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Active Acoustic Systems for the Control of Room Acoustics

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ABSTRACT

The acoustic design of auditoria involves the specification of the room geometry and boundary properties, and any additional acoustic elements such as reflectors or diffusers, to usefully direct sound to produce a desired subjective experience, quantified by measurable acoustic parameters. This design must take into account the reflection of sound within the stage area, the early reflections from the stage to the audience and the reverberant response of the room. The sound produced by the audience can also be an important consideration. Active acoustic systems provide an alternative approach to controlling subjective experience. They use microphones, electronic processors and loudspeakers to create reflections and reverberation in addition to those produced by the naturally-occurring sound field. The acoustic properties can be changed instantly, and the enhanced acoustic properties of the auditorium can typically be varied over a wider range than can be produced by variable passive techniques. The design of active acoustics follows that of passive approaches, but rather than the physical arrangement of the room surfaces, it commences with an existing passive space with some minimum acoustic condition, and requires the arrangement of microphones to detect relevant sound and the choice of processors and loudspeaker positions to direct it usefully back into the room to produce a desired set of acoustic parameters. While active systems have historically been developed with the goal of enhancing either the stage or audience sound, they must generally provide the same control of sound as passive acoustic design. This paper discusses the principles of active acoustic systems and how they are used to achieve the required range of control. A survey of current commercial systems is given and some implications for the future of live performance are explored.

INTRODUCTION

The acoustical characteristics of any room used for performing live music have a significant impact on the subjective impression of the performance. The room surfaces reflect sound generated by the performers onto the audience, producing, ideally, an enhanced subjective experience of the performance. The acoustic design of a venue is therefore an important component of its commercial success.

The study of subjective impressions of sound quality has lead to the development of a number of measurable acoustic parameters that allow acousticians to predict the subjective quality of a venue's acoustics [1-4], and which can be determined from impulse response measurements [5,6]. For example, reverberation can produce a sensation of "fullness of tone" that is desirable for some types of music, and is quantified by the reverberation time, (*RT*) [1]. The early decay time (*EDT*) is important since only the early part of the reverberant decay is audible in continuous music, the late reverberance being audible only during periods of silence. The time between the direct sound and the first reflection (the initial time delay gap) quantifies the sensation of "acoustical intimacy" [1].

More recently, measures that account for the spatial distribution of early and late energy arriving at the listener have been developed. These may be measured from impulse responses using pairs of directional microphones or a dummy head [2,7]. The spatial properties of the early sound governs the sense of broadening of the sound source without altering its localization (the Apparent Source Width *ASW*) [8-10]. The *ASW* can be quantified by the early lateral energy fraction [9]. Early sound arriving from directions other than lateral also contribute [9,11]. Early energy is also important for speech intelligibility [12].

An alternative measure of *ASW* is the inter-aural cross-correlation coefficient (*IACC*) of the early part of the binaural impulse response ($IACC_E$), [1,3,13,14]. The binaural quality index $BQI = 1 - IACC_{E3}$ increases with spaciousness and is a good predictor of subjective impression [1].

The detailed pattern of early reflections is also important to subjective quality. Typically there should be a relatively large number of reflections arriving at uniform times. This feature is termed texture [1]. Texture will be influenced by the amount of diffusion in the hall [15].

The spatial properties of the late energy arriving at the listener leads to the impression of being enveloped by the sound. The Listener Envelopment (*LEV*) is well correlated with the level of late arriving lateral energy at the listener, although *LEV*, like the *ASW*, is also increased to some extent for other directions-of-arrival of late energy [16-19].

Listener envelopment and Apparent Source Width are complementary parameters that relate to the balance of early and late energy in the hall. *LEV* tends to be higher than *ASW* in many halls [20,21].

Many of the acoustic parameters developed to quantify acoustical quality relate to the same subjective impressions. Therefore, minimal sets of independent parameters have been sought which best quantify the room acoustics [22-24]. For example in the analysis of concert halls, [22], *BQI*, EDT_{mid}, strength factor at mid frequencies G_{mid}, strength factor at 125 Hz G_{125}, surface diffusivity index *SDI* and initial-time-delay gap *ITDG* were suggested, and texture and late lateral strength were mentioned as new potentially useful parameters. In the assessment of opera houses [23] the five parameters RT_{mid}, BQI_{E3}, *ITDG*, G_{mid}, and Bass Ratio, *BR*, were suggested. The measurement of many of these parameters is now standardized [25].

The acoustics produced on stage are important for performers [26-33]. Musicians require a balance between the sound of their own instruments and that of the other performers instruments [28,30]. Acoustic parameters which quantify the quality of stage acoustics have therefore been developed [31,32]. For example, Support is a measure of early to direct energy one metre from a source which describes to what extent the early reflections assist a performer's playing.

PASSIVE VARIABLE ACOUSTICS

In many venues, a variety of performance types are hosted, and these require different acoustic conditions for an optimum audience experience. Musical styles have developed with the changes in hall designs [1,34] and modern multi-purpose auditoria would ideally recreate the acoustic conditions best suited to the style of performance [2]. With the proliferation of modern music forms such as pop and rock, the required range of acoustic conditions is even greater, particularly due to the higher levels of bass sound generated by electronic instruments. Clarity at frequencies at and below 125 Hz is important and a flat *RT* curve is desirable [35]. The acoustics below 125 Hz is also considered important in classical venues [36]. The development of multi-purpose venues must therefore include a consideration of whether the acoustics should be variable.

Variable acoustics can be provided by passive means, provided that the physical changes are significant enough to produce noticeable effects [2,36-43]. The reverberation time of an auditorium can be altered by varying its absorption or its volume. Varying the room volume will provide control of *RT* without a significant change in loudness. Reducing the reverberation time by increasing room absorption will also reduce loudness which may be undesirable [2]. Alternatively, reducing absorption in small halls to produce reverberation times sufficient for chamber music may produce excessive loudness [37,38]. The positioning of absorption elements will also allow control of the early energy and hence parameters such as clarity and lateral fraction, but in practice this can be hard to achieve [2].

Absorption can be altered by using moveable curtains or rotating panels with different absorption on each side [1,2,39]. Movable reflector panels can be used to direct early energy onto the audience, but their effectiveness is dependent on their size [2]. Significant variation is possible in cases where a large percentage of the room surface can be varied. For example, the Varechoic chamber at Bell Labs produced a reverberation time variation from 0.1 to 1.6 seconds [40].

Variable diffusion can alter the ratio of early-to-late energy in a room by dispersing reflected sound so that the energy in the specular reflection is reduced. The addition of diffusers does not significantly affect the room absorption and can change the structure of impulse responses without altering their total energy [2,44]. Many recent diffusers have been developed using the properties of number sequences such as quadratic residue, primitive root (which can suppress the specular reflection completely) and maximum length [45].

Room volume can be altered by using roof space above the ceiling [2,41], or by using reverberant chambers with variable openings [1,46,47]. The use of specially constructed chambers allows the additional acoustic space to be optimized for acoustic use. If the additional volume is poorly coupled to the main room the sound decay in the auditorium becomes that of a coupled room [48-50]. To achieve a greater variation in volume, many designs use multiple coupled spaces, which increases the coupling and the resulting room *RT* [51]. By varying the coupling area and the absorption in the secondary space(s) a variety of double-sloped decays can be produced which alter the early decay time of the room [52,53]. One of the claimed advantages of coupled systems is that they can maintain clarity with late reverberance by the deliberate use of double-sloped decays [46].

A combination of absorption and volume variation can produce a useful range of acoustic conditions. For example, the Concert Hall in Lucerne provides both variable volume and absorption, and can alter the mid-frequency *RT* from 1.5 to 2.15 seconds [1]. The Espace de Projection at IRCAM uses both volume and absorption variation to provide a variation in *RT* from 0.5 to 2 seconds [2].

The early energy properties in an auditorium can be altered by controlling the radiation of sound energy from the stage area, for example by the use of stage shells that increase the early energy and block off the fly tower. However, this can also affect the late energy. For example, altering the stage ceiling in one hall to increase stage reverberation also increased the room *RT* by 10 to 15 % [42]. Passive techniques have also been applied to stage acoustics to provide variation of ensemble and support [26,27]. However, significant mechanical changes are also required to produce noticeable changes in stage acoustics [27,43].

To summarise, passive variation of acoustics can produce useful variations of room acoustics, but requires significant variation to give appreciable acoustic change, which requires considerable time for alteration, or large mechanical actuator systems. In addition, many passive variable venues do not offer much variation of bass *RT* which would be required to accommodate both classical and modern music [35,37,39].

ACTIVE VARIABLE ACOUSTICS

The alternative to passive variable acoustics is active acoustics, in which sound in the auditorium is detected using multiple microphones, processed electronically, and broadcast back into the room via loudspeakers [54-63]. Active systems allow many of the physical limitations associated with passive acoustics to be overcome. Sound can be distributed from the room surfaces with reduced delays and larger amplitudes than passive reflections, limited by the acoustic feedback from the loudspeakers to the microphones. Loudspeakers can produce reflections over a wide frequency range and bass energy problems, which can occur with small passive reflectors, can be eliminated. Active systems can be instantly altered to a number of pre-configured settings implemented in software, which can be more reliable than the mechanical control required for passive variability. The prediction of the acoustics produced by active systems is less critical than that of passive fixed acoustics, as the parameters of active systems can be easily altered after installation, provided that the number of transducers and their layout is sufficient to produce the required range of acoustic conditions. However, prediction is also possible if passive acoustic responses can be determined from scale or computer models [57,64].

Since active systems increase the energy in the room, a multipurpose hall intended for active enhancement would be built for a minimum energy operation (ie with minimum reverberation time and strength, suitable for speech, for example) and the active system would allow the energy to be increased for other performance types. Modern active systems can typically produce *RT* gains of 2 or greater and most offer enhancement of the early energy, so that a wide range of performance types can be accommodated.

Background

The principles of electroacoustics for sound distribution, and the risks associated with feedback in a single microphone/ loudspeaker "channel", were well-established by the 1960s [65-73]. The earliest applications of electroacoustics to the enhancement of room acoustics known to the author began in the 1950s. For example, H. Olsen reported a system for enhancing room acoustics using magnetic tape and acoustic tube delays in 1959 [74], and compared the properties of passive and active acoustic systems in 1965 [54]. R. Vermeulen developed a system for enhancing stereo reproduction using a magnetic delay wheel to produce reflections [75]. This system was applied to acoustic enhancement in several halls including La Scala Theatre, Milan. G. Dutton developed a similar system for EMI in 1966 [76]. Other early systems are described in [77-81].

The technological limitations of the time meant that the audio quality in these systems was not ideal [79]. For example, Vermeulen's system was disabled after three years [59]. Barnett noted in 1988 that the use of active systems was in decline due to the "failure of these system to meet the expectation of the recipients" [55].

Modern active acoustics systems are for the most part similar in their design to early systems, but have benefited from the increased quality and reliability of audio components [61,62]. Microphones are now available with very low self-noise levels and with flat responses over a wide frequency range. The reliability of power amplifiers has been improved by the use of improved self-protection circuitry, increased integration, or by the use of techniques which reduce heat dissipation such as power supply variation (class G) or pulse-width modulation, sigma-delta or other self-oscillating switching designs (class D) [82].

Loudspeakers are available with relatively flat response characteristics, and the reduction of loudspeaker failure from overheating and overstressing at high sound levels has been achieved by improved thermal design, or by the integration of the loudspeaker driver with the amplifier which allows confinement of the driver signal to safe levels. Some powered loudspeakers have the facility for remote monitoring.

The use of oversampling and optimal dithering in analog to digital and digital to analog conversion means that digital processing is now equivalent to analogue processing with additive noise, with low phase distortion and with quantisation noise independent of the signal [83], and signal to noise ratios of modern convertors exceeding 90 dB. The use of floating point processors eliminates dynamic range and scaling issues in digital filtering, and time delays of any value can be implemented using interpolation [84]. In summary, the digital processing of acoustic signals is effectively linear, with noise levels which are low compared to acoustic background noise.

In-line and Non-in-line Systems

There have been two main approaches to the design of active systems [56]. The first developed from sound reinforcement systems which use a small number of directional microphones close to the stage area to maximise the direct-to-reverberant sound ratio. These *in-line* or *non-regenerative* systems create early reflections and reverberation from the sound sources on stage and minimize sound feedback to the microphones. Some of the earliest in-line systems were the Acoustoelectronic Auditorium developed by Olson [74], the Ambiophony system [75,87] and the patents of Graham [80] and Veneklasen [81]. More recent examples are the Delta Stereophony system [85,86] and the system in [78].

More recently, Jaffe *et al* developed some of the first digital delay systems for implementing early reflections and discussed methods for reverberation enhancement that avoided regeneration of sound, such as the Reverberation On Demand System [55,88].

The LARES system is one of the first of the current commercial systems [89-93]. It was originally developed by D. Griesinger using Lexicon time-varying reverberators to correct acoustic deficiencies in the Elgin theatre, Toronto [89]. LARES uses a small number of microphones close to the stage, and a large number of loudspeakers to achieve consistent sound distribution. The time variance is designed to minimize pitch shift artifacts. LARES has also been applied to outdoor venues [93].

The System for Improved Acoustic Performance (SIAP) was developed in the Netherlands for improving room acoustics while maintaining a balance between the visual and acoustic perceptions of the space [94-97]. The system uses a small number of microphones (typically 4 for a medium size installation) and has time-varying digital processing to control colouration, although the time-variation is not always used.

The Acoustic Control System (ACS) was originally developed as an application of wave field synthesis, based on the Kirchhoff Helmholtz integral [98-100]. Wave field synthesis is a method for sound field synthesis or reproduction [101,102], but can be combined with a microphone array to sample and modify the sound field generated by performers on stage. ACS uses an array of 18 to 24 microphones covering the stage area, and digital processors to generate early and late reflections for a number of loudspeaker outputs.

The Vivace system is an in-line system recently developed by Müller-BBM [103,104]. It uses low-latency convolution to implement reflections and reverberation. Time-variation is employed to maintain stability.

In-line systems can provide high levels of direct sound and early reflections to listeners, making them ideal for controlling early energy. They can also provide reverberation enhancement for sound sources on stage. However, natural reverberation occurs for sources at any position in the room, contributing to room ambience and the audience's experience of the hall acoustics. In-line systems do not provide this global enhancement of *RT*. This can be an advantage since audience noise is not enhanced. However, global reverberation enhancement is required for audience participation such as congregational singing in churches, and is important for supporting the audience's response to a performance.

The alternative approach to in-line systems is the *non-in-line*, or *regenerative* system, which uses microphones distributed around the room to enhance reverberation time [56]. One of the earliest of these was the Assisted Resonance system, designed to increase the low-frequency *RT* in the Royal Festival Hall by reducing the damping of individual room modes [105-112]. For each mode, this was achieved using a microphone placed inside a Helmholtz resonator connected to a loudspeaker via a phase control circuit which allowed the generation of positive feedback at the mode frequency [108]. In the Festival Hall installation 172 channels were used to cover a frequency range of 58 Hz to 700 Hz, producing an *RT* increase from 1.4 to 2.5 s in the 125 Hz octave band [106]. Subsequent systems were developed using wider band channels which allowed a reduction in their number [108].

An alternative approach was the Multichannel Amplification of Reverberation (MCR) system, which used multiple wide-band microphone-loudspeaker channels to reduce room absorption and increase reverberation time over a wide frequency range [112-117]. In the MCR approach individual room mode control does not occur. Instead, the effect of the channels on room modes is random, with some mode dampings reduced and some increased, the net effect of which is an increase in reverberation time and an increased variance in mode damping factors [118]. As the number of channels increases the variance in mode dampings reduces, creating more linear decays at the enhanced *RT*, and higher sound quality due to a decrease of modes with low damping factors compared to the mean. The MCR system does not require phase adjustment, but does require equalisation of all channels to ensure that the enhanced reverberation time is a smooth function of frequency [112,114-116].

A system that includes a regenerative component similar to the MCR approach is the Meyer Sound Constellation system, (developed from the VRA system), which, unlike MCR, includes a multichannel reverberator between the microphones and loudspeakers [118-129]. This produces an electroacoustically coupled room, and its behaviour is similar to that of passively coupled rooms [125]. The Constellation system produces a reverberation time gain which is greater than the steady state sound power gain, which allows reverberation enhancement at lower loop gains. This reduces colouration, loudness gain and amplification of the room background noise. The reverberator is time-invariant, and has a unitary property which is the multichannel equivalent of an allpass filter, so it does not degrade the stability characteristics as non-unitary reverberators would [122]. The Constellation system also includes an in-line early reflection system in which microphones close to the stage are processed by a time-invariant, unitary early reflection generator. This hybrid system aims to maintain the global property of reverberation and the local property of reflections from the stage area.

The Yamaha Active Field Control (AFC) system is a regenerative system that uses digital, time-varying finite impulse response (FIR) filters to increase the echo density of the regenerated sound [130-132]. The filters implement a multi-tap delay line and the delay of each output is varied with its own frequency modulation and time range [131]. In addition, an "Electronic Microphone Rotator" is used in which the routings from the microphones to the loudspeakers are varied in time, producing a form of spatial variation of the room transfer function matrix. This technique further reduces the risk of instability. The AFC system also uses microphones close to the stage area processed by time-invariant FIR filters to allow control of early reflections [131].

An alternative non-in-line approach arose from the idea of controlling the impedance of the room surfaces [133-135]. An "active wall" can be implemented by using a closely-spaced microphone and loudspeaker to either reduce or increase the local wall surface absorption, allowing the natural room *RT* to be either increased or reduced, respectively. However, the addition of a delay between the microphone and loudspeaker would allow a 'virtual wall' to be moved outward, creating an effective increase in room volume.

The virtual wall concept is employed in the CARMEN system [136-138]. The system operates by reducing absorption to enhance early reflections and reverberation time, and so – like other systems – it does not increase the room absorption. Since each cell in the CARMEN system has a microphone close to a loudspeaker, the cell must be made stable by control of the cell loop gain or by use of a feedback cancellation system. If the feedback is perfectly cancelled, the cell provides a single reflection but each cell loudspeaker is coupled to the microphones in the other cells via the room. Hence the system is still regenerative.

Regenerative systems are well-suited to the global enhancement of reverberation time since the microphones are beyond the hall reverberation radius from all sound sources. However, they are less suited to the enhancement of early energy as the microphones are typically far from the stage area and can not detect direct sound early enough, or with sufficient amplitude, to provide significant early energy enhancement.

Some of the more recent in-line systems were developed after experience with early regenerative systems, with the specific goal of avoiding regeneration and the risk of colouration [55,88,94,98]. However, modern regenerative systems produce reverberation gain without colouration and generally both approaches are considered to produce high quality enhancement [61,62].

Some systems adopt a hybrid approach, using microphones close to the stage for early reflection control and distributed microphones for global enhancement of reverberation [124,127,131]. Similarly, some in-line systems move microphones out into the room to produce reverberation enhancement for sources in the audience area [94]. Hybrid systems are able to control the balance of early and late-arriving energy and so produce trade-offs between ASW and LEV [127,139]. Many commercial systems also offer specific systems for enhancing on-stage acoustics.

FEEDBACK IN ACTIVE ACOUSTICS

Although the historical problems of poor audio component quality are largely eliminated, all active acoustic systems retain the fundamental physical limitations caused by acoustic feedback from loudspeakers to microphones. (Echo cancellation techniques can in principle be applied to reducing this feedback [140-144]. However, the correlation between the loudspeaker feedback path signals and the desired microphone signals is problematic [141,142,144], and no current systems use such techniques.)

At sufficiently high loop gains, any active system can become unstable, and when operating below the point of instability, colouration effects can occur which reduce the sound quality. While each type of system avoids these risks in different ways the underlying stability theory of time-invariant and time-variant systems is the same [145-150]. Since all current systems use wideband channels, we will mention some aspects of stability theory for wideband multichannel time-invariant, and time-variant systems.

The analysis of stability of multichannel systems assumes there are N independent microphones and loudspeakers.

However in most practical systems there are L loudspeakers and M microphones, and L is greater than M to avoid localizing sound to a loudspeaker (an exception is the CARMEN system for which M=L). The digital processor distributes the M microphone signals to the L loudspeakers, and can be represented as an $L \times M$ transfer function matrix $\mathbf{X}$ (Fig 1a), and the room transfer function matrix $\mathbf{H}$ is $M \times L$ (Fig 1b). At each frequency the $M \times M$ loop transfer function matrix $\mathbf{HX}$ can be represented as an uncoupled set of independent "eigenchannels", represented as a diagonal matrix $\mathbf{\Lambda}$, cross coupled by a matrix of eigenvectors $\mathbf{Q}$, $\mathbf{HX} = \mathbf{Q\Lambda Q}^{-1}$ [147,149]. The number of "channels", N, is then the number of non-zero eigenchannels (the rank of $\mathbf{HX}$) which is the minimum of M and L. For most systems, the number of channels then equals the number of microphones.

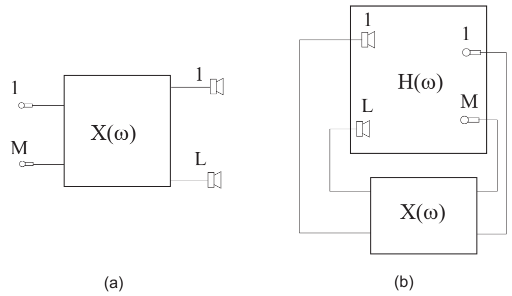

Figure 1: System processor (a) and its arrangement in an active acoustics system (b)

The stability of the active system is derived assuming that the loudspeaker-microphone distances are greater than the reverberation radius, in which case the room transfer functions behave statistically as independent, zero-mean complex normal process [151]. The statistical behaviour of the eigenchannels are in this case known [147]. Assuming that the transfer functions have a constant envelope with frequency, and that the processor is unitary so that it does not increase the variance of the loop gain (this is true, for example, for the MCR and Constellation systems) allows a derivation of the minimum risk of instability which depends only on the bandwidth of the channel B and the room reverberation time T [112,145,149]. The probability of instability for wideband channels with a BT product of 20,000 is shown in fig 2.

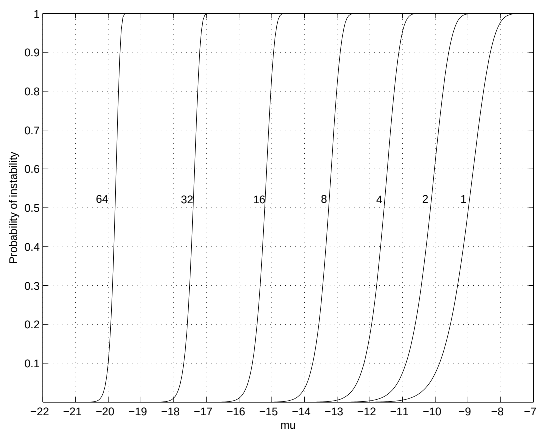

Figure 2: Probability of instability of multichannel systems assuming unitary processor, BT=20,000

We note from Fig. 2 that the transition from stable to unstable operation is more rapid for large N, which means that large systems tend to be either stable or unstable with low probability of transitioning from one state to the other. Small variations in individual channels have a small effect on the total loop gain, and – with a suitable stability margin – produces negligible risk of instability.

For example, the 50 % probability-of-instability limit for $N = 1$ is –9 dB and a margin of 2.5 dB is required to reduce the risk to 0.1 %. For $N = 16$ channels the 50 % limit is –15.2 dB and a margin of 1.2 dB is required for 0.1 % risk. The required margin reduces with the number of channels which means that for a relatively modest loop gain margin, a large installation is extremely stable. Further, the risk of colouration can be defined in a similar manner to that of instability, and also produces increasingly rapid transitions as N increases [149]. Hence, the risk of colouration in large multichannel systems is also low with sufficient loop gain margin.

Broadband equalisation is required in regenerative systems to compensate for the power response of the loudspeakers and the acoustic absorption of the room and to produce a smooth increase in reverberation time [114,126]. A single, smooth, equalisation curve may be derived for all channels from the norm of the room transfer function matrix, which eliminates the variation observed in a single transfer function [126]. Narrowband equalisation is not typically used, since active systems operate well below the loop gain where this would be necessary [70,72].

Modern systems using digital gain control and equalisation are able to produce extremely stable loop gains. Hence, time-invariant systems with large numbers of equalized channels and appropriate loop gain margins produce highly stable, colouration-free performance. Those systems which use digital reverberation can provide additional increases in RT that are independent of the loop gain, allowing the use of lower loop gains and further reductions in the risk of colouration [120].

The stability of time-varying systems has been studied in [145,150] and the behaviour of various modulation methods examined in [148]. The 50% time-invariant stability limits and the time-varying stability limits with frequency shifting from [150] are shown in table I. (Note that we give a limit of 0 dB for the single channel case as discussed in [148,150]).

N	Time-invariant	Time-variant	Difference
1	–9.0	0	9.0
2	–10.2	–2.4	7.8
4	–11.6	–5.9	5.7
8	–13.3	–9.0	4.3
16	–15.2	–12.0	3.2

Table 1: Stability limits for unitary feedback multichannel systems, from [150]

The improvement in stability limit produced by time-variation reduces with the number of channels. For $N = 1$ channels it is 9 dB and for 16 channels it is around 3 dB. Time-varying systems tend to require a larger loop gain margin [145], and the difference in useable loop gain was found to be 0 dB for 16 channels [150]. Hence, for large systems with $N \geq 16$ there is no advantage in using time variation. Systems which use time variation are in-line systems using a relatively small number of microphones where time variance is beneficial. Time-invariant systems tend to use larger numbers of channels where time variance produces no advantage.

NATURAL ACOUSTICS

Active acoustic systems may be viewed as a recent innovation that follows the historical application of technology to the live performance of music. While active systems can be exploited to produce novel acoustic conditions for modern performances, they must be capable of providing the acoustics associated with traditional performance spaces.

The risk of active acoustic systems is that they may produce unnatural artifacts that relate to their method of operation. For example, time-invariant systems can produce colouration effects caused by the greater variance of damping factors of the modes in the enhanced room, and time-varying systems can produce noticeable pitch-shifting effects.

Acoustic artifacts can be detected by listening tests, or objective measurements could be designed to detect them. For example colouration in time-invariant systems can be detected by a statistical analysis of enhanced room transfer functions [152,153] or by estimating the modulation transfer function (MTF) [58,154]. While some subjective assessments of time-varying systems have been carried out in [57] few objective measures have been proposed. Pitch-shifting artifacts could be quantified by recording the response to single or multiple tones and comparing the modulated responses with known subjective thresholds. Nielsen has also suggested that the MTF might be useful for quantifying colouration in time-varying systems [154].

The use of an active system involves achieving a desired set of acoustic conditions without producing noticeable colouration effects. If this ideal is achieved, the active acoustics are natural in that they can produce acoustics indistinguishable from a passive acoustic design. Of the systems discussed above those known to have a significant number of recent installations are MCR, CARMEN, ACS, SIAP, LARES, Constellation and AFC, and these have to a large extent earned a reputation for producing natural acoustics [61,62].

Despite this fact, there are listeners who claim that any form of electronic assistance is inherently unnatural [62]. This belief has in some cases lead to the removal or disabling of systems that were functioning adequately, and even in one recent case known to the author (not a Constellation installation) where the perception of poor sound quality was attributed to the active acoustics in a venue when in fact the active system was not operating. This echoes the statement by Vermeulen [75] that ‘Under no circumstances must the public become aware of the use of loudspeakers, for their reputation has become so bad by misuse that the mere suggestion that they are present can spoil appreciation of the performance, even when they are not in use.’ While the technology has improved, the mistrust of electronic assistance has, for some, remained.

The belief in the inherent unnaturalness of active acoustics could be countered by two arguments. Firstly, as argued above, well-designed active systems are now able to produce acoustics indistinguishable from passive concert halls. Secondly, it can be argued that the definition of “natural” as using passive architecture is arbitrary. For example, Parkin states [155] ‘How much further shall we see electronic aids spread in the future? They have obvious advantages, but equally obvious dangers, and electronic aids for music raise many ethical problems. It can be said that it is not “natural” for the acoustics to be affected in this way, but then what is natural about a concert-hall or music itself for that matter?’

More recently, Blesser states in a similar vein [59]: ‘But even in the “natural acoustics” of a concert hall without electronics, listeners hear the acoustic interventions of sound-dispersing statues, sound-reflecting ceiling panels, sound-diffusing walls and sound-absorbing panels. There is only one relevant question. Does any particular intervention benefit the aural experience of a musical space? Debates about natural versus artificial are thus spurious and misleading.’

Hence, placing active systems in the history of technological development, we find that the definition of naturalness used against active systems is obtained by setting the technological boundary “the use of electronics”. However, this boundary could just as easily be set elsewhere in history, such as “the point where complex sequences were applied to the design of diffusers” [44]. With this definition, the use of primitive root diffusers would be unacceptable as they suppresses the specular reflection that any natural planar reflector would produce [45,62].

These arguments, however, do not address the root cause of the resistance to active systems, which lies not so much in the subjective assessment of acoustic quality, but in beliefs about the role of technology in art [156-162]. For example, an argument against the use of technology in art is that art is a human endeavour, and technology risks disengaging the recipients from the reality of that art [157,158]. For example, Borgmann states, regarding stereo reproduction systems, that ‘Loudspeakers have no visible affinity to the human voice, to the brass or the strings whose sound they reproduce.’

Another argument against active systems is that any art form should be experienced in the environment in which it was originally performed [162]. While this is a valid sentiment, it is also problematic since the environment includes many factors. For example, instruments in orchestras have developed since the compositions were written, (leading in some cases to attempts to recreate period instruments). Further, listeners no longer have the aural tradition of those who heard the earlier renditions of the art [159,162]. For example: Braun [159] states that ‘sound recordings have also influenced music listeners to such an extent that many come to the concert hall with aural expectations modelled on their experience of recorded music.’ Also, modern musicians may play differently to earlier musicians. As an example, more vibrato is used in modern violin playing than was used in the past [159].

Generally, then, it is almost impossible to arrange for listeners to have the same experience of music as their forebears as the social context has irrevocably changed. Active systems might be viewed as one of the more benign of modern innovations since they can recreate the acoustics of traditional spaces as far as measurement allows, and do not raise the philosophical questions that music reproduction systems do, since they remain in the service of the human expression of music in live performance [157].

A positive argument for technology is that it simplifies the life of the user, and is a means of improving upon nature and allowing an enhanced engagement with reality [158,160]. Active systems, with their ability to supply sound at more arbitrary times and levels, may allow ideal listening conditions to be more closely approached at a greater number of seats. If this is the case, then the use of active systems can result in acoustics superior to that supplied by passive designs. As an example, the music director K. Nagano states: “Performing at Zellerbach Hall with the Constellation system, one can deeply appreciate how far technology and science have developed. The hall's acoustics come to life in response and one can tell that the audience and musicians are having a new and extraordinary experience."

Ultimately it must be accepted that some concert-goers will be unwilling to accept an active system. High quality, single-purpose, passive concert halls will always provide an alternative, and be a reference against which active systems must be measured. However, multipurpose venues with active acoustics is an emerging paradigm which offers considerable benefits to owners and to the public, and it is likely that subsequent generations of listeners will be more accommodated to the presence of electronics in live performances, and perhaps even to expect the greater range of acoustic conditions they provide [159,162].

DISCLOSURE

The author is the inventor of the VRA system, which is now the basis of the Meyer Constellation system.

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