SPEED OF SOUND
by Cor Riethof, Technical Director, RDL-Europe

INTRODUCTION
Sound travels very slowly in air compared with the speed of light. Light speed is almost instantaneous at 299337 km (186000 miles) per second; sound speed in air crawls along at about 342 meters (1120 feet) per second depending on the temperature. To be exact 343.59 m/s (1127.29 ft/s) at a temperature of 20 degrees C (68 degrees F) and humidity of 60%. It is the relatively slow speed of sound which causes audible echoes to bounce back from a flat surface like a wall, building or mountain. The closer you are to the wall the shorter the echo. As you move further away the echo time lengthens. If you check echo lengths with the chart below, remember to double the length shown because the sound has to travel there and back.

The repeating sound you often hear from multiple speaker public address systems give an excellent illustration of time delay and the problems it creates for the sound designer or engineer. It's a common occurrence in railway stations because there are usually speakers placed in a straight line down the length of a platform. Other examples include most distributed audio systems in sport arenas, multi purpose halls and outdoor audio systems.

Even in theaters and churches one might experience problems with audio timing with the sound of under balcony or other loudspeakers remote from the main loudspeakers.

A typical situation is a system that uses a main loudspeaker cluster high above the stage or altar (and in fact almost above the first rows of the audience) that is set for a reasonably high sound pressure level.

The sound system-designer does not want amplified sound coming back at the stage or altar, because this will almost certainly cause undesired feed-back. The result is that the first rows have very bad intelligibility because the direct sound, especially the high frequencies from the cluster, does not reach the audience while the direct sound (speech) from the priests or actors is too soft to solve this problem. Also in this case there is a relatively small but very important timing problem to make it worse. We like the audience to have the illusion that the sound arrives directly from the priest or actors and not from above, although some priests might appreciate that, but probably not if it is their own speech.

In cases like this it is recommended to fit a horizontal row of small loudspeakers (maybe about 1.5 meter apart depending on the distance to the first row) in front of the audience to cover the bad reception from the amplified sound cluster and to create the illusion of louder direct sound from the actors or priest for visitors in the front rows.

This introduces the next problem. Timing problems between the direct sound from the priest or actors, the amplified sound from the cluster and the sound of the small front speakers will cause bad intelligibility because the ears of the audience get confused. We can solve this problem by delaying the signal fed to the small front speakers. In most cases about 15 to a maximum of 35 milliseconds will be enough.

Sometimes, depending on the sound system-design, it is desirable to delay the main loudspeaker system to synchronize it with the audio source. It may also be beneficial to delay a part of the main loudspeaker system, to time-align the loudspeaker combination. This is especially the case if all the speakers used are not in line. This happens often with the woofers and subwoofers.

After the primary sound from the nearest speaker reaches your ears, the repeats you hear are the sounds arriving from speakers further away. Hearing multiple sound sources can result in the message becoming indistinct. The use of audio delay units in sound systems can contribute to perfect intelligibility.

RDL AUDIO DELAY UNITS

The standard method for improving system synchronization and intelligibility is with the use of signal time delay units. *RDL* devices (for example, see RU-ADL2) use a full bandwidth digital processor, to "hold up" the signal to the secondary speakers for a fraction of a second until it is in sync with the main source. Mono and multiple units must be set-up carefully (the chart below might be a big help).

Many other applications are possible and most of them are based on the theory of Dr. Helmut Haas published in 1946. The so-called Haas effect or precedence effect, in short:

> *If two successive sounds are heard as fused, the location of the total sound is determined largely by the location of the first arriving sound. This, among other things, stops you getting confused when a sound comes at you from two speakers or audio sources at once.*

STI AND RASTI

Performance indicators now used are called STI and RASTI, which prescribe the desired intelligibility of announcements; those have replaced the formerly used % AlCons (Articulation loss of Consonants) method.

STI means Speech Transmission Index and RASTI means Rapid Speech Transmission Index. Both are measured the same way, but the STI measurement covers a wider frequency range. The graphs are almost identical although the frequency differences can produce a different result. The graphs will also be different depending on the volume of the space.

STI and RASTI values vary between 0 and 1.00. 0 means very bad intelligibility; 1.00 means excellent intelligibility. In a normal meeting room values above 0.70 to 0.80 are considered to indicate good intelligibility.

Time Delay Table for loudspeakers

Meters	Delay in milliseconds		Feet	Delay in milliseconds
5	14.6		20	17.9
10	29.2		30	26.8
15	43.8		40	35.7
20	58.5		50	44.6
25	73.1		60	53.6
30	87.7		70	62.5
35	102.3		80	71.4
40	117.0		90	80.4
45	131.6		100	89.3
50	146.2		110	98.2
55	160.8		120	107.1
60	175.4		130	116.1
65	190.1		140	125.0
70	204.7		150	133.9
75	219.3		160	142.9
80	233.9		170	151.8
85	248.5		180	160.7
90	263.2		190	169.6
95	277.8		200	178.6

THE HAAS EFFECT

The Haas Effect, also called the precedence effect, describes the human psycho acoustic phenomena of correctly identifying the direction of a sound source heard in both ears but arriving at different times. Due to the head's geometry (two ears spaced apart, separated by a barrier) the direct sound from any source first enters the ear closest to the source, then the ear farthest away. The Haas Effect tells us that humans localize a sound source based upon the first arriving sound, if the subsequent arrivals are within 25-35 milliseconds. If the later arrivals are longer than this, then two distinct sounds are heard. The Haas Effect is true even when the second arrival is louder than the first (even by as much as 10 dB!). In essence we do not "hear" the delayed sound. This is the hearing example of human sensory inhibition that applies to all our senses. Sensory inhibition describes the phenomena where the response to a first stimulus causes the response to a second stimulus to be inhibited, i.e., sound first entering one ear cause us to "not hear" the delayed sound entering into the other ear (within the 35 milliseconds time window). Sound arriving at both ears simultaneously is heard as coming from straight ahead, or behind, or within the head. The Haas Effect also describes how full stereophonic reproduction from only two loudspeakers is possible. (After Helmut Haas's doctorate dissertation presented to the University of Gottingen, Gottingen, Germany as "Über den Einfluss eines Einfachechos auf die Hörsamkeit von Sprache;" translated into English by Dr. Ing. K.P.R. Ehrenberg, Building Research Station, Watford, Herts. England Library Communication no. 363, December, 1949; reproduced in the United States as "The Influence of a Single Echo on the Audibility of Speech," J. Audio Eng. Soc., Vol. 20 (Mar. 1972), pp. 145-159.)