 shows a side discharge configuration, which is the preferred arrangement, particularly for small installations. Where possible the ducts should penetrate the roof at a duct shaft or other nonsensitive area, since there is noise generated by the branch takeoffs. Where the building is located in a high exterior noise environment, the silencers can be located vertically at the roof penetration to reduce break-in problems.

Figure 16.34 illustrates a rooftop unit in which considerable effort has been expended to address the noise problems before the ducts penetrate the roof. The equipment support system is mounted on columns. The air handling unit is raised to allow the silencers to be installed vertically. Round ducts are used to control breakout and rumble.

Fan Coil and Heat Pump Units

In buildings such as hotels, where there is a need for temperature control in individual zones, a split system is used, which consists of an exterior condensing unit and cooling system, and an interior fan coil or heat pump unit located near the conditioned space. The fan in these units is the source of noise; as a general rule, fan coil units should not be located above or adjacent to occupied spaces when there is only an acoustical-tile or louvered-grille separation.

Fan coil units are available in two configurations, horizontal and vertical. The sound power levels vary with the fan capacity but in general a 3’ (1 m) medium pressure drop silencer or equivalent is required on both the supply and return to reduce noise levels to an NC 30. Horizontal units can be located above a closet, which allows access through the closet ceiling for maintenance. Vertical units can also be placed in a closet with a silencer in the supply duct. The return can be ducted through a return air plenum in the ceiling or through a transfer duct silencer mounted to the back of a solid core or rated door (depending on the sound power level of the unit). Lined plenum returns located beneath vertical units are usually not sufficient to achieve an NC 30, but can achieve an NC 40 to 45 in the region adjacent to the return air grille. This may be sufficient for nonsensitive spaces but is not

FIGURE 16.35 Vertical Fan Coil Units

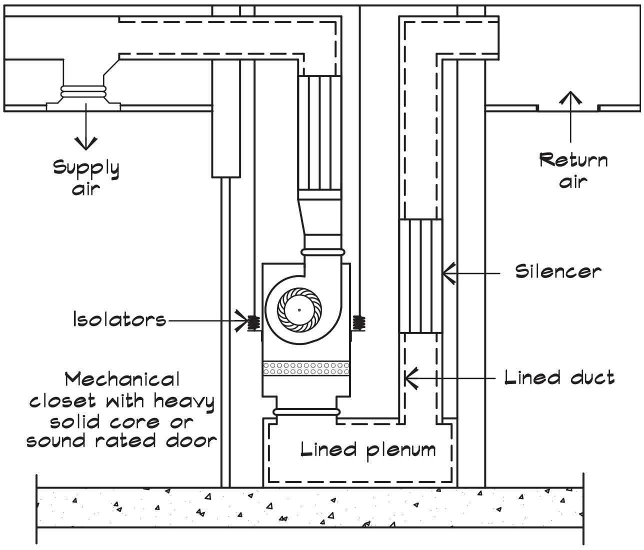
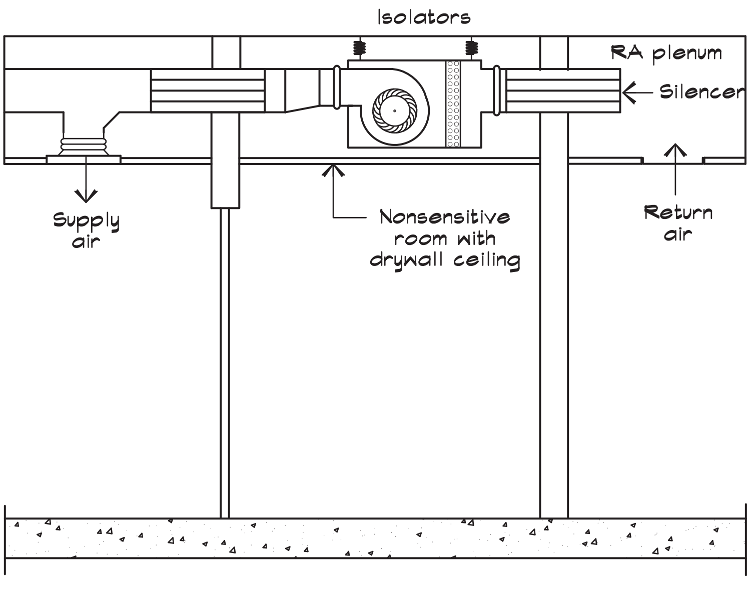


FIGURE 16.36 Horizontal Fan Coil Units

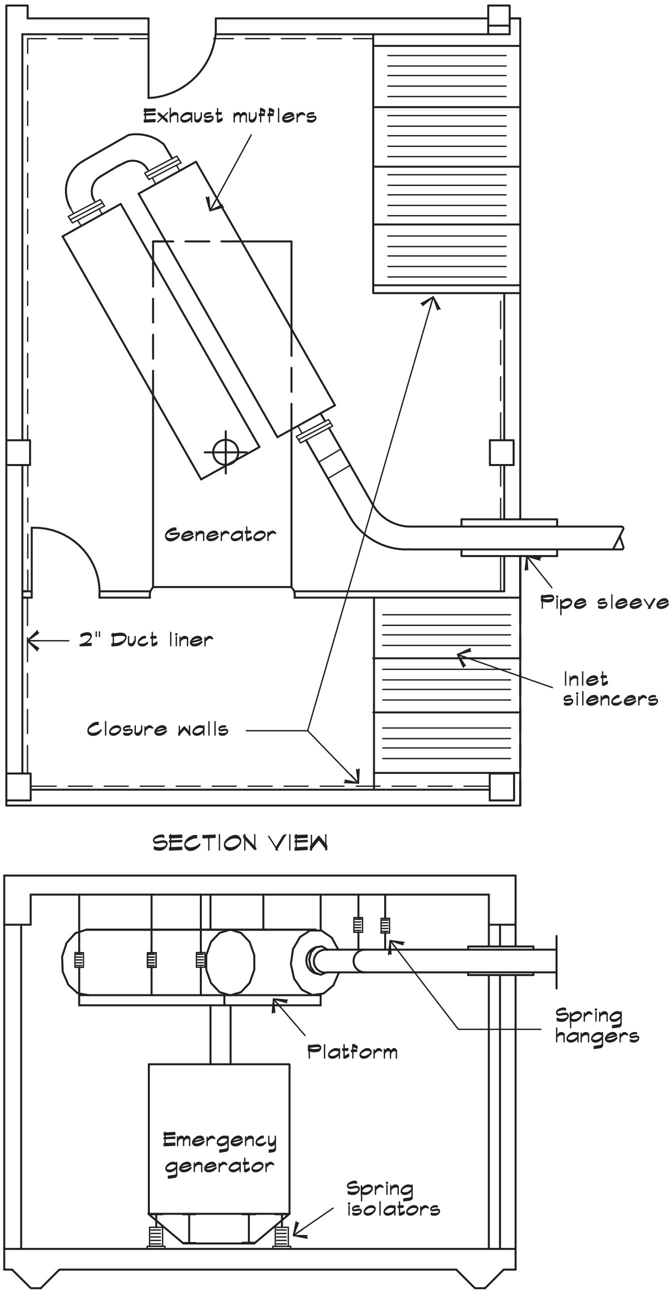


recommended for residential installations. Figures 16.35 and 16.36 illustrate these two types of fan coil installations.

Emergency Generators

Emergency generators are included in large buildings to supply power to selected equipment when the main power is lost. Although they are used infrequently, they must be tested

FIGURE 16.37 Plan of an Emergency Generator Room



periodically, typically an hour per month, so they need to be acoustically isolated. This requires treatment of both the exhaust and the inlet/cooling air. Inlet air is drawn in through the radiator by a fan and is used to cool the engine as well as to provide the combustion air. It circulates through the generator room and exhausts out again through silencers. The air intake has a large open area since the fan can accommodate only a small back pressure. The exhaust passes through one or more large mufflers. A design is shown in Fig. 16.37.

17 DESIGN OF ROOMS FOR SPEECH

17.1 GENERAL ACOUSTICAL REQUIREMENTS

General Considerations

Intelligibility depends on the masking effects of extraneous sounds on the speech we hear. Masking can be caused by noise from background sources or by reflections of the original spoken words. Speech combines the quick high-frequency sounds of consonants with the broader tones of the vowels. It is the recognition of consonants that correlates most closely with speech intelligibility, so the transmission of undistorted high-frequency information is critical. Figure 17.1 gives an illustration of a level versus time plot of the spoken word "back." Since the first part of the word is louder than the rest, its reverberant tail can mask the consonant ending.

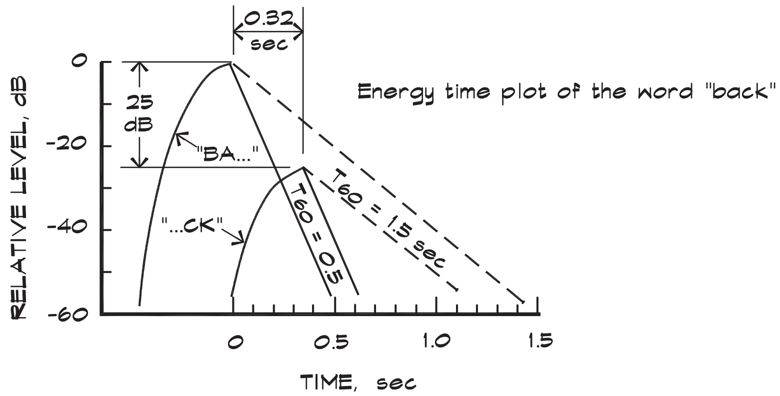
In the design of classrooms, conference rooms, and auditoria, the ability to understand speech is very important. The architectural components of these rooms—size, shape, surface orientation, and materials, as well as the background noise level—all influence intelligibility. There are several fundamental requirements in designing rooms for speech (Doelle, 1972), each of which contributes to achieving a high signal-to-noise level at the receiver:

1. There must be adequate loudness.
2. The sound level must be relatively uniform.
3. The reverberation characteristics of the room must be appropriate.
4. There must be a high signal-to-noise ratio.
5. Background noise levels must be low enough to not interfere with the listening environment.
6. The room must be free from acoustical defects such as long delayed reflections, flutter echoes, focusing, and resonance.

Adequate Loudness

For adequate loudness in a room, there must be a high direct field level. In unamplified spaces such as classrooms, the distance between the source and the receiver should be controlled.

FIGURE 17.1 An Illustration of the Effects of Reverberation on the Intelligibility of Speech (Everest, 1994)



Beyond 30 to 40 feet it is difficult to understand unreinforced speech, especially in a reverberant space. The volume per seat should be low, no more than 80 to 150 cu ft (2.3 to 4.3 cu m) per seat with an optimum value of 110 (3.1 cu m) (Doelle, 1972). By reducing the volume per seat, the loudness is increased and the reverberation time decreased for a given area of absorptive material. In general the smaller the seating capacity the larger the volume per seat can be within this range. Grazing attenuation should be controlled by raising the talker height and by sloping the floor. Beneficial reflections, preferably from overhead, should be designed in to help offset the effects of geometric spreading and grazing attenuation.

Making the audience seating area more circular minimizes the source-receiver distance. As the seating circle expands, there is a region beyond which the human voice cannot extend without physical or electronic reinforcement. These limits define the shape of a simple outdoor amphitheater and with the addition of walls and ceiling, they also contribute to the shape of a classroom or small lecture hall. For an auditorium, the semicircular seating area drives the shape toward a width that is greater than the depth as in Fig. 17.2. With a hard ceiling, the depth can be increased and the length-to-width ratio can exceed one. A balcony allows more of the audience to be seated close to the talker, as it brings the center of mass of the audience forward.

Nonacoustical considerations such as sight lines also influence the choice of room shape. The included angle between the outermost seats should be less than 140°, as shown in Fig. 17.3. This seating arrangement allows a clear view of the lecturer and writing boards. For projection screens the included angle should be limited to 125°, or about 60° from the screen centerline. Multiple angled screens can improve sight lines and reduce the effects of off-axis screen gain (loss).

Floor Slope

The floor of a large auditorium should be sloped to provide adequate sight lines. Good sight lines result in good listening conditions. Sight lines are set so that the audience can see the lowest point of interest on stage, called the arrival point of sight (APS), over the head of a person sitting in front of them. Even though it is theoretically desirable to design a theater with every-row clearance, from a practical standpoint this yields floor slopes that are too steep.

FIGURE 17.2 General Shapes of Auditoria (Doelle, 1972)

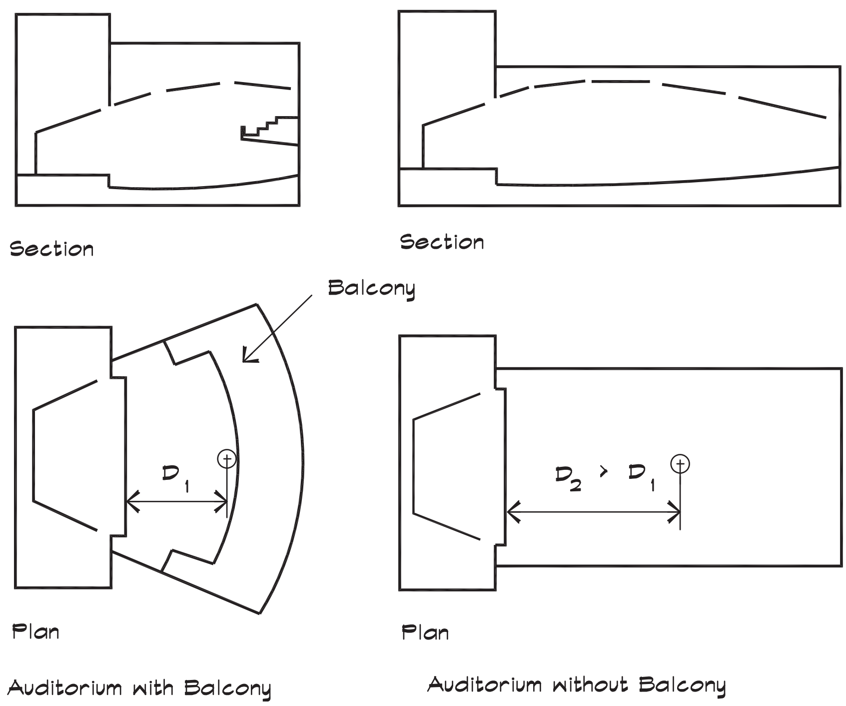
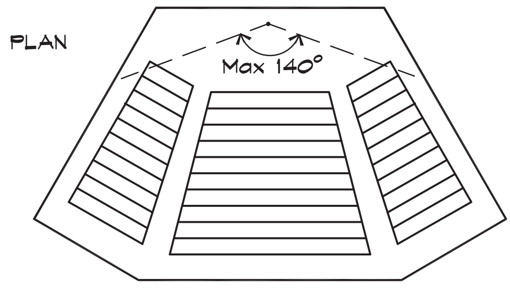


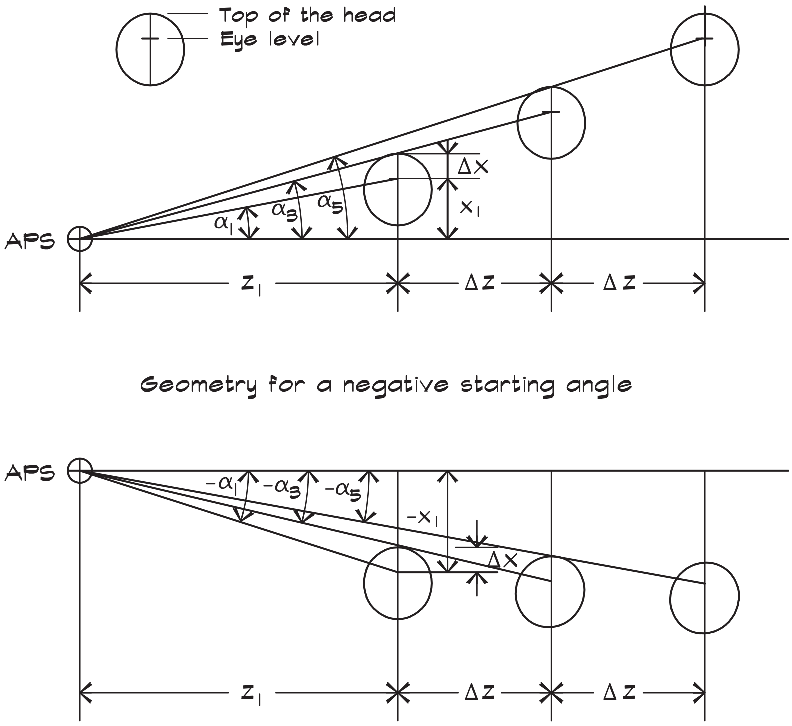
FIGURE 17.3 Seating Layout for a Lecture Hall



It is assumed that a person will adjust his position to look between the patrons seated in the next row so most theaters are designed for every-other-row visibility.

Figure 17.4 shows a typical sight-line design problem. The slope of the floor will depend dramatically on the APS that is selected. A high APS such as that found in a movie theater will still be visible, even with a relatively shallow floor slope. Stage floor heights are set low enough that a person sitting in the front row can see the actors' feet, but high enough that the APS does not force excessive floor rake. The eye height of a seated person ranges from 44 inches (1.12 m) for an adult female to 48 inches (1.22 m) for an adult male (Ramsey and Sleeper, 1970). Stage heights are fixed at between 40 and 42 inches (1.02–1.07 m) above the floor. The floor slope is determined by drawing a series of sight lines from the APS to a patron's ear level using standard anatomical data or by iteratively

FIGURE 17.4 Geometry of Theatrical Sight Line Calculations Every Other Row Sight Lines



applying a mathematical relationship

$$\tan \alpha_1 = \frac{x_1}{z_1} \tag{17.1}$$

where the index 1 refers to the first row and each subsequent index to the next row. The odd numbered rows are calculated for every-other-row sight lines. For this case the third row angle is

$$\tan \alpha_3 = \frac{x_1 + \Delta x}{z_1} \tag{17.2}$$

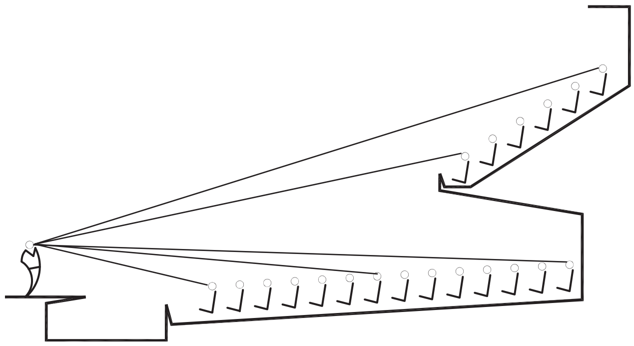
Subsequent odd numbered rows are calculated iteratively

$$\tan \alpha_n = \frac{\left\{ z_1 + \left[\frac{(n-1)}{2} - 1 \right] \Delta z \right\} \tan \alpha_{n-2} + \Delta x}{z_1 + \left[\frac{(n-1)}{2} - 1 \right] \Delta z} \tag{17.3}$$

where n = 5, 7, 9, and so forth, Δx = 5” (12.7 cm) for two rows, and Δz = 2· (row spacing).

Where there are fixed seats, grazing effect produces an attenuation that depends on the angle of incidence. The lower the angle, the greater the effect. The angle can be increased by

FIGURE 17.5 Good Sight Lines Yield Good Direct Sound



raising the talker on a platform or by raking the angle of seating as in Fig. 17.5. Seating rake is set by the sight-line requirements that are fixed by the APS and by the relative heights of each row of seats. In general the higher the APS the lower the seating rake. The rise of each row of seating can be calculated using Eq. 17.3. In the orchestra level seating a 1:9 rake for the first ten rows, and thereafter a 1:8 slope, yields a good result for a theater stage having a normal 42” (1.07 m) height. Building codes, which require no more than a 1:12 slope for handicapped access, may dictate the floor design.

In lecture halls, where the APS is selected to be at or above the waist of the lecturer or at the bottom of the writing board, the rake can be modest. In a large flat-floored classroom, a platform of 1 ft (0.3 m) height can improve the sight lines significantly. In small classrooms, seating fewer than 50 people, a platform is not necessary.

Sound Distribution

Unamplified speech can be augmented by physically placing hard surfaces in positions where they can distribute sound to the audience. Reflectors must have sufficient size that they scatter the frequencies of interest and should be close enough so that the reflection delay time is less than 30 to 50 msec. To provide this support, a hard ceiling is preferred in a lecture hall and auditorium (50–500 seats). In small classrooms (< 50 seats) the direct field, along with support from the walls, provides sufficient loudness and control of reverberant noise using an absorptive ceiling as the normal choice.

The orientation of a reflective element is determined by the required coverage area of the scattered sound. For specular reflection, the deflected angle is determined by locating the mirror image point of the sound source and by then drawing a line from the image point through the point of reflection toward the receiver. An example is given in Fig. 7.1. When the reflecting surface has a finite size, it is not an effective reflector over its entire length. Simple reflection studies illustrate the procedure. In these cases a reflection should be considered only if it occurs at a point more than one-half wavelength from the end of the reflector. Where the reflector is curved, the reflection angle is determined by mirroring the incident ray about the line connecting the center point of the curved surface with the intersection point of the ray and the surface. This is relatively straightforward in a CAD program; however, simple ray tracing does not tell the whole story since the scattered intensity falls off rapidly with increasing included angle, as was illustrated in Eq. 7.37.

The shape of the ceiling can be used to distribute sound evenly throughout an auditorium. Figure 17.7 shows a simple example of a flat ceiling. In this example the reflected rays

FIGURE 17.6 Reflection Studies from Ceiling Panels

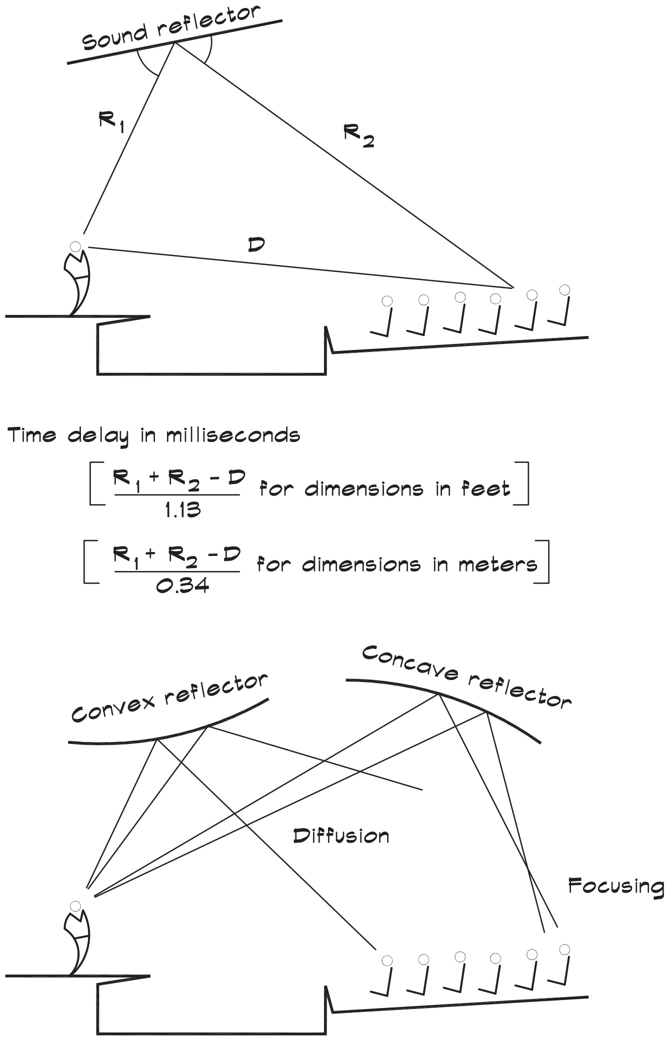


FIGURE 17.7 Reflections from a Flat Ceiling Section

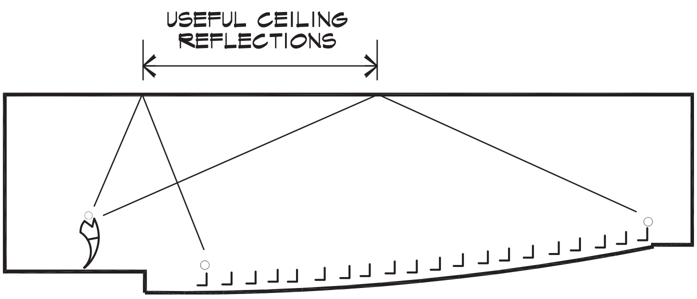


Figure 17.8 Reflected Sound from a Segmented Ceiling (Doelle, 1972)

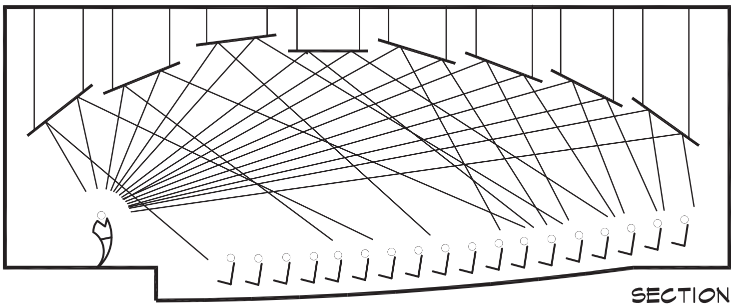
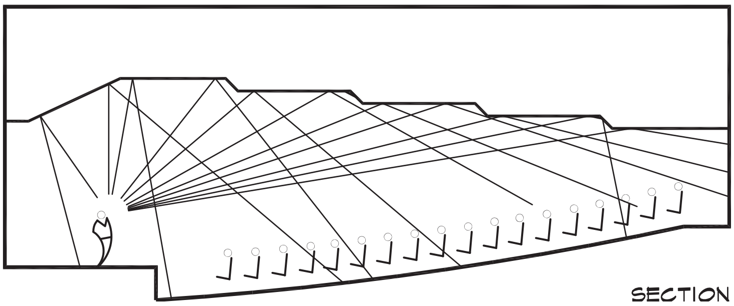


Figure 17.9 Reflected Sound from a Stepped Flat Ceiling



illuminate the front and middle portions of the space but much of the energy falling on the rear portion of the ceiling is grounded out on the absorptive rear wall.

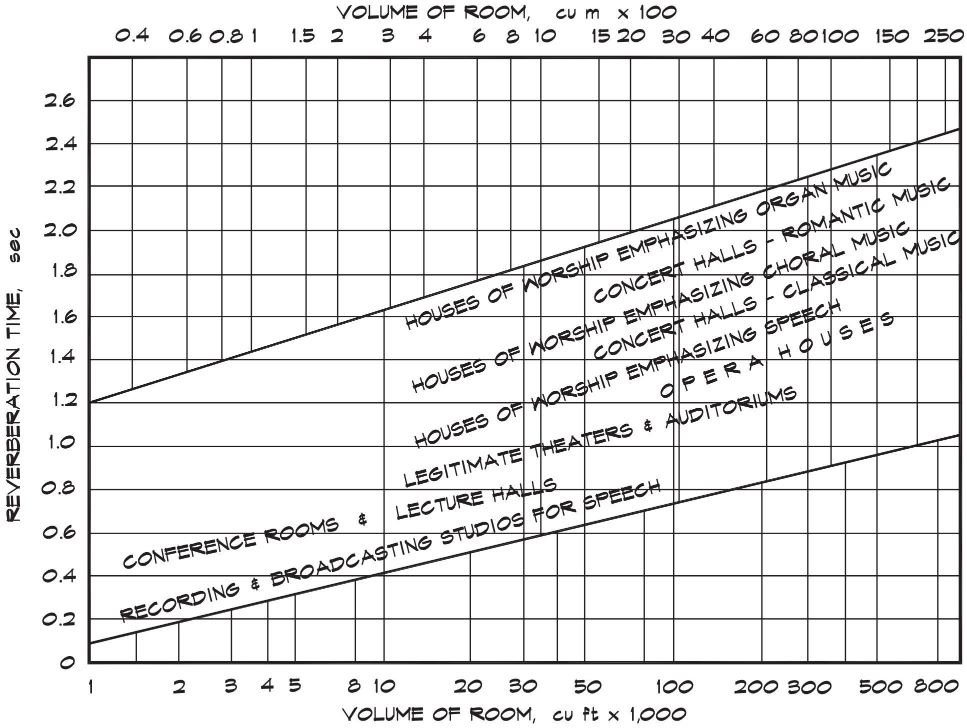
To improve the design, the ceiling can be segmented as in Fig. 17.8 or the seating raised and the ceiling stepped as in Fig. 17.9. Note that only about half of the ceiling provides useful specular reflections in both Fig. 17.7 and 17.8 since the ends of the segmented reflectors are diffusive.

The energy distribution is dependent on the location of the talker, which may vary, so slightly convex panels may be used to provide additional flexibility. Panels should not be used to reflect sound directly down, or back to the listener from behind, since this shifts the perceived source location overhead.

Reverberation

Reverberation can be the boon or the bane of the acoustical performance of a room. In general, the more speech content there is to the sound, the lower the ideal reverberation time. For classrooms and small lecture halls times at or below one second are preferred. Longer reverberation times are desirable for music; the ideal length depends on both the room size and the type of music. For light opera such as Gilbert and Sullivan, where understanding the complicated play of words is critical, a time of 1.0 to 1.2 seconds would not be too low. For a Mozart opera preferred reverberation times might range from 1.2 to 1.5 seconds. A Wagnerian opera is ideal in a 1.5 to 1.6 second room, and romantic symphonies sound best in a 1.7 to 2.1 second hall. For organ concerts and chant, reverberation times between 2.5 to 3.5 seconds

FIGURE 17.10 Reverberation Times vs Room Volume



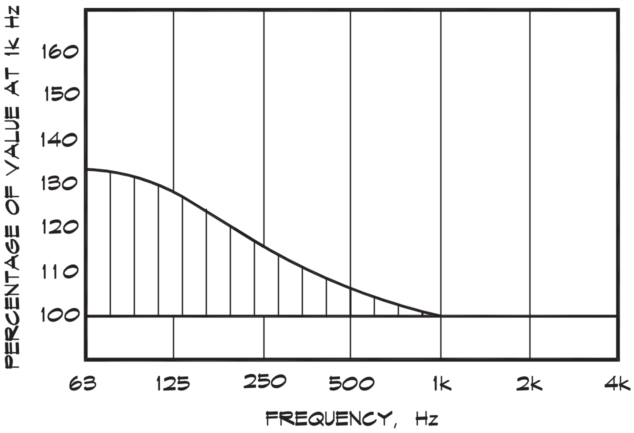
are not too long. Clearly there is no single reverberation time that is perfect for all uses of a given room. Variations of 5 to 10% from the ideal values are commonplace.

Various authors have made recommendations on ideal reverberation times for different types of spaces. Figure 17.10 shows a synthesis of a number of these graphs (e.g., Doelle, 1972; Knudsen and Harris, 1950; Long, 1999). Recent trends, particularly in the design of churches for electronically reinforced music, have driven the desirable reverberation times in large spaces downward, since reverberation can be added back electronically. Reverberation time recommendations for motion-picture theaters are given in Fig. 17.26.

Authors Knudsen and Harris (1950) and Doelle (1972) have recommended that for music the reverberation times at frequencies below 500 Hz rise to a number higher than the mid frequency value. Beranek (1996), citing measured results from various halls, recommends a factor of about 1.2 times the 500 to 1000 Hz value at 125 Hz. A recommended graph is shown in Fig. 17.11. A rising bass reverberation is good practice for performance rooms used for unreinforced music but not necessarily desirable in spaces where the low-frequency is provided by loudspeakers. As a practical matter it is difficult to achieve a rising reverberation time at very low frequencies, due to the weight and thicknesses of the materials required.

Since single-layer 16-mm (5/8”) gypsum board is nearly 30% absorptive at 125 Hz, a rising bass requires the use of multiple-layer gypboard or thick plaster construction. In the best concert halls the use of 25 to 50 mm (1”–2”) plaster is common. For speech the reverberation time behavior with frequency should be flat. In large rooms this is also difficult to achieve due to air attenuation, and the times fall off above 1 kHz. In large concert halls the HVAC system must include humidity control to reduce the high-frequency losses.

FIGURE 17.11 Ratio of the Bass to Mid-Frequency Reverberation Time



Recommended variation of T_{60} with frequency for music. At frequencies lower than 1000 Hz the T_{60} should increase according to the values inside the shaded area. For speech the T_{60} should remain flat with frequency.

Signal-to-Noise Ratio

Background noise levels in small classrooms and lecture halls are designed to an NC 30 (35 dBA) and larger auditoria to an NC 25 (30 dBA). The difference is due to the greater loss of loudness in the larger space. Some authors (Peutz and Klein, 1974) recommend that the received level be at least 25 dB higher than the background noise level for adequate intelligibility. Others (Bradley, 1986) hold that a 10–15 dB margin is a more reasonable choice. The latter value is consistent with an NC 30 background level and a direct-field level of 45 dBA, based on a speaking voice sound power level of 75 dBA and a source to receiver distance of 25 feet (7.6 m).

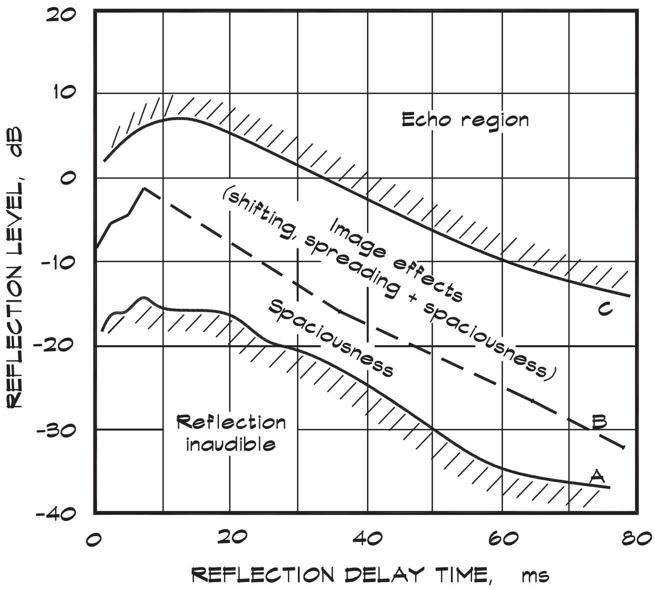
When the reverberant field is the masking noise a higher level can be tolerated. In these cases signal-to-noise ratios are rarely positive and a signal-to-reverberant-noise of −6 dB can still yield good intelligibility. This is discussed in more detail in Chapt. 18.

Acoustical Defects

The presence of acoustical defects can contribute to poor intelligibility and general discomfort in rooms. The principal defects, in addition to those already discussed, are multiple or long delayed reflections, focusing, coloration, and low-frequency phenomena such as room resonances and locally high-amplitude sound fields. In large auditoria there are also shadowed areas under balconies, coupled spaces with mismatched reverberation characteristics, and excess attenuation due to grazing incidence. Not all these defects are important in every room and some may be present without affecting the room's primary use.

There are a number of phenomena, associated with single or multiple reflections, that can detract from good intelligibility in rooms and should be avoided. These include long-delayed reflections, echoes, and flutter. Echoes occur when a sound of sufficient loudness arrives later than the direct field by more than a given time. The cause might be a single reflection from a rear wall of an auditorium, particularly if it is concave. Figure 17.12 shows

FIGURE 17.12 Perception of Lateral Speech Reflections (Everest, 1994)



the effect of reflections for various amplitudes and delay times, as simulated in an anechoic environment. Below curve B the echo increases the perception of spaciousness, while below curve A the reflected sound is reduced to inaudibility. Above curve C the reflection is perceived as an echo.

Echo and reverberation are not the same thing. Echo is a repetition of the original sound that is distinctly perceptible, whereas reverberation is a prolongation of the sound through multiple reflections, which is frequently beneficial for music. Long-delayed reflections are like echoes, but have a somewhat shorter delay time. They are not perceived as separate sounds, but blur the understanding of the original sound. Flutter echoes are sounds that persist locally due to multiple reflections between parallel planes, concave, or chevroned surfaces. They can be caused by two, three, or more reflections. Figure 17.13 gives several examples of acoustical defects.

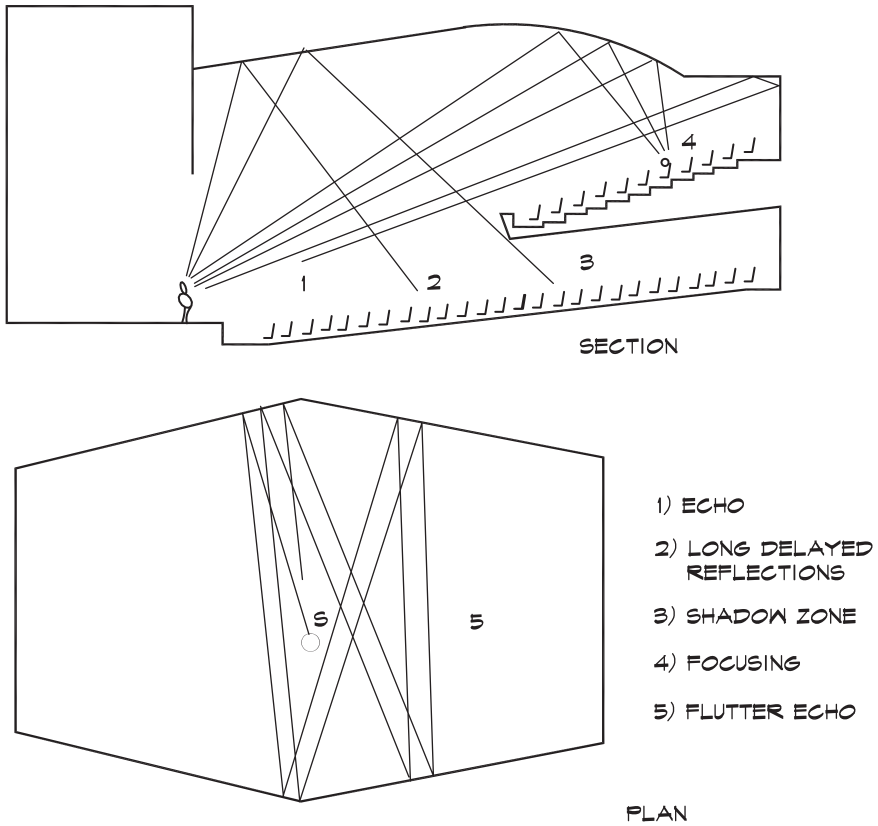
Coloration is the emphasis of certain frequencies or frequency bands over others. It can be caused by room-mode buildup or by absorptive materials that only absorb in certain frequency ranges. Focusing is the buildup of sound energy in localized regions of a room, due to concave surfaces. Shadowing is the blockage of sound traveling from the source, or from a significant reflecting surface, to the receiver. Each of these defects can detract from the overall acoustical environment in a room and each can be avoided with careful design.

17.2 SPEECH INTELLIGIBILITY

Speech-Intelligibility Tests

Speech-intelligibility tests for an unamplified talker are carried out using a single loudspeaker, ideally having a directivity similar to that of the human voice. Prerecorded words are presented in a neutral context carrier phrase such as, “Word number __ is ___”, at one or more calibrated levels, in rooms exhibiting a variety of acoustical conditions. As we gleaned

FIGURE 17.13 Examples of Acoustical Defects (Doelle, 1972)



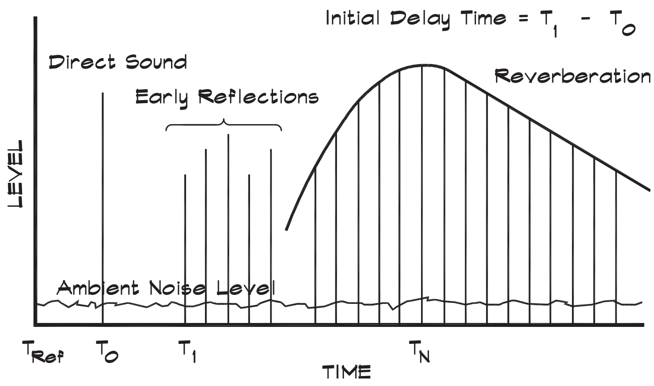
from Fig. 3.19, there is not just one test that gives the single answer for speech intelligibility. Rather there are many different results that depend on the details of the test and the type of material presented to the listener. The prediction of speech intelligibility in an enclosed space thus combines the results of listening tests with knowledge of the room's acoustical properties in such a way as to produce a predictable outcome.

Energy Buildup in a Room

When a sound is generated by a single source, the listener receives, in rapid succession, the direct-field signal followed by individual early reflections, and a rising swell of merged reflections whose sum becomes the reverberant field, which finally decays at a rate characteristic of the space. Figure 17.14 shows an example of the idealized pattern. In this figure the three temporal regions are neatly separated; in practice the divisions are not so distinct. The early reflections and the reverberation may be merged. If long-delayed reflections are present they may arrive during the reverberant decay. Sometimes individual reflections are louder than the direct sound when focusing or grazing attenuation is present. The reverberant field can be louder than the direct sound when the receiver is a relatively long way from the source.

The time between the arrival of the direct sound and the first major reflection is called the initial delay gap. If this gap is short enough, early reflections can contribute to increased

FIGURE 17.14 Idealized Acoustic Response of a Room to an Impulse Excitation



intelligibility, a broadening of the sound image, and a pleasant augmentation of the sound level. If it is too long, its effect will be to decrease intelligibility.

Background noise, along with long-delayed reflections and persistent reverberation, serve to decrease intelligibility. Background noise that interferes with the comprehension of speech can originate from many sources: people, HVAC systems, exterior noise sources such as traffic, or electronically generated masking noise, which is purposefully introduced to increase speech privacy. When words are spoken in a room, the reflections off the walls and other surfaces will eventually have a negative effect on speech intelligibility, either through long delayed individual reflections or as part of the general buildup of background noise. Thus the reverberant field of speech itself can also become the source of masking noise.

Room Impulse Response

Although it is possible to measure speech intelligibility directly in an existing room, it is also useful to have algorithms to predict it before a room is constructed. As we discussed in Chapt. 11, the impulse response completely defines the properties of a system, and we can predict the result of introducing an arbitrary forcing function (speech) by convolving (integrating) the input with the room's impulse response (Eq. 11.40). An exact formulation of a room's response is not available a priori, but it can be approximated by using the simplifying assumptions or by ray tracing.

A simple model assumes that the room is diffuse and that there exists a reverberant field characterized by a reverberation time. This model ignores common acoustical defects such as long delayed reflections, flutter echo, focusing, and the process of reverberant sound buildup. More complicated analyses utilizing ray tracing can describe these effects, but they are not expressible in a closed-form equation and are time consuming. The approximate methods yield results that are sufficiently accurate, as long as steps are taken to avoid the acoustical defects, which they do not include. The approach is to use the direct and reverberant sound energy densities previously discussed. The direct field (Eq. 2.56) energy density is given by

$$D_d = \frac{E}{S\, c_0\, t} = \frac{W_S}{S\, c_0} = \frac{p^2}{\rho_0\, c_0^2} \tag{17.4}$$

and

$$D_d = \frac{Q\,W_S}{4\,\pi\,c_0\,r^2} \tag{17.5}$$

The steady state reverberant field (see Eq. 8.82) energy density is

$$V\,D_r = \frac{4\,W_S\,V}{c_0\,S_T\,\overline{\alpha}} \tag{17.6}$$

or

$$D_r = \frac{Q\,W_S}{4\,\pi\,r_c^2\,c_0} \tag{17.7}$$

where r_c is defined as the critical distance, the point at which the direct sound pressure level is equal to the reverberant field level.

$$r_c = \sqrt{\frac{Q\,R}{16\,\pi}} \tag{17.8}$$

where W_s = source sound power (W)
V = room volume (m^3 or ft^3)
Q = source directivity in the receiver direction
c_0 = speed of sound in air (m / s or ft / s)
ρ_0 = density of air (kg / m^3 or lbs / ft^3)
$\overline{\alpha}$ = average room absorption coefficient
R = room constant (m^2 or ft^2)
S = area of the control surface (m^2 or ft^2)
S_T = total surface area of the room (m^2 or ft^2)
T_{60} = reverberation time (s)

We can approximate the impulse response of a room by assuming that the sound field is made up of only a direct and perfectly reverberant field that decays at a rate defined by the reverberation time. This idealized model, illustrated in Fig. 17.15, ignores all individual reflections.

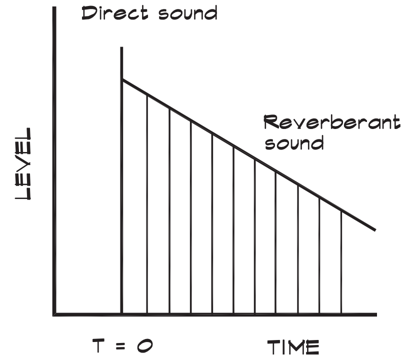
The sound power density as a function of time ($t \geq 0$) is (Houtgast et al., 1980)

$$w\,(t) = \frac{W_S}{4\,\pi\,c_0}\left[\frac{Q}{r^2}\,\delta(t) + \frac{Q\,\kappa}{r_c^2}\,e^{-\kappa t}\right] \tag{17.9}$$

where $\delta(t)$ = Dirac delta function at $t = 0$
κ = decay rate = $13.82 / T_{60}$ (1/s)

and the terms within the brackets are the impulse response of the room.

FIGURE 17.15 Simplified Room Impulse Response



Once the impulse response is known, the sound energy density arriving at a receiver during the time period from t = 0 to t = T is found by integrating over time

$$D_{0-T} = \int_0^T w\,(t)\,dt \qquad (17.10)$$

The steady-state energy density for a continuous driving function is found by setting T = ∞. In this way we recover Eqs. 17.5 and 17.7.

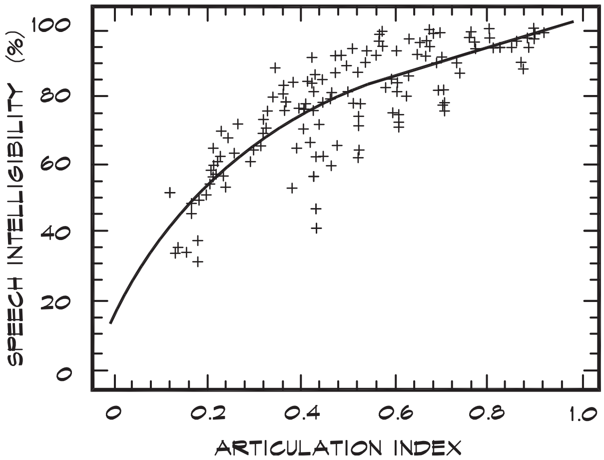
Speech-Intelligibility Metrics—Articulation Index (AI)

There are several metrics currently enjoying use for the prediction of the intelligibility of speech in rooms: the Articulation Index (AI), the Articulation Loss of Consonants (AL_{cons}), the Speech Transmission Index (STI), and the various signal-to-noise ratios including the Useful to Detrimental Energy Ratio (U_τ) and the Useful to Late Energy Ratio (C_τ).

Much of the pioneering work in communication acoustics was done at Bell Laboratories, where engineers studied methods of improving the intelligibility of telephone conversations. Harvey Fletcher (1884–1981) was one of these early pioneers. Although Fletcher's work dates from the 1920s and 1930s, much of it was not revealed publicly until the publication of later papers and his classic book in 1953.

Fletcher (1921) proposed to quantify the speech distortion in telephone systems by relating it to articulation scores. He defined the "articulation," which ranged from 0 to 1, as an overall measure of the intelligibility of speech transmitted through a system. One of Fletcher's contributions was the discovery of the probabilistic nature of intelligibility, and indeed the definition of articulation is the probability of understanding an individual sound. If, for example, a syllable consists of a consonant-vowel-consonant (cvc) sequence, the probability of understanding the whole sequence would be the product of the probabilities of understanding each separate consonant or vowel. When this was combined with the realization that the probabilistic approach carried over into the analysis of separate frequency bands (published later in Fletcher and Galt, 1950), the basis for the Articulation Index was established. French and Steinberg (1947) formalized the method of measurement and Kryter (1970) published a method of calculating the expected speech intelligibility in rooms using the sum of weighted signal-to-noise ratios in third-octave frequency bands (Chapt. 3).

Figure 17.16 Measured Intelligibility vs Articulation Index (Bradley, 1986)



In 1986 Bradley published a study comparing the accuracy of various articulation metrics, and Figure 17.16 shows his result for Articulation Index.

Articulation Index (AI) is like virtually all other intelligibility prediction schemes in that it uses a signal-to-noise ratio as part of the calculation. The differences among the various schemes are how the terms "signal" and "noise" are defined. In AI calculations, the signal is the long-term rms average speech level (direct + reverberant) plus 12 dB, and the noise is the steady background noise level in each frequency band. AI is difficult to use as an intelligibility prediction methodology since it does not have a built-in way of accounting for reverberant noise. In the ANSI standard (ANSI S3.5-1969) there is an empirical correction table for reverberation time (Knudsen and Harris, 1950) but no way of dealing with the contribution of the reverberant field. Where an electronic masking system generates the steady background noise, AI yields good results in the assessment of privacy. Typical results of intelligibility scores versus AI values were given in Fig. 3.19.

Articulation Loss of Consonants ($\mathbf{AL_{cons}}$)

Early researchers (e.g., Knudsen, 1932) found that intelligibility was based on the recognition of consonants rather than vowels and developed metrics based on this concept. Maxfield and Albersheim (1947) at Bell Laboratories examined the measured articulation-loss-of-consonants data published by Steinberg (1929) and Knudsen (1932) and plotted them versus a steady-state direct-to-reverberant energy ratio. They found that the data did not lie along a straight line and subsequently developed the concept of a liveness factor, for use with microphone pickups, which they defined as

$$L = \int_0^\infty D_r(t)\, dt \,/\, D_d \tag{17.11}$$

where $D_r(t)$ is the reflected-energy density at any time, t, and D_d is the direct-field energy density. Under the assumption of a reverberant field and unit directivity, the liveness can be written (Bistafa and Bradley, 2000) in metric units as

$$L = \frac{T_{60}\, D_r}{13.82\, D_d} = \frac{4\,\pi}{(13.82)^2} \frac{c_0\, T_{60}^2\, r^2}{V} = 22.6\, \frac{T_{60}^2\, r^2}{V} \tag{17.12}$$

which is the reverberant-to-direct energy ratio multiplied by the reverberation time. Bistafa and Bradley (2000) fitted a curve to the Maxfield and Albersheim plots, which related articulation loss (AL) to liveness based on Steinberg's data

$$AL = 4.5\, L^{0.67} \tag{17.13}$$

In 1971 Peutz measured the speech intelligibility in rooms using cvc phonetically balanced words in Dutch. Like Knudsen, he also found that the articulation loss was much smaller for vowels than for consonants, so that consonant loss probabilities controlled cvc recognition. Unlike Maxfield and Albersheim, however, he found a linear relationship between the measured articulation loss of consonants and liveness. His relationship, in metric units, is shown in Eq. 17.14, and assumes a directivity of one and negligible background noise.

$$AL_{cons} = 8.9\, L + a = 200\, \frac{T_{60}^2\, r^2}{V} + a \tag{17.14}$$

where a is a correction factor that can vary from 1.5% for a "good" listener, to 12.5% for a "bad" listener. This equation is said to hold as long as the listener is no more than a limiting distance r_ℓ away, where

$$r_\ell = 0.21 \sqrt{V/T_{60}} \quad (m) \tag{17.15}$$

This is the distance at which the direct-field level is 10 dB below the reverberant-field level for a directivity of one. Beyond that point Peutz states that the articulation loss is given by

$$AL_{cons} = 9\, T_{60} + a \quad \text{for } r \geq r_\ell \tag{17.16}$$

In terms of the limiting distance, the equation is

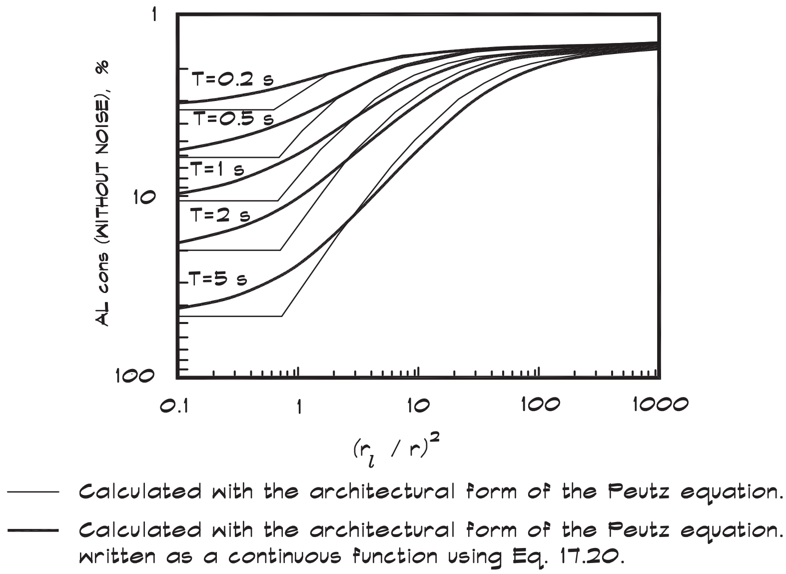
$$AL_{cons} = 9\, T_{60}\, \frac{r^2}{r_\ell^2} + a \quad \text{for } r < r_\ell \tag{17.17}$$

Equations 17.16 and 17.17 are known as the architectural versions of the Peutz equations. They do not account for early reflections, discrete echoes, background noise, or frequency dependence of the variables.

Bistafa and Bradley (2000) published a continuous version of the noiseless equations based on work by Peutz (1974) and Peutz and Kok (1984).

$$AL_{cons} = 9\, T_{60} \left[\frac{1}{1 + (r_\ell / r)^2} \right] + a \tag{17.18}$$

FIGURE 17.17 ALcons–N vs $(r_\ell/r)^2$ with Reverberation Time as a Parameter (Bistafa and Bradley, 2000)



and Klein (1971) added back the directivity by defining the limiting distance in terms of the room constant.

$$r_\ell = 0.45\sqrt{Q\,R} \quad \text{(m)} \tag{17.19}$$

A comparison of Eqs. 17.16 and 17.17 to 17.18 is shown in Fig. 17.17.

In 1974 Peutz and Klein published a graphical method of accounting for the presence of noise. This was curve fitted by Bistafa and Bradley (2000) and in its continuous form is

$$AL_{cons} = 9\,T_{60}\left\{\frac{1}{1+(r_\ell/r)^2}\right\}\left[1.071\,T_{60}^{-0.0285}\right]^{(25-L_{SN})} + a \tag{17.20}$$

where L_{SN} is the signal-to-noise ratio $L_{SN} = L_s - L_n$ in dB. According to Eq. 17.20, when the signal is less than 25 dB above the background noise there is a reduction in speech intelligibility, which becomes progressively worse as the signal level decreases. If the signal level is greater than 25 dB above the noise, there is no degradation due to background noise and the noise term is dropped. Here the signal level is the direct plus reverberant speech level, and the noise level is the steady background level having the same spectral shape as the speech level. Peutz and Klein did not include information on the spectrum of the background noise or the frequency at which the calculations are to be carried out. Standard practice is to use the 2000 Hz octave band.

Davis and Davis (1987) recommend AL_{cons} for general use in sound-system design, although in this form there is no single value of the directivity when multiple loudspeakers are used. Jacobs (1985), experimenting with single high, medium, and low-Q loudspeakers, found a poor correlation between the predicted and measured intelligibility, particularly in highly reverberant rooms. His data indicated that AL_{cons} underpredicted the speech intelligibility for low- and medium-Q loudspeakers and overpredicted with a high-Q device. Bistafa and Bradley (2000) also found a poor correlation between AL_{cons} predictions and those based on STI and U_{50} metrics. They recommended that its use be limited to classrooms and small meeting rooms. This would seem to be a good approach. AL_{cons} includes the reverberant field as part of the signal in a signal-to-noise ratio, but switches to a different formulation when the reverberant field dominates the direct field.

Speech Transmission Index (STI)

Researchers in optics (Baker, 1970), seeking to quantify the distortion of light received from stars, developed the optical transfer function, which was based on a mathematical formulation called the modulation transfer function (MTF). Houtgast, Steeneken, and Plomp (1980) reasoned that stars are the spatial equivalent of an acoustical impulse source and this approach could be useful in evaluating distortion in rooms. As we discussed in Chapt. 4, the MTF uses a modulated sinusoidal input

$$I_{in}(t) = I_{in}(1 + \cos \omega_m t) \tag{17.21}$$

which is introduced into a room. It is convolved with the room's impulse response g (t') to obtain an output

$$I_{out}(t) = \int_0^\infty I_{in}(t - t')\, g(t')\, dt \tag{17.22}$$

which has the general form

$$I_{out}(t) = I_{out}\left\{1 + m\left[\cos \omega_m (t - \theta)\right]\right\} \tag{17.23}$$

The modulation transfer function is defined as

$$m(\omega_m) = \frac{\left| \int_0^\infty g(t)\, e^{-j\omega_m t}\, dt \right|}{\int_0^\infty g(t)\, dt} \tag{17.24}$$

and $\omega_m = 2\pi f_m$ is the modulation frequency. Schroeder (1981) pointed out that this is the normalized Fourier transform of the power density impulse response. Assuming a diffuse field, the impulse response for both the direct and reverberant field components is

$$g(t) = \frac{Q}{r^2}\,\delta(t) + \frac{Q\kappa}{r_c^2}\, e^{-\kappa t} \tag{17.25}$$

where κ is the exponential decay constant of the reverberant energy, $\kappa = 13.82 / T_{60}$.

When background noise is added to the mix the output intensity is

$$I_{sum} = I_{out}(t) + I_n \tag{17.26}$$

and using Eq. 17.23,

$$I_{sum} = I_{out}\left[1 + m\cos\omega_m(t-\theta)\right] + I_n \tag{17.27}$$

which can be written

$$I_{sum} = (I_{out} + I_n)\left[1 + m\frac{I_{out}}{I_{out}+I_n}\cos\omega_m(t-\theta)\right] \tag{17.28}$$

So the modulation factor due to background noise is

$$m_n = \frac{I_{out}}{I_{out}+I_n} = \left[1 + 10^{-0.1\,L_{SN}}\right]^{-1} \tag{17.29}$$

and the signal-to-noise ratio in dB is $L_{SN} = 10\log(I_{out}/I_n)$.

This yields the overall modulation transfer function including both

$$m(\omega_m) = \frac{\left|\int\limits_0^\infty g(t)\,e^{-j\,\omega_m t}\,dt\right|}{\int\limits_0^\infty g(t)\,dt}\left[1 + 10^{-0.1\,L_{SN}}\right]^{-1} \tag{17.30}$$

room distortion and noise. We substitute the impulse-response function (Houtgast et al., 1980)

$$m(\omega_m) = \frac{\left(A^2 + B^2\right)^{1/2}}{C} \tag{17.31}$$

with

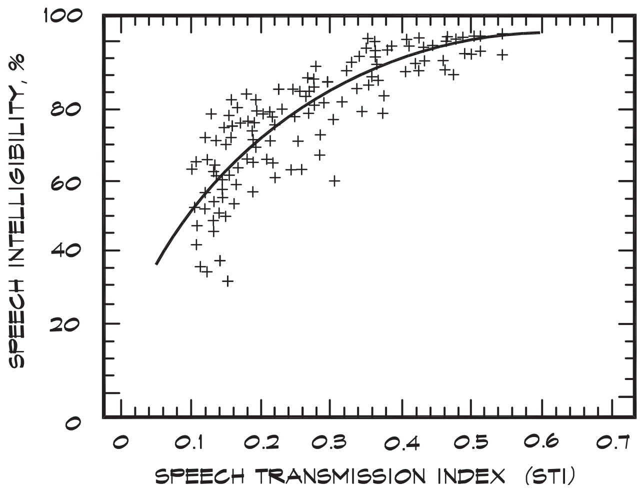
$$\begin{aligned} A &= \frac{Q}{r^2} + \frac{Q}{r_c^2}\left[1 + \left(\frac{\omega_m T_{60}}{13.8}\right)^2\right]^{-1} \\ B &= \frac{\omega_m T_{60}}{13.8}\frac{Q}{r_c^2}\left[1 + \left(\frac{\omega_m T_{60}}{13.8}\right)^2\right]^{-1} \\ C &= \frac{Q}{r^2} + \frac{Q}{r_c^2} \end{aligned} \tag{17.32}$$

and the critical distance defined in Eq. 17.8. This can be simplified in the far field, $\frac{r_c^2}{r^2} \to 0$ for unit directivity to

$$m(f_m) \cong \left\{1 + \left(2\pi f_m \frac{T_{60}}{13.8}\right)^2\right\}^{-1}\left[1 + 10^{-0.1\,L_{SN}}\right]^{-1} \tag{17.33}$$

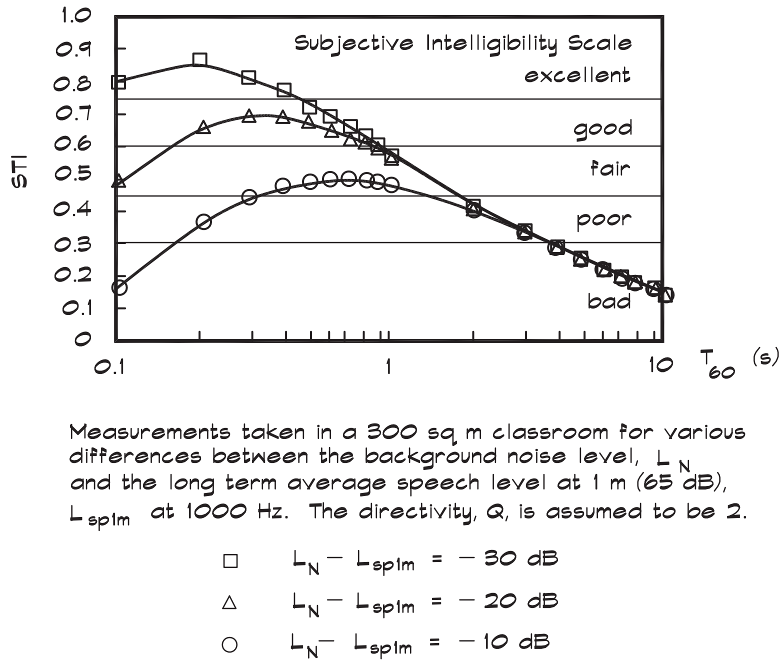
For a given modulation frequency an apparent signal-to-noise ratio and speech transmission index (STI) is calculated from the modulation index described in Eqs. 4.23 through 4.25, and from this an intelligibility can be determined.

FIGURE 17.18 Measured Speech Intelligibility vs Speech Transmission Index (Bradley, 1986)



In a study of classroom intelligibility, Bradley (1986) measured speech-intelligibility scores, including the effects of noise, and compared them to calculated STI values. The results are given in Fig. 17.18. Bistafa and Bradley (2000) plotted STI values versus reverberation time for unamplified speech in classrooms, which are reproduced as Fig. 17.19. Here we see that for a given signal-to-noise ratio the intelligibility can be maximized as a function of reverberation time.

FIGURE 17.19 Speech Transmission Index (STI) vs Reverberation Time (Bistafa and Bradley, 2000)



***Signal-to-Noise Ratios* (C_t *and* U_t)**

In 1935 two researchers, F. Ainger and M. J. O. Strutt, reported on the property of the ear that combines early-reflected sounds with the direct sound so as to increase the apparent strength of the whole. They suggested an energy ratio formula to quantify the effects of background noise and room acoustics on intelligibility. They called this ratio *impression*, which they defined as

$$Q = \frac{E_d + E_e}{E_\ell + E_n} \tag{17.34}$$

where E_d = direct field energy (N m)
E_e = early part of the reflected energy (N m)
E_ℓ = late portion of the reflected energy (N m)
E_n = constant noise energy (N m)

They set the dividing line between early and late reflections at 1/16 second and set a lower limit of 1 for a satisfactory value of Q. If we write Eq. 17.34 in terms of energy densities we obtain a similar expression

$$Q = \frac{D_d + D_e}{D_\ell + D_n} \tag{17.35}$$

where D_d = direct field energy density (W s / m^3)
D_e = early part of the reflected energy density (W s / m^3)
D_ℓ = late part of the reflected energy density (W s / m^3)
D_n = constant noise energy density = $\dfrac{p^2}{\rho_0\, c_0^2}$ (W s / m^3)

Using this model and the impulse response from Eq. 17.25 we can calculate the value of the impression (Bistafa and Bradley, 2000)

$$Q = \frac{1 + (r_\ell / r)^2 - e^{-0.86/T_{60}}}{e^{-0.86/T_{60}} + 10^{0.1\,(L_n - L_r)}} \tag{17.36}$$

where L_n = steady background noise level (dB)
L_r = steady reverberant signal level (dB)

The metric is seldom encountered now but is interesting, not only for its historical significance, but also as an introduction to more recent versions of the same concept using different cutoff times.

In the 1950s, Thiele (1953) published one of the earliest attempts at relating early to total sound energy ratio to intelligibility, which he called the definition, D. He considered the useful energy to be the direct plus the reflected energy that arrives within 50 msec of the direct sound. The definition can be written (Bistafa and Bradley, 2000) as

$$D_{50} = \frac{1 + (r_\ell / r)^2 - e^{-0.69/T_{60}}}{1 + (r_\ell / r)^2} \tag{17.37}$$

Definition does not account for the contribution of the background noise to the detrimental energy. It represents another early attempt to quantify speech intelligibility in terms of room acoustics.

Bradley (1986) used variations of the Q metric in his study of speech intelligibility in classrooms. These included the useful-to-detrimental noise ratio

$$\mathrm{U}_\tau = 10 \log \left[\frac{R_\tau}{(1 - R_\tau) + 10^{-0.1\, \mathrm{L}_{\mathrm{SN}}}} \right] \tag{17.38}$$

where R_τ is the ratio between the early and the total energy

$$R_\tau = E_\mathrm{e} \,/\, (E_\mathrm{e} + E_\ell) \tag{17.39}$$

and the early-to-late signal-to-noise ratio

$$\mathrm{C}_\tau = 10 \log \left[\frac{R_\tau}{1 - R_\tau} \right] \tag{17.40}$$

which is obtained by setting the second term in the denominator of Eq. 17.38 equal to zero.

When these expressions are evaluated using the diffuse-field impulse response and a cutoff time of 50 msec we obtain

$$\mathrm{U}_{50} = 10 \log \left[\frac{1 + (\mathrm{r}_\ell/\mathrm{r})^2 - \mathrm{e}^{-0.69/\mathrm{T}_{60}}}{\mathrm{e}^{-0.69/\mathrm{T}_{60}} + 10^{\,0.1\,(\mathrm{L}_\mathrm{n} - \mathrm{L}_\mathrm{r})}} \right] \tag{17.41}$$

and

$$\mathrm{C}_{50} = 10 \log \left[\frac{1 + (\mathrm{r}_\ell/\mathrm{r})^2 - \mathrm{e}^{-0.69/\mathrm{T}_{60}}}{\mathrm{e}^{-0.69/\mathrm{T}_{60}}} \right] \tag{17.42}$$

Bradley (1986) published intelligibility versus U_{80} values in his study of classrooms, which are given in Fig. 17.20. Bradley worked with several cutoff times: 35, 50, 80, and 95 msec. He found (1998) that the differences using cutoff times between 50 and 95 msec are not great, for example, $\mathrm{C}_{80}(\mathrm{A}) \cong \mathrm{C}_{50}(\mathrm{A}) + 2$. The results are plotted in Fig. 17.21.

Weighted Signal-to-Noise Ratios ($\mathbf{C}_\mathbf{t}^\alpha$ *and* $\mathbf{U}_\mathbf{t}^\alpha$)

Early-to-late ratios were also the basis of work by Lochner and Berger (1964) in the Afrikaans language. These authors identified and separated the early sound energy, arriving at less than a certain time after the direct sound, from the later reflected sound. In their system the early arrivals are weighted and integrated over the time period and compared to the sound energy arriving after that time. They defined a useful-to-late energy ratio as

$$\mathrm{C}_\tau^\alpha = 10 \log \left[\frac{\int_0^\tau \alpha(\mathrm{t})\, w(\mathrm{t})\, \mathrm{dt}}{\int_\tau^\infty w(\mathrm{t})\, \mathrm{dt}} \right] \tag{17.43}$$

where $\alpha(\mathrm{t})$ is the average fraction of the energy of an individual reflection that is integrated into the useful early energy sum. This weighting term depends on the amplitude of the

Figure 17.20 Speech Intelligibility vs U_{80} Values (Bradley, 1986)

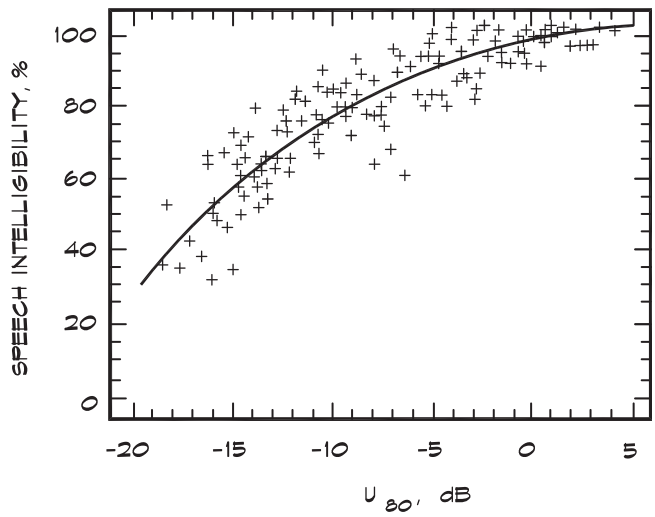
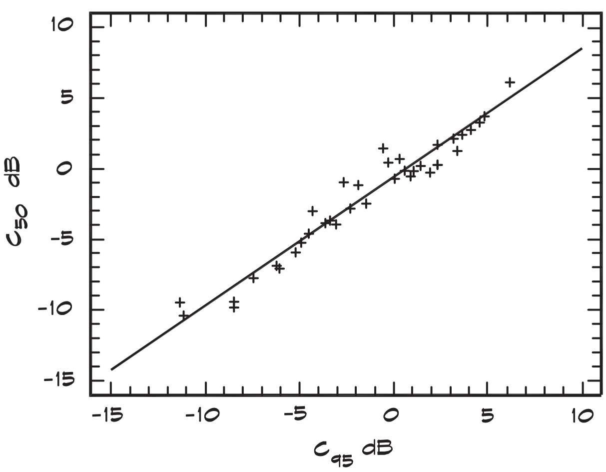


Figure 17.21 Measured C_{50} vs C_{95} values at 1 kHz (Bradley, 1986)



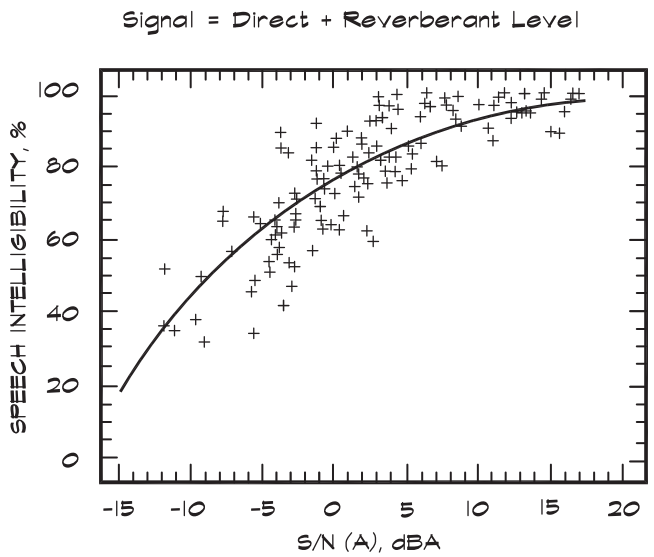
reflected energy, relative to the direct sound and the time of arrival. The $\alpha(t)$ term was included because the unweighted method proved highly sensitive to individual reflections arriving just before or just after the cutoff time. The weighting factor was set to 1 at a start time and to 0 at the finish time, and decreased linearly between them. Various algorithms have been used as a weighting function. Among them are

$$\alpha(t) = 1 \quad \text{for } 0 \le t < t_1$$

$$\alpha(t) = \frac{t_2 - t}{t_2 - t_1} \quad \text{for } t_1 \le t \le t_2 \tag{17.44}$$

$$\alpha(t) = 0 \quad \text{for } t > t_2$$

FIGURE 17.22 Measured Speech Intelligibility vs A-Weighted Signal-to Noise-Ratio (Bradley, 1986)



with $t_1 = 0.035$ s and $t_2 = 0.095$ s. For a diffuse field and a 95 ms cutoff time, Lochner and Burger's useful-to-detrimental ratio is

$$U_{95}^{\alpha} = 10 \log \left[\frac{1 + (r_\ell / r)^2 + 1.21\, T_{60} \left(e^{-1.31/T_{60}} - e^{-0.48/T_{60}}\right)}{e^{-1.31/T_{60}} + 10^{0.1\,(L_n - L_r)}} \right] \qquad (17.45)$$

The useful-to-late ratio C_{95}^{α} can be obtained by setting the noise term in the denominator equal to zero.

A-Weighted Signal-to-Noise Ratio

Bradley (1986) also worked with a simple metric, namely the A-weighted steady-average-speech level (55 dBA at 1 m for a normal voice and 63 dBA for a raised voice in this study), based on anechoic measurements of speech. He calculated the direct plus reverberant-field level and used it to test intelligibility for various background-noise levels. The results were very similar to those found with more complicated metrics, and its ease of use makes it attractive. Figure 17.22 shows his results in terms of a signal-to-noise ratio. It is interesting to note that these data support his assertion that signal-to-noise ratios significantly less than 15 dB yield very satisfactory intelligibility.

Comparison of Speech-Intelligibility Metrics

Bradley (1986), in his comparison of several methods of predicting speech intelligibility in rooms, examined metrics in three categories: AL_{cons}, STI, and the various signal-to-noise ratios. His studies were carried out using a Fairbanks rhyme test, which gives a result similar to that obtained with nonsense syllables. He found that there was close agreement between STI and the early-to-late ratios, but poor correlation between AL_{cons} and the other metrics. Jacobs (1985), using loudspeakers of differing directivities, found a similar result with AL_{cons}, yielding errors on the order of 20% in intelligibility. In his work the use of

STI lead to a slight (5%) underprediction of intelligibility, whereas a weighted signal-to-noise ratio, similar to Eq. 17.45, yielded an overprediction of the same order of magnitude. Bistafa and Bradley (2000) found a linear relationship between STI and U_{50}

$$U_{50} \cong 31\,\mathrm{STI} - 16 \tag{17.46}$$

indicating that these metrics are essentially equivalent. A similar relation was deduced for Lochner and Burger's signal-to-noise ratio

$$U_{50}^{\alpha} \cong 1.25\,U_{50} + 3.4 \tag{17.47}$$

The research cited in this section was done with single, as opposed to distributed, loudspeakers and is best utilized in analyzing rooms with unamplified talkers or single-source reinforcement systems. The complications introduced by multiple loudspeakers with different directivity characteristics and delay times are not addressed here.

17.3 DESIGN OF ROOMS FOR SPEECH INTELLIGIBILITY

The interior design of a given room depends on the use, interior décor, and the acoustical goals for the space. In many rooms such as restaurants or private homes, the noise may be generated by conversations other than those of interest. In these cases the addition of absorbing materials can control reverberant noise but must be balanced against the interior design goals. Acousticians must be sensitive to the appearance of their work and architects must accept the fact that design is not only visual.

The Cocktail Party Effect

The cocktail party effect* is an interesting and amusing exercise in the buildup of a sound field in a room. Let us assume that we are giving a party in a relatively reverberant room and invite a number of people to come. Let us say that the room has a carpeted floor, hard walls and ceiling, and some furniture, which contribute 93 metric (1000 sq ft) sabins of absorption. Before the guests arrive two hosts are having a conversation in the living room. They are polite so that only one speaks at a time with a sound power level of 70 dB. For purposes of this calculation let us assume that the direct sound, which is transmitted between the talker (with $Q = 2$) and the listener, is the signal, and the reverberant sound reflected from the surfaces of the room is the noise. Clearly some of the reflected sound contributes to intelligibility but we are going to ignore that for this simple analysis. Using Fig. 17.23, let us say that for barely adequate (60%) intelligibility, we need a signal-to-noise ratio of at least −6 dB to understand sentences.

The reverberant field level in our living room is

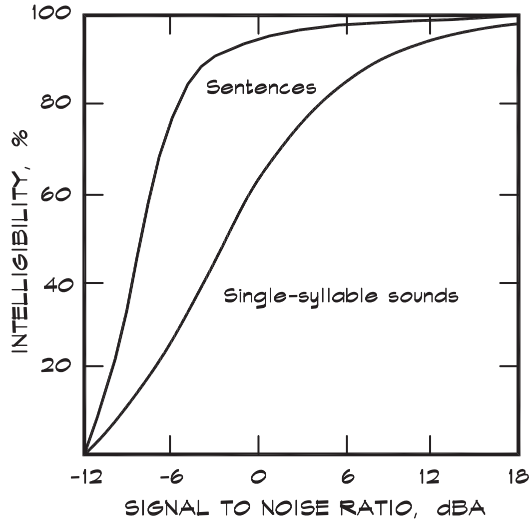
$$L_p \cong L_w + 10\,\log(4\,/\,R) \tag{17.48}$$

so

$$L_p = 70 + 10\,\log(4\,/\,93) = 56.3\,\mathrm{dB} \tag{17.49}$$

*Cocktail party effect is also used to describe our ability to understand an individual talker in the presence of a noisy combination of other conversations and background noise.

FIGURE 17.23 Percent of Words and Sentences Correctly Identified in the Presence of Background Noise (Kinsler et al., 1982)



This means that speech can be understood at a direct field level of 50.3 dB. Assuming the background noise due to other sources is low, the two people can converse comfortably at a separation distance of 3.9 m (13 ft).

Our first pair of guests arrives and the two groups begin talking, only now two people, one from each group, are talking simultaneously. The reverberant level increases by 3 dB, but the direct-field remains the same, so the minimum conversation distance drops to 2.7 m (9 ft). When two more couples arrive and pair off, the comprehension distance drops to 1.9 m (6 ft). When four more arrive the distance drops to 1.3 m (4 ft), and so forth.

In practice what happens is that people may choose not just to move closer, but also to talk louder. This raises the background noise and forces everyone to elevate their voices so at the end of the evening they all go home with sore throats—a corollary of the cocktail party effect. The point of this example is that more absorption in the room allows a higher signal-to-noise ratio and more people can talk comfortably before the increasing-volume spiral begins to kick in.

Restaurant Design

Restaurant design includes a similar problem in speech intelligibility since we want patrons to be able to talk comfortably across a table, but we do not want their conversations overheard by someone at a neighboring table. Consequently we need sufficient absorption that we do not have to raise our voices to be understood at a distance of 1 to 2 m (4 to 6 ft), but we want masking at a distance of, say, 3 m (10 ft) and beyond.

Let us imagine a restaurant that has a hard ceiling and walls and some absorption in the furniture for a total of around 20 metric sabins. A normal conversational level ($L_w = 70$ dB) will produce a direct field of 60 dB at 1.2 m (4 ft). With 20 metric sabins, our self-generated reverberant-field noise is 63 dB, our signal-to-noise ratio is -3 dB, and we achieve 75% intelligibility. If there are 20 tables in the room, with one person talking at each table, the reverberant noise level rises to 76 dB, a very uncomfortable level, and we can no longer have an intelligible conversation. This simple calculation tells us something useful—in hard-surfaced restaurants it is very difficult to have a normal conversation across a table.

People who enjoy talking to their dinner companions do not come back to these establishments and the owners ultimately suffer. Yet for some unfathomable reason countless restaurants are designed in this way.

We treat the problem by adding absorption. For example, assume that we cover the ceiling with an absorbent material. If it has an absorption coefficient of 0.9, this adds 170 metric sabins to the 13.7 × 13.7 m (45 × 45 ft) room. The 20 table reverberant noise level drops to 66 dB, which is low enough to carry on a cross-table conversation. At an adjacent table 3 m (10 ft) away, the direct field level from our conversation is about 54 dB and so it is not understandable. Off-axis directivity losses also may provide some additional isolation.

What we see from these relatively simple calculations is that unless we add absorptive treatment with an area approximately equal to the restaurant ceiling area, when the room is full of patrons, conversation across a table will be difficult and the background noise level will be uncomfortable. Second, even when we add this amount of absorption, the environment is not so dead that conversations are easily overheard at a neighboring table. More formally, these two conditions can be stated as follows.

$$L_p(\text{signal}) = L_w + 10 \log \left[\frac{Q}{4 \pi r^2} \right] \tag{17.50}$$

and

$$L_p(\text{noise}) = L_w + 10 \log N + 10 \log \left[\frac{4}{N R_t} \right] \tag{17.51}$$

where N is the number of simultaneous talkers (or tables) in the room and R_t is the absorptive area per table. The signal-to-noise ratio is the difference between these two equations

$$L_{SN} = 10 \log \left[\frac{Q}{4 \pi r^2} \right] + 10 \log \left[\frac{R_t}{4} \right] \tag{17.52}$$

To insure adequate communication for a cross-table distance equal to r_s we apply the condition that $L_{SN} > -6$ dB. This leads to the requirement that the amount of absorption per table in terms of the cross-table separation distance must be

$$R_t > 6.33 \, r_s^2 \tag{17.53}$$

To insure privacy between tables, we apply the condition that the signal-to-noise ratio $L_{SN} < -9$ dB. This leads to the requirement that the amount of absorption per table, in terms of the separation distance r_t between tables, be limited to

$$R_t < 3.16 \, r_t^2 \tag{17.54}$$

For a talker-to-listener distance of 1 m, our analysis suggests at least 6.3 or more square meters (68 sq ft) of absorption per table. If we treat the ceiling with a highly absorptive material, the minimum spacing between tables becomes about 2.5 m (8 ft), based on filling the room evenly. At that distance the maximum allowable absorption from Eq. 17.54 should be no more than 20 sq m (215 sq ft) per table. Normally we design based on Eq. 17.53 since the requirement in Eq. 17.54 is easily met.

Conference Rooms

Small conference rooms have become increasingly sophisticated primarily due to the audio-visual and computer interface requirements. Even with such systems in place face-to-face communication must still take place within a room, and the natural acoustical characteristics of the space are very important. Strong overhead reflections aid in cross-table communications so the ceiling above the table should be hard and flat. The area of reflective ceiling does not need to extend beyond the seating area. Outside this area the ceiling may be absorptive, diffuse, or recessed. In the central ceiling area acoustical diffusers are not particularly helpful. Above the conference table the ceiling should be low, preferably less than 3 m (10 ft) so the distance loss is minimized.

Most conference rooms are set up to have a table in the middle of the room with people seated around it. The shape of the conference table can help improve intelligibility. A lenticular shape allows people to see everyone seated at the table and also see plans and diagrams in the center. Horseshoe-shaped tables should be avoided, particularly if people are seated on both sides of the U, since they may face away from people on the other side.

Floors should be carpeted and absorption applied to the middle and upper portions of the walls in the form of cloth-wrapped panels, preferably with a tackable surface of 3 mm (1/8") dense fiberglass or cork between the cloth and the fiberglass. Reverberation times may be selected in accordance with the recommendations in Fig. 17.10.

Sound systems are often included in conference rooms if only to present recorded or transmitted material. Where there is a projection screen at one end of the room, loudspeakers should be located on either side. If there is also a speech-reinforcement system, loudspeakers are best located overhead with an electronic delay to maintain the correct impression of source direction. The loudspeaker system associated with the screen should not be used for speech reinforcement in order to minimize feedback.

Classrooms

The architectural design of a classroom begins with the seating layout, which is driven by the number of seats, code requirements, and the location of the audio-visual elements. Typical classrooms are relatively small, perhaps 25 feet wide by 30 feet deep, which will accommodate 30 to 40 students. Control of classroom noise, including exterior, mechanical, and reverberant, is of particular concern. For small classrooms, an NC 30 is an appropriate background level and noise from exterior sources such as traffic or aircraft should be limited to an L_{eq} of 35 dBA.

As a general rule reverberation times should be less than 0.8 seconds for good intelligibility. In his study of intelligibility in classrooms, Bradley maximized intelligibility as a function of reverberation time. The result appears in Fig. 17.19. He found that intelligibility was maximized at a reverberation time that depends on the signal-to-background-noise ratio in a range from 0.2 to 0.8 seconds. This requires a ceiling material of acoustical tile having an NRC of 0.8 or above for a ceiling height of between 9 and 12 feet. In small classrooms the ceiling is the only absorptive surface. When this is the case, the mid-frequency reverberation time can be estimated using the approximation

$$T_{60} \cong \frac{h_c}{20\,\overline{\alpha}_{NRC}} \tag{17.55}$$

where h_c is the ceiling height in feet and $\overline{\alpha}_{NRC}$ is the NRC value of the ceiling material. For a 10-foot ceiling, an NRC of 1.0 produces a half-second reverberation time.

Similar recommendations have been memorialized in an ANSI standard (ANSI, S12.60-2002) which sets background noise levels for spaces of less than 20,000 sq ft (283 cu m) to 35 dBA. Reverberation times for classroom volumes less than 10,000 sq ft are < 0.6 sec and for rooms having between 10,000 and 20,000 sq ft (566 cu m) are < 0.7 sec.

Small Lecture Halls

In a small lecture hall the choice of room shape is between a fan and a rectilinear form with a range of floor plans between the two. Fan-shaped rooms bring the seats closer to the front whereas a rectilinear shape provides a more frontal view of the display areas. Background levels due to HVAC systems and exterior sources should be limited to no more than an NC 30 or an L_{eq} of 35 dBA. Carpeted aisles are helpful in controlling the footfall noise due to latecomers. Automatic door closures without latches help to muffle the sounds of entry doors.

As a room grows larger the direct field should be augmented with early reflections from hard surfaces. Overhead reflections are preferred since the human ear is easily fooled as to the source direction, when the image source is located above (or below) the actual source. Lateral reflections smear the perceived source direction particularly when the reflection is louder than the direct sound. This condition occurs when there is grazing attenuation due to the presence of an audience that results in a direct sound being weaker than the reflected sound. In the case of large conference rooms, small auditoria, lecture halls, and legitimate theaters, a relatively low hard ceiling is preferable to an absorbent one.

A stepped or sloped floor, along with a raised platform for the talker, aids in the useful reflections and reduces grazing attenuation. Absorptive panels should be applied to the rear and side walls of the room to control reverberation and lateral reflections. The ceiling above the podium and the side walls surrounding the podium should be slanted (a 1:12 slope is sufficient) to avoid flutter echo.

A floor plan of a typical small lecture hall of about 120 seats is shown in Fig. 17.24. This hall is typical of several designed by the author and combines the audio-visual program with the acoustical requirements of the space. Moveable writing boards can be incorporated into the front walls along with projection screens for slides or video. The side walls at the front of the room are canted to accommodate the screens and to reduce the flutter echo from the side walls on either side of the lecturer.

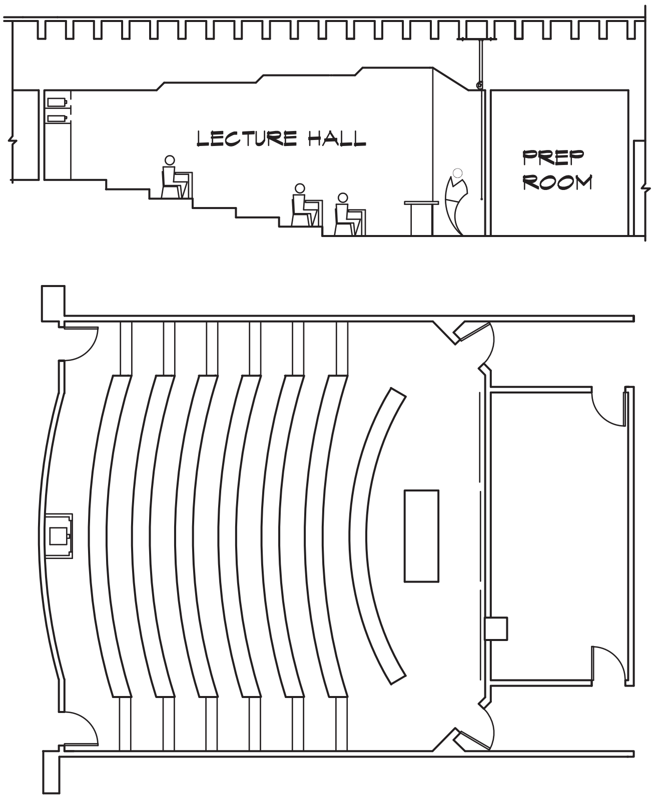
The ceiling is a series of flat-stepped elements, which provide beneficial early reflections. Flat ceiling elements are both more practical to build and better for intraclass discussions than more complicated ceiling shapes. The rear and side walls are treated with absorptive panels, which can be made tackable if classroom activities require the posting of student work.

The design of small lecture halls is increasingly influenced by the audio-visual requirements of the space. As room size increases the size of the projection screens must increase proportionately, and they tend to dominate the front surface of the room. A projection screen, which can be raised and lowered, is preferred to a fixed screen, since it discourages lecturers from writing on it, although surfaces are available that can be used for both functions.

Large Lecture Halls

In large lecture halls the design techniques are similar in principle to that of small halls. The distance from the source to the receiver should be short, which requires some widening of the seating area. Unsupported speech is not intelligible more than 30 to 40 ft (9 to 12 m)

FIGURE 17.24 Design of a Small Lecture Hall

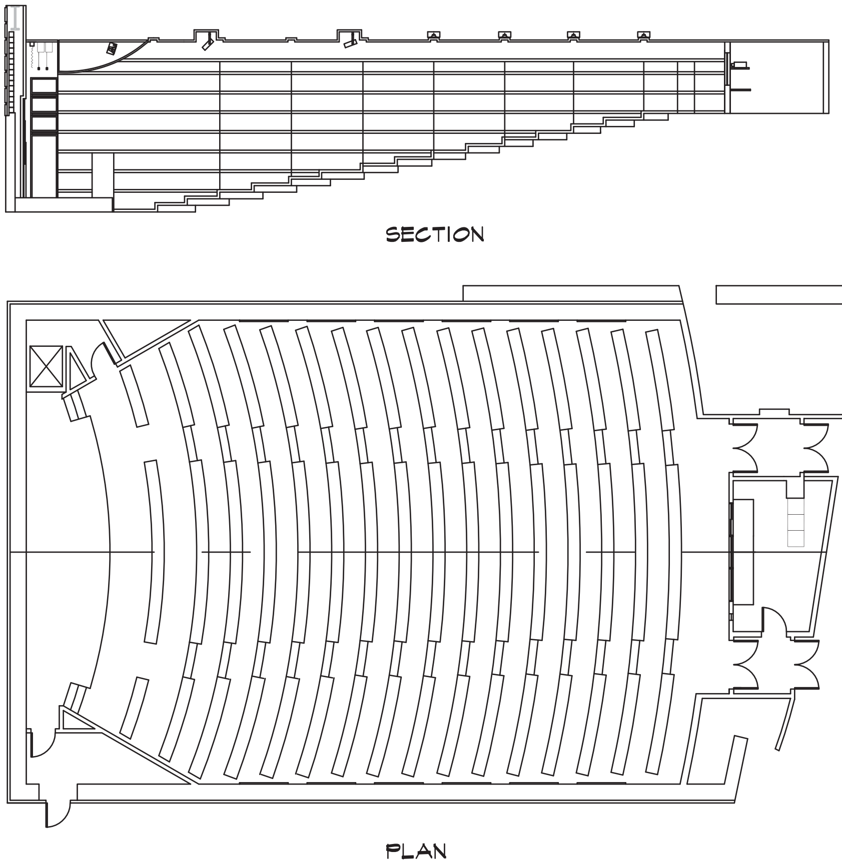


away unless considerable care is taken with the design. A fan-shaped configuration brings the audience close to the platform; however, the seating layout should be contained within a 125° maximum included angle if there is a projection screen on the front wall. The first-reflected-sound path should also be kept short. To that end the ceiling should be hard and relatively low so that the room volume (Doelle, 1972) is between 80 and 150 cu ft/seat (2.3 to 4.3 cu m/seat). Background-noise levels should be limited to no more than an NC 25 and exterior noise to an L_{eq} of 30 dBA, somewhat lower than the requirements for small halls.

A sound system should be included as part of the design and the loudspeakers should be integrated into the appearance of the room. The reverberation times can be selected from Fig. 17.10. If opera chairs are used, they should have padded seats and backs to reduce the variation in reverberation between the empty and full conditions. Reflections from the lower side walls can be helpful; however, reflections from the rear wall should be controlled with absorption.

An example of a successful lecture hall design at the Applied (Acoustics) Research Laboratory at Penn State University is shown in Fig. 17.25. This auditorium seats about 500 people and has extensive audio-visual capability. Although the room has a sound-reinforcement system, amplification is unnecessary, due to the drywall ceiling, but convenient for most lecturers. Loudspeakers are located behind a curved perforated metal screen as well as in the ceiling in the rear half of the seating area. The lower portion of the screen is backed with clear plastic to provide an overhead reflecting surface, which has the same appearance as the absorptive portion. Even though a flat ceiling yields good results when

FIGURE 17.25 Design of a Large Lecture Hall
Penn State Applied Research Laboratory (Acoustical Engineer, Marshall Long Acoustics) (Architect, The King Lindquist Partnership)

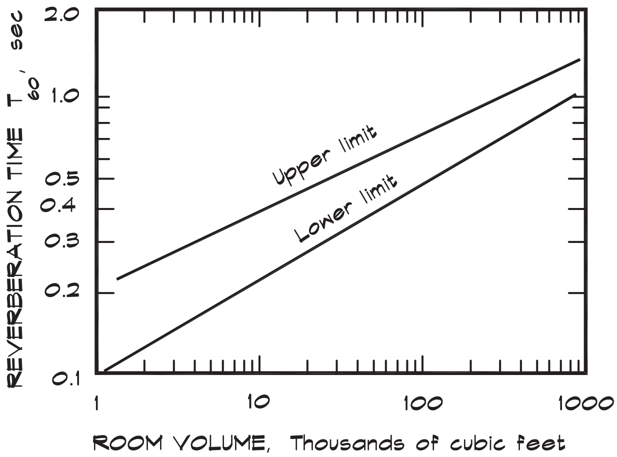


the seating is raked, a shaped ceiling is necessary in a room having a flat or shallow-angled floor. Lecture halls seating more than about 100 people should be designed with a sound reinforcement system and a hall having any type of projection or audio playback system needs to have sound reinforcement. In council chambers or courtrooms there may be a need for a recording system or for simultaneous translations. In these cases all talkers are miked and both loudspeakers and headphone feeds should be provided. The specifics of sound reinforcement design are discussed in Chapt. 18.

17.4 MOTION PICTURE THEATERS

Although motion pictures include speech and music, the design of movie theaters is driven by speech intelligibility considerations rather than by the need to provide reverberant support for unamplified music. The theater itself is an important link in the production chain since a film, as a mass-produced entertainment medium, is most effective if it is viewed in a controlled environment that yields the same auditory experience for every patron. Not all movie theaters are the same but they should be designed to achieve a consistent listening environment.

FIGURE 17.26 SMPTE Standard Reverberation Time vs Room Volume (SMPTE, 1989)



Reverberation Times

By and large, motion picture theaters are built to be acoustically dead, with absorptive material on virtually every surface except the floor, which must be washable. Ceilings are dark-colored acoustical tile and the side and rear walls are covered with minimum 1” thick cloth wrapped fiberglass panels or heavy pleated drapes. Curved rear walls should include 6” of fiberglass batt behind the panels to reduce focusing.

Recommended standards have been issued by SMPTE (Society of Motion Picture and Television Engineers) and by THX, a private company founded by George Lucas, on the preferred background noise levels, reverberation times, and sound system equalization curves. Motion picture theaters are designed to an NC 30 background-noise level and to the reverberation times shown in Fig. 17.26. THX recommends a minimum transmission loss rating (STC 65) for walls separating theaters, as well as a list of approved sound system components.