ASSESSING THE POTENTIAL INTELLIGIBILITY OF DEAF AID LOOP AND OTHER ASSITIVE AUDIO SYSTEMS

Peter Mapp Peter Mapp Associates Colchester, UK.
(peter@petermapp.com)

1 INTRODUCTION

Around 9 million members of the UK population (~ 14%) suffer from a noticeable degree of hearing loss and would benefit from some form of hearing assistance or deaf aid. Recent DDA legislation and requirements mean that many more hearing assistive systems are being installed – yet there is evidence to suggest that many of these systems fail to perform adequately and provide the benefit expected. This paper reports on the results of some trial acoustic performance testing of such systems. In particular the effects of system microphone type, distance and location are shown to have a significant effect on the resultant performance. The potential of using the Sound Transmission Index (STI) and in particular STIPa, for carrying out installation surveys has been investigated and a number of practical problems are highlighted. The requirements for a suitable acoustic test source to mimic a human talker are discussed as is the need to the need to adequately assess the effects of both reverberation and noise. The findings discussed in the paper are also highly relevant to the installation and testing of classroom 'sound field' systems.

The electronic and electromagnetic radiation characteristics and performance of audio frequency inductive loop systems designed to assist hard of hearing listeners (AFILS) are well catered for in various codes of practice and standards (eg BS 7594 & IEC 60118-4) [1, 2]. However the acoustic and intelligibility performance of such systems is given scant regard. Equally alternative systems such as Infra Red and FM wireless transmission are also widely used, particularly outside the UK – but again little or no consideration has been given to the potential intelligibility and clarity of these systems and few, if any, standards are available for hearing assistance applications.

There are for currently about 2 million regular hearing aid wearers in the UK, but approximately 4-6 million members of the UK population would benefit from the use of a hearing aid. In terms of population percentages, there is nothing to suggest that other European countries and the USA are particularly different to the UK. DDA & national building regulations have had a significant impact on the provision & installation of facilities for the hard of hearing, though the degree to which such regulations and local requirements are enforced (or heeded) across Europe appears to vary markedly. Most public buildings in the UK now provide assistive listening facilities and they are also extensively available in theatres, lecture theatres, cinemas, churches and even some football stadiums are beginning to install systems. London Underground (LUL) has just embarked on a programme of providing deaf-aid loop systems (AFILS) at all of its 270 stations (both at ticket office counters and on the platforms and routeways). Many main line railway stations either also provide or are beginning to provide similar facilities. However surveys carried out by a number of user groups and the National Institute for the Deaf (RNID) regularly find installed assistive hearing facilities to be far from optimal, if working at all [3]. One of the major problems that is regularly encountered is associated with the acoustic and intelligibility performance of the systems. To date, no standards and little guidance exist as to the required intelligibility performance for systems for the Hard of Hearing (HOH) or Assistive Hearing Systems (AHS) or how such facilities should be designed or tested to ensure that adequate intelligibility is achieved. One of the aims of this paper is to provide some of the necessary background information required in order to progress this process.

2. BACKGROUND

Assistive Hearing Systems (AHS or HOHS – Hard of Hearing 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 or hearing aid and earpiece/earphone acoustic transducer. 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 FM wireless broadcast or Infra Red (IR) transmission. As all National Health scheme hearing aids in the UK currently incorporate a means for picking up an audio induction signal, this is by far the most commonly used technique within the UK, although Infra Red systems are also common in theatres. However, in the USA & Europe AFILS usage varies greatly as indeed does the use of any of the systems - if used at all. Each method has its advantages and disadvantages dependent on the technology involved, but there are many common elements and associated problems.

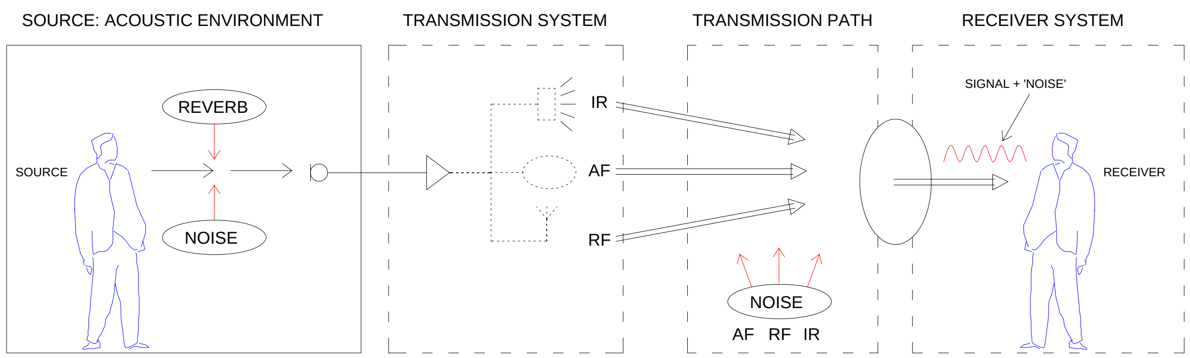


Figure 1 AHS Signal Transmission Path

3 ACOUSTIC PICK UP OF THE SOURCE (SPEECH) SIGNAL

The desired speech signal may be picked up either by a body worn microphone (eg Lavaliere, tie clip microphone, or head worn microphone – most frequently with wireless transmission), local tabletop or lectern microphone or more distant remote microphone. In the latter case this is usually a directional microphone (Cardioid or Rifle mic) and may be up to 10m or more away from the talker in venues such as theatres, concert halls or similar auditoria. In conference rooms or courts, the speech pick up can also be by means on ceiling mounted boundary microphones or suspended overhead microphones. Teleconference / conference / meeting rooms may also employ ceiling microphones. In theatre & concert hall systems, the signal is also often shared with, or more usually, derived from the 'show relay' pick up for the backstage areas and dressing rooms. In the author's experience, these systems are rarely formally tested and even more rarely with a hard of hearing user in mind. One of the objectives of this paper is to review how such systems might be tested, particularly bearing in mind the recent introduction of portable, direct reading STI (STIPa) test equipment and the increasing impact of DDA legislation.

A point to note is that where a sound reinforcement system is installed, a feed of the programme material is also often taken directly from the mixer output. This can produce excellent results where body worn microphones are used, as the signal to noise ratio and direct to reverberant ratio are as good as can be obtained – though care needs to be taken with some microphone positions as the resulting speech spectrum pickup response can be less than optimal – as illustrated in figure 2. However it is normally possible to equalise the response and this would usually be implemented in order for the main acoustic reinforcement to sound natural / intelligible. (Whilst this is the case in professional or professionally operated systems, many 'set and forget' systems will not exhibit this luxury. Figure 2 shows a number of responses measured by the author for different body worn microphone positions [4]. As can be seen, considerable EQ is required for some positions if vocal clarity is to be maintained.

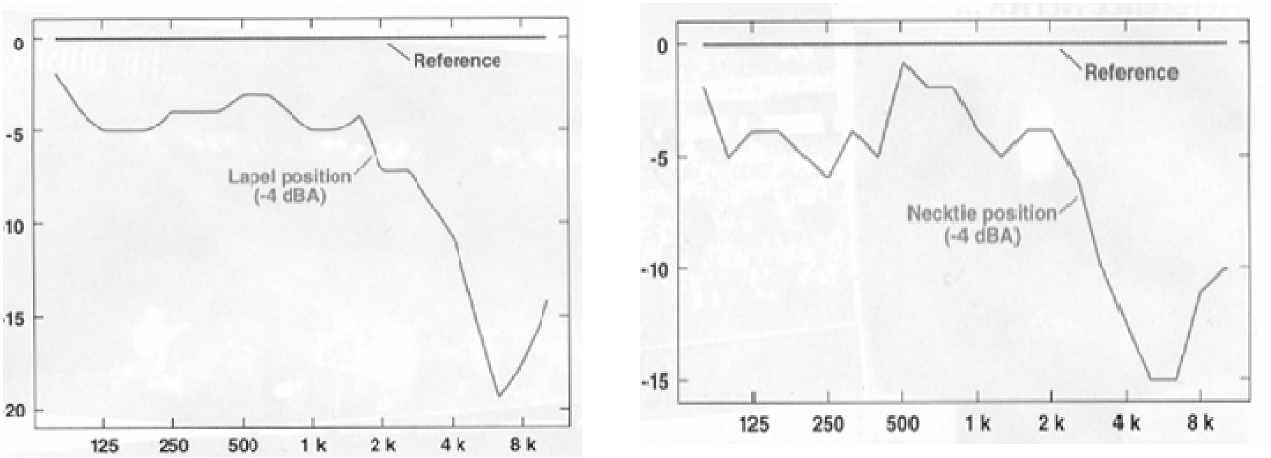


Figure 2 (a & b) Microphone response relative to on axis position for Lapel and Tie / front torso mounted microphone.

With local, but non body worn microphones, the possibility exists for noise and frequency response degradations to occur depending upon the local acoustic conditions. For example table mounted microphones can suffer from comb filtering distortions that can severely affect the resultant tonal quality as shown in figure 3, which shows the response of a microphone located approximately 200 – 300 mm from a reflective boundary.

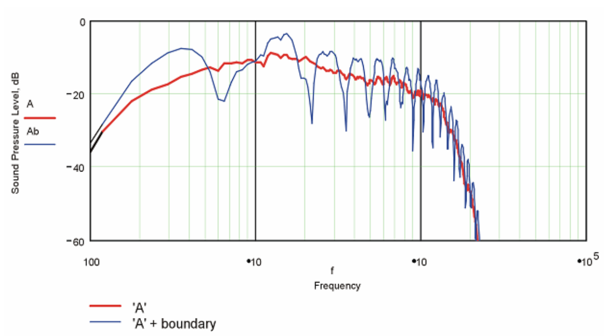


Figure 3 Microphone comb filtering caused by reflection form adjacent boundary

Using a cardioid microphone can help to reduce the problem but not for shallow incident angles. Boundary microphones are usually the best choice, provided that they can be sited close enough to the talker to ensure a high signal to noise ratio as well as an adequate direct to reflected (reverberant) ratio.

As the distance between the talker and pickup microphone increases, the direct to reverberant sound ratio rapidly decreases, as this ratio and the resultant intelligibility generally follow an inverse square law relationship. (ie for each doubling of distance away from the source that the microphone undergoes, the potential intelligibility reduces by a factor of 4 times). This occurs until the 'critical' distance for the microphone – room combination is reached, after which a more or less consistent level of intelligibility occurs. Figure 4 shows a typical plot of measured total sound level and intelligibility (as measured by the Speech Transmission Index [STI]), as a function of distance for an acoustically well controlled conference room / classroom (RT = ~ 0.4 sec). (Note in this instance, the measurements were made using an omnidirectional microphone).

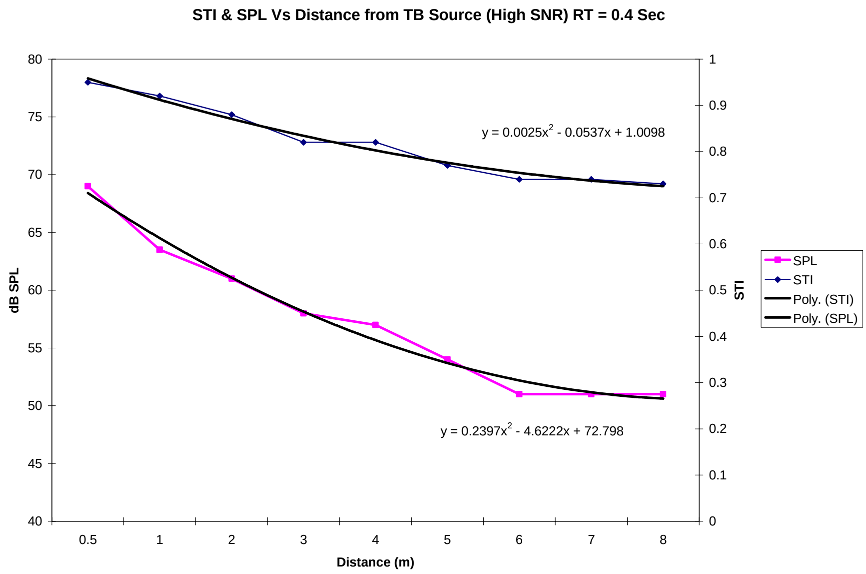


Figure 4 SPL & STI Reduction with distance for Talker source in low RT room.

As mentioned above, in theatres & concert halls and similar auditoria, the speech pick up microphone may be located up to 10 metres or more away from the stage / speech source (talker). Figure 5 shows a plot of Intelligibility (STI) versus distance measured in an empty and reverberant mid size recital hall (RT = 2.2 seconds). Again, the fall off approximately follows an inverse square law characteristic and settles to a steady value at around 8 m where the reverberant field begins to dominate. As has already been mentioned, it is normal to use either a cardioid or even a rifle mic for speech pick up in auditoria. However, the performance between microphones can vary enormously.

Figure 5 shows the results of a set of STI measurements carried out in a 2.2 second recital hall using omni, cardioid and rifle microphones. Although these directional microphones follow a similar trend to the omnidirectional microphone, the potential intelligibility (as measured in terms of STI) is markedly improved, with the cardioid microphone being on average 0.09 STI higher and the rifle microphone 0.17 STI higher.

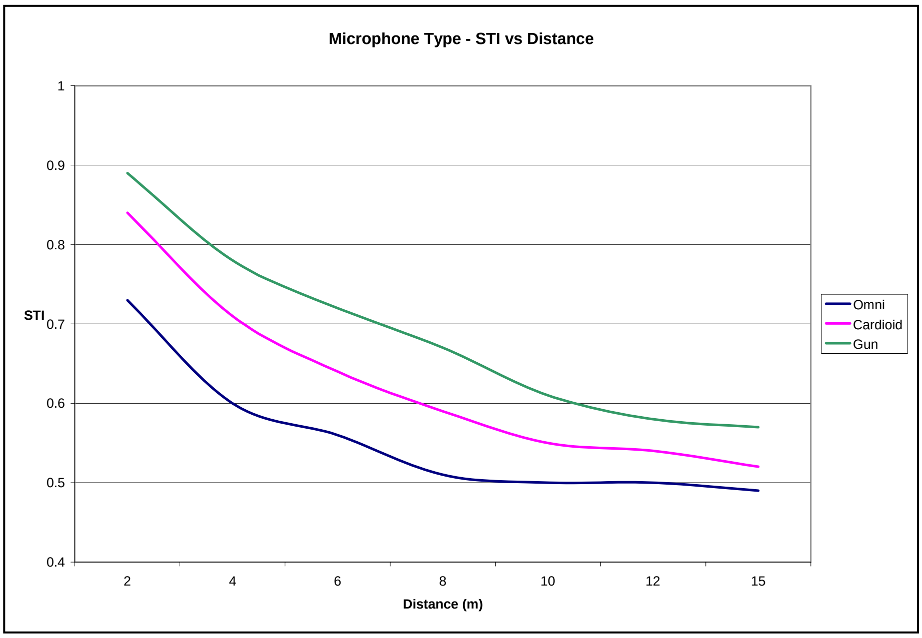
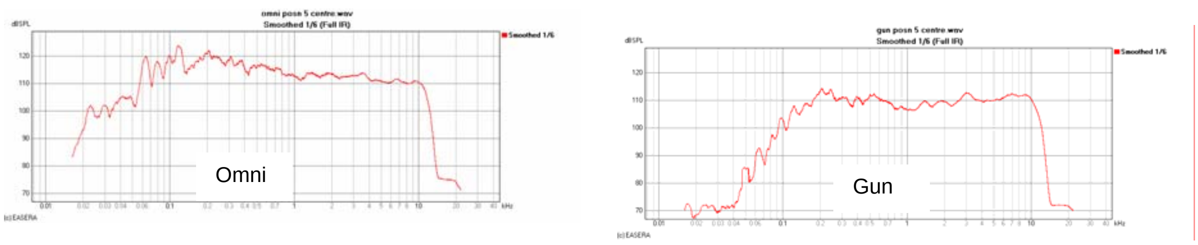


Figure 5 STI Vs Distance for 2 sec RT Recital hall

To put this into context, assuming that a minimum target of 0.7 STI is required for a AHS [5], this can be achieved up to a distance of 7m for the rifle microphone, up to 4m with the cardioid and up to just 2.5m with the omnidirectional microphone. For a theatre of a similar size, but exhibiting a typical reverberation time of around 1 second for speech auditoria, these values would increase substantially and could generally be expected to more than double, so that the cardioid mic at 8 - 10m distance may be viable, whereas clearly the rifle mic could theoretically operate out to more than 14m – assuming that the high signal to noise ratio could be maintained.

A further insight into how the directivity of the pick up microphone can affect the perceived frequency response is illustrated by figure 6. Here, the in-situ acoustic responses measured using a special calibrated test loudspeaker with a three different microphones are shown. In each case, the on axis microphone response is essentially flat. This nominal response however becomes significantly altered by the different reverberant contributions picked up by each of the three microphones



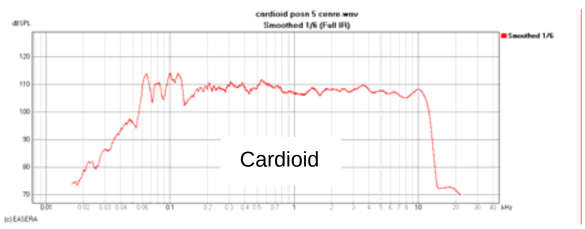


Figure 6 In-room responses for Omni directional, Cardioid and Gun microphones located at 6m for test 'Talker' loudspeaker in 1.2 sec RT auditorium.

4 TALKER TEST LOUDSPEAKER RESPONSE AND SIMULATION

One of for the most challenging obstacles to carrying out formal testing of microphone placement intelligibility is the ability to accurately simulate the acoustic characteristics of a talker. Whilst IEC 268-16 [6] states that a 100mm driver in a small enclosure may be employed to simulate a talker, a series of tests conducted by the author clearly show that simply using this size of drive unit and a 'head sized' loudspeaker does not replicate the sound field of a human talker and can lead to significant intelligibility measurement errors. Figure 7 shows a plot of intelligibility (STI) versus distance for a number of test talker loudspeakers carried out in the same recital hall as the microphone tests presented in figures 5 & 6 above. The tests were carried out using an artificial mouth (B+K 4227), a single cone 4 inch loudspeaker and a single 5 inch loudspeaker, together with a 2 way 5inch loudspeaker and a new special head simulator of the author's design.

Whereas differences of up to 0.07 STI were found to occur for the on axis case as shown by figure 7, greater variations and frequency response changes were found to occur for off axis positions simulating a talker turned sideways to a listener.

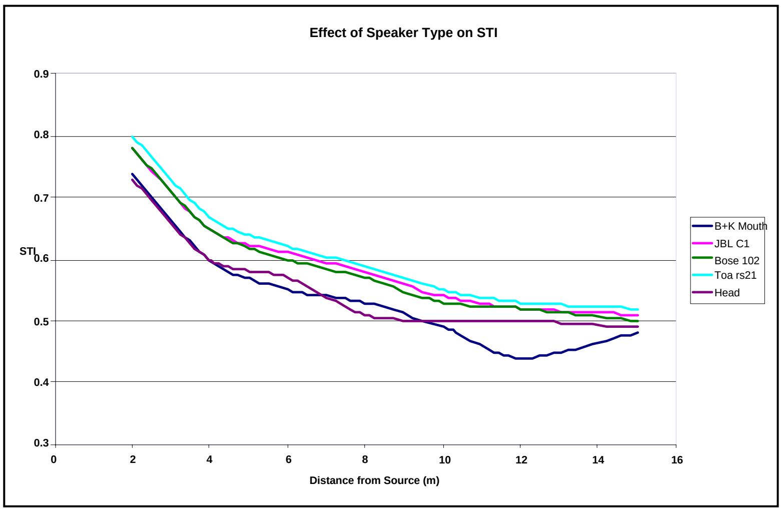


Figure 7 STI Vs Distance for 2 sec RT Recital hall, comparing different talker speaker types

A practical demonstration of the potential error that this can produce is shown in figure 8. This shows the predicted radiation from a human talker (red lines) and 4inch cone loudspeaker (blue lines) in an acoustic computer model of an auditorium. Whilst the sound from the human talker is shown to strike the overhead reflectors, the sound from the talker simulator loudspeaker clearly does not. The early reflection patterns will therefore be very different and the resultant intelligibility can also be expected to be affected.

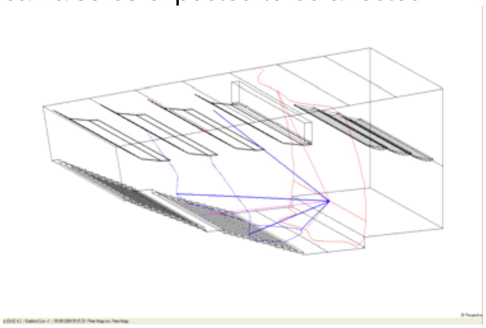


Figure 8 Comparison simulation of the acoustic radiation pattern from a human talker and a 4 inch loudspeaker in a theatre

The front of the lighting bridge is a common position for an AHS / show relay pick microphone in many auditoria, clearly testing the effectiveness of such a microphone using a 4 inch loudspeaker on the stage would give rise to erroneous results.

5 FREQUENCY RESPONSE OF THE TRANSMISSION SYSTEM

Both FM and IR systems generally exhibit a fairly wide frequency response, however, AFILS are deliberately band limited from 100Hz to 5 kHz, as hearing aids provide an even narrower response with no useful information being obtainable outside these limits. Whilst reducing the fidelity of reproduction, as heard by a normal hearing listener, this band-limiting has virtually no effect as far as a hearing aid user is concerned – although there is a theoretical reduction in intelligibility of up to approximately 14%. For applications other than deaf aid transmission, the frequency response of Induction Loop Systems (AFILS) can readily be extended by the application of appropriate filtering. Whereas in the past, voltage driven, multi-turn loops have caused poor frequency response results and given loop systems an undeservedly bad reputation, modern single turn, current driven systems can provide a more than adequate response. Figure 9 compares the frequency response for a typical, large area loop system using both a single turn current drive loop and a multi-turn voltage driven loop.

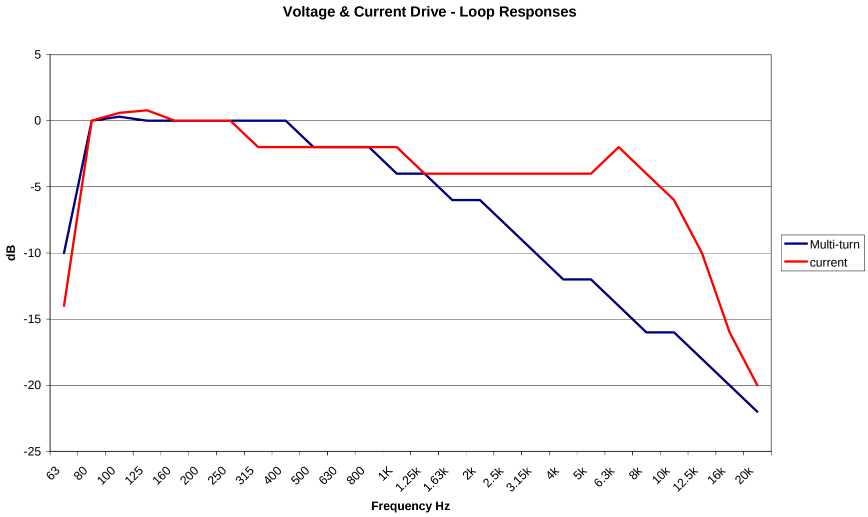


Figure 9 multi-turn voltage drive and single turn current dive loop responses

6. SIGNAL TRANSMISSION & BACKGROUND NOISE

A common problem encountered with loop systems is the presence of undesirably high levels of background magnetic noise. This may be being radiated from mains power cabling, power transmission lines, transformer equipment, power supplies or lighting circuits and fittings. The noise from these sources is usually limited to 250 Hz and below, though fluorescent lighting can give rise to higher frequency problems. Data transmission / information signaling (particularly on railway systems) can also be a problem and often occurs within the speech bands critical for intelligibility. It can be quite common in railway station and industrial environments for the background magnetic interference level to apparently exceed the desired signal level. However, in order to evaluate the situation correctly, the response of the hearing aid or receiver must also be taken into account, for as we shall see the inductive response of most hearing aids at low frequencies is such as to generally filter out many mains power related problems.

7 RECEIVER RESPONSE CHARACTERISTICS

Figure 10 shows the inductive pick up response of a typical analogue hearing aid. As can readily be seen, at high frequencies the aid attenuates steeply above 5 kHz, whilst at the lower frequencies there is little or no useable pick up below 500 Hz or even below 1000 Hz with many models.

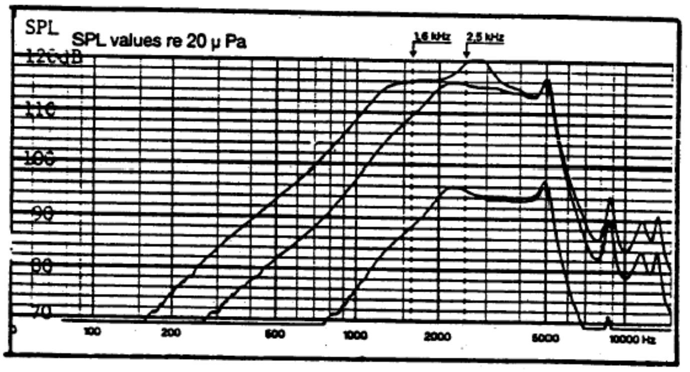


Figure 10 Typical analogue hearing aid inductive pick up response

Clearly therefore, in order to assess the potential effects of both background noise and reverberation on the received signal, the response and bandwidth of the hearing aid must be fully taken into account in any measurement procedure or assessment. The most logical approach would be to use a filter that attempts to replicate the general response of a hearing aid. IEC 60118-4 however requires the background noise level to be measured using the standard 'A' weighting characteristic. Whilst this is undeniably a well established and universal curve, it is a gross mismatch to the response of an analogue hearing aid. (The author understands that current generation of digital hearing aids that have a 'T' coil fitted have an improved response and it may be that the A weighting curve could work better with these).

Interestingly the 'A' weighting is not used by IEC 60118-4 to measure the field strength of the desired speech signal, instead a linear (though band limited) response is used. It is therefore not possible to make true direct signal to noise ratio comparisons.

IEC 60118-4 sets the 0 dB reference level to 0.4A/m. This is the maximum rms ('peak') value to be achieved. In practice therefore, typical speech signal levels will be around -12 to - 6 dB below this for a well set up loop system (ie 0.1 to 0.2 A/m) depending on the amount of signal compression applied. The standard suggests that the Awtd magnetic noise level should be at least 47 dB below the 0 dB reference level. In practice this equates to an Awtd SNR of around 30 – 35 dBA. At ratios below 32 dB (ie 15 – 20 dBA on real speech real speech) the standard requires that the ratio be reported and remedial measures be considered. It also notes however, that if the magnetic noise has no significant tonal quality or is mostly at low frequencies than a ratio as low as 22 dB may be tolerable (ie 5 to 10 dBA with real speech). The overestimation of low frequency noise by the A weighting therefore appears to have been recognized, as the above figures seem reasonable when all are converted to A weighted values.

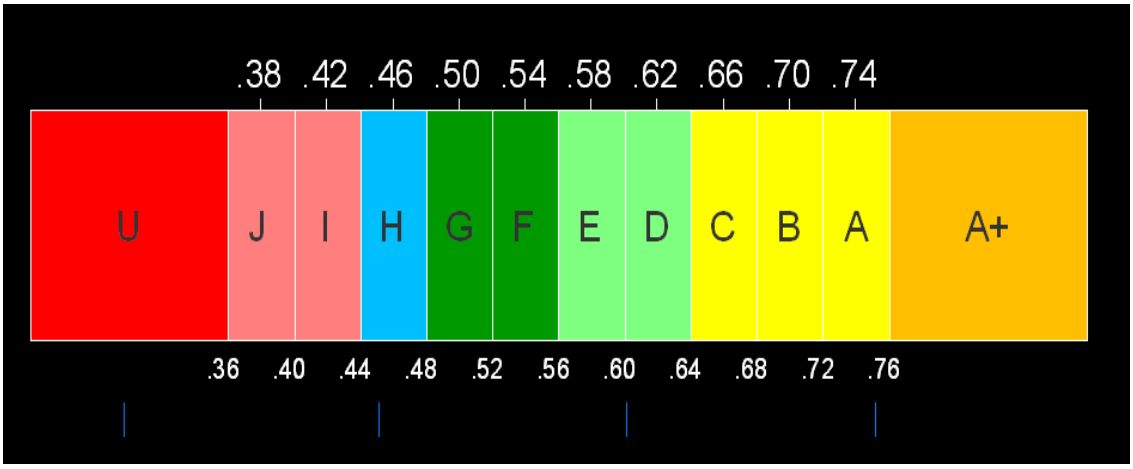
From the above analysis, It can be expected that magnetic noise levels greater than -32 dB may well have an effect on intelligibility – they almost certainly will have an effect on the 'ease of listening', system acceptability and listener fatigue. When assessing the complete transmission chain and the intelligibility / sound quality of the signal that will be heard by a listener, the detrimental effects of any acoustic noise pick up at the source microphone must be taken into account and added to the magnetic noise component. An 'end to end' measurement should therefore be carried out.

8 USING STI & STIPA TO EVALUATE AFILS AND ASSISTIVE LISTENING SYSTEMS

STIPa (STI for PA systems) is rapidly becoming the preferred method for evaluating sound system intelligibility performance [9,10] and should also be potentially suitable for assessing AFILS or other assistive listening systems. Some modification to the procedure however is required. Firstly the signal analyser will need to be modified to have a similar bandwidth and frequency response to that of a normal hearing aid - or at least to that of an 'A' weighting curve, as exemplified in figure 18. Secondly, the redundancy factors and masking factors require adjustment as they are no longer valid under these conditions. Thirdly an accurate (but affordable) 'talker' simulator loudspeaker is required to generate the necessary acoustic input to the system. However, in many applications it may not always be feasible to input the signal acoustically and so a means of also injecting a calibrated electronic STIPa signal is also required.

From an inspection of the frequency response curves presented in figure 10, it can be seen that a hearing aid will receive no useful speech signal contributions in the 125Hz, 250 Hz and 8 kHz bands, and probably only limited information within the 500 Hz band. This reduces the maximum possible intelligibility to an equivalent of approximately 0.70 STI (or lower - ie assuming there to be no noise (acoustic or magnetic) or reverberant degradation whatsoever). Reducing the target criterion previously proposed by the author [5] from 0.7 to the equivalent to 0.6 STI (which equates to the performance of a good concert hall or reasonable theatre sound system) enables there to be just 0.1 STI loss within the transmission chain. From figures 4 & 5 it can be seen that microphone distances of less than 2-3m would be required for the well controlled conference room and 2m or less for the reverberant recital hall (4-5m for a theatre). If the target criterion is lowered further to 0.5 STI then these distances could be increased to a maximum of 4m for the conference room and recital hall and possibly 8m for an equivalent sized theatre. Whilst this provides more scope it may well also mean that the AFILS is not capable of providing sufficient intelligibility to a hard of hearing listener. However, this is somewhat speculative as very little research has been carried out relating hearing aid / HOH user requirements with STI. (Nor indeed has even the measurement protocol for making accurate bandwidth restricted STI measurements been formulated and standardised).

Setting a definitive STI or related requirement may be a difficult and controversial task. In order to assist in setting a suitable goal, the author has produced a new form of STI scale, which operates in bands rather than with fixed values. Application specific values can then be applied as required. Figure 1 and Table 1 present a prototype scale and application table.



CATEGORY	**TYPICAL USE**	**COMMENT**
> 0.76		
A 0.74	Communication Systems	Very High speech intelligibility
B 0.70	Theatres, speech auditoria, HOH/AFILS systems	High Speech Intelligibility
C 0.66	Theatres, speech auditoria, teleconf	High Speech Intelligibility
D 0.62	Lecture Theatres, classrooms, concert halls, Modern churches	Good Speech Intelligibility
E 0.58	Concert halls, Modern churches	High Quality PA systems
F 0.54	Shopping malls, public buildings offices, VA systems	Good Quality PA systems
G 0.46	Shopping malls, public buildings offices, VA systems	Lower Target requirement for VA/PA
H 0.42	VA & PA Systems in difficult acoustic environments	Lower limit useful for PA
I 0.38	VA & PA Systems in V difficult spaces	
J 0.36	Not suitable for PA systems	
U < 0.36	Not suitable for PA systems	

Table 1 Proposed STI Application / Banding Categories

9. CONCLUSIONS & FURTHER WORK

Clearly much further research needs to be carried out into both measurement techniques and HOH / Hearing aid user requirements. In addition to this, there are a number of other issues relating to signal processing effects and their impact on test stimulus aberrations that need further investigation (Eg effects of compression and the crest factor of real speech as compared to test signal dynamics). The ability to import microphone directivity characteristics and information into acoustic simulation programs would enable the potential siting of microphones to be improved and the potential intelligibility of an Assistive Hearing System to be evaluated prior to installation.

10 REFERENCES

[1] BS 7594 Audio Frequency Induction-Loop systems
[2] EN 601184 Electroacoustics – Hearing Aids- Part 4 : Induction loop systems for hearing aid purposes – Magnetic field-strength.
[3] RNID : Induction Loop System User Assurance Testing May 2007
[4] Mapp, P Microphones on the sly, S&VC magazine 1997
[5] Mapp, P (2003) The Acoustic and Intelligibility Performance of Assistive Listening and Deaf Aid Loop (AFILS) Systems. AES 114th Convention.
[6] IEC 268-16 Objective rating of speech intelligibility by speech transmission index
[7] Mapp, P (2005) Is STIPa a robust measure of speech intelligibility performance? AES 118th Convention.
[8] Mapp, P (2004) Systematic & Common Errors in Sound System STI and Intelligibility Measurements. AES 117th Convention.