



Room Acoustics and Speech Reception: A Model and Some Implications

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Introduction

The speech signal reaching the ear of a listener is conditioned by the acoustic properties of the surrounding space. In essence, the room “couples” the talker to the listener in the same way that an earmold couples a hearing aid to an ear canal. A model of this process is developed in the present chapter. The model is based on four premises:

- a) The effective signal at the listener is a combination of the direct speech signal and the early components of reverberant speech.
- b) The effective noise level at the listener is a combination of the actual background noise and an equivalent noise generated by the late components of reverberant speech.
- c) The useful information in speech covers a range of 30 dB within any frequency band.
- d) The percentage of the effective signal that exceeds the effective noise provides an index of speech perception that is independent of the characteristics of the message or the listener.

Effective signal

When communication takes place out of doors, the only signal of interest is the direct speech signal, which travels in a straight line from talker to listener. In a room, however, the sound is trapped. It bounces around the room, making repeated reflections. Energy is lost at each reflection and the sound eventually becomes inaudible. But the sound persists for a while as it dies away. This persistence of sound in a room, after the source has stopped generating energy, is what we call *reverberation*.

The early components of reverberation arrive at the listener after only a few reflections. (In fact, they are more commonly referred to as early reflections. Both terms will be used here). Because they must travel farther than the direct signal, they are delayed in relation to it. But the delays are short enough that the human hearing system is able to ignore them. In the auditory system, therefore, the early reflections are integrated with the direct signal, and with each other, and contribute to the effective signal.

One problem with the direct signal is that its level falls by 6 dB every time the distance from the talker is doubled, as illustrated in figure 1. In contrast, the level of the early reflections is more or less constant throughout the room (also illustrated in figure 1). This level depends on four factors (Davis and Davis, 1997).

1. The level of the original signal
2. The source’s directionality, or *Q* (a measure of its tendency to radiate sound in a forward direction).
3. The volume of the room.
4. The room’s reverberation time (the time taken for sound level to fall by 60 dB after the source is turned off).

The room illustrated in figure 1 is assumed to measure $30 \times 20 \times 10$ feet and to have a reverberation time of 0.6 seconds. Talker level is assumed to be 72 dB at 1 foot. Talker *Q* is assumed to be 4 – meaning that radiation efficiency in the forward direction is 4 times that averaged over all directions. Under these assumptions the predicted level of the early reflections is around 56 dB SPL.

Because of the way the decibel scale works, the level of the combined signal is, essentially, that of the stronger component. Only when the two levels are

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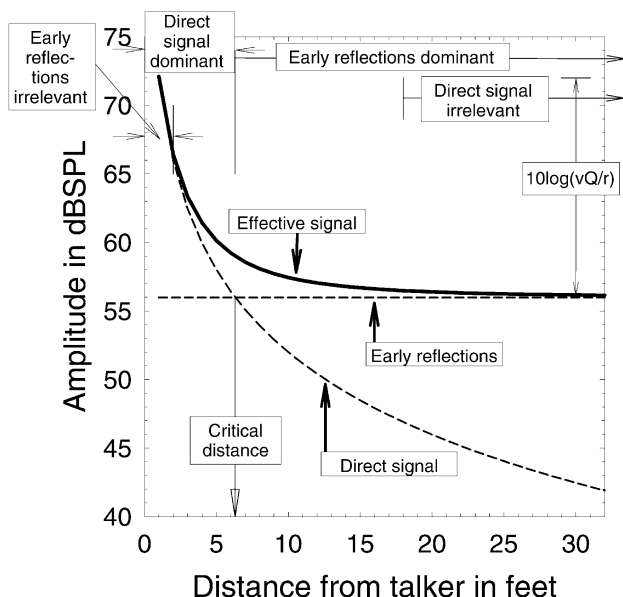


Figure 1. Estimates of the direct, early-reflected and combined (effective) speech signal as functions of distance in a room measuring $30 \times 20 \times 10$ feet ($v = 6000 \text{ ft}^3$) with a reverberation time (r) of 600 milliseconds. Talker Q is assumed to be 4. Average speech level at 1 foot = 72 dBSPL.

similar does this simple rule break down. And even then, if the two levels are identical, the combined level is only 3 dB higher than that of either one.

Critical Distance

The distance from the talker at which the levels of the direct signal and the early reflections are equal is known as the *critical distance*. In figure 1, this distance is around 6 feet. Beyond the critical distance the effective signal level is determined mostly by the early reflections. Beyond 3 times the critical distance (18 feet in this example) the contribution of the direct signal becomes negligible. Within the critical distance, the effective signal level is determined mostly by the direct signal. And within 1/3 of the critical distance (2 feet in this example) the contribution of the early reflections becomes negligible.

It is often assumed that any reverberation is bad. But listeners who are beyond the critical distance may rely on the early components of reverberation to keep the signal level above that of background noise (Bradley, 1986a, 1986b; Bradley, Sato and Picard, 2003). Consider, for example, a student at the back of the room illustrated in figure 1. Without early

reflections, the received signal, at only 42 dBSPL, could easily become lost in background noise. The early reflections, however, provide a 14 dB boost of level – albeit with an “echoic” quality.

Critical distance depends on the room’s dimensions and reverberation time and the directionality of the source. Table 1 provides estimates of critical distance as a function of reverberation time and room volume. As in figure 1, the talker is assumed to have a Q of 4.

Effective noise

Noise, by definition, is any sound that interferes with what the listener wants to hear. As with the effective signal, the effective noise has two components. The first consists of the actual background noise, which is present regardless of whether the talker is speaking. The second consists of the late components of reverberation, which are generated by the speech itself.

The actual background noise in a room can come from many sources, some external and some internal. Two of the commonest internal sources are heating, ventilating and air-conditioning units and the students themselves. The level of the actual noise is usually measured in dBSPL, using the A setting of the sound-level meter. This level tends to vary with location – especially for internal sources.

The late components of reverberation arrive at the listener after many reflections. They are so delayed in relation to the direct signal that the human ear cannot integrate them with it, or with the early reflections. Instead, the late components of reverberation interfere with the perception of speech sounds that started after them. The difference between early and late reflections is illustrated in figure 2, using text. A second version of each sentence (a “reflection”) is superimposed on the first. If it is shifted (delayed) by only a small amount (early reflection), perception is enhanced. But if the shift is too great (late reflection), perception is impaired.

Because they interfere with what the listener wants to hear, the late components of reverberation can be considered noise. The equivalent level of this noise, in relation to the early reflections, depends only on the reverberation time of the room (Peutz, 1977) as illustrated in figure 3. This level can be assumed to be constant throughout the room. For a listener who is far from the talker, therefore, the effective signal-to-noise ratio, even in a quiet room, is limited by late

Table 1. Estimates of critical distance as a function of room size and reverberation time.

Volume (cubic ft.)	Reverberation time (seconds)							
	0.3	0.4	0.5	0.6	0.7	0.8	0.9	1.0
2000	5.2	4.5	4.0	3.7	3.4	3.2	3.0	2.8
4000	7.3	6.3	5.7	5.2	4.8	4.5	4.2	4.0
6000	8.9	7.7	6.9	6.3	5.9	5.5	5.2	4.9
8000	10.3	8.9	8.0	7.3	6.8	6.3	6.0	5.7
10000	11.5	10.0	8.9	8.2	7.6	7.1	6.7	6.3
12000	12.6	11.0	9.8	8.9	8.3	7.7	7.3	6.9
14000	13.7	11.8	10.6	9.7	8.9	8.4	7.9	7.5
16000	14.6	12.6	11.3	10.3	9.6	8.9	8.4	8.0
18000	15.5	13.4	12.0	11.0	10.1	9.5	8.9	8.5
20000	16.3	14.1	12.6	11.5	10.7	10.0	9.4	8.9

Critical distance (feet)



Figure 2. A simulation of the different effects of early and late reflections.

reverberation. In the example shown in figure 3, for example, the effective signal-to-noise ratio for a listener at the back of the room, in the absence of actual noise, is only around 4 dB.

As with the effective signal, the decibel level of the effective noise is, essentially, that of the stronger of the two components – as illustrated in figure 4. This figure also demonstrates that the reduction of late reverberation, by acoustic treatment, will be of little or no benefit once its level falls below that of the background noise. This point will be discussed later under implications.

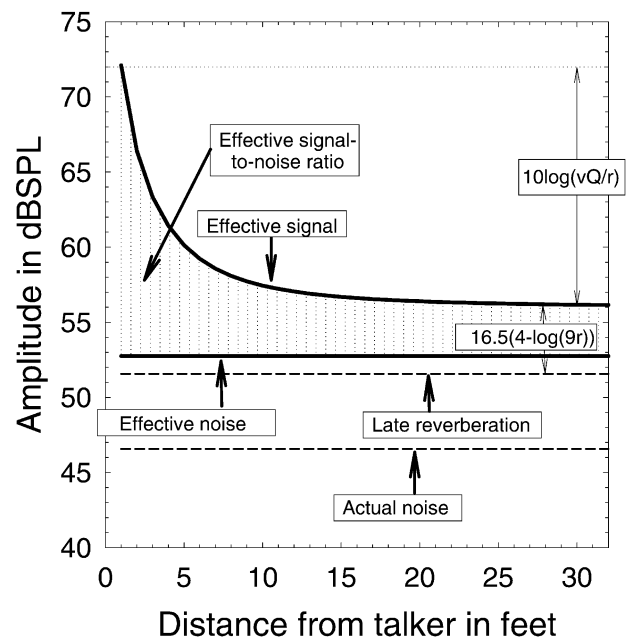


Figure 3. Estimate of equivalent noise level of late reverberation in the same room illustrated in figure 2. Actual noise is arbitrarily assumed to be 5 dB below the late reverberation. The effective noise is the logarithmic sum the two.

Effective signal-to-noise ratio

By subtracting the average level of the effective noise from the average level of the effective signal we arrive at an effective signal-to-noise ratio. This ratio is relatively constant for listeners who are well beyond the critical distance, but as will be seen from

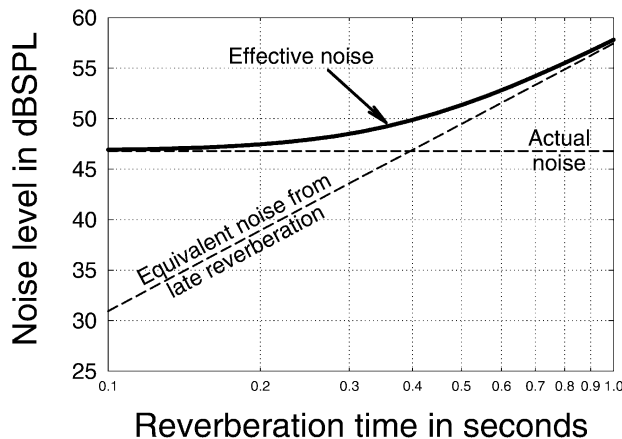


Figure 4. The effective noise is the logarithmic sum of the actual noise and the equivalent noise generated by the late components of reverberation. The latter is a function of room size, talker Q, and reverberation time. This illustration is for the same room as in figures 1 and 3, with actual noise held constant at 47 dBSPL and reverberation time varied between 0.1 and 1.0 seconds.

figure 3, it increases dramatically for listeners who are close to the talker.

The illustrations in figures 1 and 3 show only the long-term average level of the effective signal. But the short-term level of speech varies over quite a wide range (see, for example, Boothroyd, Erickson and Medwetsky, 1994). Figure 5 shows short-term amplitude as a function of time for a sample of male speech. In this figure, it will be seen that the level of the stressed vowels is some 15 dB above the average level while the level of the soft sounds sometimes drop to 15 dB below it. The result is an overall range of around 30 dB. These measurement were made with an integration time of 50 milliseconds and were limited to the two-octave frequency band between 707 and 2828 Hertz – this being the region of most importance to speech intelligibility.

Because of the 30 dB variation in speech level, full audibility of the acoustic signal requires a signal-to noise ratio of at least 15 dB. That is to say, the average speech level must be at least 15 dB above the average noise level. By the same token, the speech will be completely inaudible when the signal-to-noise ratio is –15 dB or less. That is to say, the average speech level is 15 dB or more below the average noise level. When the average levels of the effective speech and effective noise are equally (signal-to-noise ratio = 0 dB), roughly 50% of the speech signal will be audible.

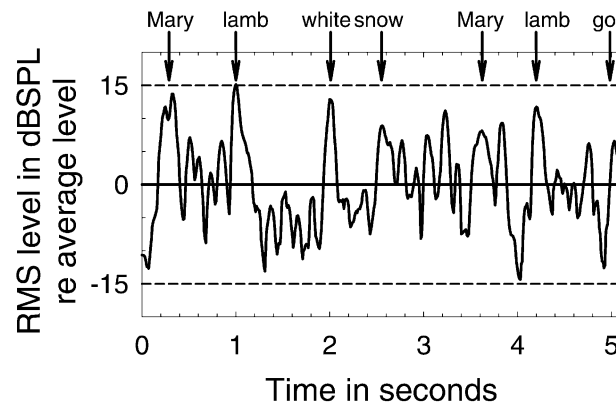


Figure 5. Short-term speech level, as a function of time, for a sample of speech produced by a male talker. Level is shown for the frequency band between 707 and 2828 Hz. Text is: “Mary had a little lamb; its fleece was white as snow. And everywhere that Mary went, the lamb was sure to go”.

Speech Audibility Index

Speech Audibility index is defined, here, as the percentage of the effective signal that is not masked by the effective noise. The transform from effective signal-to-noise ratio (s/n) to Speech Audibility Index (SAI) is:

$$SAI = (s/n + 15)/30 * 100 \dots\dots\dots (1)$$

Where:

SAI = Speech Audibility Index in %

s/n = effective signal-to-noise ratio in dB

Several indices have been developed for estimating the efficiency of speech transmission. These include, Articulation Index (AI), Speech Intelligibility Index (SII), Speech Transmission Index (STI) and Rapid Speech Transmission Index (RASTI) (American National Standards Institute, 1995, 2002a, 2002b; Fletcher, 1953; French and Steinberg, 1947; Steeneken and Houtgast, 1973). What these indices have in common with each other, and with Speech Audibility Index, is that they are all attempts to express speech reception in purely physical terms, without reference to the abilities of the listener or the nature of the communication. In the case of Speech Audibility Index, the physical variables are the original speech level, talker directionality, distance, noise, and reverberation time. SAI is comparable to STI in that it accounts for reverberation. The two indices are, however, derived from different models.

Predicting speech perception

Although Speech Audibility Index makes no reference to how listeners interpret the received speech signal, it can be transformed into estimates of speech perception. Any such transforms, however, must take account of the hearing and language abilities of the listener and the content and complexity of the language (Boothroyd, 1978, 1985, 2002; Boothroyd and Nitttrouer, 1988; Miller, Heise and Lichten, 1955; Nitttrouer and Boothroyd, 1990).

Figure 6 provides an example of the possible effect of hearing status on word recognition. The solid line shows predicted word recognition in fairly complex sentences for a person with normal hearing. The broken line shows predictions for a person with a sensorineural hearing loss in the region of 55 dB who is wearing a hearing aid. The assumption is that the poor resolution resulting from cochlear damage has reduced Speech Audibility Index by a factor of around 2. It is also assumed that the two listeners have the same world and language knowledge and perceptual skills so that the only difference is auditory capacity. This analysis predicts that the normally hearing person can function effectively with Speech Audibility Index as low as 60%. In contrast, the person with a hearing loss needs Speech Audibility Index to be as close to 100% as possible.

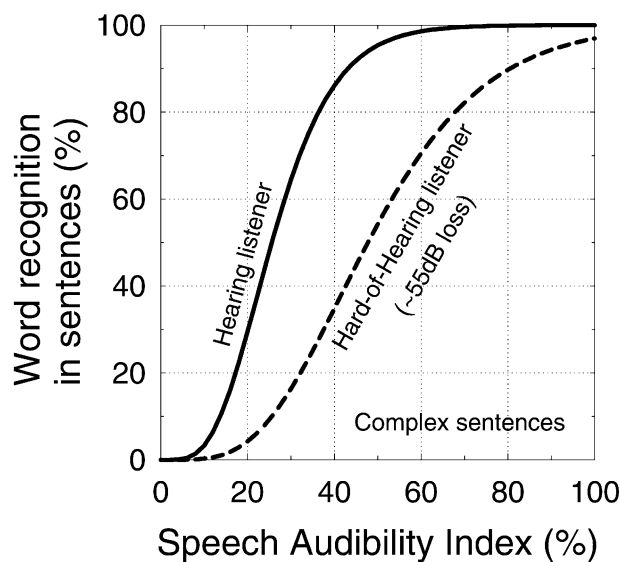


Figure 6. Estimated word recognition in complex sentences, as a function of Speech Audibility Index, for a hearing and hard-of-hearing listener.

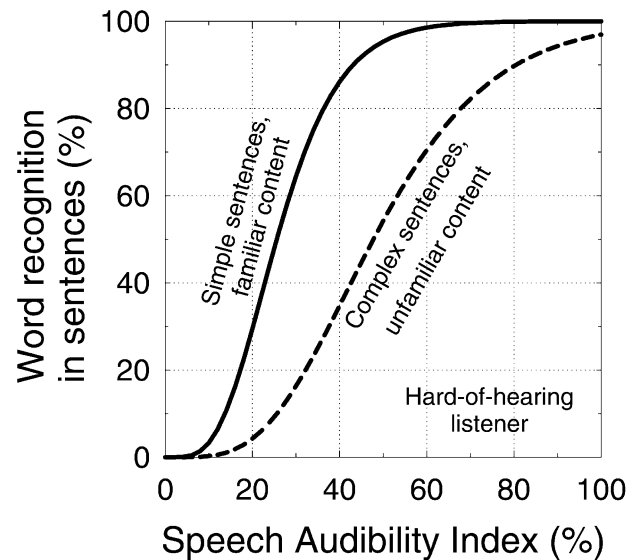


Figure 7. Estimated word recognition in simple and complex sentences, as functions of Speech Audibility Index, for a hard-of-hearing listener (approximately 55 dB loss).

Figure 7 provides an example of the possible effects of language complexity on word recognition by the same person with a hearing loss, wearing a hearing aid, as predicted from Speech Audibility Index. In this case, the solid line shows predicted word recognition in simple sentences with familiar content. The broken line shows predictions for complex sentences with unfamiliar content such as might be expected in a learning situation. This analysis predicts that, in casual conversation, this individual will be able to function very effectively with a Speech Audibility Index as low as 65%, but the same acoustics will be totally inadequate in a learning situation.

The message, here, is that criteria for the adequacy of an acoustic environment cannot be absolute but must take account of the capabilities of the listener and the nature of the intended communication¹. (See, also, American Speech-Language Hearing Association, 1995; Crandell and Smaldino, 2000; Grant and Seitz, 2000; Johnson, 2000; Neuman and Hochberg, 1983; Nitttrouer and Boothroyd, 1990; Picard and Bradley, 2001).

¹ The predictions in figures 6 and 7 were obtained using *j* and *k* factors selected to reflect the benefits of lexical and sentence context (Boothroyd, 1985; Boothroyd and Nitttrouer, 1988).

Some Implications of the Model

1. Directional Microphones

When the effective signal consists mainly of early reflections, directional microphones will be relatively ineffective. The directional microphone is designed to attenuate signals that do not come from directly in front – such as the actual noise and the late components of reverberation. Unfortunately the early reflections also come from non-frontal directions and are suppressed. The result is little or no gain in effective signal-to-noise ratio. Maximum benefit from directional microphones occurs when the listener is at or within the critical distance (see, also, Surr, Walden, Cord and Olson, 2002).

2. Optimal Reverberation Time

Acoustic modifications of classrooms must address noise and reverberation together. Reducing the late components of reverberation will reduce the effective noise, but only to the point where actual noise becomes dominant. Further reductions will simply reduce the level of the early reflections and, therefore, the effective signal-to-noise ratio, for listeners who are far from the talker.

An example is shown in figure 8. Note that the best signal-to-noise ratio, for a listener who is relying on early reflections, is obtained when the late reverberation is lowered slightly below the point at which its equivalent noise equals the actual background noise. The full benefits of further reductions will only occur when background noise is also lowered.

Table 2 provides an estimate of optimal reverberation time as a function of room size and background noise. Reducing reverberation time below these values, without simultaneously reducing background noise, will cause a loss of effective signal-to-noise ratio for those listeners who are far from the talker.

3. Measured Signal-to-noise Ratio

The third implication is that empirical measures of signal-to-noise ratio should be interpreted with caution. Signal-to-noise ratio is often used as a criterion for the acoustical adequacy of classrooms. But such measures compare the effective signal level only with the *actual* noise level (Latham, 1979). The contribution of the late components of reverberation is

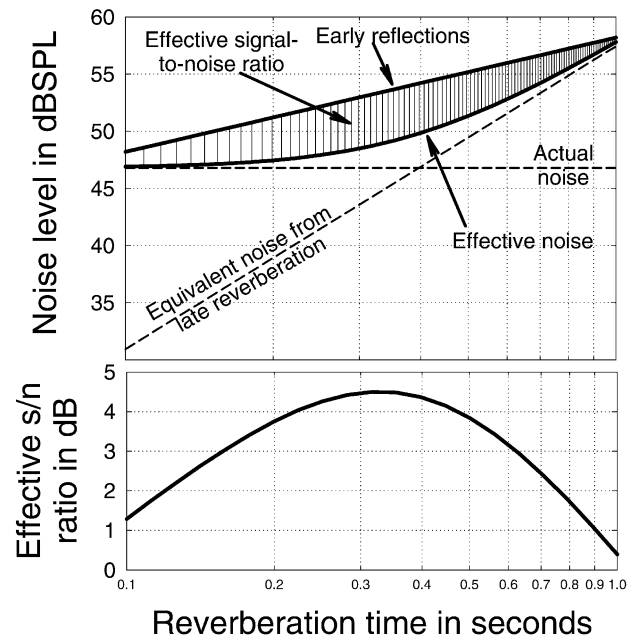


Figure 8. Effective signal level, noise level and signal-to-noise ratio as functions of reverberation time, for a listener whose effective signal consists entirely of early reflections. The room dimensions, talker Q and actual noise level are the same as in figure 3.

ignored. In Figure 3, for example, the measured signal-to-noise ratio at the back of the room would be around 9 dB. But the *effective* signal-to-noise ratio at this location is only 3 dB.

4. Sound-field Amplification

Sound-field amplification offers one way to enhance coupling between the talker and the listener. A noise-free and reverberation-free signal is picked up by a close-talking microphone and delivered to one or more loudspeakers. Unfortunately, the problem of coupling the talker to the listener is now replaced by the problem of coupling the loudspeaker(s) to the listener.

Amplification of the signal boosts three things:

1. The level of the direct signal (as delivered by the loudspeaker).
2. The level of the early components of reverberation
3. The level of the late components of reverberation

Actual background noise, however, is not boosted. If, therefore, the effective noise is dominated by actual

Table 2. Estimated optimal reverberation times as a function of room size and actual background noise level.

Actual noise (dB SPL)	Room volume in cubic feet									
	2000	4000	6000	8000	10000	12000	14000	16000	18000	20000
20	0.03	0.03	0.04	0.04	0.05	0.05	0.05	0.06	0.06	0.06
25	0.04	0.05	0.06	0.07	0.07	0.08	0.08	0.09	0.09	0.09
30	0.06	0.08	0.09	0.10	0.11	0.12	0.13	0.13	0.14	0.15
35	0.09	0.12	0.14	0.16	0.17	0.18	0.20	0.21	0.22	0.22
40	0.15	0.19	0.22	0.24	0.27	0.29	0.30	0.32	0.33	0.35
45	0.22	0.29	0.34	0.38	0.41	0.44	0.47	0.49	0.51	0.53
50	0.35	0.45	0.52	0.58	0.64	0.68	0.72	0.76	0.79	0.82
55	0.53	0.69	0.81	0.90	0.98	1.05	1.11	1.17	1.22	1.27
60	0.82	1.07	1.25	1.39	1.51	1.62	1.72	1.81	1.89	1.97
65	1.27	1.65	1.93	2.15	2.34	2.50	2.65	2.79	2.92	3.04
70	1.97	2.56	2.98	3.32	3.61	3.87	4.10	4.31	4.51	4.69

Optimal Reverberation time in seconds

noise, then amplification enhances signal-to-noise ratio. If, however, the effective noise is dominated by late reverberation, any amplification of the effective signal is offset by an equal amplification of the effective noise.

The message, here, is that sound-field amplification offers a straightforward way to enhance coupling of talker to listeners (at least for the talker with the microphone) so long as the primary problem being

addressed is background noise. If, however, the primary problem being addressed is reverberation, then sound-field amplification offers less benefit. To optimize benefit, the equipment must be selected, installed and adjusted with great care – taking maximum advantage of directionality of the loudspeakers and proximity of the listeners to a loudspeaker.

For purposes of reference, table 3 gives the estimated equivalent noise level of late reverberation

Table 3. Equivalent noise level, generated by the late components of reverberation, as a function of room size and reverberation time.

RT (secs.)	Room volume (cubic feet)									
	2000	4000	6000	8000	10000	12000	14000	16000	18000	20000
0.15	40	37	36	34	33	33	32	31	31	30
0.20	44	41	39	38	37	36	35	35	34	34
0.25	46	43	41	40	39	38	38	37	37	36
0.30	48	45	44	42	41	41	40	39	39	38
0.35	50	47	45	44	43	42	42	41	41	40
0.40	52	49	47	46	45	44	43	43	42	42
0.45	53	50	48	47	46	45	45	44	43	43
0.50	54	51	49	48	47	46	46	45	45	44
0.55	55	52	51	49	48	48	47	46	46	45
0.60	56	53	52	50	49	49	48	47	47	46
0.70	58	55	53	52	51	50	50	49	49	48
0.80	60	57	55	54	53	52	51	51	50	50
0.90	61	58	56	55	54	53	53	52	51	51
1.00	62	59	57	56	55	54	54	53	53	52
2.00	70	67	65	64	63	62	62	61	61	60
5.00	81	78	76	75	74	73	72	72	71	71

Equivalent noise level (dB SPL)

as a function of room size and reverberation time. If the actual noise is 10 dB below the value shown here, it may be assumed that the effective noise consists mainly of late reverberation.

5. Personal FM or Loudspeaker

As already stated, a microphone within a few inches of the talker's mouth can pick up a noise-free and reverberation-free signal. If this signal is transmitted directly to a listener's hearing aid or personal loudspeaker – by an FM wireless link, for example – then optimal access is available anywhere in the room. The coupling of talker to the listener's ear or microphone is perfect (See, Ross, 1992).

The benefits of this approach are, of course, limited to the speech of the person with the microphone. And if a listener with a personal FM system uses the hearing aid in the Aid+FM mode, incorrect adjustment of the relative gains via the Aid and FM microphones will negate the benefits of the close-talking microphone (American Speech-Language, Hearing Association, 2002). Nevertheless, if properly used, the close-talking microphone with direct transmission to a personal aid or loudspeaker is by far the best solution to problems caused by inferior room acoustics – a fact that has been known to the entertainment and broadcast industries for decades.

Multi-band Implementation

A full implementation of the model described here must account for the spectra of the speech and background noise as well as the frequency-dependence of reverberation time. Estimates of Speech Audibility Index are made for several frequency bands and then combined; using weightings based on the relative importance of those bands to speech perception (see figure 9). A single-band implementation can be used, however, if the spectra of the source and background noise are similar and if reverberation time is estimated for the frequency band between 707 and 2828 Hz – this being the region of most importance to speech perception.

Summary

In summary:

1. The effective signal received by a listener is a combination of the direct speech signal and those

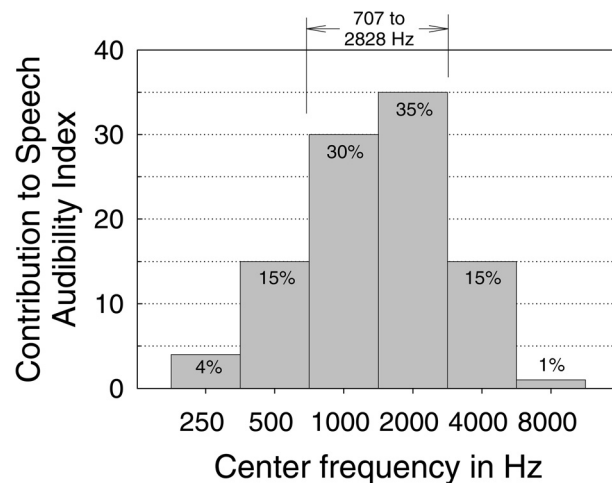


Figure 9. Octave-band contributions to Speech Audibility Index (the Frequency vs. Importance function). These data are averaged across three male and three female talkers and are based on the work of Mulhearn (1999).

speech reflections that arrive early enough to be integrated with it (early reverberation).

2. The level of the direct signal falls by 6 dB for every doubling of distance.
3. The level of the early reflections depends on the level and directionality of the source and the volume and reverberation time of the room.
4. Critical distance is the distance from the source at which the direct signal and the early reflections make equal contributions to the effective signal level.
5. Effective noise is a combination of actual noise and those speech reflections (late reverberation) that arrive too late to be integrated with the effective signal.
6. The effective signal-to-noise ratio is the difference of level between the effect signal and the effective noise.
7. The level of the late reverberation falls below that of the early reverberation by an amount that depends only on the reverberation time of the room.
8. Reverberation time, therefore, places an upper limit on the effective signal-to-noise ratio for a listener who is beyond three times the critical distance.
9. Speech Audibility Index is a physical measure of speech perception that rises from 0% to 100% as the effective signal-to-noise ratio rises from -15dB to +15 dB.

10. Speech Audibility Index can be used to predict speech perception, but only if the effects of language complexity and listener hearing status, knowledge and listener skill are accounted for.
11. Speech Audibility Index should be calculated separately for several frequency bands and the results summed according to the relative importance of those bands to speech perception.
12. If a single-band implementation is needed, it should use the two octaves centered around 1000 and 2000 Hz (707 to 2828 Hz).

Implications of the model developed here include:

1. The effectiveness of directional microphones falls when the listener is beyond the critical distance.
2. For a given room volume and noise level, there is an optimum reverberation time below which Speech Audibility Index falls.
3. Direct measures of signal-to-noise ratio do not account for late reverberation and are overly optimistic in reverberant rooms.
4. Sound-field amplification easily increases signal-to-noise ratio when the effective noise is dominated by actual noise.
5. The benefit of sound-field amplification is reduced when the effective noise is dominated by late reverberation, and that benefit depends on the effective control of source directionality and listener-loudspeaker distance.
6. Direct transmission of the signal from a close-talking FM microphone to a personal hearing aid or loudspeaker is the most effective way to address problems of room acoustics for an individual listener. But if a personal aid is set to Aid + FM, incorrect adjustment of relative Aid and FM gain will negate the benefit.

We have many options for optimizing the coupling of talker output to the ear or microphone of a listener. A thorough understanding of the underlying processes will help to ensure that that we make full and effective use of these options.

Acknowledgements

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The model described in this presentation is incorporated into Sound Field Wizard – a simulation

program demonstrating the effects of acoustics and amplification on speech reception and perception in a room. The software is downloadable, free of charge, from either www.phonicear.com or www.arthurboothroyd.com.

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