ARE WE MAKING BEST USE OF ASSISTIVE LISTENING SYSTEMS IN THEATRES?

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1 INTRODUCTION

Assistive listening systems (ALS), such as audio frequency induction loop (AFIL) and infra-red systems, are now commonplace in most theatres. However, simply installing such systems to tick the ‘compliance’ box does not in itself ensure that deaf and hard-of-hearing audience members have a satisfactory quality of experience at the theatre. Sound designers and technical staff should be aware that audio requirements for ALS differ from those for the main PA system. Audio delivered through the ALS needs to provide a greater signal-to-noise ratio than that typically available acoustically in most seats. The common practice of feeding the ALS from a show relay microphone rarely achieves this.

While some guidance exists to address this issue, it can be hard to find as many mainstream theatre design and sound design textbooks make little mention of assistive listening systems. BS 7594:2011 ‘Code of practice for audio-frequency induction-loop systems’[1] provides some guidance on suitable audio input for ALS, but this is not a resource with which most end users can realistically be expected to be familiar. Peter Mapp’s 2014 paper ‘The Acoustic and Intelligibility Performance of Assisted Listening Systems’[2] sets out valuable guidance on ALS microphone placement in auditoria, which this paper seeks to build upon.

This paper instead aims to distil guidance from a range of sources into a practical overview to assist sound designers and operators in providing clear and intelligible audio mixes for ALS.

2 WHY ARE ASSISTIVE LISTENING SYSTEMS NECESSARY?

2.1 Equality Act 2010

The Equality Act 2010 and its predecessor the Disability Discrimination Act 1995 have arguably been significant drivers in the adoption of assistive listening systems in theatres and other venues. The Equality Act crucially introduced the concept of ‘substantial disadvantage’. Previously, ‘reasonable adjustments’ only had to be made by service providers where it would otherwise be ‘impossible or unreasonably difficult’ for a disabled person to use the service. Under the Equality Act, adjustments must be made ‘where a disabled person would, but for the provision of an auxiliary aid, be put at a substantial disadvantage in relation to a relevant matter in comparison with persons who are not disabled, to take such steps as it is reasonable to have to take to provide the auxiliary aid.’[3]

In this respect, the Equality Act effectively brings into consideration the ‘quality of experience’ of deaf or hard-of-hearing audience members. In the case of theatres and performance venues, such individuals may not necessarily be prevented from attending a theatre if an adequate assistive listening system is not provided, but the theatre could nevertheless be in breach of the Act if these audience members are put at a substantial disadvantage compared to those with normal hearing.

2.2 Acoustic issues encountered in theatres and auditoria

Theatres and other auditoria can be challenging listening spaces for hard-of-hearing individuals. An annual survey conducted by the RNID (now Action on Hearing Loss) of their members in 2005 reported that 44% of the respondents did not attend the theatre as often as they would like. Of this group, the majority, 61%, gave the reason that they cannot hear the performance well enough.[4]

The key problem is one of signal-to-noise ratio; that is the ratio of wanted sound, in this case speech or music from the stage, against unwanted noise and reverberation. For audience members sitting at some distance from the stage, unamplified speech levels from the actors on stage can often be relatively low, and easily overwhelmed by seemingly innocuous sources such as fidgeting or coughing from nearby audience members. Reverberation in the space can also have a substantial impact on the ability of hard-of-hearing individuals to perceive speech. A 1990 paper by Harris and Swenson[5] demonstrates that in otherwise quiet conditions, individuals with normal hearing are able to cope relatively well in reverberant environments and their ability to recognise speech reduces only slightly in a reverberant room compared to a dry studio. Hard-of-hearing individuals generally show a marked reduction in their ability to recognise speech in reverberation, with the reduction becoming more pronounced with increasing severity of hearing loss.

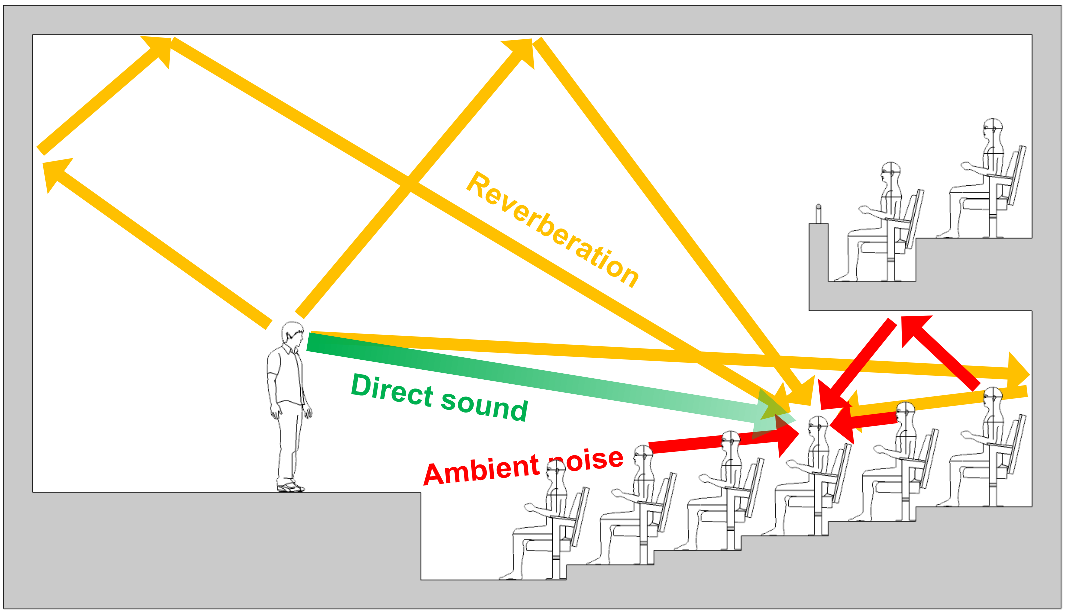


Figure 1 – Illustration of conflicting noise sources in an auditorium

Locating sounds can also be more difficult for hard-of-hearing individuals, and particularly for hearing aid wearers. In normal binaural hearing, minute delays between the arrival of sound to the left and right ear are used to pin-point where the sound is coming from; this can be a vital tool for identifying and focusing on specific sounds in a noisy environment. Wearing a hearing aid in one or both ears can disrupt this function. Cochlear implant users, even those with implants in both ears, may also lose much of the benefit of binaural hearing as a great deal of the complex phase and delay information of the sound field is lost in the voice coding process.[6]

2.3 Benefits of an assistive listening system

A misconception commonly encountered by the author is that assistive listening systems are simply about 'making things louder' for hard-of-hearing users. This however is only part of the purpose of such systems. Principally, assistive listening systems provide a means of delivering audio directly to the listener, reducing the influence of reverberation and ambient noise from other sources in the space and effectively offering hearing aid users the 'best seat in the house'.

For an assistive listening system to achieve these ends effectively, the audio input to the system must be highly intelligible. As sound engineers will often recount, even with the best quality loudspeakers and sound system equipment, it is all too easy for these systems to sound dreadful in the wrong hands. Similarly, while it is vitally important to have an AFILS or similar assistive listening system which works correctly and provides good field coverage to the audience, unless adequate consideration is given to the quality of the audio input, these systems will be of little benefit to hard-of-hearing patrons.

In the author's experience, the quality of audio input into an ALS is seldom adequately considered by installers. Often the default position is to place a microphone in the middle of the room with the intention of 'picking up everything'. In theatres and auditoria, this line of thinking usually results in the show relay microphone, typically located on a balcony front or lighting bridge and often at some distance from the stage, being used to feed audio to the ALS. However, such an approach is unlikely to provide adequate intelligibility for hard-of-hearing ALS users. For example, audio from a cardioid show relay microphone mounted on a balcony front 10 metres from the stage in a 500-seat theatre is unlikely to yield a speech transmission index (STI) of more than 0.5, well short of the 0.7 STI criterion discussed in Section 3.3.1 of this paper.

3 PROVIDING AUDIO TO ALS

3.1 Challenges of capturing audio for ALS in theatres

Theatres can be uniquely challenging environments for capturing intelligible audio for ALS. Even in relatively conventional proscenium theatres, there can be a very large stage area to cover. The playable area of a well-proportioned proscenium stage tends to be between 9 and 12 metres wide and up to 9 to 10 metres deep. The modern trend towards more immersive presentation formats such as thrust, in-the-round and traverse can introduce further complications. Although such formats tend to employ a smaller playing area, actors have to command a much wider audience area on several sides of the stage. In playing to one of several sides, actors will inevitably find themselves with their back turned to at least part of the audience. Beside the inevitable acoustic issues, this can create additional problems for audience members who rely in part on lip-reading.

With recent developments in digital audio production technology, sound designs have inevitably become more ambitious. This is something of a double-edged sword. On one hand digital technology has enabled increasingly refined and 'transparent' speech reinforcement. An excellent example of this witnessed by the author was the Bridge Theatre's 2018 production of Julius Caesar, a production combining elements of promenade and theatre in-the-round, from which several of the author's fellow theatre consultants emerged blissfully unaware that the actors had been amplified. On the other hand, sound design on some productions has become increasingly complex and 'busy' through a desire to create a more 'cinematic' theatrical experience, with actors' dialogue competing against an underscore of sound effects and music.

In addition to issues of noise and reverberation discussed earlier, the dynamic range of the audio should also be carefully considered. Speech in a typical theatrical production may cover a wide range of vocal effort from a whisper to a shout. A 2002 paper by Zeng et al, 'Speech dynamic range and its effect on cochlear implant performance'[7] suggests that the dynamic range of an audio input necessary to accommodate a variety of talkers could be as wide as 60dB. By contrast, the dynamic range producible by cochlear implants is typically between 3dB and 20dB[8], depending on the type of implant; this is primarily governed by the strength of electrical impulses which can be safely applied to the auditory nerve. A similar order of dynamic range is usually necessary for hearing aid wearers to provide audio at a consistent level. A significant degree of compression is therefore needed to reduce the wide dynamic range of the audio input to match the necessarily narrow output range. Most modern loop amplifiers incorporate automatic gain control (AGC) circuits to achieve this, but for more demanding theatrical applications it may be advisable to implement dynamic control at the FOH (front-of-house) console.

The suggested approaches to providing audio to assistive listening systems for amplified and unamplified performances are discussed in the following sections.

3.2 Amplified performances

For amplified performances such as musicals, rock concerts or even stand-up comedy, providing a 'clean' mix for an assistive hearing should be a relatively straightforward matter. Close microphone placement is frequently used to maximise the available gain before feedback and this will inherently provide good rejection of reverberation and extraneous background noise for the assistive listening system. For such performances, a direct line-level feed should be provided from the FOH console into the assistive listening system. For simple, primarily speech-based performances such as stand-up comedy, the ALS may be satisfactorily fed with a copy of the FOH main mix. For more complex shows such as musicals or dramatic productions involving prominent sound effect cues, a dedicated mix should be provided for the ALS.

The advent of digital mixing consoles has made mixing out to multiple channels significantly more accessible and even many entry-level mixers provide a relatively large number of discrete configurable outputs. Most sound engineers are familiar with providing monitor mixes to performers on stage and the task of providing a dedicated mix to an ALS system should be approached in a similar manner.

Speech and musical vocals should be at the 'top of the mix' for the ALS audio feed and should be clearly audible above musical accompaniment. Where underscoring or ambient sound effects are used beneath dialogue, careful consideration should be given to how this affects the assistive listening system audio. Ideally, a ducking circuit should be used to reduce the level of other audio when speech is present. The effect of a ducker with a fast attack and slow release (approximately 1 second) is illustrated in Figure 2.

A similar effect may be achievable by mixing the sound effect track at a substantially lower level than the speech track (ideally around -15dB) and applying an AGC or slow-release levelling compressor to the mixed signal. This approach can however raise the noise floor when gain reduction is not applied. It is recommended that compression or ducking is applied to the mix before sending to the ALS so that the sound engineer can more easily monitor the effect of applied compression and avoid excessive 'pumping and breathing' artefacts. While the AGC circuits provided in many modern AFIL drivers can also provide the necessary gain control, the engineer will have less control over how these operate. In all cases, it is important that the sound engineer is provided with means of monitoring the loop signal within the auditorium, using either a telecoil-based loop monitor or a monitoring headphone output from the AFIL driver, if this is available within reach of the mix position.

It is also important that an 'ambient' microphone is provided to pick up audio from the room when there is no direct feed from the FOH console. Where a performance is only partially amplified, for example a musical where the lead singers are provided with body-worn microphones but the chorus are unamplified, microphones for ambient pickup should be positioned following the guidance set out in the next section. Where a performance is fully amplified, a simpler microphone in the middle of the room' may be employed to capture audience reactions. This is vital for audience cohesion and to ensure that hard-of-hearing audience members feel that they are fully included in the collective experience of live theatre. It is important to note that audio from an ambient microphone may potentially add a considerable amount of reverberation and ambient noise to the mix so it may be necessary to apply ducking to the ambient microphone audio when a 'direct feed' signal is present.

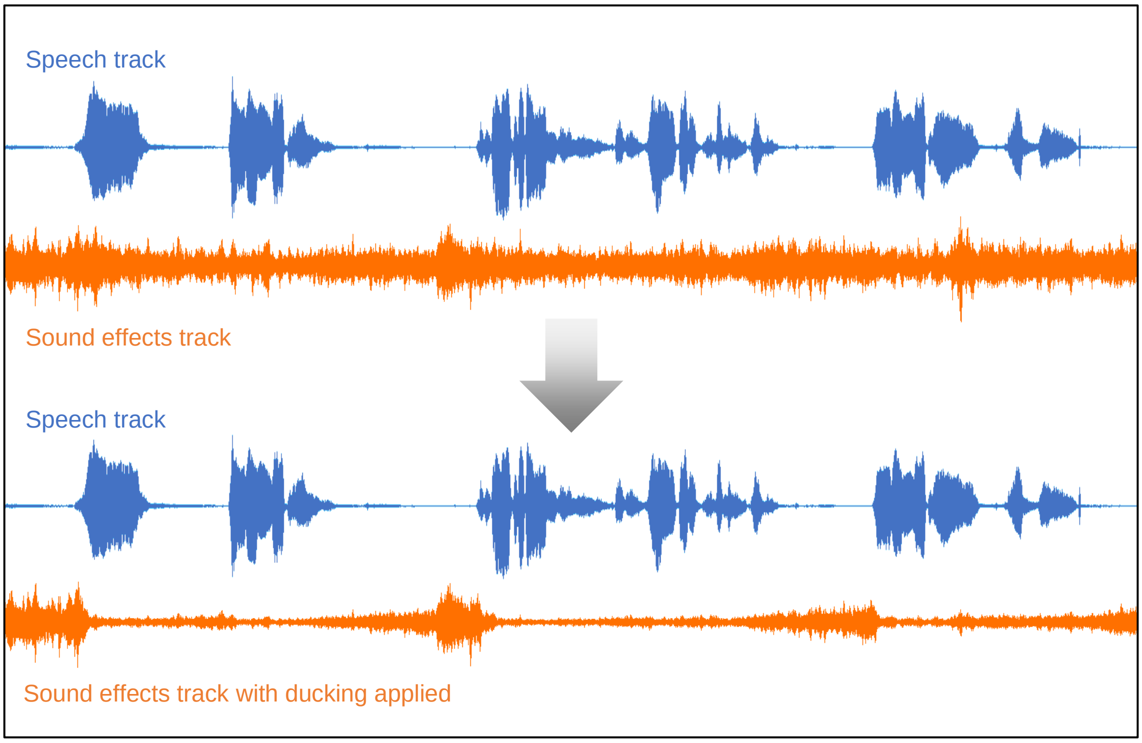


Figure 2 – Illustration of ducking applied to a sound effects track during speech

3.3 Unamplified performances

Unamplified theatrical performances arguably present a greater challenge in providing a clean, intelligible audio feed for an assistive listening system. Selection and location of microphones to pick up sound from the stage with an acceptable signal-to-noise ratio requires very careful consideration. At distances of more than a few metres from the actors in most theatre spaces, reverberation can begin to significantly colour the picked-up sound and will often be the primary limitation to achieving a good signal-to-noise ratio.

3.3.1 Discussion of numerical criteria

BS 7594:2011 suggests a simplified method of determining the achievable direct-to-reverberant sound ratio, K (dB), at a given distance between the talker and a single microphone, as follows:

$$\mathbf{K = 10\ Lg} \left[\frac{\mathbf{QV}}{\mathbf{314D^2\ (RT)}} \right]$$

Where Q is the directivity index of the microphone
V is the volume of the room (m3)
D is the distance between sound source and microphone (m)
RT is the reverberation time in the room (s)

It is the author's belief that Q in the above equation should in fact be the directivity factor, rather than directivity index as stated in the Standard. BS 7594:2011 suggests that K should be at least +4dB to +6dB for hard-of-hearing listeners.

Speech Transmission Index (STI) can be used to evaluate the intelligibility of AFIL systems. While BS 7594:2011 recommends an STI of 0.65, based on a recommendation of 0.65 RASTI in the previous 1993 version of the Standard, BS 60268-16:2011[9] suggests a minimum STI of 0.7 for AFIL systems.

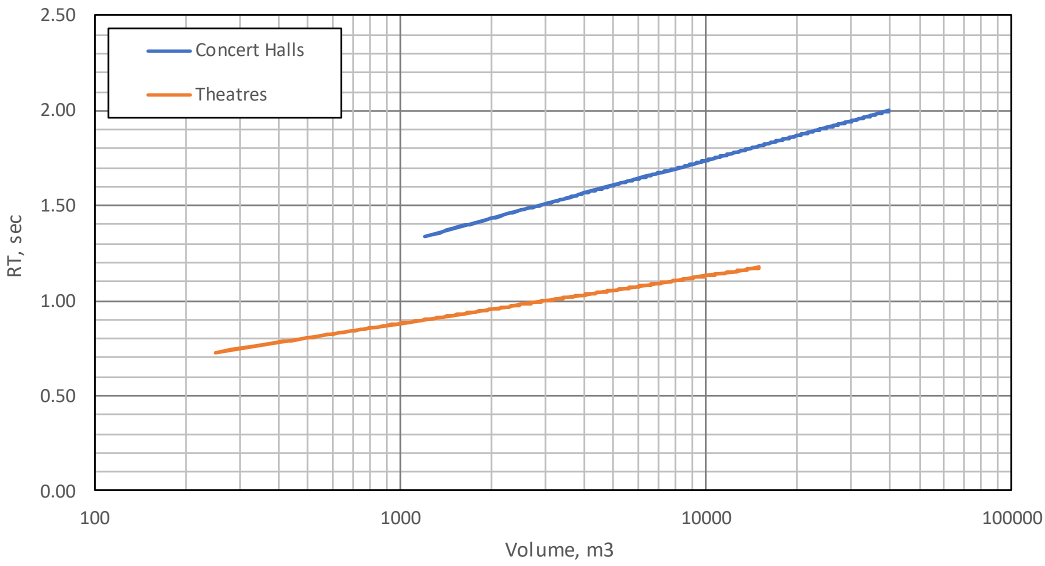
Appendix L of BS 60268-16 sets out a procedure for calculation of STI using statistical methods, which can be used to estimate STI based on the ratio of speech to reverberant and ambient noise. For the range of auditoria volumes and RTs considered in this paper, calculations indicate that an STI of 0.7 or better is generally achievable where K is at least +3dB and the speech-to-noise ratio (SNR) at the microphone is at least +23dB(A). Although the relationship between STI, K and SNR is variable, calculations indicate that it is generally necessary to increase K to at least +5dB and the SNR to +26dB(A) to achieve an STI of 0.8.

3.3.2 Microphone positioning

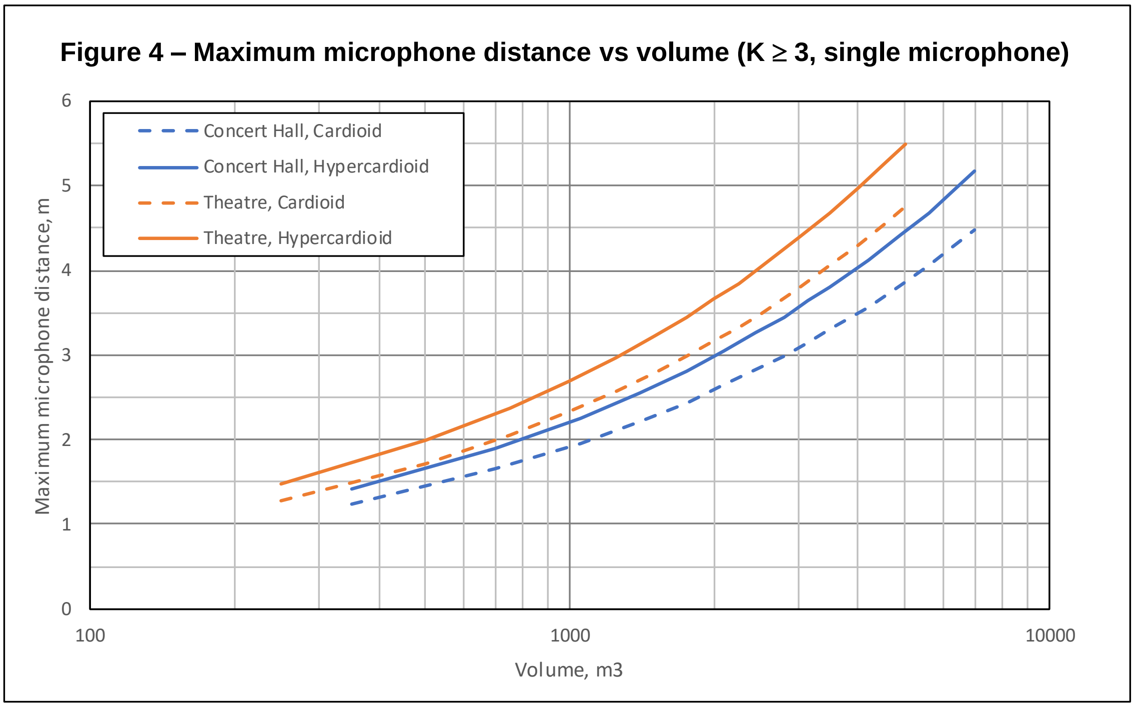
So how might we distil the above criteria into some practical rules of thumb for end users? As acousticians, we sometimes forget that the rest of the populace does not have access to expensive integrating sound level meters which make measuring RT very simple. Similarly, measuring the room volume of a multi-tiered, irregularly shaped auditorium can be a time-consuming challenge. Can general guidance for microphone placement be provided in other terms?

Typical RTs could be assumed based on established design criteria for auditoria for speech and orchestral music. Various textbooks including [10] and [11] set out familiar graphs of mid-frequency RT versus volume for music and speech. While the general shape of such graphs is broadly similar from source to source, the specific figures vary. Figure 3 presents an interpretation of typical RT criteria for theatres (based on speech criteria) and concert halls (based on music criteria).

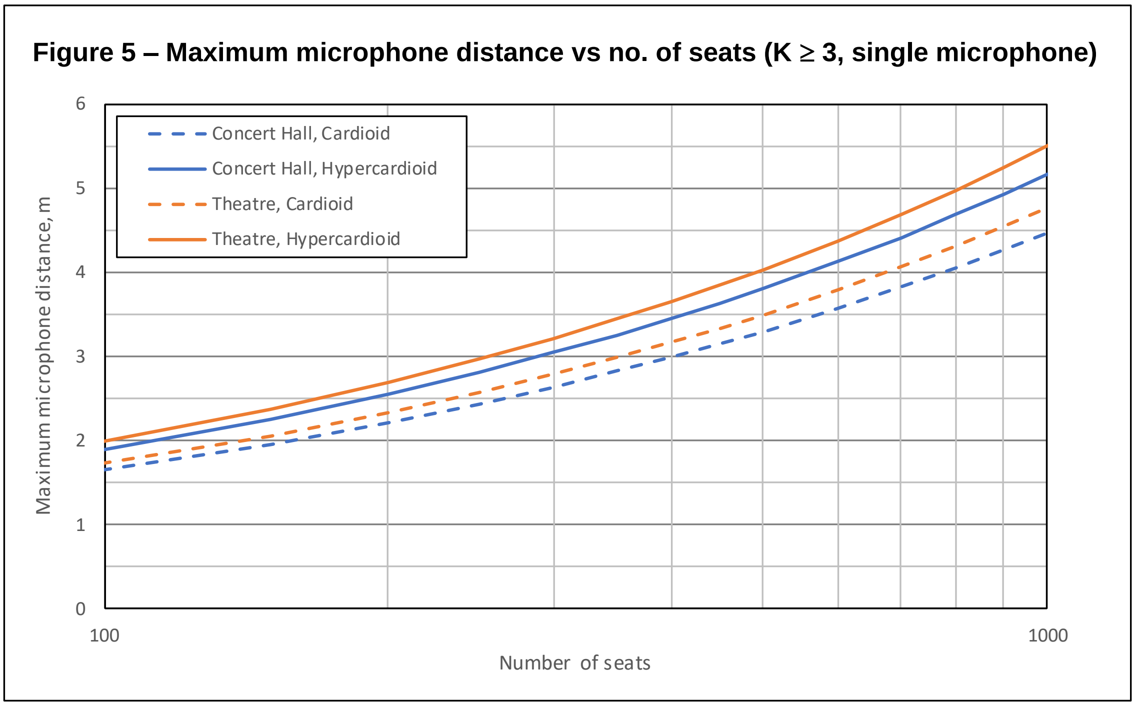
Figure 3 – Assumed typical mid-frequency RT vs volume in auditoria



Based on the assumed RTs set out in Figure 3, Figure 4 sets out the calculated maximum source-to-microphone distances to achieve a direct-to-reverberant ratio, K, of +3dB. Results are presented for a cardioid microphone, with an assumed directivity factor (Q) of 3 and directivity index (DI) of 4.8dB, and for a hypercardioid microphone with an assumed Q of 4 and DI of 6dB[12].



At the risk of over-simplification, this point can be stretched further. If the volume of the auditorium is unknown, we can again use established design criteria to derive an assumed volume per seat and thus express the maximum microphone distance relative to the number of seats in the auditorium. Figure 5 shows the maximum source-to-microphone distances to achieve a direct-to-reverberant ratio, K, of +3dB, assuming a volume of $7m^3$ per seat for concert halls and $5m^3$ per seat for theatres.



It is important to note that range of these graphs is deliberately limited to auditoria of up to 1000 seats. While it may be possible to achieve adequate speech intelligibility with a microphone at a distance greater than 6 metres from the actor in a quiet, unoccupied auditorium, the acoustic signal-to-noise ratio of a raised voice at this distance is likely to fall below the suggested +23dB(A) criterion in an occupied theatre.

The use of multiple microphones requires careful consideration. A pair of microphones near the sides of the stage may usefully pick up an actor's voice over a wider angular range compared to a single central microphone[4], but the reverberant level of the summed signal from both microphones would be up to 3dB higher than that for a single microphone. In practice, both microphones may be receiving speech, depending on the actor's position on stage, so the resulting reduction in the direct-to-reverberant ratio compared to a single microphone may be less than 3dB. Microphones should ideally be positioned as close to the actor's head height as possible. Microphones on short floor stands at the front of stage may provide good pickup of voices but can be susceptible to high levels of footfall noise, even with anti-vibration bases or shockmounts. Suspended microphones can be effective if placed low enough to pick up sufficient frontal sound from the actor. Care must however be taken to position these away from noisy equipment such as projectors and fan-cooled stage lighting fixtures.

While the guidance on microphone placement presented in this section is very generalised and ignores many of the nuances of auditorium acoustics, the resulting recommendations are expected to err on the conservative side. Ideally, the calculated results presented here should be validated with measurements in a range of auditoria.

4 REFERENCES

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