>1.7 seconds:	Directional loudspeakers required (churches, multipurpose auditoriums and highly reflective spaces).
>2 seconds:	Very careful design required. High quality directional loudspeakers required. Intelligibility may have limitations (concert halls, churches, treated sports halls/arenas).
>2.5 seconds:	Intelligibility will have limitations. Highly directional loudspeakers required. (large masonry churches, sports facilities, arenas, atriums, enclosed railway stations and transportation terminals).
>4 seconds:	Very large churches, cathedrals, mosques, large and untreated atriums, aircraft hangars, untreated enclosed ice sports arenas/stadiums. Highly directional speakers required and located as close to the listener as possible.

When designing or setting up systems for highly reverberant and reflective environments, the main rule to follow is: aim the loudspeakers at the listeners and keep as much sound as possible off the walls and ceiling. This tends to maximize the D/R ratio, though in practice it may not be quite so simple.

7. Echoes and Late Reflections:

As mentioned earlier, speech signals arriving within approximately 35 ms of the direct sound generally integrate with the direct sound and aid intelligibility. In most sound system applications, and particularly in distributed systems, considerable numbers of early reflections will be received at a given listening position, and some bridging (sequential masking) may occur which can extend the useful arrival time out to perhaps 50 ms. The way in which single or discrete reflections affect intelligibility and our perception has been studied by a number of researchers. The best known is probably Haas.

Haas found that under certain conditions, delayed sounds (reflections) arriving after an initial direct sound could in fact be louder than the direct sound without affecting the apparent localization of the source. This is often termed the Haas effect. Haas also found that later arriving sounds may or may not be perceived as echoes, depending upon their delay time and level. These findings are of significant importance to sound system design and enable delayed in-fill loudspeakers to be used to aid intelligibility in many applications ranging from under or over-balcony fills in auditoriums and pew back systems in churches to large venue rear fill loudspeakers. If the acoustic conditions allow, improved intelligibility and sound clarity can be achieved without loss of localization.

Figure 24 presents the basic Haas curve. Haas showed that with delays of about 15 to 25 milliseconds a secondary signal had to be about +10 dB higher in level than the primary for it to be judged as being as equally loud. This is quite an astonishing result and further helps to explain the integration characteristics of the ear. The curve should not be confused with the true precedence effect, which was studied by Wallach and Meyer & Shodder. This describes an effect in which listeners lock onto the direction of the first arriving sound and ignore, either partially or completely, subsequent short term delayed sounds or reflections. In this effect the delayed sounds integrate or fuse completely with the initial or direct sound, as shown in Figure 25.

We therefore have two useful phenomena which can be used to advantage in sound system design:

* Listeners tend to lock onto the first arrival sound and its direction, ignoring short-term delayed secondary sounds.

* Secondary sounds arriving within approximately 30 - 35 ms of the initial or primary sound fuse with it to produce one apparent sound of increased loudness. Furthermore, the secondary sounds may be up to 10 dB louder than the primary before being judged as equally loud.

Figure 24. **Basic Haas effect curve.**

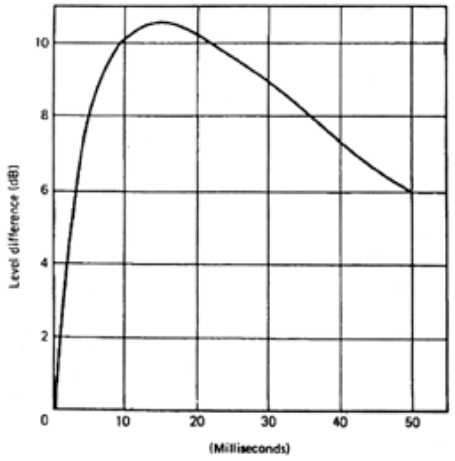
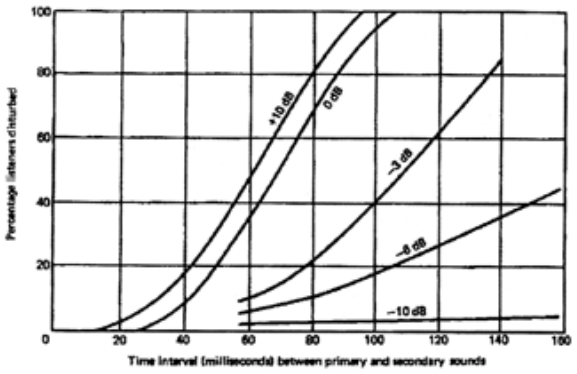


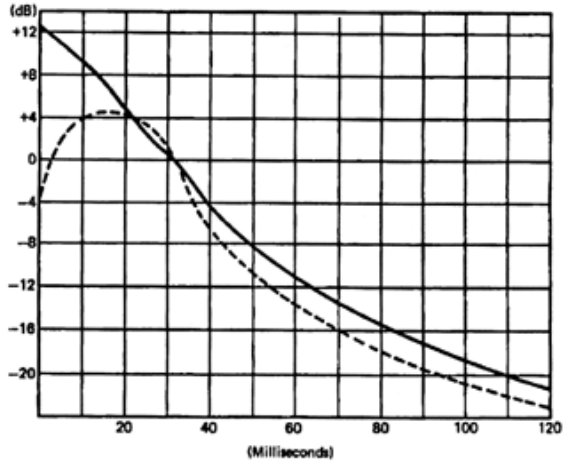
Figure 25. **Haas echo disturbance curves.**



Unfortunately, a widely held and often quoted misconception has arisen as a result of the above findings. It implies that a secondary or delayed signal can be up to 10 dB louder than the primary or direct sound before it is perceived as a secondary source with loss of localization of the primary source. The inference is that one can increase the local level in a sound system by up to 10 dB from a secondary delayed loudspeaker (e. g., under the balcony) before it is detected and heard in its own right without a shift from the primary source. This is not the case; at +10 dB Haas predicts that the secondary source will sound "equally loud" which, by definition, means that the secondary source is clearly being detected as such. In fact, for the secondary source or signal to be just imperceptible for time delays of between around 10 to 25 ms, the secondary signal can only be about 4 to 6 dB higher than the level of the primary signal at the listening position.

In Figure 26, the dotted line (after Meyer and Shodder) presents a curve of 'echo' perception versus delay and source level differences. This useful curve shows that delayed sounds become readily discernible at delays in excess of 35 ms. For example, at 50 ms delay, a secondary signal has to be more than 10 dB lower before it becomes imperceptible, and more than 20 dB lower at 100 ms. The solid curve in Figure 26 tells us when a delayed sound will be perceived as a separate sound source and ceases being integrated with the direct sound.

Figure 26. **Echo perception curve.**

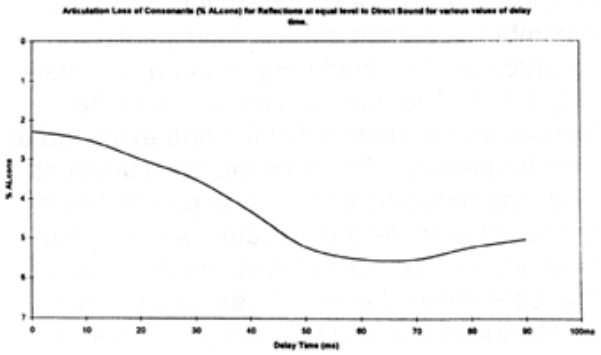


A third useful piece of data relating to time delay and secondary source echo perception is shown in Figure 25. The graph provides an insight into the likelihood of an echo or secondary source causing disturbance to a listener. For example, when the primary and secondary sources are of equal level and have a 30 ms delay between them, less than 5% of

listeners will be disturbed. With secondary sources more than 10 dB lower, Haas determined that only a few percent of listeners would be disturbed, although from Figure 25 it can be seen that such echoes would be clearly heard.

Although potentially annoying, echoes may not degrade intelligibility as much as is generally thought. Figure 27, based on the work of Peutz, shows the reduction in $\%Al_{cons}$ caused by discrete late sound arrivals or echoes. The figure starts at just below 2% with no delay, since this was the residual loss of speech information for the particular talker and listening groups used in conducting the experiment. As the figure shows, typically only 4 - 5% loss was measured. This means that a correction factor of only around 2 - 3% additional loss needs to be added to the total Al_{cons} for a given system.

Figure 27. **Effect of echoes on $\%Al_{cons}$.**



The data discussed above are based on the effects of single reflections. In large reverberant spaces, a series of reflections will occur and may integrate together, becoming potentially more disruptive. Such late reflections can degrade intelligibility even though they may not be perceived as discrete echoes in themselves. This is particularly true when the intelligibility is already degraded by the normal reverberation of the space. For example, if the reduction in $\%Al_{cons}$ is 10% due to reverberation, it could be further reduced by say 3% due to an echo, resulting in 13% Al_{cons} and creating a condition that is unsatisfactory. This assumes no loss of intelligibility due to the person talking, which may add 1 - 2% further reduction, even for well articulated speech. This may result in borderline articulation.

Too many widely spaced loudspeakers in distributed systems can cause undesirable delays when their spacing exceeds approximately 50 feet (17 m). This can also occur in systems where a local in-fill loudspeaker is used to provide coverage without signal delay. In the first case, the spacing between speakers should be reduced, or a back-to-back mounting employed. In the remote in-fill case, the arrival times should be brought back into synchronism by means of a delay line.

8. Talker Articulation and Rate of Delivery:

As has already been mentioned, the individual articulation of talkers can have a profound effect upon intelligibility as perceived by the listener. Peutz found that even good talkers could cause 2 - 3% additional loss, and poor ones up to 12.5% additional loss. It is therefore prudent to design in a margin for such potential losses. The rate at which a person speaks is also an important factor in reverberant spaces. Considerable improvement in intelligibility can be achieved by making announcements at a slightly slower than normal rate in difficult environments, such as large churches or other untreated venues.

The importance of announcer training can not be overstated, but it is often ignored. Prerecorded messages loaded into high quality, wide bandwidth digital systems can overcome certain aspects of this problem. For highly reverberant spaces, the speech rate needs to be slowed down. This may be difficult to accomplish during normal use, but carefully rehearsed slower recordings can be very effective.

Research has shown that intelligibility is improved when the lips of the talker can be seen. At low levels of intelligibility (0.3 to 0.4 Al) visual contact can produce improvements up to 50%. Even with reasonably good intelligibility (0.7 to 0.8 Al) improvements up to 10% have been observed. This suggests that paging systems may have a more difficult task

than will typical speech reinforcement systems with visual cues.

No sound reinforcement system is without its limitations, and they need to operate in acoustically acceptable environments. It has been found that feeding back a slightly delayed or reverberated voice signal via headphones to the talker can be a very effective way of slowing the talker's rate of speech.

9. Uniformity of Coverage:

When working in noisy and/or reverberant spaces, it is essential to provide uniform direct sound coverage level. While a 6 dB variation (±3 dB) may be acceptable under better conditions, such a variation in a reverberant space can lead to intelligibility variations of 20 to 40%. A 40% degradation of clarity under such conditions is usually unacceptable. Again, the off-axis performance of the selected loudspeakers becomes of critical importance. Where the listeners are free to move around, as in a shopping mall, then it may be possible to relax the variation in intelligibility within the space. However, with a seated audience in an enclosed space, no such luxury can exist if all are to adequately hear and understand.

An allowable variation in direct sound coverage of 3 dB or less should be aimed for, particularly over the range 1 - 5 kHz. This is a stringent and often costly requirement. To put this into perspective, assume that in a given space with a reverberation time of 2.5 seconds, a preliminary design shows that an on-axis value of 10% Al_{cons} can be achieved. From Figure 23, it can be seen that reducing the D/R ratio by 3 dB will reduce the intelligibility to 20% Al_{cons} - an unacceptable value. It is therefore vital to consider the off-axis positions and the uniformity of coverage, particularly when we recall that the S/N ratio will also be similarly degraded and will contribute additionally to reverberation based losses.

10. Equalization:

Even loudspeakers with nominally flat frequency response, when turned on for the first time, may produce anything but flat response on the analyzer screen. Any major discrepancies in frequency response will of course have an adverse effect on the intelligibility and clarity of the sound system. The five main reasons for response anomalies in sound systems in enclosed spaces are:

1. Local boundary interactions (see Figure 8)
2. Mutual coupling or interference between loudspeakers
3. Irregular sound power interacting with reverberation characteristics of the space
4. Incorrectly loaded loudspeaker (e. g., a ceiling loudspeaker with a too-small back can and/or highly resonant enclosure)
5. Inadequate coverage, resulting in dominant reverberant sound off-axis.
   To these may be added abnormal or deficient room acoustics exhibiting strong reflections or focusing.

An example of the effects listed in items 1 - 3 above is shown in Figure 28. This is a measurement made on a sound system comprising a number of a short column loudspeakers mounted on the structural piers of a reverberant church. The anechoic response of the loudspeaker is reasonably flat and extended at high frequencies. Because the measurement (listening) position is beyond critical distance, the reverberant field dominates, and it is the total acoustic power radiated into the space that determines the overall response. Figure 29 illustrates the concept. Many distributed sound system loudspeakers do not exhibit flat power response but almost always have a rising response at low frequencies. This coupled with the longer reverberation time at low frequencies characteristic of many stone or dense structure buildings results in an overemphasis at low frequencies. The peak in the response at 400 Hz is due to a combination of mutual coupling of loudspeakers and local boundary interactions. The resultant response causes a considerable loss of potential intelligibility as high frequency consonants are masked. Equalizing the system, as shown by the dotted curve. improves clarity and intelligibility significantly. Figure 30 presents a *house curve* response suitable for distributed systems. This is a guideline only, but has proved

to be a good starting point for many systems. Interestingly, the equalized response shown in Figure 29 does not exactly match the recommended curve, since extending the high frequency response produced an overly bright, shrill sound. This is due to the discrepancy between the sound power (reverberant response) and the nominal direct field sound response (traditional reference response) of the loudspeaker.

Figure 28. **Response of distributed system in reverberant space.**

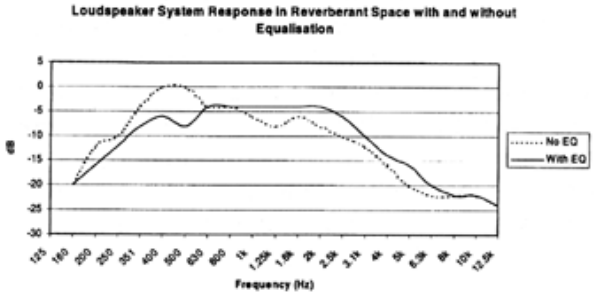


Figure 29. **Loudspeaker coverage, direct and reverberant contributions.**

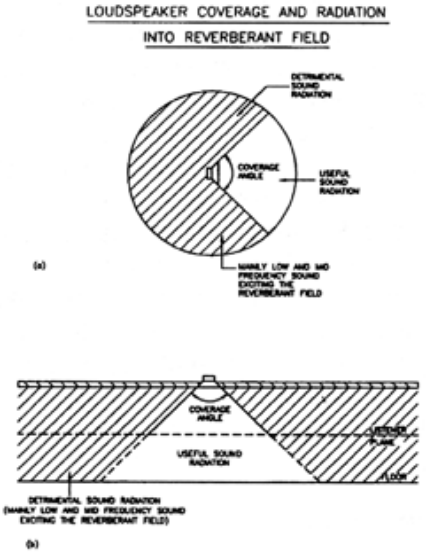
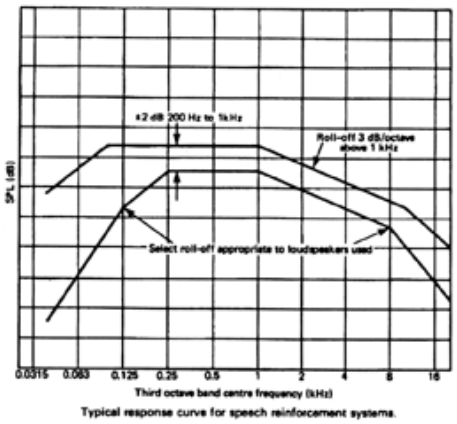


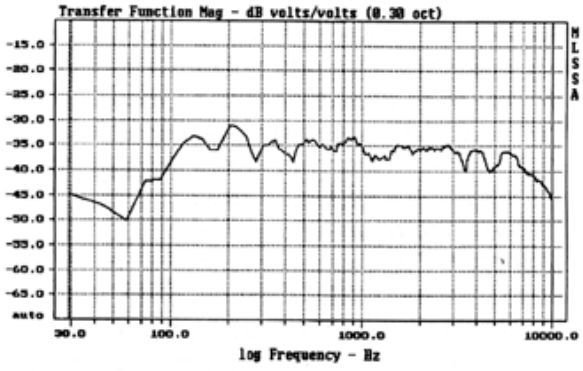
Figure 30. **Speech reinforcement "house curve."**



Typical response curve for speech reinforcement systems.

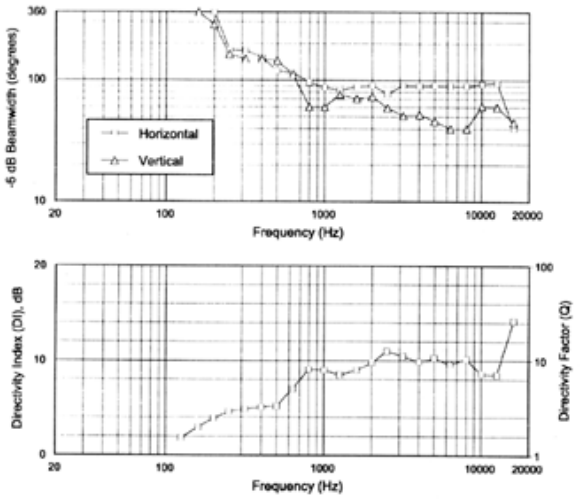
When a loudspeaker with smooth power response that essentially follows its direct on-axis response is used in a sound system, less high frequency rolloff will occur, and it may thus be possible to extend the high frequency equalization curve. This is often the case in well controlled environments where the listener is within critical distance of the loudspeaker. An example of this is shown in Figure 31, which shows the *in situ* measurement of a laterally distributed system using 2-way loudspeakers. The reverberation time is 0.6 s and C_{50} is clearly positive.

Figure 31. **Distributed system with high D/R response.**



The off-axis and sound power responses of a loudspeaker are often forgotten parameters. Speakers exhibiting a well controlled, smooth response without excessive attenuation within the nominal coverage angle should be used. As target examples of proper response, Figure 32 shows the directional data for the JBL SP215-9 full range sound reinforcement loudspeaker.

Figure 31. **Distributed system with high D/R response.**



Electronic equalization is a powerful tool and can make a remarkable difference in the clarity of a system. However, it needs to be carried out carefully and with a full understanding of what is happening acoustically. Remember that response peaks can usually be attenuated, but sharp response notches are generally acoustic in origin and cannot be 'fixed' by simple frequency domain equalizers.

Adding bass to a sound system might make it sound impressive but will do nothing for its clarity and intelligibility. Indeed, in a reverberant or even semi-reverberant space, too much bass will adversely affect clarity. Nevertheless, many operators, DJs and announcers actually think that more bass makes their voices sound better! In some cases it may be useful to design a system with two signal input paths; one for speech with a carefully determined LF rolloff, and the other tailored for music. Designers have long known that flat response in sound reinforcement systems often causes more problems than it solves, and the traditional house curves used in the industry invariably show a controlled rolloff above about 1 or 2 kHz.

11. Speech Intelligibility Criteria: Measurement, Assessment and Estimation Techniques:

Sections 5 & 6 introduced a number of criteria for judging system performance in the areas of speech intelligibility and clarity. We will now present a more detailed and comprehensive discussion of this information:

Word Scores and Sentence Intelligibility:

The fundamental measurement of intelligibility assessment is of course speech itself. Over the years a number of techniques have been developed to directly rate speech intelligibility. The initial work was carried out in the 1920s and 30s and was associated with telephone and radio communication systems. From this work the effects of noise, S/N ratio and bandwidth were established and subjective testing methods were formulated. The sensitivity of the various test methods was also established, and it was found that tests involving sentences and simple words, while easiest to conduct, were the least sensitive to corruption and did not always provide sufficiently detailed information for system analysis.

The need to ensure that all speech sounds were equally weighted led to the development of phonemically balanced (PB) word lists. Lists with 32, then 250 and finally 1000 words were developed. Tests using nonsense syllables (logatoms) were also produced. These provide the most sensitive measure of speech information loss but are complex and very time consuming in application. The MRT (Modified Rhyme Test) was developed as a simpler alternative to PB word lists and is suitable for use in the field with only a short training period. (The more sensitive methods can take several hours of training before actual testing can begin.) The various methods and their inter-relationships are shown in Figure 11, where the Articulation Index (AI) is used as the common reference.

Articulation Index:

The Articulation Index was one of the first criteria and assessment techniques developed to use acoustical measurements and relate these to potential speech intelligibility. AI concerns the effects of noise on speech intelligibility and was primarily developed for assessing telephone communications channels. Later, corrections were added in an attempt to take

account of room reverberation time, but the method is not considered sufficiently accurate for present-day needs. AI remains a very accurate means of assessing the effects of noise on speech. ANSI Standard S3.5 1969/1988 specifies the methods for calculation based on measurements of the spectrum of the interfering noise and the speech signal. The index ranges from 0 to 1, with 0 representing no intelligibility and 1 representing 100% (total) intelligibility. See Kryter (1962).

Articulation Loss of Consonants (%Al_{cons}):

This method, developed by Peutz during the 1970s and further refined in the 1980s, takes into account noise and reverberation effects and is the most widely used method currently available. The criteria for the various degrees of intelligibility are set out in Figure 22, however, the measurement of a system to determine whether it has met the desired criteria is not as well established. The TEF analyzer incorporates a sub-routine for doing this and is one of the few instruments to do so. The intelligibility module of JBL SmaartPro is also able to do this.

A significant limitation of the measurement method is that it uses only the 2 kHz band. For natural speech, where there is essentially uniform directivity among talkers, single-band measurements can be quite accurate. Sound system response is usually far from linear, and considerable errors can occur, particularly when measuring high-density distributed sound systems. However, where devices exhibiting a uniform directivity are involved, good correlation between the calculated data and word score intelligibility has been established.

Direct to Reverberant and Early to Late Energy Ratios:

Measures such as C_{50} and C_{35} have been established for the assessment of natural acoustics in theaters and concert halls. A well defined scale has not been developed, but it is generally recommended for good intelligibility in such venues that a positive value of C_{50} is essential and that a value of around +4 dB C_{50} should be targeted. (This is equivalent to about 5% Al_{cons}). Measurements are usually made at 1 kHz, or may be averaged over a range of frequencies. The method does not take account of background noise (usually not a problem in auditoriums), and is not particularly suitable for sound system measurements due to the lack of a defined scale and the fact that it is usually measured only at 1 kHz. See Lochner and Burger (1964).

Speech Transmission Index (STI) and Rapid Speech Transmission Index (RASTI):

The STI technique was developed in Holland during the 1970s and 1980s. While the %Al_{cons} method has become popular in the USA, the STI method is far more widely used in Europe. It has been adopted in a number of European Standards relating to speech intelligibility and Sound Systems design and assessment.

The technique considers the source/room/listener as a transmission channel and measures the reduction in modulation depth of a special test signal as it traverses the channel. Schroeder showed that this was related to the room's impulse response, and that a wide range of test signals could be used to determine its value. A number of instruments are now available that can measure either STI or RASTI (the simpler, shorter method), including TEF, MLSSA, JBL Smaart and some models of B&K equipment. The full STI technique is a very elegant analysis method and is based on the modulations occurring in normal speech. Measurements are made in the main speech frequency octave bands from 125 to 8 kHz, with 14 individual modulation signals in each band ranging from 0.63 to 12.5 Hz. Because the STI method operates over the entire speech band, it is well suited to assessing sound system performance. The RASTI method is restricted to fewer modulation signals in only the 500 Hz and 2 kHz bands; even so, it still takes account of a wider frequency range than the D/R or %Al_{cons} techniques.

The STI/RASTI scale ranges from 0 to 1; zero represents complete unintelligibility and 1 represents perfect sound transmission. Good correlation exists between the STI scale and subjective word list tests. As with all objective electroacoustic measurement techniques, STI does not actually measure the intelligibility of speech, but only certain parameters that correlate strongly with intelligibility. It also assumes that the transmission channel is linear. For this reason an STI measurement can be 'fooled' by certain system nonlinearities or time-variant processing. For example, discrete, late arriving echoes can corrupt the results, as they can with any of the measurement systems we have described. A significant advantage of STI is that it automatically takes into account room background noise directly in the measurement itself. The basic equation for STI modulation reduction factor m(f) is:

$$M(f) = \frac{1}{\sqrt{1 + [2\pi fT/13.8]^2}} \bullet \frac{1}{1 + 10^{(-S/N)/10}}$$

Unfortunately, this equation cannot be solved directly, making STI prediction a complex procedure requiring a detailed computer analysis of the sound field. The approximate relationship between RASTI and $\%Al_{cons}$ however has been established and is given below. Figure 33 relates the two scales.

The data presented in Table 2 shows the approximate correspondence between RASTI measurements and $\%Al_{cons}$ calculations or measurements:

Figure 33. **STI/RASTI versus Al_{cons} scale.**

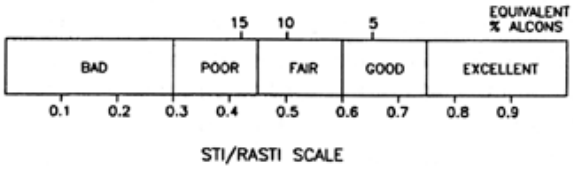


Table 2. **RASTI and Alcons Measurements.**

	RASTI	%Alcons		RASTI	%Alcons
BAD	0.20	57.7	GOOD	0.66	4.8
	0.22	51.8		0.68	4.3
	0.24	46.5		0.70	3.8
	0.26	41.7		0.72	3.4
	0.28	37.4		0.74	3.1
	0.30	33.6		0.76	2.8
	0.32	30.1		0.78	2.5
	0.34	27.0		0.80	2.2
				0.82	2.0
POOR	0.36	24.2		0.84	1.8
	0.38	21.8		0.86	1.6
	0.40	19.5			
	0.42	17.5	EXCELLENT	0.88	1.4
	0.44	15.7		0.90	1.3
	0.46	14.1		0.92	1.2
	0.48	12.7		0.94	1.0
	0.50	11.4		0.96	0.9
				0.98	0.8
FAIR	0.52	10.2		1.0	0.0
	0.54	9.1			
	0.56	8.2			
	0.58	7.4			
	0.60	6.6			
	0.62	6.0			
	0.64	5.3			

The SII (Speech Intelligibility Index) - 1997) is closely related to Articulation Index (AI). SII calculates the effective signal-to-noise ratio for a number of frequency bands related to speech communication. Several procedures with different frequency domain resolutions are available, including 1/3 octave and one octave, a 21-band critical bandwidth analysis, and an analysis based on 17 equally contributing bands. The method is suitable for direct communication channels rather than sound reinforcement systems.

12. Speech Intelligibility Optimization: Summary of Main Techniques:

It is worth remembering that sound quality and intelligibility are not the same thing. Often, a deliberately shaped system response with little bass and perhaps an accentuated HF range can be clearer than a ruler-flat, distortionless system. The following tips will prove useful in optimizing the intelligibility of a system:

* Aim the loudspeakers at the listeners, and keep as much sound as possible off the walls and ceiling.
* Provide a direct 'line of sight' between loudspeaker and listener.
* Ensure an adequate bandwidth, extending from at least 250 Hz to 6 kHz, preferably to 8 or 12 kHz.
* Avoid frequency response anomalies. Roll off the bass and ensure adequate but not excessive high frequency response. Try to avoid mounting loudspeakers in corners unless local boundary interactions can be effectively overcome.
* Minimize the distance between the loudspeaker and listener.
* Ensure a speech S/N ratio of at least 6 dB-A, preferably >10 dB-A.
* Ensure the microphone user is adequately trained and understands the need to speak clearly and slowly in reverberant environments.
* Provide a quiet area or refuge for the announcement microphone or use an effective close talking, noise canceling microphone with good frequency response.
* Avoid long path delays (>50 ms). Use electronic delays and inter-speaker spacing of less than 45 feet (15 m).
* Use automatic noise level sensing and gain adjustment to optimize S/N ratios under varying noise conditions.
* Use directional loudspeakers in reverberant spaces to optimize D/R ratios. Models exhibiting flat or smoothly controlled sound power response should be used if possible.
* Minimize direct field coverage variations. Variations of as little as 3 dB can be detrimental in live spaces.
* Consider making improvements to the acoustic environment. Do not design the sound system in isolation; remember, the acoustical environment will impose limitations on the performance of any sound system.
* Under very difficult conditions, use simple vocabulary and message formats.

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