Institute of Acoustics

STEREO RECORDING OF LIVE AMPLIFIED MUSIC

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INTRODUCTION

The fundamentals of stereo recording technique were devised during an era when almost all music was purely acoustic, without use of electronic amplification. Today, however, almost all live music (with notable exceptions in classical and folk musics) uses amplification in live performances. Indeed, much music is inconceivable without amplification - notably that using synthesisers, electric guitars and loud bass guitars, as well as any music using electronic effects and modifications of acoustic sources.

Such amplified music has introduced new factors into live recording. In many cases, it is no longer adequate simply to attempt to capture as accurately as possible the live sound heard by an audience. This is for several reasons:

(i) Amplification equipment suitable for live venues is often necessarily of much poorer technical quality than modern domestic hi-fi reproduction equipment, since it is designed to meet quite different technical constraints.

(ii) In much amplified live music, live reproduction sound levels are higher than would generally be used in the home, thereby changing the perceived quality and effect of the music. In particular, levels that might be acceptable for one-off performances would risk hearing damage if maintained on a regular basis for repeated listening. Also, such high levels in the home would risk excessive neighbour disturbance problems.

(iii) Currently, as discussed by the author in [1], P.A. ("public address") amplification equipment is used in a manner that degrades the intelligibility and presentation of sound to the audience, not because this sound is considered desirable, but because few P.A. operatives have the knowledge, time or technical facilities to get the sound they would ideally like to have.

(iv) Audiences for many kinds of amplified music often make noises that are musically distracting on repeated listening, although capturing some audience response is usually essential if live atmosphere is to be preserved in a a recording.

These factors mean that a good recording of live amplified music would, ideally, capture an idealised version of the live sound in which the inherent technical defects and unwanted losses of intelligibility are reduced and rebalanced for domestic playback, without losing those virtues of the live "ambience" and "atmosphere" that create the unique "feel" of a live musical performance. The basic problem in devising recording techniques for live amplified music is the almost mutual exclusiveness of the two main desiderata : good technical quality, and good "live" atmosphere.

The conflict between these requirements is illustrated by two extreme techniques: (i) recording with a stereo pair of microphones from the middle of the audience - a technique often used to make bootleg live recordings, and (ii) recording with a multitrack machine, putting each instrument (recorded either by a very close microphone or by a direct injection of its electronic output) on a seperate track, along with a pair of room microphones to pick up atmosphere and ambience, all to be mixed down to stereo later in the studio.

The latter technique is the one usually used for commercially-released live recordings of amplified music.

The first technique, competently used, generally gives superb atmosphere and "feel", since it accurately documents what was heard live, but often gives appalling technical quality and poor intelligibility, whereas the second technique generally gives good technical quality and balance, but a very unconvincing impression of the original live atmosphere and feel.

This paper aims to suggest techniques that achieve a better balance between technical quality and the original atmosphere than most current approaches. These techniques are based on a systematic application of psychoacoustic and theoretical results about directional perception of complex multisource sounds, along with suggestions for practical compromises that arise out of the author's experiences in recording literally hundreds of concerts of amplified music in many different styles.

In this paper, we summarise the kinds of problems created by typical amplification arrangements encountered in live music. It is hoped that this summary will, in itself, be useful to many readers who may be unaware of the complexity of much current P.A. practice. The psychoacoustics of the resulting multiple sound images heard by the audiences and microphones will then be discussed, based on the classic Haas or Precedence effect, but with a discussion of the limitations (as well as uses) of these principles in complex polyphonic musical situations. We shall then review the three traditional approaches to stereo recording that were developed in the 1930s, namely coincident, spaced and multimicrophone techniques. We then introduce new techniques based on the previously outlined principles that help to overcome the quality and intelligibilty problems in recording amplified music. Such techniques vary according to the specific problems encountered with specific amplification arrangements.

This paper is based on the thesis that, with rare exceptions, the prime aim of stereo is not to place instruments at predetermined locations, but to provide an enhancement of intelligibility when several sounds play simultaneously, improving a listener's ability to hear subtle details of quiet sounds in the presence of loud ones. We shall place considerable emphasis in this paper on the importance of such multi-sound intelligibility as being musically important, and shall emphasise the role of electronically-induced multiple sound images of each sound source caused by modern P.A. systems.

LIVE AMPLIFICATION AND MULTIPLE IMAGING

There is no single uniform way of applying amplification to live sound, and it is important to distinguish typical different modes which may require different approaches to recording :

(i) Instruments to which no amplification at all is applied - depending on the music, this is usually the louder instruments, such as drums and percussion, but also (in smaller venues) often brass and saxophones.

(ii) Instruments that are very quiet acoustically (e.g. synthesisers, solid--body electric guitars) whose main acoustic output emerges only from a (so--called "combo") amplifier/speaker unit dedicated to that instrument only.

(iii) Instruments that have significant or substantial acoustical outputs which are also fed only to their own dedicated combo loudspeaker unit.

(Sound sources that are purely acoustic or that emerge from a combo amplifier dedicated to a single instrument are generally referred to as the "backline" in P.A. parlance, and such sounds as in cases (i)-(iii) above will be frequently referred to as such in the rest of this paper).

(iv) Instruments (among which we include vocals!) whose sounds are amplified mainly by being mixed in with other sounds and then fed to an overall P.A. system, normally implemented nowadays as a pair of speaker systems placed at either side at the front of the stage or performance area, in front of the performers and backline sound sources. This case may be subdived into: (iva) when the instrument has much quieter acoustic output than its sound emerging from the overall P.A., (ivb) when the instrument also has a significant or substantial purely acoustic output, and (ivc) when the instrument is fed to its own dedicated combo amplifier, which is in its turn amplified via the overall P.A..

(v) Any of the above situations (i) - (ivc) in which additionally the sound of the instrument is mixed with others and then fed to additional "foldback" loudspeakers which are aimed at the performers so that they can hear each other more clearly. Such foldback loudspeakers are usually designed and oriented so as to minimise sound projection towards the audience. Nevertheless, some sound from them usually reaches the audience and, importantly for live recording, tends to become significant at any microphone placed among or above the group. Foldback loudspeakers generally give a less good sound quality than other sound sources mentioned above.

It is quite possible for most of the above possibilities to occur at the same time for different instruments in a musical group. For example, parts of a drumkit may be purely acoustic (i), and parts (ivb) fed, along withacoustically quiet vocals(iva) through the P.A., while bass guitar might be fed only to its own combo amplifier (ii); electric guitar might also be fed to its own combo system (ii), which might then be miked and fed (iii) to the overall P.A. (ivc). If all performers are placed close to the drumkit, drums might not be fed to the foldback, but the vocals almost certainly will be since they would not form a part of the backline easily audible to the performers. If the music is not too loud, a saxophone might not have any P.A. amplification (i), but might nevertheless be mixed into the foldback(v).

In any case, almost any of the 12 basic situations considered in (i)-(v) above can occur, although some combinations will be uncommon in particular musical styles. More complex situations (e.g. involving multiple P.A. arrays fed with different mixes) are sometimes encountered, but the situations described already incorporate most of the difficulties we need to consider. There are also other ways of setting up P.A.s than the conventional front-of-stage stereo arrangement with foldback, as discussed in [1], but for simplicity, we do not consider them further here.

As can easily be seen, a given sound might emerge simultaneously from several sources widely spaced from each other. This leads to a multiplicity of acoustic images for each sound, each generally having a different sound quality and coming from different directions and distances. This makes any use of overall stereo microphones much more difficult than for purely acoustic music, where (apart from natural acoustic reflections and diffractions) all sounds have only one source.

In general, multiple acoustic images degrade the intelligibility of sounds, especially when one is considering the audibility of subtle details of quiet sounds in the presence of simultaneous louder sounds. We shall discuss possible exceptions to this rule in the next section. The basic problem lies in the precise amplitudes, times of arrival and directions from which each replica of a sound arrives at a given position. It is helpful, when choosing microphone position for recording, to bear in mind the basic facts of physical acoustics for the various sources from which sounds arrive.

In particular, sound travels through the air at about 340 metres/sec (so that every metre of distance from a sound source contributes 3 msec delay, and every foot 0.9 msec. Direct sound also propagates according to the inverse square law of energy, i.e. every doubling of distance reduces direct sound energy by 6.02 dB. (It is actually possible to show that these two elementary principles together are actually equivalent to the conventional wave equations for propagation of sound in air, so that they can be used confidently at least for point sources).

It also helps to be consiously aware of the directional pattern of radiation of individual wanted or unwanted sound sources, as this can be used in placing microphones to help reject undesirable sounds and to reinforce desired ones. For example, human voices tend to have strongly attenuated treble from above or behind the person, so that microphones above or behind noisy audience members will be less sensitive to this annoyance. Similarly, most loudspeakers have a rapid loss of treble and even middle frequencies once a microphone is placed sufficiently far off their axis, which again can be useful in rejecting some speaker signals. Also certain acoustic instruments, notably trombone, french horn and trumpet, but also saxophones, are highly directional and so necessitate care in microphone placement if they are to be picked up acoustically.

TIME DELAYS, IMAGING AND INTELLIGIBILITY

There has been a large literature on the subjective imaging of sounds arriving from different directions with different intensities, tonal qualities and time delays. The best-known and most widely used, result [2],[3] is the Haas Effect, or Precedence Effect, whereby, provided that the first sound arrival is not too quiet (typically not below 10 dB quieter than the second arrival), and that the next arrival is delayed by not less than about 4 msec or more than about 30 to 40 msec, then the apparent sound position is that of the first sound to arrive. In principle, this should determine directional effect heard from a microphone position or audience for most amplified music situations. According to the Haas effect, subsequent sound arrivals within a specified time window(which depends on the relative amplitudes of first and subsequent arrivals [2.3]) tend to be perceived as an increase of loudness of the first-arrival sound. However, contrary to the oversimplified use of the Haas effect sometimes encountered, there are important side-effects of even delayed arrivals within the Haas time window which have to be considered. Early reflection simulation programs in some commercially available digital sound effects units,which only add reflections within the Haas time window, would not be considered advantagous by users were such side-effects not to exist or be audible.

The importance of perceived directional positioning (whether live or in a stereo recording)is rarely the precise perceived position of each instrument (this will only be important musically in the rare case that the position of sounds is of structural musical significance, as for example in the 40 part motet Spem In Allium by Thomas Tallis for eight choirs of 5 voices, or Kontakte by Stockhausen for 4 loudspeakers). However, different positions do help the ears more easily to pick out a quiet sound in the presence of a much louder one at the same time - i.e. positioning can be an important aid to improved intelligibility for music with many seperate lines. Moreover, for acoustic music performed in a good live acoustic, it is not merely the directions and the levels of direct sounds from instruments that matters to the ears, but also the levels, directions and times of arrival of all the reflected sounds as well. This additional information, which I have termed "acoustic labelling" of the individual sound sources,is subjectively important. Peter Craven (unpublished communication) has noted that, assuming omnidirectional sources and plane perfectly reflective room boundaries, then it is possible to deduce from the relative levels and time delays of early reflections what the distance of a sound source is; since each reflection provides such information seperately, it is reasonable to suppose that several reflections in a real acoustic environment will allow the ears to estimate distance despite a degree of acoustic absorbtion and departures from perfect plane boundaries. Certainly, experiments by James A. Moorer (private communication) at Bell Labs have confirmed that computer simulations of a few early relections permit perception of distance.

It can be shown by informal experiments that acoustic labelling can permit a sound to be audible and intelligible in the presence of much louder sounds, corresponding to an effective increase in level of the quiet sound of around 16 dB in the case with no acoustic labelling. In other words, acoustic labelling allows the ears to perceive a 16 dB better subjective seperation between sounds.

This is probably the reason why many people [4-6] have advocated the use of simple near-coincident stereo microphone techniques for recording acoustic music in good acoustics. Properly implemented, such microphone techniques can approximately capture the directions, amplitudes and times of arrival of all reflected sounds making up the acoustic labelling.

However, except for cases (i) and (ii) of the previous section (i.e. acoustic--only instruments and amplified instruments whose only sound emerges from a dedicated combo amplifier), amplified music generally generates additional delayed (or even time-advanced!) sounds from different directions quite unlike those encountered in acoustic music. For such sounds, acoustic labelling ceases to work both for live listening and for simple stereo microphone techniques.

In this situation, it is still possible to use the Haas effect to improve intelligibility. In general (and this applies both to live listening and stereo recording), the best intelligibility arises if the first sound arrivals at a listening (or recording) position are not too quiet relative to later arrivals and if every instrument (and, where appropriate, every seperately--sounding part of each instrument) has a different localisation of first arrival. The common (although not universal![1]) P.A. practiceof placing speakers in front of a band means that, except for backline not fed through the P.A., all sounds arrive first at the audience from the P.A., so that every everything's first arrival is from the same position - i.e. the nearest P.A. speaker (unless one is practically equidistant from the two P.A. speakers).

This causes a substantial loss of intelligibility.

Since sound is delayed by 3 msec for every metre of distance, live P.A. results can be improved either by placing the P.A. speakers behind the backline (a procedure termed "post-fade monitoring", since the P.A. speakers also provide foldback monitoring for performers [1]), or else by using conventional P.A. but with the P.A. (and possibly foldback) speaker feeds time delayed by 10 to 30 msec using a high quality stereo digital delay unit. Such a proposal will provide a Haas localisation by the backline, except for sounds only going through the P.A. and foldback. Failing such (usually desirable but often not practical - good delay lines are still expensive) measures, a recordist can attempt to obtain a good spread of stereo positions for first arrivals by positioning the microphones so that the backline sounds arrive first or such that any first arrivals from P.A. or foldback speakers are substantially attenuated by using loudspeaker and microphone polar patterns.

Such a technique can also give significant improvements in the perceived quality of the backline, since most combo amplifiers have a much better subjective quality than P.A. or foldback speakers. This is partly because there is no intermodulation distortion between different instruments, but mainly because combo amplifiers are generally designed as musical instruments to enhance the sound of the instrument they are used with. For some instruments, such as intentionally-distorted electric guitar, the sound reproduced direct from hi-fi speakers is unlistenable, but is often very satisfying through a combo amplifier.

We note several situations in which the Haas Effect is an unreliable guide to perceived results: (i) It tends to fail above 3,500 Hz, where the ears tend to perceive each arrival as a seperate image; this particularly affects instruments with no low or mid-frequency fundamentals, e.g. cymbals, or having very high transient high-frequency output, e.g. soprano saxophones. (ii) Even where one does not locate delayed sounds as seperate sources, one is still aware of the general stereo directionality and spread of such sounds, although the direct sound is clearly localised in the position of first arrival. (iii) The Haas effect notably can also break down if the delayed sound arrives from a direction far from the first arrival, e.g. 180°. (iv) The Haas effect tends to work best on mid-frequency transient sounds. Generally, sustained sounds tend to localise much more vaguely unless special precautions (too complex to discuss here) are taken. As I shall show in a future paper, it is even possible to contrive a pattern of delayed sounds within the constraints of the Haas effect whereby transients clearly localise at one position and sustained sounds clearly localise at another.

CONVENTIONAL STEREO RECORDING TECHNIQUES

There are three basic classes of traditional techniques of stereo recording of acoustic music, all dating back to the 1930s. The first is near-coincident microphone techniques, using a pair of closely-spaced directional microphones pointing in different directions [4-6]. This class of techniques originated in 1931 with A.D. Blumlein [7], and achieves localisation of sounds primarily through the amplitude ratios with which sound are fed to the two speakers, although a small spacing between the two microphones also introduces small interspeaker time delays (generally under 1 msec for commonly used spacings [6]). It is well known [5,7] that at low frequencies (below around 700 Hz), such an amplitude-ratio stereo technique gives good phantom stereo images. However, the quality of such images varies (assuming the use of good microphones with accurately controlled directional sensitivity patterns over a broad frequency range) according to two main factors : (i) the spacing between the microphones, which produces direction-dependent interspeaker time delays, and (ii) the way the stereo pair handles (in terms of amplitude, phase and group delay) the acoustic labelling information caused by reflected sounds in the original environment. In practice, the stereo imaging quality is degraded by the effects of microphone spacings outside the range 0 to 8 cm, where the spacing is such that both capsules look "away from" the other rather than towards it. Capsules "looking towards" each other definitely degrade localisation. For cardioid microphones angled 120° from each other, the optimum spacing for good phantom image quality is around 5 cm as in figure 1. This is because, when reproduced via a standard stereo speaker array subtending a 60° angle at the listener, such an array gives for each front-sector direction a close match to the relative phases and amplitudes at the two ears of a listener occuring with live sound sources up to about 2 kHz. The demonstration of this requires detailed use of models of the propagation of sound from speakers to the two ears, but a similar result has been independently noted for spaced hypercardioid microphones by Olson [8]. Although such 120° angled 5cm spaced cardioids also capture acoustic labelling information from the front quite well, they are liable to cause substantial amplitude errors on acoustic labelling information from the rear. A better technique for handling acoustic labelling is a substantially coincident 90°-angled pair of figure-of-eight microphones [4], as originally suggested (and tried in 1934) by Blumlein [7].

The second traditional class of stereo microphone techniques is spaced microphone technique. Originally developed at Bell Telephone Laboratories in the early 1930s [9], at its simplest, this uses two widely spaced (typically 3 to 6 metres) omnidirectional microphones, one fed to each stereo channel. This technique can give a very "big" and spacious sound with a good sense of depth, but since it attempts to localise phantom images mainly by large interspeaker time delays of the order of 4 to 20 msec, it gives very bad phantom imaging. The Haas effect tends to cause most of the sounds to bunch at one speaker or the other, although sustained sounds can sometimes wander erratically between the two speakers - see our comment (iv) at the end of the last section. The spaciousness of this technique is largely (although not solely) due to the large interspeaker time delays. Nowadays, digital delay effects units are often used in studio popular music recording to simulate similar time-delay effects artificially, and it is clear that the spaciousness thus obtained is not related to any sense of space of the original recording environment.

Unfortunately, spaced microphone techniques tend to lose much of the intelligibility enhancement that stereo can give, because the delayed sounds are not directly related to the original acoustic labelling, and because only two positions are clearly localised, rather than the spread of stereo positions given by nearly coincident microphone techniques. As a stereo technique, therefore, spaced microphones particularly tend to exacerbate the marginal intelligibility of much live amplified music, resulting in a confused muddled sound, whatever advantages it might confer in spaciousness and depth.

However, spaced microphone techniques have one clear advantage over near--coincident techniques when factors such as audience noise or unwanted P.A.

speaker sounds require the use of microphones relatively near the backline sound sources. This advantage is that it is much easier to get a good balance between the levels of different instruments. If a coincident microphone pair is placed relatively near a line of wanted sound sources (see figure 2a), then the inverse square law means that sounds from the middle are picked up much more loudly than those at either end. However, a widely spaced pair (see figure 2b) placed at the same distance but towards each end of the same line of sources will pick up all sources at more nearly identical total energy levels, since the point at the middle most distant from the two microphones is covered by both.

The third traditional class of stereo microphone technique, originally used in recording Walt Disney's Fantasia in 1939 on multitrack film, is the multimicrophone technique, where each instrument or small group of instruments has its own mono microphone placed sufficiently close to largely reject the sounds of other instruments. Each mono sound is then laid on a seperate track for subsequent mixdown to stereo; each mono sound is positioned in the stereo image by using a pan-pot, an electrical means of simulating stereo position by amplitude ratios between the stereo channels. Such a mixdown can alternatively be mixed down straight to a stereo recorder provided one has adequate monitoring conditions and an adequate pre-performance soundcheck to set up the mixing controls. Such a technique completely loses all trace of the original acoustic (and, where relevant, audience reaction), so it is often supplemented by a pair of so-called ambience microphones to sample the sound of the audience and hall, which is then mixed in with the multimic mix.

Unfortunately, adding in spaced microphones tends to result in a very muddled sound due to the time-delay problems of spaced microphone technique mentioned earlier. Even if near-coincident ambience microph ones are used and if the pan-potted images of the multimic mix are laid on top of the stereo images of the stereo pair, any acoustic labelling information from the stereo pair is destroyed by the superposition of mono close-miked sounds on top, since the resulting time-delay and amplitude relationships between first-arrival and later sounds is markedly altered. Passing the multimic mono sounds through delay lines to more-or-less match the tmes of arrival at the stereo microphone can reduce these anomalies, but not eliminate them, although the resulting sound becomes less coloured.

Additionally, the size and spread of real acoustic sources is lost by the use of mono close microphones, reducing the naturalness of sounds. The sound heard by a close microphone in an instrument's near field is often very different to its far-field sound, so that multimic technique tends to distort tonal balances markedly in a way that cannot be compensated for by equalisation. These problems mean that the results are often heard as being very close mono instruments with no individual size, to which is added an unconvincing ambient murkiness. One can either suppress the original ambience totally, substituting for it a synthetic ambience from reverberation units which never sounds much like the original ambience, or one can turn down the multimicrophones so much that one virtually has a slightly augmented stereo pair.

The virtues of multimicrophone technique are well known : it has the ability to sound very tight and punchy and clean, and one has a high degree of control over the precise recorded balance. It also has excellent rejection of exteraneous sound. But it cannot capture a good impression of the original live ambience.

Various hybrids of the above three classes of technique exist, e.g. a coincident stereo pair mixed with a few spot microphones for solo instruments, or mixed in with an "outrigger" spaced pair at low level. Although many such hybrids have particular virtues of space, balance or clarity, they tend not to improve on intelligibility of subtleties of sound, and tend to detract from a sense of the original live ambience.

GENERAL PRINCIPLES OF LIVE AMPLIFIED RECORDING

Based on the above discussions, we can formulate some general principles for recording live amplified music when capturing atmosphere, intelligibility of subtleties and technical quality are all important. These principles have been evaluated via the practical recording experience of many hundreds of concerts of amplified music by the writer, covering a wide range of styles including jazz and contemporary improvised music, pub rock, indie pop, electric folk-rock and folk music as well as much music which is difficult to classify, e.g. avant-garde pop (?).

Recording techniques that best capture the atmosphere and intelligibility of an original live performance vary according to the precise circumstances encountered. We describe what we believe are the most appropriate techniques on a case by case basis.

The first case is that with no overall P.A. or foldback. A simple near--coincident stereo pair of microphones as described earlier is then almost always adequate providing a number of precautions are taken. It must be carefully positioned so as to give a good balance and stereo positioning, and preferably so that any direct acoustic sounds arrive at the microphones before the associated individually amplified sounds, ideally by at least 3 msec. The ratio of direct to amplified sound levels received by the stereo microphone desired should be judged musically on a case-by-case basis, but a first-arrival direct acoustic image helps to enrich an amplified sound even if it is too weak to make the Haas effect work fully, and often a microphone can be positioned so that the ratio of distances to the amplified and acoustic sound sources is much larger than one, i.e.so that the microphone is much closer to the acoustic sound than to its amplified replica.

The second case is when P.A. and/or foldback is used with a group whose backline sounds in themselves would give a more-or-less well balanced sound. In such groups (e.g. many jazz groups), P.A. is merely used to help make the group more audible over a larger space, but not otherwise to modify its basic sound. With such a group, a single near-coincident stereo pair can again often be used, but now placed behind or well to the sides of the main P.A. speakers, and close to the acoustically weakest sound sources (e.g. saxophone and piano for a jazz quartet also using drums and amplified double bass). The effect of the main P.A. speakers on the recording should be attenuated by the stereo microphone being well off the P.A. speaker axes, and closer to the acoustic and backline sources than to the P.A. speakers. The foldback speakers can be attenuated partly by careful positioning of both microphones and speakers, and partly by using a stereo pair fairly high up, typically 2 ft above the saxophone for good balance.

In this application, careful microphone positioning is absolutely essential, and is best done by listening to the group with one's head at proposed microphone positions during a pre-performance soundcheck - otherwise one has to rely on previous experience with this approach, which can yield truly excellent results. In cases where it is impossible to find a microphone position that balances a group adequately, e.g. too weak a piano, one can either modify the sound of the instrument (e.g. by pointing piano-lid reflections straight at the microphone), or one can try using pick-up from the foldback to reinforce this sound. Alternatively, one can try using the new microphone technique described in the next section.

The third case is when recording a system with P.A., but where one has to record down to stereo with inadequate monitoring conditions and without a proper pre-performance soundcheck, so that careful mixing is not practical. This may seem an unfavorable set of circumstances, but is commonly encountered. The only safe method here is to refine the "bootleg" approach of a simple stereo pair from the audience. With conventional stereo P.A., the best stereo microphone position is generally placed symmetrically at that location at which central P.A. images seem central (lateral placement here is critical to within a few cm to avoid lop-sided images due to the precedence effect). The microphones should also generally be positioned as close as possible to the P.A. speakers without going so far off their axis that one loses treble; normally the P.A. speakers subtend about 90° at this position, although this varies greatly according to the dispersion of the P.A. speaker stacks. Normally,one would suspend the microphones over the audience at a height that picks up just the desired amount of audience reaction, using s stereo technique that rejects sounds from the back of the room (e.g. 120°-angled 5 cm-spaced cardioids as in figure 1, or MS microphone technique [6]), so that audience noises from the back are attenuated - those from the front are attenuated by the absorption of bodies!

Alternatively, one can hand-hold the microphone from the body of the audience, recording either on portable equipment or via a cable run under seating (or well taped to the floor) to mains equipment. Although this approach still suffers from poor P.A. speaker sound, and risks audience noise, the careful central positioning can yield a convincing portrayal of any stereo effect conveyed via the P.A., minimises Haas-effect confusions from the P.A. speakers, and (if hand-held) one can use one's body as a shield for audience noise from behind. At its best, this technique can give superb live atmosphere, good balance and good portrayal of backline localisations. As always with stereo pair techniques, the microphones should have accurately controlled broadband polar diagrams for good stereo imagery.

The fourth case considered is when some of the sounds of a group go through a P.A., whereas others emerge only from the backline. Here, providing one is using at least a 4-track recorder so that the most critical mixdown problems can be postponed, the preferred technique is to use a near-coincident stereo microphone to capture the backline instruments, and to take a direct feed from the P.A. mixing desk <u>only</u> of those instruments fed to the P.A. speakers which are picked up by the stereo microphone primarily from the P.A. or foldback. The idea is, as far as possible, to preserve genuine ambience and localisation cues for the backline, and only to mix in close-miked sounds for those sounds whose acoustic labelling has already been substantially destroyed by the P.A.'s mixing together and multiple imaging of sounds. The stereo microphone sounds and the selected P.A. sounds are recorded on seperate tracks and mixed down later.

Within this basic case, there are two alternative approaches to positioning the stereo microphone. The first possibility is to carefully position a stereo microphone in or above the audience as in case three immediately above, following the positioning precautions there described. In this case, the stereo microphone is picking up all of the P.A.'s acoustic output, so that the recorded mix from the mixing desk should contain most or all of the sounds fed to the P.A., but no others. The simplest possibility here is simply to record the stereo output of the P.A. desk on a second stereo pair of tracks for subsequent mixdown, and I have obtained excellent results this way. However, the actual stereo used for the P.A. feed is usually not ideal for subsequent mixdown. For reasons to do with good audience coverage, most P.A. mixes are basically mono, with only subsiduary lines and effects being panned in stereo (although it is common to feed an acoustic piano into a P.A. in stereo). But we have seen that stereo positioning can improve intelligibility, as well as giving a more generally interesting sound, so that it is often worth preparing a seperate stereo version of the P.A. mix for recording purposes. Alternatively, the P.A. engineer can be asked to produce P.A. speaker feeds with a small stereo spread (which will not significantly compromise the P.A.'s audience coverage), and this can be electronically widened (e.g. by increasing the stereo difference gain) either before or after recording on tape. (Widening a recording requires good phase coherence of the recording channels to avoid anomalous results.)

A stereo version of the P.A. mix can also be achieved by a simple passive stereo mix without gain adjustments of post-fade channel outs from the P.A. desk if these are available, so that the P.A. engineer's live decisions on level and equalisation are retained, with only the stereo positioning modified. A stereo version of the mix not only improves intelligibility and interest, but it can also allow a degree of readjustment, rebalancing and repositioning post-tape when mixing the P.A. and stereo microphone signals. Also, those P.A. sounds also emerging from the backline can be positioned on top of the backline positions of the same sounds. For more flexibility, the seperate components of the P.A. mix can be laid down seperately on an 8-, 16- or 24- track recorder for later mixdown, although one might be tempted then towards the multimic approach by also laying down close-mic tracks for backline-only sounds; if these were to be used in mixdown, they would damage the acoustic labelling of such sounds picked up by the stereo microphone.

With this approach, it is also possible to move the stereo microphone further forward, to the point where the tonal quality of the sounds it picks up from the P.A. speakers becomes muffled due to being off-axis, relying on the direct P.A. feeds to restore the lost tonal quality of these sounds.

As one moves the stereo microphone in between the P.A. speakers, one gets to a position where the backline sounds dominate in the microphone pickup as in case two above. One then only need suplement this "backline" stereo sound by those parts of the P.A. mix that are <u>only</u> reproduced through the P.A., so that the recorded P.A. mix will omit those P.A. sounds that are picked up by the stereo microphone predominantly (or on first-arrival) from the backline.

A problem with using a stereo microphone in front of the P.A. is that it is only possible to turn <u>up</u> the levels of sounds fed through the P.A., and one cannot turn up the relative levels of backline-only sounds, since these are not incorporated in the tracks derived from the P.A. desk feeds, but are only picked up by the stereo microphone which also incorporates the live reproduced P.A. sound. Nevertheless, it is often possible to achieve reasonable balances by careful application of gentle graphic equalisation to the stereo microphone tracks during mixdown.

The use of a stereo micrphone behind the P.A. speakers, positioned so as to pick up mainly the backline, avoids this problem, since to a large extent, it does not incorporate the P.A.-only sounds, which can therefore be balanced up or down in relative level. Unfortunately, due to the operation of the inverse square law, it is not always possible to place the stereo microphone close to the backline while achieving a good balance among backline sounds ; often this microphone will pick up the closest backline sound too strongly and the furthest too weakly. Thus this class of live recording technique still has problems of getting ideal balance, largely because one cannot rebalance that information which has been recorded with proper acoustic labelling,but only that whose acoustic labelling has been destroyed by the P.A.. In the next section, we describe a new microphone technique designed to reduce some of these problems.

In mixing down down 4- or more track recordings made by these techniques, it is wise to avoid the use of any artificial reverberation or delay effect, since these will mask the original acoustic labelling information on the backline, but to rely entirely on the original live acoustic (although any artificial delay or reverb effect used in the original P.A. can still be used since it is already present in the live sound - however, it is often found that the levels of effect used in the P.A. often sound excessive under domestic conditions unless turned down). The direct P.A. feed tracks can often be usefully subjected to a degree of dynamic processing, notably compression to control vocal peaks, and expansion of low-level signals to diminish the effect of low-level buzzes and noise often encountered in P.A. outputs.

One very useful adjustment that can markedly improve such mixdowns is to insert a high-quality stereo digital time delay into the stereo P.A. mix tracks, so as to delay it to (more-or-less) match the (later) time of arrival of sounds at the stereo microphones. Experience suggests that relative delays of 20 msec or more on the stereo microphone tracks tend to give the mix an unpleasant coloured quality due to the comb-filter effects caused by adding delayed and undelayed signals. This unpleasant coloration is also quite often heard on commercial live recordings due to time delays between "ambience" and close microphones. These comb filter colourations are much less objectionable when the relative delays are reduced, and it then becomes practical to mix P.A. mix tracks and stereo microphone tracks at comparable levels without coloration becoming unacceptable.

SPACED STEREO MICROPHONES

We observed above that our preferred techniques for recording P.A. had problems in optimising backline balance. We also noted in our earlier examination of spaced omni microphones that, when used fairly near a backline, they could achieve a better balance than could near-coincident stereo pairs. We describe here a new microphone technique that retains this latter advantage while giving an improved stereo image. This is achieved by replacing each of the two spaced omnidirectional mono microphones by a coincident stereo microphone pair. There is, of course, nothing new in the idea of mixing together the outputs of more than one stereo microphone pair (e.g. see [10]), but our proposal has several unusual features which, when combined, lead to improved results.

First, we propose that the stereo pairs be placed symmetrically at each side of the axis of left/right symmetry of the performing stage, just as is done with the spaced omnidirectional technique. Secondly, we propose to adjust the stereo pickup of each pair so that, over the main area of wanted sound pickup, they both pick up sounds at roughly the same stereo positions. This ensures that the two stereo images (one usually time-delayed relative to the other by the extra distance sound has to travel from the source to that stereo microphone) are roughly compatible, so that one will not get the poor stereo imaging of spaced omnidirectional microphones.

Thirdly, we propose to use, for each of the two stereo pairs, highly asymmetric microphone pairs - i.e. the left microphone of each pair has a different polar response than the right. Previously described uses of stereo pairs have been with left/right symmetrical stereo microphones. However, we shall still retain overall left/right symmetry by requiring that the right stereo microphone arrangement be a mirror image of the left stereo microphone arrangement.

Overall left/right symmetry is considered important in stereo microphone techniques, since otherwise, no matter how favorable might be the balance and positioning of direct sounds, the acoustic environment is heard as being reproduced asymmetrically, with a tendency to hear more "acoustic" from one side of the stereo image than the other.

The technique we propose is designed for maximum rejection of sounds behind the microphones, so as to minimise excessive direct pick up from such unwanted sounds as P.A. or foldback speakers or audience noise. We use a pair of MS stereo microphones, but not in their usual left/right modes. More precisely (see figure 3), we place two stereo pairs (usually on stands $1\frac{1}{2}$ to 3 metres high to avoid acoustical obstruction of wanted sound sources) around 3 metres apart (the precise spacing will vary on a case-by-case basis). The "left" stereo microphone consists of a forward-facing cardioid microphone M_L (whose polar amplitude response for sounds arriving from an angle θ measured anti-clockwise from due front is $1+\cos\theta$), and a rightwards sideways-facing figure-of-eight microphone S_L (whose polar amplitude response is $-\sqrt{3}\sin\theta$). Similarly, the "right" stereo microphone consists of a forward-facing cardioid M_R (with polar pattern $1+\cos\theta$) and a leftwards-facing figure-of-eight microphone S_R (polar pattern $+\sqrt{3}\sin\theta$). It has been found that for best portrayal of the sense of acoustic space, the gains of the figure-of-eight microphones (in their directions of maximum sensitivity) should be approximately $\sqrt{3}/2$ (i.e. - 1.25 dB) relative to the maximum gains (at the front) of the cardioids. This can partly be explained by computing the effective stereo distribution of reflected reverberant sound as described in [11] for this technique; the reverberation is captured roughly equally for all in-phase and out-of-phase stereo positions.

One derives the recorded left and right channels L_T and R_T respectively by mixing the two pairs thus :-

$$L_T = M_L + S_R \quad , \quad R_T = M_R + S_L \quad .$$

If the S-microphones were to be faded out, this would simply become a spaced pair of forward-facing cardioids, and the Haas effect tends to cause sounds closest to one microphone to be reproduced from that channel. However, with the S signals present, the Haas effect causes sounds away from the centre of the stage to be positioned mainly by the closest stereo pair. For example, if a sound is left-of-centre on the stage, it is positioned by the M_L and S_L microphones. A sound arriving from the front of the left microphones is reproduced at pure left, whereas a sound arriving 60° clockwise from due front (see figure 3) from towards the centre of the group is recorded and reproduced at stereo centre. Sounds beyond this angle but positioned live left of stage centre are reproduced from the right side of the stereo image. Sounds arriving from ouside the area between the two stereo microphones (see figure 3) are reproduced as antiphase stereo images. Thus it is normally important to confine important direct sound sources within this intermicrophone area. The M_R and S_R stereo pair position sounds similarly for sources on the right side of the stage area.

This spaced stereo-microphone technique has many advantages : each stereo pair rejects sounds from behind the microphones (as the nulls of both the M_L and S_L microphones lie in the same direction behind the microphones), and sounds from nearly behind are reduced in level and panned over towards the opposite channel (i.e. for the left pair, they are panned to the right and vice-versa), so that audience sounds (and unwanted P.A. and foldback speaker sounds) are attenuated but have a wide spacious stereo spread. Each individual stereo pair has a roughly uniform total energy gain for sounds from anywhere in their frontal 180° sector, meaning that for stage sounds, the spaced pair of stereo microphones gives almost the same balance of levels as an identically-situated spaced pair of omnidirectional microphones.

For sounds near the middle of the stage area, there is generally (except for the point X in figure 3) some discrepancy between the localisations they are given by the two stereo microphones, but such differences tend to create a not unpleasant spread image just covering the range between the left--microphone and right-microphone stereo images. This is a lot less disturbing than the huge uncertainties covering the whole stereo width encountered with spaced omnidirectional microphones. There is some comb-filter coloration produced by the superposition of two stereo images of each source, captured with differing times of arrival, but this has actually not been found too serious in practice. This is partly because sounds to one side are picked up predominantly by one stereo microphone only, and partly because the centsl image "spreading" tends to make any coloration more acceptable.

A more serious problem is that sound sources more-or-less between the two stereo microphones are actually reproduced at the wrong side of the stereo stage. This does not matter much if the sources are foldback speakers and the like, but need attention if any direct sound sources occur in this area. This anomaly can be reduced by reducing the gain of the S signals, but this would reduce the overall spaciousness and would result in widely spread images towards the back of the stage area, since the stereo positions of rear-stage sources would differ markedly for the two stereo microphones.

Like all widely spaced microphone techniques, that of figure 3 tends to distort the original acoustic labelling information of the original environment. However, unlike conventional spaced microphones, the good stereo imaging tends to enhance intelligibility beyond that heard by the more distant live audience. The relatively close microphone positioning used with this technique means that it portrays well the stereo size and distribution of individual large sound sources, so that, for example, one hears each part of a drumkit clearly seperated from the others. Additionally, because each stereo microphone individually captures the acoustic labelling of sounds near it reasonably well, one finds that that part of the acoustic labelling caused by the earliest reflections (from the floor, ceiling, the back and the near-side wall) is actually preserved for those sounds picked up predominantly by one microphone. This appears to be enough for the ears to still benefit, in particular in its ability to perceive distance, but also in improved intelligibility, for sounds at the two sides of the stage.

The antiphase pick-up of reflections from outside the stage area with this technique undoubtedly helps to create the spaciousness of this technique due to the sense of space beyond the stereo speakers. Indeed, it is instuctive to compare this technique with simple spaced forward-facing cardioids,by fading the S signals up and down, since it reveals a far more convincing spaciousness than that given by spaced microphones.

A minor modification of the spaced stereo-microphone technique can cope with backline sources that occupy a different width to that between the stereo pair pairs. This is simply to "toe" the two stereo pairs in or out, so that they are each pointing at their respective edge-of-stage sound-source, as illustrated in figure 4.

If one has a multitrack recorder, one can lay each of the signals M_L, S_L, S_R, M_R on 4 seperate tracks for later fine adjustment of mixdown, although recording $L_T = M_L + S_R$ and $R_T = M_R + S_L$ onto 2 tracks works well, since this combination usually has the best sense of naturalism for reasons described above. Owing to its good balance, good rejection of audience, P.A. and foldback, this technique is well suited to recording the backline, and as in the last section, one need only lay down on tape tracks for those P.A. sounds going <u>only</u> to the P.A., e.g. vocals. The results, recorded and mixed down from a 4 track recorder, can closely approximate the ideal of a good intelligible sound with a good sense of live atmosphere.

Apart from discussing rejection, we have in the above made very little reference to how audience sounds should be captured. However, it is found (certainly in venues with audience capacities of around 200) that audience pick-up from the rear of the microphones is enough to capture a good atmosphere, often spectacularly so. A part of the reason for this may lie in the fact that this technique picks up the very front of the audience. The front of the audience tends to be those people most actively interested and involved with the music, so that their audible reactions tend to be more in keeping with the music than that of members of the audience at a greater distance, such as is usually captured by "ambience" microphones.

APPLICATIONS TO SURROUND-SOUND

Most of the techniques of this paper can be extended to the recording of live music using Ambisonic surround-sound technology [12-14]. This can be done by replacing stereo pairs in the above by a Soundfield microphone, and replacing the stereo P.A. mix by a front-sector surround P.A. mix using the technology described in [13] and [14]. However, it is worth observing something that has not, as far as I am aware, been explicitly noted in the literature except in an inaccessibly mathematical form : namely that an MS microphone, with sum signal with cardioid polar diagram 1 + cos θ and with difference signal S with gain sinθ such that sounds to either side of the microphone are recorded in their respective stereo channels without crosstalk, already is a surround-sound recording technique; it encodes direction according to the Japanese Regular Matrix surround-sound specification , although sounds from the rear are attenuated. Any MS microphone with a 90° setting will record in this mode. When fed to a UHJ transcoder [13],[14], such a microphone gives a good approximation to UHJ surround-sound encoding, but with the often-desired attenuation of unwanted rear sounds.

Thus if one uses an MS pair as the main stereo microphone, one can remix the recording for UHJ using a transcoder, and it will give a true surround-sound effect. It is also possible to adapt the spaced pair of MS microphones described in the previous section to surround-sound use, but the full theory is more comlex than we can go into here. Nevertheless, the techniques described in this paper need not be confined to stereo, but can be developed for full surround-sound.

CONCLUSIONS

The complex and ever-developing techniques of live amplification technology means that there is never going to be a perfect all-purpose technique of live recording of amplified music, and one is always going to need to use musical judgements and compromises in recording. However, we have shown that there are several principles that often allow improved results over standard practices. In particular, the use of mixing and of several microphones does not mean that we have to abandon the use of "acoustic labelling" information that helps the ears place the sound in a live acoustic, and which also helps to improve intelligibility. The awareness of the basic principles of physical acoustics and of directional psychoacoustics often allows appropriate techniques to be devised, particularly the precedence effect and the use of clear stereo seperation between distinct sounds.

Suitable recording techniques can be devised that work well with a 4-channel recorder - I often use a Sony 501 digital processor with a Beta Hi-Fi video machine, recording the stereo microphones on the digital tracks and the recorded P.A. mix on the Beta Hi-Fi tracks, although this particular method gives an 11 msec time-delay on the stereo microphone outputs unless delay compensation is used. However, these techniques can be used with more post-production control using machines with more tracks, although in general, acoustic labelling cues tend to be adversely affected by noise reduction systems.

Even if the particular methods suggested above do not meet particular requirements, it is hoped that the discussion of the psychoacoustic and physical problems of live amplified recording will stimulate others to devise new techniques overcoming the problems of existing approaches.

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FIGURES

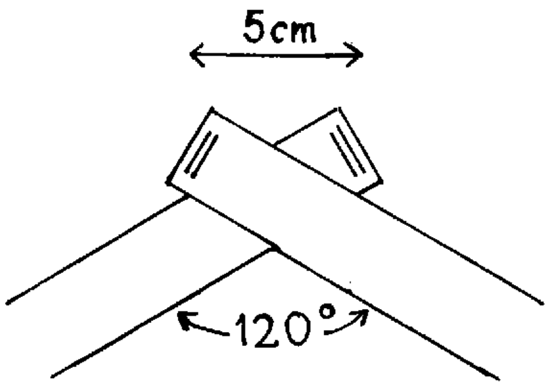

Figure 1. 5 cm-spaced 120°-angled cardioid microphones

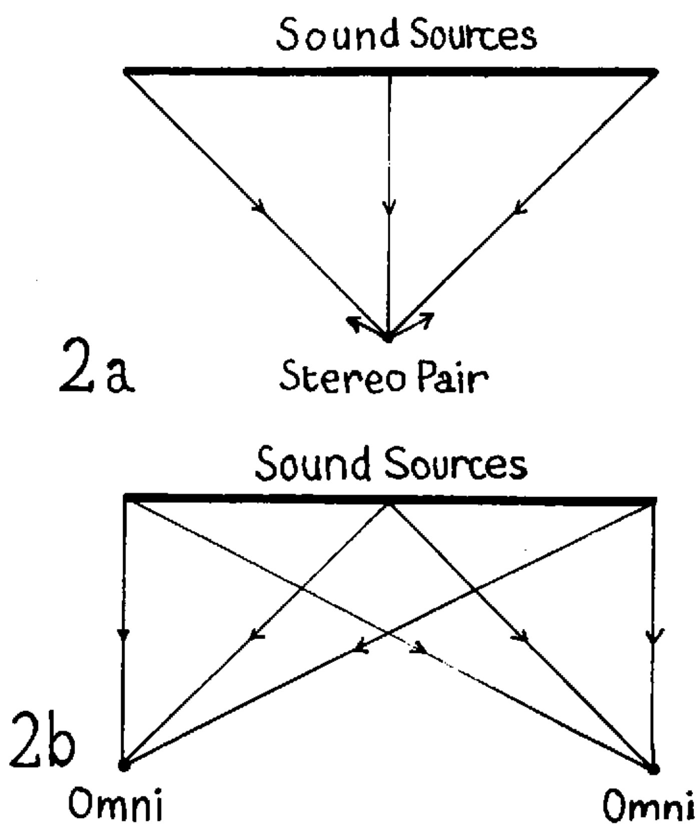

Figure 2. Operation of the inverse square law, giving poor balance for a fairly near coincident stereo pair (2a), and good balance for similarly near spaced omnidirectional microphones (2b).

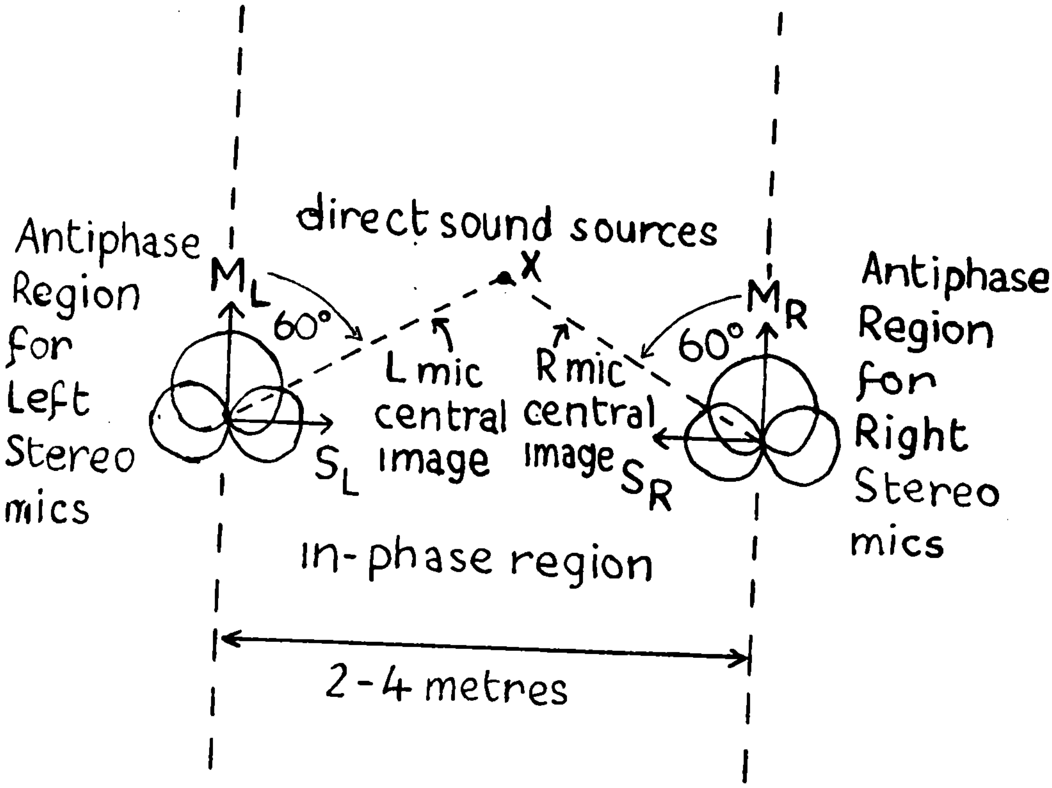

Figure 3. Spaced stereo-microphone technique.

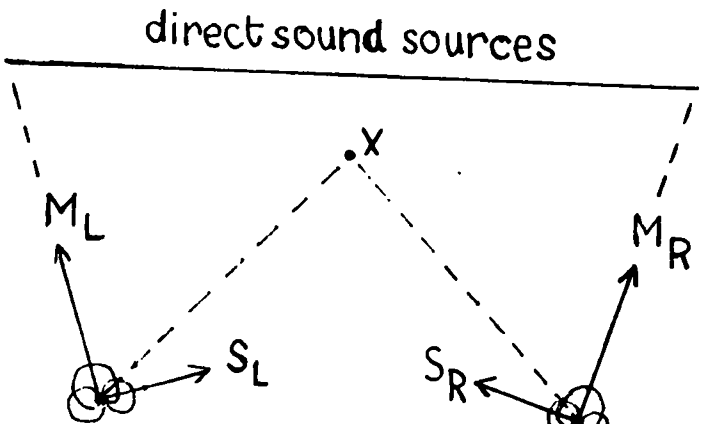

Figure 4. Toeing out the stereo pairs to cope with a sound-source stage wider than the microphone spacing.