AN EVALUATION OF THE SOUNDSHOWER™ LISTENINGLAMP™ TV SOUND ENHANCEMENT SYSTEM

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INTRODUCTION

The SoundShower™ ListeningLamp™ (SSLL) TV sound system has been designed to help people with hearing loss to hear and understand the sound from a television more clearly and with greater ease. The device consists of an infrared transmitter which transmits the TV sound, an infrared receiver / amplifier with volume and tone controls, and a small loudspeaker that is positioned just above the head of a seated listener, by means of an adjustable tripod stand with boom. (Fig. 1 shows a picture of the SSLL receiver and loudspeaker, as set up for use.) Since it first became available in 1997KU, more than KU units have been sold, and apparently users find the device to be effective in its intended application (see, e.g., testimonials at www.audiologyproducts.com). The purpose of this paper is to present a technical assessment of the device, with the aim of explaining, mostly in scientific terms but also with informed impressions, why the device is effective. Several sets of electroacoustic measurements are described, the results of which support the notion of the effectiveness of the device.

Fig. 1 – The receiver and loudspeaker of the SSLL, as set up for use.

SOME PROPERTIES OF HEARING LOSS THAT AFFECT TV LISTENING

It is fair to say that people with normal hearing usually can listen comfortably to a television set located at a "typical" distance away in a "typical" living room. However, broad clinical experience also makes it fair to say that people with hearing loss very often have considerable difficulty when listening under the same conditions. In general, two sources of signal degradation can contribute to such difficulty: 1] the acoustic conditions in the listening room (to be discussed in a later section), and 2] the psychophysical (perceptual) degradations caused by hearing loss. Regarding the latter, perhaps the most obvious is an elevation of hearing threshold levels: a hearing impaired person may require elevated sound levels just to hear the sound from a television set. Unfortunately, there are at least two reasons why turning up the volume may not provide satisfactory help: 1] Normal-hearing listeners in the same room may not be willing to listen at the elevated sound levels desired by a hearing-impaired listener. 2] Certain attributes of hearing loss can make it difficult for the hearing-impaired listener to be comfortable with understanding speech and with discerning other sounds from a television set, even at elevated levels.

Perhaps the most important of these attributes is the broadening of auditory frequency filters that is typical of cochlear hearing loss (Moore, 1998). The broadening of auditory filters in cochlear hearing loss leads to an increased susceptibility to off-frequency masking (Crandell, 1991). Masking, in general, is the phenomenon by which the threshold for hearing a "target" signal can be elevated by the presence of another signal within the same auditory filter. Off-frequency masking is the phenomenon by which a sound at one frequency can mask an otherwise audible sound at another, somewhat distant frequency.

One form of off-frequency masking is called "upward spread of masking" (USM). With USM, low-frequency sounds mask higher-frequency sounds. Normal-hearing listeners experience this phenomenon (especially at high sensation levels), but the situation can be far worse for those for those with cochlear hearing loss, because of broadened auditory filters. With broadened auditory filters, it is more likely that a low-frequency masker and a higher-frequency "target" signal will fall within the same auditory filter, thereby increasing the chance that the masker will be an effective one (Moore, 1998). In fact, one of the consequences of USM is that the lower frequencies of vowel sounds in speech can mask the higher frequencies of upper formants and consonants, and thus one part of a speech signal can mask another. This phenomenon is sometimes called “self-masking.”

Unfortunately, no amount of “raw” amplification (simply turning up the volume) can make a self-masked speech-sound more audible. The same can be said regarding the failure of raw amplification to increase the audibility of sounds masked by reverberation (to be discussed in detail, later). Indeed, self-masking, reverberation, and other adverse acoustical properties of TV listening rooms – such as background noises and high-frequency sound absorption – can combine to make masking a very difficult problem for hearing-impaired listeners to overcome.

A solution?

One of the ways to make a masked sound audible is to increase the signal-to-noise ratio (SNR) – that is, to amplify the masked sound without amplifying the masker (see, e.g., Egan and Hake, 1950). Consequently, in helping hearing impaired listeners overcome the effects of USM, general clinical practice is to provide amplification with high-frequency emphasis, toward improving the audibility of high-frequency sounds without amplifying the low-frequency sounds which can mask them. Most prescriptive strategies for hearing-aid fittings (e.g., Byrne, et al., 2001), in fact, call for high-frequency emphasis in the amplified frequency response, for both downward-sloping and flat audiograms.

However, in hearing impairment, the perceptual effects of broadened auditory filters can vary greatly, depending on the individual listener. For example, off-frequency masking can be worse than normal in either direction, not only in the upward direction (as with USM). That is, some listeners may be more susceptible than normal to “downward spread of masking” (Crandell, 1991). For such listeners, especially for those who may have upward-sloping hearing losses, low-frequency emphasis can potentially make low-frequency sounds more audible, in a manner similar to the way high-frequency emphasis can help with USM. It will be shown that the SSLL provides something of an antidote to masking, in the form of increased SNR with a range of choices for emphasis of one frequency range or another.

RANGE OF FREQUENCY RESPONSES

The graphs in Fig. 2 show the range of frequency responses available by adjustment of the tone control of the SSLL. To obtain these graphs, a pink noise signal was fed electrically to the transmitting unit of the SSLL via a “Minirator” hand-held signal generator from NTL. A ¼-inch measurement microphone, Earthworks M30L, was placed approximately 6 inches from the loudspeaker of the SSLL and was connected to an NTL “Minilyzer” hand-held signal analyzer, which was set to perform 1/3-octave band analysis. The graphs in Fig. 2 show the resulting 1/3-octave band analyses for three positions of the tone control of the SSLL. The analyses show that the emphasis in the frequency response can be varied from the low frequencies (at 250 Hz and above), to the high frequenciesy , as shown in graphs (a), (b), and (c), respectively. From an audiological perspective, these frequency responses represent a range of sound tailoring that targets improving audibility for a wide range of hearing-loss configurations.

Range of Frequency Responses of the SSLL

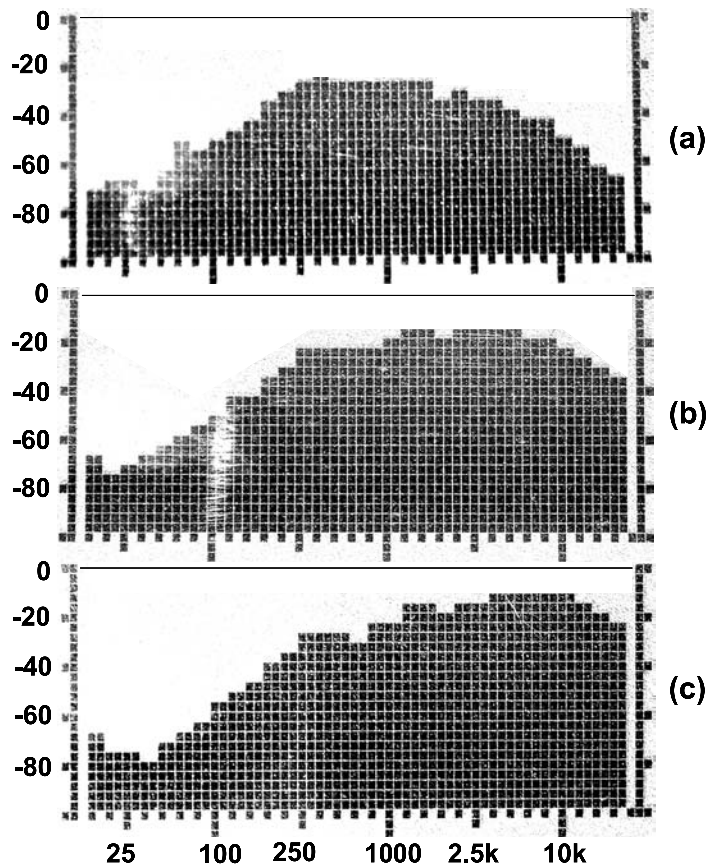

Fig. 2 – 1/3-octave band analyses of listening lamp frequency responses for three positions of the tone control. Continuous variation is possible between the lowest (a), middle (b), and highest (c) positions of the control. The vertical scale is in decibels; the horizontal scale is in Hertz. (The above graphs are actually photographs of the screen of the Minilyzer analyzer; the white shady areas in the upper and middle graphs were caused by reflections of light from the screen.)

OVERCOMING THE ADVERSE EFFECTS OF BACKGROUND NOISE AND HIGH-FREQUENCY ACOUSTIC ABSORPTION

This section describes how the function of the SSLL targets improved audibility, with consideration given to some of the typical acoustic conditions in TV listening rooms. Extraneous background noises, such as electric fans, people talking in the next room, or traffic noise from an open window, can decrease the SNR in the listening area, thus increasing the chance for the masking of the sounds coming from a television loudspeaker. Research shows that people with hearing impairments require a higher SNR than do normal-hearing people to avoid the adverse effects of masking on speech understanding (Boothroyd, 2002). One way a hearing-impaired person could potentially overcome the adverse affects of masking from background noises would be to increase the SNR, by turning up the signal – in this case, the TV volume. However, normal-hearing listeners in the same room may be made uncomfortable by the increased sound level. Moreover, turning up the TV volume would offer no remedy for self-masking.

In addition to background noise, which may or may not be present in a listening room, almost all domestic listening rooms have acoustic conditions that disfavor high-frequency sound transmission compared with lower-frequency sound transmission. Physically soft objects, like carpets, padded upholstery, drapes and the like, absorb high-frequency sound energy much more readily than low-frequency sound energy. Also, the air in a room itself is an absorber of high-frequency sound energy (Doelle, 1972). The acoustic absorption factors mentioned above can attenuate the high-frequency parts of speech sounds, potentially making them inaudible, and/or more susceptible to masking, especially for listeners with high-frequency hearing losses. Again, a potential solution would be to turn up the TV sound until all speech sounds were above the masked hearing threshold for the hearing-impaired listener. But again, doing so could require that the TV sound become very loud and likely annoying to anyone in the same room with normal hearing. The SSLL addresses the above issues by turning up the sound substantially for the hearing-impaired listener, and by providing high-frequency emphasis that can help make masked sounds audible, without increasing the overall sound level in the room appreciably.

To illustrate, the author performed the following informal experiment using himself as a single, hearing-impaired subject. The subject has a severe hearing loss, but for the purpose of this experiment he wore hearing aids which were set to leave him with a mild-to-moderate hearing-threshold deficit (see Table 1). The experiment was carried out in the living room of the author's home – 15 ft long by 12 ft wide by 8 ft from floor to ceiling, with the end of the room opposite the smaller wall open to the dining area, which has a cathedral ceiling. The TV set, at the center of the small wall, was turned on and tuned to "The Antiques Roadshow." The subject's wife, who has normal hearing thresholds, adjusted the TV sound to a comfortable level. Although the subject, an experienced sound engineer, observed that the sound from the TV set was generally audible, he observed that many words were difficult to understand. He observed that he would have preferred the sound level to be higher, especially at high frequencies, for comfortable listening. He then turned on the SSLL and added just enough sound level at his listening position, with the tone control set for a degree of high-frequency emphasis, the settings having been chosen to enable him to hear comfortably.

Table 1 – Subject LJR's aided HTLs

Frequency (Hz)	250	500	1000	2000	4000	6000	8000
HTLs (dB)	30	65	65	70	90	85	85
Insertion gain (dB)*	14	19	22	33	32	27	27
Residual Hearing loss (dB)	16	46	43	37	58	58	58

*Insertion gain was measured with a broadband composite signal at 50 dB SPL rms. This measure can be taken to be equivalent to functional gain, as the compression thresholds of the hearing aid were above the level of the test signal in every band (Dillon & Murray, 1987).

Sound-level readings were taken at the subject's listening position (with the loudspeaker 8.5 inches directly overhead) and at the subject's wife's listening position (4 ft away from nose to nose), with and without the SSLL turned on. Three successive sets of sound-level readings were taken, each one starting with the normal-hearing listener's adjusting the TV sound to a "normal" level, followed by the hearing-impaired subject's adding the desired enhancement using the SSLL.

For the three repeated measures, the mean A-weighted sound-pressure level (SPL) from the TV set, before the SSLL sound was added, was 57.5 dB at both listening positions (see Fig. 3). After the introduction of sound from the SSLL, the mean A-weighted SPL at the hearing-impaired subject's position increased by 7.3 dB, to 64.8 dB. The subject observed a substantial increase in loudness and listening comfort. At the normal-hearing listener's position 4 ft away, the mean A-weighted SPL increased by only 1.5 dB (from

57.5 to 59 dB, which corresponds to only a very small difference in loudness for the normal-hearing listener. (Stevens, 1955).

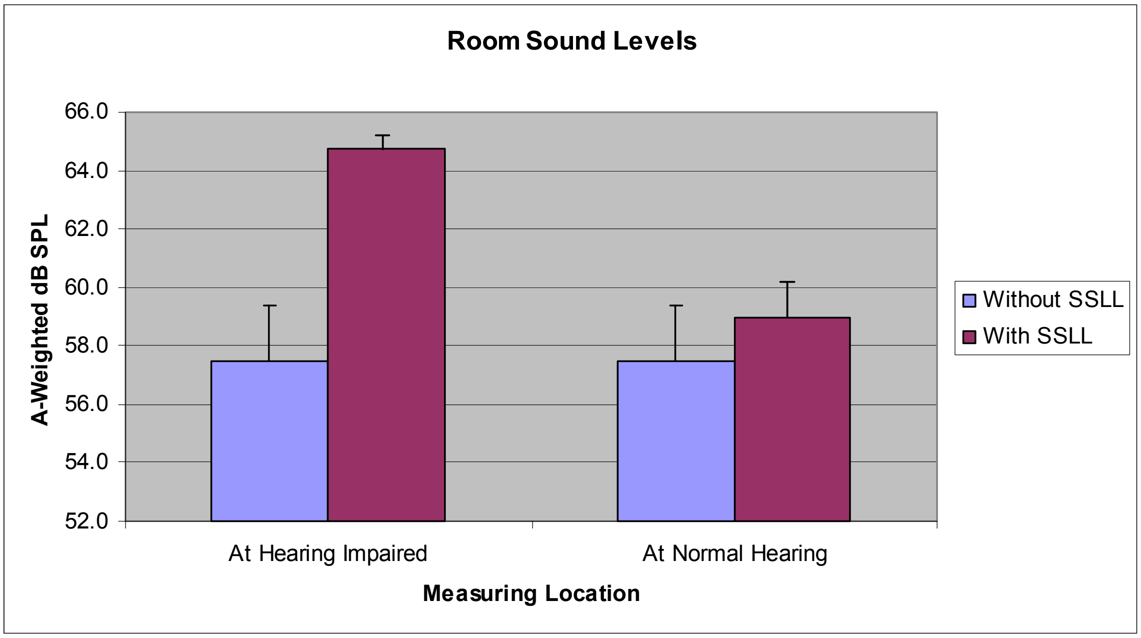

Fig. 3 – A-weighted SPL at listening positions for one hearing-impaired and one normal listener, before and after the addition of enhancement from the SSLL. Columns show means of three measures at each location, with standard deviations shown as error bars.

In brief, the introduction of sound from the SSLL increased the SPL (and loudness) substantially at the SSLL listening position, while providing the desired high-frequency emphasis, all while not appreciably increasing the SPL (and loudness) elsewhere in the room. This informal experiment was for only one hearing-impaired subject (the author), but its results can be considered typical for a person having a mild-to-moderate hearing loss.

OVERCOMING THE ADVERSE EFFECTS OF REVERBERATION

An especially important acoustical property that causes masking in typical TV-listening rooms is reverberation. Reverberation is a collection of sounds that reflect from the surfaces in a room, persisting beyond the time that a sound is propagated. In other words, after a sound is propagated, it literally bounces around the room for a time, being reflected by the floor, the ceiling, the walls, and other surfaces. The reverberated sound eventually dies out because of acoustic absorption (the sound energy is dissipated as heat) (Haughton, 2002). All rooms are prone to reverberation to some extent, and reverberation can be severely detrimental to speech understanding, especially for those with hearing impairments (Boothroyd, 2004).

Reverberated sounds tend to mask the details of speech signals. The literature in psychoacoustics (e.g., Moore, 1998) tells us that the effectiveness of a masker depends on its similarity to and synchronicity with the sound being masked. Taken to the extreme, the most effective masker of a sound is the sound itself! But effective maskers can occur not only in synchronism with the sounds they mask; they can be non-simultaneous, occurring forward or backward in time, as well (Moore, 1998). A reflected (reverberated) sound is a slightly delayed version of the initial sound itself, and therefore can be a very effective backward masker. Additionally, the continuing reverberation of a given sound serves as a simultaneous masker to subsequent sounds.

The relative amount of reverberation in a room, compared with the direct sound from a TV set, does not depend on how loud one turns up the sound level of the TV set. It depends only on the acoustic characteristics of the room and on the distance from the TV set at which one listens. In fact, once a listener is a beyond a "critical distance" from the sound source, the power of the reverberant sound actually exceeds that of the direct sound (e.g., Walker, Dillon, & Byrne, 1984). When a viewer turns up the sound from a television set, the reverberation of that sound becomes louder, right along with the direct sound from the loudspeaker of the TV set. And so even if higher sound levels would not disturb others in the listening room, the adverse effects of reverberation can make it difficult for a hearing-impaired listener to understand the sound from a TV, no matter how loud one turns up the sound level.

The solution to the problem of reverberation is similar to that for other masking problems: one must increase the SNR. Considering reverberation as a masking noise, increasing the SNR normally means getting close enough to the source of the direct sound (i.e., well within the critical distance), so that the direct sound is predominant in what reaches the listener's ear. But instead of the listener's having to move very close to the TV set, the SSLL can bring the direct sound very close to the listener – thereby reducing the adverse effects of reverberation in a convenient fashion.

To quantify the reduction in reverberation at the listener's position afforded by the SSLL, the author measured the reverberation time ("RT-60") at the listener's position in his living room, with and without the SSLL turned on. The RT-60 is a conventional measure of reverberation, defined as the time it takes for a sound to decay by 60 dB, once the propagated (direct) sound stops suddenly. Pictured in Fig. 4 is an oscilloscope trace measured in the author's living room, the same room that was used for the sound-level experiment described earlier. (See Fig. 5 for a photograph of the actual setup in the room.) The oscilloscope trace was captured as follows: The measuring microphone that was used in the earlier experiment was placed near the TV set, at a distance of 9 ft from the loudspeaker of the SSLL. The loudspeaker was positioned between the two TV-viewing chairs in the room. In other words, for the requirements of the experiment, the listening position and the sound-source position were reversed: the "listening position" (the position of the measuring microphone) became the location of the TV set, while the SSLL loudspeaker, near the viewers' chairs, served as the sound source. A pink noise signal (equal energy per octave) from the NTL Minirator signal generator was introduced into the transmitter of the SSLL system, and the measuring microphone was connected, via a Behringer UB802 mixer, to a Tektronix TDS 1002 storage oscilloscope. The output of the SSLL was then adjusted for its maximum possible level before overload, just under the level for which a clipped waveform was observed on the oscilloscope. Then the gain and display settings of the measuring system were adjusted such that frequent peaks of the pink-noise signal from the microphone just reached the edges of the display screen, at values of plus- and minus-8 volts. The oscilloscope was then set to its single-trace, storage mode. The plug connecting the pink-noise source with the preamplifier was then disconnected, and then momentarily reconnected, creating the trace seen in Fig. 4. The measurement cursors of the oscilloscope were then placed at the plus- and minus-240 millivolt positions, which correspond to the 30-dB-down points re plus- and minus-8 volts, respectively. The arrows on the graph indicate where the trace dips within the +/-240 range.

Note that the horizontal (time scale) positions of the upper and lower arrows differ, because the waveform was not symmetrical about the 0-volt axis. Therefore, the overall 30-dB-down point, or RT-30, was taken as the average of the 30-dB-down points for the positive and negative phases of the waveform. The ambient noise in the room prevented measuring the actual 60-dB-down point, or RT-60. But because the decay of reverberation is logarithmic, as are decibels, a good estimate of the RT-60 is simply twice the RT-30 (e.g., Doelle, 1972).

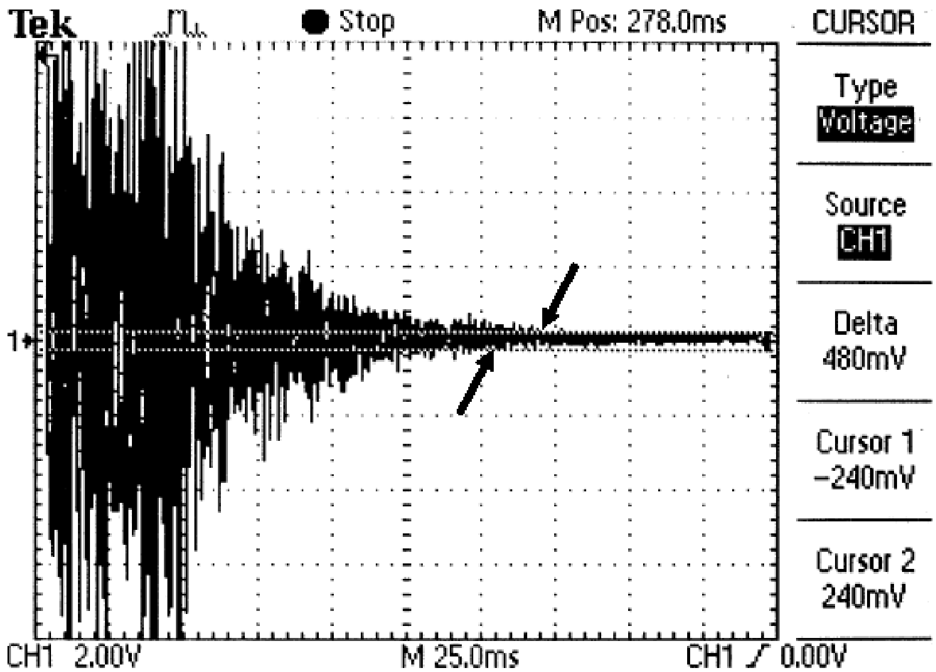

Fig. 4 – Oscilloscope trace: SSLL loudspeaker between the two listening chairs; measuring mic near the TV set, 9 ft away (as in Fig. 5). RT-60 = 2 x RT-30 = 2(105 + 120)/2 = 225 milliseconds.

Fig. 5 – Setup for the measurement shown in Fig. 4.

For the conditions of the trace of Fig. 4 (with a 9-ft distance of sound source to measuring microphone), the observed RT-60 was 225 milliseconds. In contrast, the trace of Fig. 6 was taken with the measuring microphone just above the right ear of the subject (LJR) while he was seated at his usual listening position (in the blue chair at front-left in Fig. 5), with the SSLL loudspeaker in its normal position, 8.5 inches above the top of his head, 12 inches from the ear. Once again, the gain of the measuring system had been adjusted for frequent peaks of the trace to be just reaching the edges of the display of the oscilloscope. In this case the observed RT-60 was 92.5 milliseconds – a substantial reduction in reverberation time.

Further observation of the two traces reveals perhaps an even more important difference. For the purposes of this experiment, the horizontal position of the display was adjusted so that, for both traces, the time where the direct signal stops is the 50-millisecond point, represented as two major horizontal scale divisions from the left edge of the graph frame. The beginning of the reverberation tail in Fig. 4 (9-ft distance) appears to start, on average, at about the plus- and minus-5.5 volt level, which is about 3.3 dB down from the peaks of the direct signal. In other words, this measurement was taken near the critical distance point, where the peaks of the direct and reverberated sound components sum to +3 dB, and consequently where the SNR (direct sound versus reverberation) is close to 0 dB. In the trace of Fig. 6, however, the reverberation tail begins at about plus- and minus-1 volt, or -18 dB with respect to the peaks of the direct sound, a much more favorable signal-to-reverberation ratio.

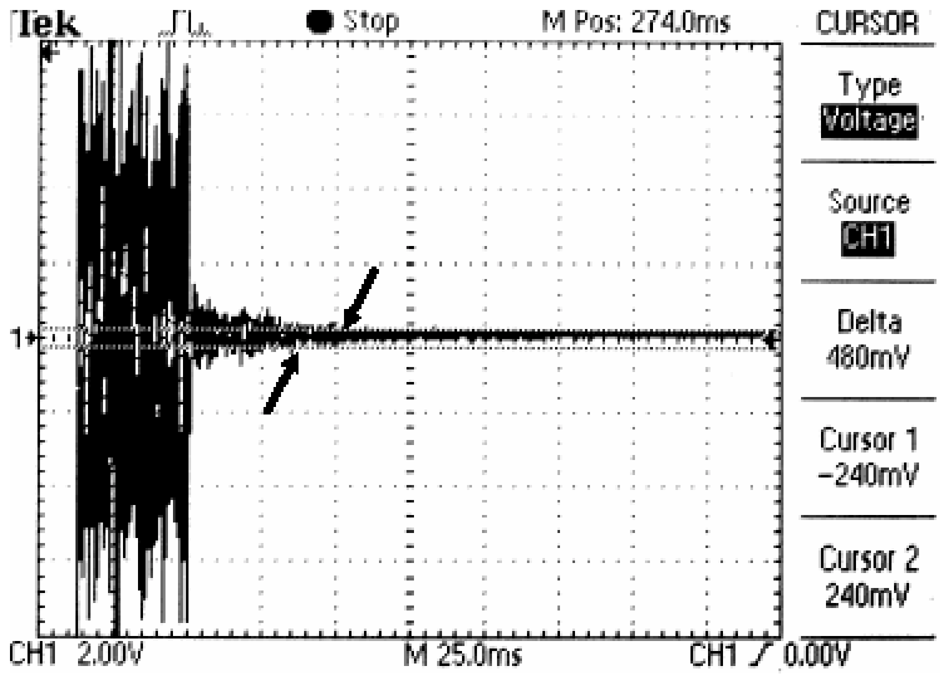

Fig. 6 – Oscilloscope trace: SSLL loudspeaker 8.5 inches above the top of the listener's head; measuring mic at his right ear, 1 ft from the SSLL loudspeaker. RT-60 = 2 x RT-30 = 2(37.5 + 55)/2 = 92.5 milliseconds.

As a control condition, an additional measurement was made with the microphone just one inch from the loudspeaker, to demonstrate that the previous samples were truly capturing room reverberation and not just the artifactual ringing of the loudspeaker or microphone. Fig. 7 shows a trace taken with the measuring microphone just 1 inch from the SSLL loudspeaker. The measured RT-60 was 16 milliseconds, only a fraction of the RT-60s for the other conditions. The reader should be aware that this and the other oscilloscope measurements represent but a single sample. The following experiment, however, used multiple samples, giving the reader an idea of the degree the variability of these measurements.

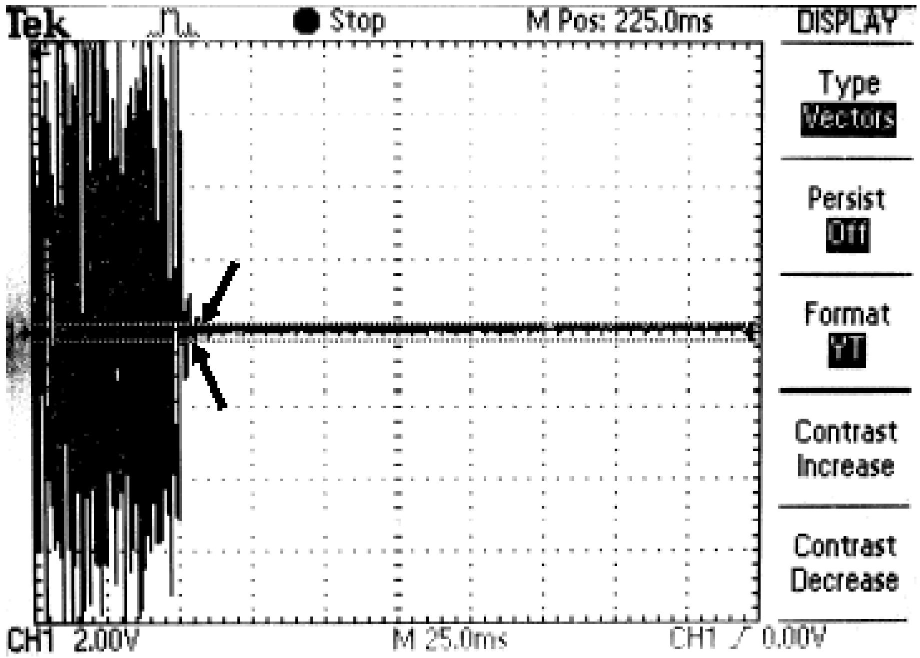

Fig 6 – Oscilloscope trace: Control condition: loudspeaker 1 inch from mic. RT-60 = 2(7 + 9)/2 = 16 milliseconds.

How close is "close enough"?

There have been other proposed TV listening systems which place a loudspeaker to the side of or slightly behind the listener at a distance of 1 m (3.3 ft) from one of the listener's ear (KUref?). But is a 1-m loudspeaker distance as effective as the 1-ft distance recommended for the SSLL in reducing the reverberation of TV sound? A third experiment addressed this question.

For all of the following measurements, the measuring microphone was placed just above the pinna of the subject's (LJR's) right ear, in the same listening room as was used for the previous measurements. Three sample measurements were made for each of three locations of the SSLL loudspeaker: 1 ft, overhead; 1 m, 45 degrees behind; and 1 ft, 45 degrees behind. For each location, the gain of the measurement system was initially adjusted for frequent peaks of plus- and minus- 8 V on the oscilloscope, and a single reverberation trace was then recorded by unplugging and momentarily reattaching the signal source, as for the previous experiments. Table 2 shows the results.

Table 2 – RT-60 measurements for three loudspeaker locations. For each, the measurement microphone was just over the ear of the listener.

Location	**o'head, 1 ft**	**45 deg, 1 m**	**45 deg, 1 ft**
Sample 1 (ms)	110.0	155.0	90.0
Sample 2 (ms)	95.0	172.5	88.0
Sample 3 (ms)	92.5	170.0	87.6
Mean (ms)	**99.2**	**165.8**	**88.5**
***t*-test** significance level versus o'head, 1 ft		**0.01**	0.08

Distance was the important factor for reduction of reverberation at the ear. At a 1-ft distance, there was no significant difference in reverberation time, whether the loudspeaker was overhead or 45-degrees to the side and behind. But the 1-m distance had a significantly longer reverberation time compared to the 1-ft overhead location.

ADVANTAGES OF THE OVERHEAD POSITION OF THE LOUDSPEAKER

The overhead position of the loudspeaker may offer several important advantages compared with any other location. Most of these stem from the fact that, when the loudspeaker is overhead, both ears get essentially the same signal. Of course, that would also be the case if the loudspeaker were directly in front of or directly behind the listener.

But for TV viewing, the directly-in front location would have an obvious drawback (blocking the view to the TV screen), and having the loudspeaker directly behind could make it impossible for the listener to rest one's head on the back of one's easy chair. But, perhaps most importantly, when the sound source is overhead it presents no lateral directional cues to the listener. In other words, one can turn one's head from side-to-side with no apparent change in the relative amplitude or spectrum of the sound reaching the two ears. The only lateral directional information, therefore, comes from TV loudspeaker, not the SSLL loudspeaker. Consequently, as long as one is not nodding one's head up and down (creating elevation cues), the perception of the sound from the SSLL loudspeaker tends to disappear as a separate sound source, blending in very well with the sound from the TV loudspeaker.

This blending of the SSLL sound with the sound from the TV loudspeaker happens not only because the sound reaches both ears equally – thereby presenting no conflicting lateral directional cues – but also for the following reason. Sound arriving from overhead is virtually indistinguishable, spectrally, from diffuse sound (sound that arrives equally from all directions – or "coming from everywhere at once"). To illustrate, the bold, solid curve of Fig. 8 shows the frequency response of an overhead loudspeaker at the eardrum of a KEMAR manikin in an anechoic chamber. This eardrum response curve was taken with respect to the response of a reference sound-field microphone located just above the test ear (Revit, 1987). The dashed line shows the response at the KEMAR eardrum using a diffuse sound field signal (Killion & Monser, 1980), again with an over-the-ear reference. The maximum differences between the two frequency responses are only 2 to 4 dB through 10 kHz.

The diffuse-field-like quality of reverberation comes into play here (Doelle, 1972). Spectrally at least, there is little for the listener to distinguish between the sound delivered from the overhead position and that of the reverberant field of the sound from the TV loudspeaker (aside from the chosen SSLL frequency-range emphasis), and so the two sound components in the room blend together well. However, as has been shown earlier, the sound from the close-by SSLL loudspeaker does not create additional reverberation at the listening position; if anything, the favorable SNR of the SSLL sound would tend to mask (or perceptually "replace") the actual reverberation of the TV sound.

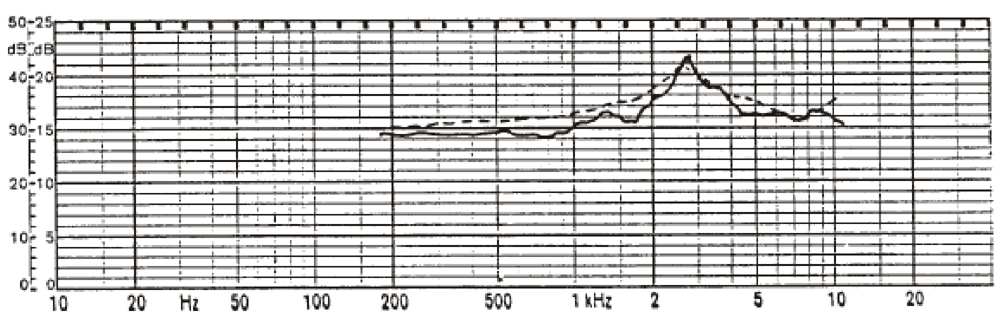

Fig. 8 – The anechoic chamber response to an overhead loudspeaker (bold curve) at the eardrum of a KEMAR manikin with respect to an over-the-ear reference microphone, compared with the diffuse-field response (dashed curve) at the eardrum of the manikin. (Use the far left dB scale, in terms of relative dB.)

CONCLUSIONS

The SSLL provides an aid to TV listening by targeting the affects of self-masking, background noise, high-frequency absorption, and reverberation. It does this by substantially increasing the direct sound level at the listening position, with adjustable low- to high-frequency emphasis, but without a substantial change in the overall sound level in the listening room. The overhead position of the loudspeaker is optimal for blending the assistive sound with the diffuse sound from the TV speaker.

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