The Acoustic and Intelligibility Performance of Assisted Listening Systems

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ABSTRACT

Approximately 10 - 14% of the general population of the UK & Northern Europe suffer from a noticeable degree of hearing loss and would benefit from some form of hearing assistance or deaf aid. However, many assistive listening systems (ALS) do not provide the benefit that they should, as they are often let down by their poor acoustic performance. The paper investigates the acoustic and speech intelligibility requirements for ALS performance and examines a number of microphone pick-up scenarios in terms of their potential intelligibility and sound quality performance. The results of testing carried out in a number of rooms and venues are presented, mainly in terms of the resultant Speech Transmission Index (STI) measurements. The paper concludes by providing a number of recommendations, ALS performance criteria and 'rules of thumb' for successful microphone placement and testing.

1. INTRODUCTION

The Commission on Hearing Loss report, published in July 2014[1], is predicting that almost 20% of the UK population could be living with hearing loss by 2031. However, many assistive listening systems do not provide the benefit that they could or that hard of hearing listeners require. This applies to all types of system, regardless of the underlying technology (eg Infrared, Audio Frequency Induction Loop or Wireless). One of the primary causes of this is poor pick up of the desired sound. This in turn is caused by poor microphone placement and poor understanding of the hard of hearing listener's requirements. In many theatres and concert halls, it is common to take the ALS feed from the 'Show Relay' microphone which is rarely positioned for optimal intelligibility. Its primary purpose is to provide a 'cue' feed to the dressing rooms and back-of-house areas as opposed to a high quality signal with optimal intelligibility for the hard of hearing. Even where microphones are well positioned, the associated signal processing is often inadequate for an ALS system [2]. However, little guidance is provided within the relevant standards or codes of practice to assist system designers, installers or equipment manufacturers. An exception to this is BS 7594 the code of practice for audio frequency induction-loop systems[3]. Often, temporary systems are required and in many cases a sound system is not available to provide a 'clean feed' requiring the use of a remote pick-up microphone. A number of microphone configurations have been tested and the results of both objective measurements and subjective assessments are presented. The work highlighted the need for an accurate 'talker' simulator loudspeaker and this has been extensively researched and will be reported at a future time. A number of microphone types, formats and positioning have been investigated and it is hoped that the research and findings reported may be used to improve existing standards and could eventually form part of the basis of a guide or code of practice for the installation of Assistive Listening Systems in Auditoria & Meeting Rooms.

2. BACKGROUND

Even though digital hearing aids can provide significant assistance to hearing impaired users, in many environments, they can not provide sufficient intelligibility enhancement for the user to fully participate in an event. A large auditorium or noisy classroom would be two such situations. (It is little understood than in many situations the effective range of a hearing aid is only around 2-3m). The only way to improve the clarity of the wanted signal is to reduce the ‘effective distance’ between the listener and source of sound and / or also reduce the amount of received noise. It is the aim of an assistive listening system to provide the means to do this. Put more precisely, it is the objective of an ALS to improve the direct to reverberant sound ratio and acoustic signal to noise ratio. This is achieved by positioning a suitable pick-up microphone or microphones close to the wanted source of sound and then feeding this, hopefully high intelligibility signal, directly to the listener.

Assistive Listening Systems essentially consist of three basic elements : (1) an acoustic pick up of the desired speech (or other sound) information (2) a means of transmitting this to the listener (3) a suitable receiver & earphone or hearing aid. Whilst this represents a fairly simple signal transmission chain, signal degradation can readily occur at any of the associated interfaces or transmission paths. A number of transmission techniques are currently in use including Audio Frequency Induction Loops, (AFILS) dedicated Radio Frequency FM wireless broadcast (RF) or Infra Red (IR) transmission. As all National Health scheme hearing aids in the UK incorporate a means for picking up an audio induction signal (‘T’ coil), this is by far the most commonly used technique within the UK, although Infra Red systems are also common in theatres. In Europe AFILS usage varies greatly as indeed does the use of any of the systems. Recently, AFILS are being adopted in the USA in a rapidly growing number of venues and the number of Assistive listening systems is about to soar. Each method has its advantages and disadvantages dependent on the technology involved, but there are many common elements and associated problems.

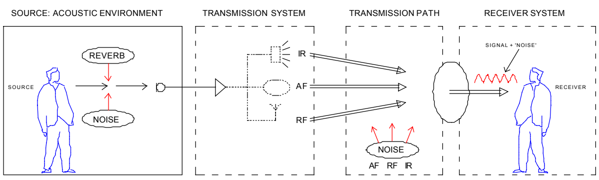


Figure 1: ALS Signal Transmission Path

To the author it has become clear that one of the biggest issues relates to the format and placement of the pickup microphone in order to provide a suitably intelligible signal. There is very little information or guidance available for installers or even manufacturers relating to microphone placement and the intelligibility target that an Assistive Listening System should achieve. The author has previously published information on the subject [2,4] and suggested that the Speech Transmission Index (STI) might be a suitable measure for assessing the potential intelligibility of AFILS and ALS. Mapp concluded that the target STI should be > 0.7 for such systems. BS7594, the UK Code of Practice for Audio Frequency Induction Loop Systems, currently recommends that the intelligibility of the system should achieve at least 0.65 STI [3]. However, this guidance dates back to 1993 and was originally conceived in terms of the now obsolete RaSTI method. Backke [5] reported that the STI should be at least 0.84, which, as the later sections of this paper will show, is an onerous and difficult target to achieve. The objective of this paper is to provide some further insight as to what is required to meet these targets and how practical it is to reliably measure the target intelligibility performance of AFILS and ALS systems. The reported research primarily relates to reverberation and room acoustic based detriments though some comments on background noise and background noise pick up performance are also included. The frequency response of the ALS system is also of critical importance. Interestingly, an extended response is often not required nor indeed may be desirable. This has significant impact on the resultant STI measurement / calculation.

3. TALKER DIRECTIVITY & SPEECH RADIATION

In order to understand how to optimise the location and placement of microphones for Assistive Listening Systems, it is necessary to understand how speech is radiated from a human talker. Again there is little readily available information on this – particularly in a format useful to potential systems designers. In view of this, the author has carried out a number of studies and published some background and guidance information [6,7].

In the horizontal plane, the acoustic output of a human talker covers quite a wide angle, with a 3dB down point of approximately 70 to 80 degrees occurring at 2 kHz. Interestingly, in the vertical plane, more sound radiates in directions just above and just below the centre of the mouth than does directly on axis. At 2 kHz the acoustic radiation is 3 dB down at approximately 40-45 degrees above the axis of the mouth.

A simple way to view this information is presented in figure 2. Here the solid (blue) area represents the useful speech radiation – taken as the -3dB points. (see Mapp[7] for further information).

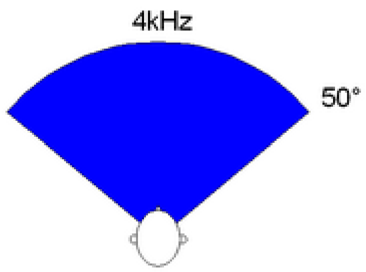


Figure 2: Solid angle of 'useful speech radiation' at 4 kHz for the horizontal plane.

Clearly therefore, when siting an ALS microphone, it is essential to ensure that it is generally located within the solid angles indicated above. However, the effect of distance to the source must also be considered and in the author's experience it can often be better to locate a microphone just outside these angles if this means it can be closer to the source.

4. AUDITORIA AND LARGE ROOMS

Figure 3 shows how the intelligibility (STI) for a talker decreases with distance in a medium size (500-600 seat) recital hall measured when with a number of different microphone types. The hall had a mid frequency reverberation time of approximately 2 seconds and so is typical of many concert halls and similar venues. The graph shows the effect of increasing microphone directivity on the measured STI as a function of distance from the 'talker' test loudspeaker.

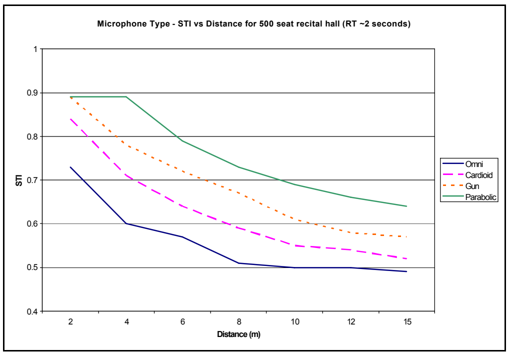


Figure 3 : Effect of microphone type (directivity) on resultant ALS intelligibility.

As can be seen from the graph, in order to meet a target of 0.7 STI an Omni directional microphone needs to be located within 2m of the talker. This increases to 4m for the cardioid microphone and approximately 6m for a 450mm 'Gun' microphone. The power of microphone directivity can therefore readily be seen, though of course, the more directional the microphone, the less the area of pickup becomes.

In many theatres and concert halls, the, provision for the Assistive Listening System (ALS) is merely a signal feed taken from the 'Show Relay System' as shown in figure 4. The primary purpose of the show relay microphone(s) is to provide a basic relay of the sound on the stage to the dressing rooms and back of house production offices / areas, to provide cues and to allow staff to track the progress of an event. The fact that this can also provide a feed to the ALS is seen as a bonus and little or no thought generally goes into optimising its performance for hard of hearing users.

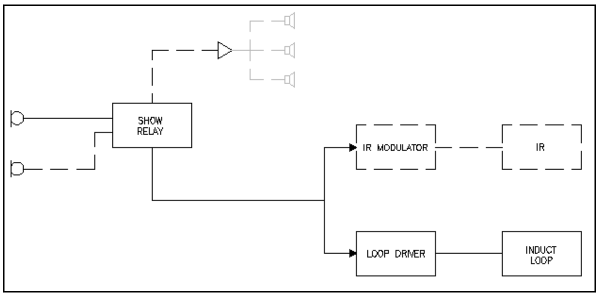


Figure 4: Block diagram of basic theatre / auditorium show relay system with an output for an ALS. (Both AFILS and IR systems are shown – but generally only one system would be installed).

The positioning of the show relay microphone (it is generally termed this rather than the 'ALS' microphone) is normally not given too much thought and certainly no acoustic criteria or design is applied. Many theatres for example provide a simple cardioid microphone on the front of the balcony or on a lighting bridge. This approach, in the author's experience, can lead to resultant ALS potential intelligibility values barely exceeding 0.50 to 0.60 STI. More recent installations generally employ a more directional gun microphone. Again this is often positioned at a central balcony or lighting bridge position. Alternatively, a pair of gun microphones may be located either on side balconies or other auditorium side positions.

Where a pair of microphones is provided, these are generally directly summed rather than fed to an auto-mixer which can either gain share or level switch. Whereas the potential intelligibility can theoretically decrease with two open microphones as compared to one, this is often more than compensated for by the closer positioning of the microphones to the source and better off axis pick up. Equally, when the talker turns away, a centrally located microphone will not adequately pick up the higher frequencies. Figure 5 shows these auditorium microphone approaches, with the 1 and 4 kHz 'useful' talker speech angles included. As the figure clearly shows, two side microphones are required in order to pick up the higher frequency components of speech as a talker turns to the side.

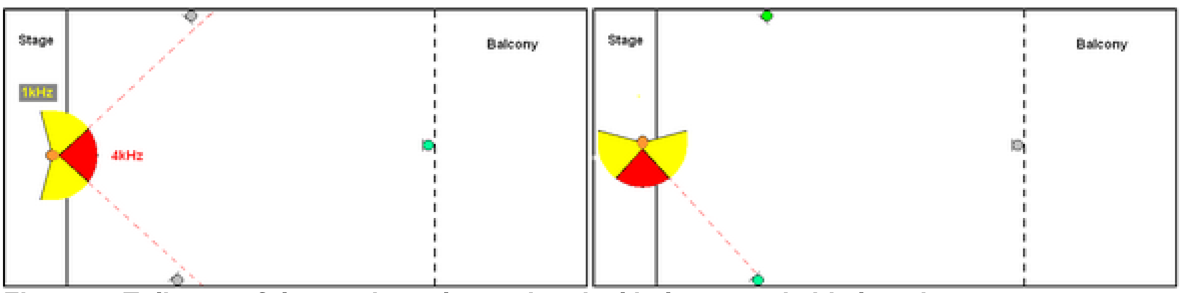


Figure 5: Talker useful speech angles and typical balcony and side locations.

In the author's experience, once a talker turns by more than approximately 45 degrees, speech intelligibility and voice quality are noticeably reduced, with the high frequencies becoming attenuated / muffled. As can be seen, this subjective view ties in well with the 50 degree horizontal radiation shown in figure 2.

The limiting useful angle of vertical sound radiation is about 40 degrees for a human talker. Care therefore needs to be taken to ensure that microphones are not placed above this angle. A review of the general geometry of most theatres and similar venues suggests that this should not be a problem, except where microphones are placed on lighting bridges or upper balconies. The polar pattern of the pick-up microphone must also be taken into account when considering the resultant sound reception. Whilst many microphones may attenuate sound from the sides and rear, this can lead to significant colouration and potential listener fatigue.

An important, yet rarely considered factor to take into account when deciding potential microphone locations, is the signal delay that will be introduced. For example locating the ALS microphone centrally on a balcony may mean this is some 20 metres away from the stage, resulting in a transit time of approximately 58ms. This means that the signal fed to the assistive listening system will be 58ms behind what is happening on stage or from the front of house sound system. This is a significant delay and can lead to an echo being perceived or visual mis-synchronisation of the sound. Whilst this is not a problem for the show relay, it can be disturbing to audience members using the ALS - particularly if the ALS signal is only fed to one ear. By locating the microphones closer to the stage, this problem can be minimised, though in large venues there will inevitably be some timing discrepancy. (In systems where two microphones are employed without auto mixing, a similar delay effect can also be noticed).

It should be noted that the dynamic range of hard of hearing listeners is also generally reduced in comparison to the normal listening population. Optimising the signal level and keeping this relatively constant is therefore most important and the use of an efficient AGC circuit within the ALS system is essential. Although many ALS systems incorporate a signal compressor, few systems incorporate an AGC. However, most proprietary european induction loop amplifiers (drivers) do incorporate AGC as the necessity for this has long been recognised and is a requirement of BS 7594 and IEC 60118. An AGC range of at least 20 dB is required but in the author's experience, for many applications, such as musical theatre, a capture range of 30 dB is required [4]. The effect of an AGC circuit on the measured STI is discussed in [6].

5. MEETING ROOMS, CONFERENCE ROOMS & CLASSROOMS

It might be logical to imagine that smaller rooms would pose fewer acoustic issues from an ALS point of view. However, this is not necessarily the case, as the early sound field is usually very much more complex in such rooms [8].

A survey, by the author, of 15 meeting rooms or rooms where an Assistive Listening System might be installed, showed that the majority were highly reflective spaces and several suffered from external noise break-in. Although in many cases the reverberation time of the spaces was reasonable, the reflective nature of the wall surfaces produced a poor acoustic environment. The meeting rooms can be divided into three categories, Small – e.g. for 2-6 people occupancy, Medium for occupancy of up to around 12-15 people and Large for occupancy of 15-30 people. Rooms of greater capacity were also measured but have been classed as lecture theatres, halls or auditoria. On average the small and medium sized rooms all had a mid frequency reverberation time of around 0.7 seconds. All the rooms had thin pile carpet and padded seats but few had acoustically absorbing ceilings. A typical reverberation time characteristic is shown in figure 6 below.

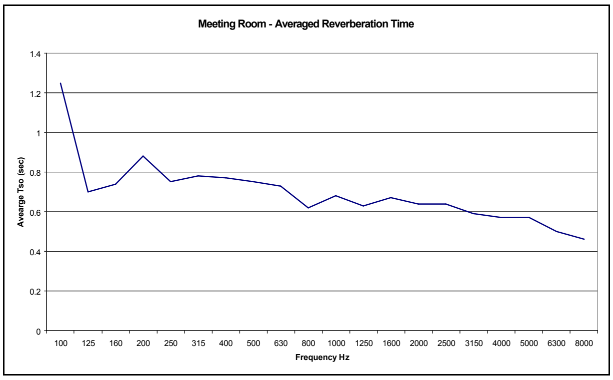


Figure 6: Reverberation Time (T20) for typical meeting room (6.7 x 3.3 x 3m) with hard, sound reflecting walls.

Modern buildings often employ lightweight wall and partition constructions and so tend to exhibit lower bass reverberation time characteristics due to the associated diaphragmatic absorption of sound.

The relatively small sizes of the rooms mean that strong model behaviour and excitation can occur, often leading to significant colouration of the sound. Figure 7 for example shows the measured frequency response of a 4 inch calibrated 'flat response' loudspeaker when measured in a medium size meeting room (7 x 9 x 2.5m). Clear modal response peaks can be seen to be occurring at approximately 180, 230 and 310 Hz.

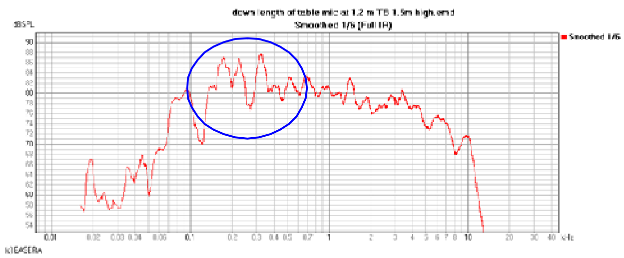
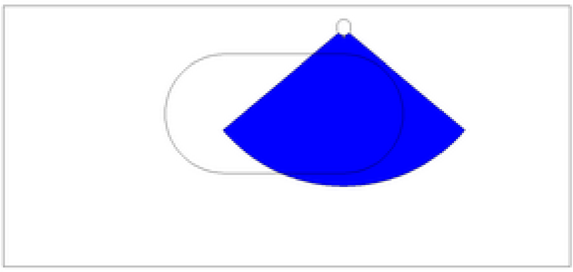
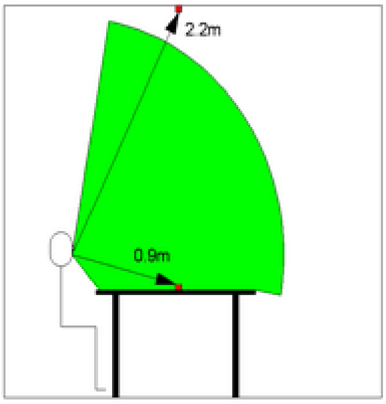


Figure 7: Effect of room acoustics on a calibrated 'flat response' talker test loudspeaker

In many meeting rooms, the reflected sound far outweighs the level of the direct sound. This has significant implications for AFILS and Assisted Listening systems. Such an acoustic environment also tends to increase the background noise level. A detailed study of microphone types and positioning was carried out in the room noted in figures 6. The results of this study are described below.

Fifteen different combinations of microphone type and location were studied. A single talker location was employed, which was approximately ¾ down the length of the 3m meeting room table. A range of different microphone types and locations was set up and measurements of the corresponding STI and other metrics were made. Figure 8 shows the talker location in a plan view of the room with a talker speech radiation pattern at 4kHz superimposed.





Figures 8 & 9: Plan & sectional views of meeting room & table showing typical talker speech radiation at (Room dimensions are 6.3m L x 3.3m W x 3.0m H). The locations of a ceiling mounted and a table top boundary microphone are also shown.

A range of microphone types and locations were measured, including table top mounting as well as ceiling and suspended microphones. Several ‘talker’ test loudspeakers were also employed. The combination of test loudspeakers, microphones and locations provided a range of very useful insights into not only ALS requirements but also the complexity of the soundfields in small, reflective environments.

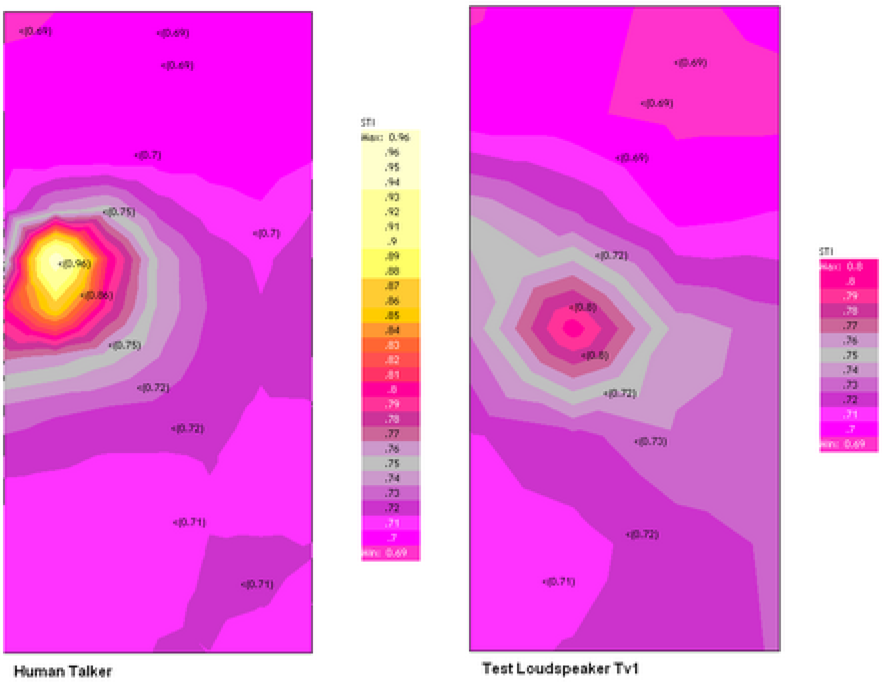


Figure 10: Predicted STI plot for Human talker & Test ‘Talker’ loudspeaker

Figure 10 (a) shows an STI plot mapped onto a listening plane within the room for a seated talker and seated listeners. (listening height = 1.2m). As previously described, the talker was located approximately ¾ the way down the table on the left hand side.

As would be expected, the intelligibility (STI) decreases rapidly as the distance away from the talker increases, before plateauing out and reaching a consistent level of approximately 0.7 STI. Interestingly, none of the test talker loudspeakers managed to mimic this exact pattern. Figure 10(b) for example shows the corresponding plot for a purpose made 2.5 inch test loudspeaker with head-like dimensions. Interestingly, once into the reverberant field, the test loudspeaker was able to produce very similar STI values to the human talker. (The critical distance for the room was calculated to be just under 1m).

From figure 10, it can be seen that all locations in the room should meet the suggested criterion of 0.70 STI. (A factor not realised until after the room had been measured and computer modelled).

A problem with meeting room systems, is that on the one hand it needs to be realised that the closer the microphone is to the talker, the better the intelligibility but on the other hand, where eight or more people are present around the table, the number of required microphones becomes both expensive and difficult to manage. (This is particularly the case for temporary systems). An example of this is shown in Figure 11 where a table top boundary microphone solution is being employed. One of the objectives of the measurements was to see if, in a smaller meeting room, with a 3m conference table, whether a single, centrally located microphone could work satisfactorily. In many rooms with moveable or non permanent furniture, table microphones are not always practical, so a further test was therefore conducted to see if centrally mounting a ceiling boundary microphone could also work.

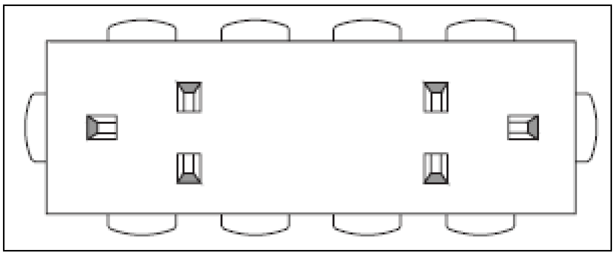


Figure 11 : Meeting Room with table mounted Boundary Microphones (one mic shared between two participants)

Initially it was thought that the room (considered by most of its users to be acoustically poor) and range microphones tested would produce a wide variation in measured values and subjective ratings. It was therefore surprising to find that the STI only varied by 0.1 (0.70 to 0.80) for the 15 microphone combinations tested. (A later test with a hypercardioid microphone increased this range to 0.15 as it measured 0.85 STI). Later consideration of the computer modelling study showed this small range of STI results to be not so surprising given that the minimum predicted STI for the talker in the room is 0.7. This latter aspect also affected the range of subjective ratings that the various microphones elicited. The natural speech recordings, made via the microphones, were evaluated at a later date, monaurally (diotic), over headphones. A mean opinion test rather than a word score test was employed. Figure 12 compares the measured STI values for the 15 microphone combinations with their subjective rating.

In addition to Speech Transmission index measurements, corresponding C50 measurements were also made and are shown in figure 13.

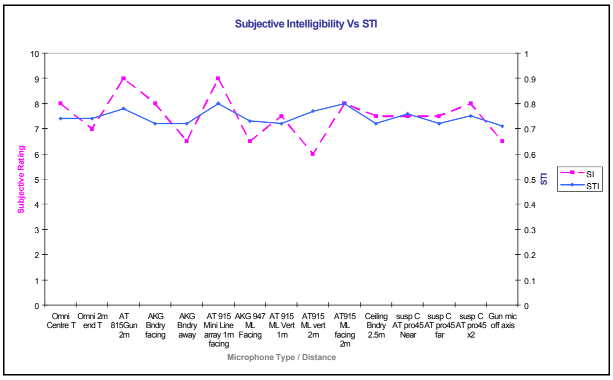


Figure 12: Comparison of measured STI & subjective intelligibility scores

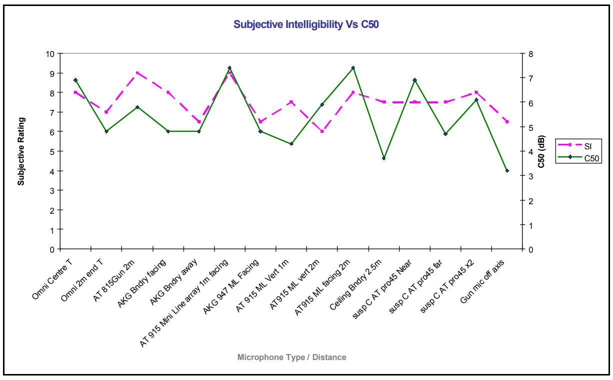


Figure 13: Comparison of measured averaged C50 & subjective intelligibility scores

As can be seen from figure 13, a much clearer variation in the measured parameter is indicated by the C50 scores as compared to the STI. From the two figures above, it can bee seen that there is not always particularly good agreement between subjective opinion and the objective measures. The microphone recordings were also subjectively rated for overall sound quality / ease of listening. The results of this rating versus STI are shown in figure 14.

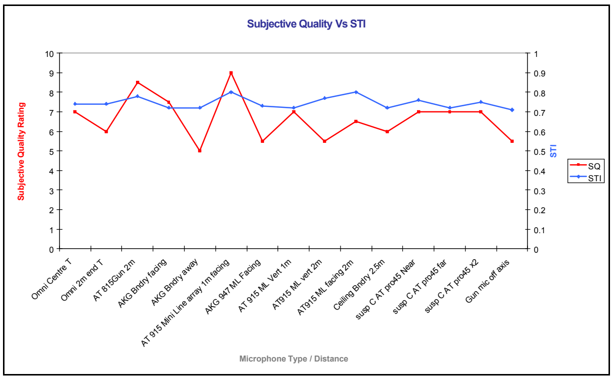


Figure 13 : Comparison of measured STI Vs subjective Sound Quality rating

As can be seen from figures 12 -14, the Hypercardioid microphone at 1m or the Gun Microphone at 2m were rated highest. The average STI for all microphone combinations was 0.75, whilst the mean subjective intelligibility rating was 7.5.

Figure 15 compares the subjective intelligibility versus the subjective sound quality rating. As can be seen, the two graphs track each other quite closely, though there are some interesting exceptions (e.g. the Hypercardioid microphone at 2m). The ceiling mounted boundary microphones and suspended cardioid microphones, whilst being rated as average, exhibited quite different objective measurement results.

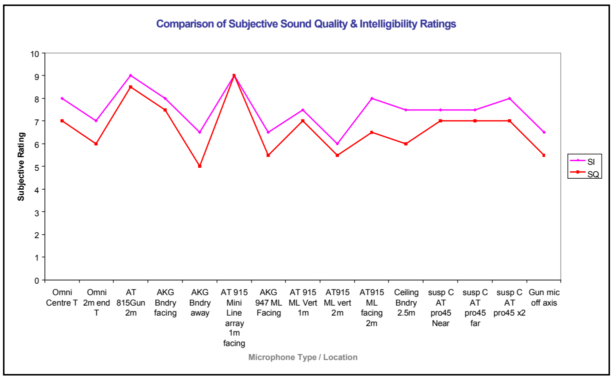


Figure 15: Comparison of subjective Intelligibility Vs Sound Quality rating

6. COMMENTS & CONCLUSIONS

The research has shown STI to be a useful measure for determining the performance of Assistive Listening Systems. However, as has been reported elsewhere it is not necessarily a descriptor of perceived sound quality.

The meeting room tests suggest that the current BS7594 intelligibility target of >0.65 STI (and the author's previous recommendation of ≥0.70 STI) may require reviewing as the microphone combinations achieving > 0.8 STI were considered to be both far better and desirable. (This would also tie in rather more closely with Bakke's conclusion that a value of > 0.84 STI is required). However, great care needs to be exercised, as the test listeners were not hearing impaired and were listening to full bandwidth recordings. (The STI measurements were also not bandwidth restricted).
If a value of 0.8 STI is to be targeted, this has very significant implications for ALS microphone requirements, both in small meeting / lecture rooms and in auditoria. Clearly considerable further research is required before suitable criteria can be firmly set.

Whilst it is dangerous to read to much into the results of a limited set of measurements carried out (albeit in detail) in one room, studies of other meeting rooms show that the situation evaluated is quite common. It may be that additional acoustic criteria need to be set for rooms that hard of hearing listeners may use.

The meeting room tests again show how difficult it is to make a test loudspeaker that accurately mimics not only the directivity but also the soundfield of a human talker.

A factor not directly considered within the tests was the effect of background noise on the objective and perceived intelligibility. In the primary meeting room tested, road traffic noise persistently intruded. This was considerably more noticeable via the ALS microphones than to a normal binaural listener. The highly reflective wall surfaces also exacerbated the problem. Some further research into the effects of acoustic noise break-in for Assistive Listening System users may be required. [It is interesting to note that BS 7594 and IEC 60118-4 recommend a signal to noise ratio of 32 dBA as the minimum for communication, noting that for short periods, a value of 22 dBA might be tolerated]. This implies that an ALS microphone needs to be relatively close to the talker. Certainly in rooms prone to external noise break-in or internal equipment noise (eg air-conditioning) this suggests that ceiling mounted boundary microphones are best avoided.

The meeting room tests clearly show that additional work is required in finding an optimal microphone solution but for a reflective environment, it would appear that a reasonably directional microphone within 1m of the talker would be required. (In less reflective rooms, it may be possible to relax this criterion). Further research is clearly required to consider a greater range of acoustic environments.

Although directional microphone arrays and other advanced technological solutions are available, it must be recognised that most Audio Frequency Induction Loop and Assistive Listening Systems rarely command significant budgets and so straightforward and economical solutions need to be found and applied.

The use of STI as a criteria and measure of the potential intelligibility of Assistive listening systems, although showing considerable promise, requires further work and validation and indeed, perhaps some modification.

7. REFERENCES

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