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DIGITAL STEREO & ITS APPLICATION IN MUSIC PLAY SYSTEMS

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The use of stereophonic sound in installed music play systems is a relatively recent phenomenon which is odd when one considers how long stereo has been around. The use of digital equipment at the front end in such installations is very recent indeed. While there may be some clearly predictable benefits arising from the use of digital front ends in both background and foreground music play systems, perhaps the most significant benefit of all has been less obvious.

It is largely to attempt to explain this unexpected virtue of digital sound that I am concerned here, but in the process, the trials, tribulations and findings which have occurred while taking the theories into field use will be described. What has emerged from the development work undertaken in the U.K. during the past eighteen months, in association with Viewfax Digital Ltd., is a perceived response to the use of digital source material in music play systems which is better than originally envisaged by the system's creators.

The improvements over analogue which we expected related to such matters as dynamic range, absence of wow and flutter, distortion and degradation of the signal through wear and mis-alignment. All of these exist of course, and they are in themselves very worthwhile, but they are only a part of what digital has to offer.

It is clearly true to say that, to hear stereophonic sound it is necessary to be within earshot of at least two loudspeakers, appropriately distanced from each other so that signals from them can be placed by the ear into a 'realistic' space-time continuum. Even in a domestic environment this is a difficult enough matter, and true stereo is generally only available to the listener in a very small area of the listening room, close to the locus of points equidistant from both speakers. Small wonder then that many have held (and some still do) that installing a stereophonic system in a pub, cafe or hotel is a waste of time. What such statements really reflect is the degree of difficulty in obtaining a credible, satisfying stereo effect. The reasons for this are complex, and it is necessary to trace back to some of the fundamental research into acoustics carried out in the 1930's to gain an insight into the issues and the solutions.

Specifically, in 1933, F. K. Baker (1), working in the Bell Telephone Laboratories, described for the first time what he called the Precedence Effect. What Baker noted was that when a single sound is reproduced from **two** loudspeakers, and the sound

coming from one of them is delayed by a few milliseconds, the listener will hear the sound as if it came only from the speaker where he first heard it. The listener would, he found, judge the other speaker to be silent. At this point it is important to note that Baker was using two speakers and experimenting with stereo.

More familiar to speaker designers and acousticians is the Haas Effect (2), which states that the first sound to be heard takes command of the ear, and sound arriving up to 50 milliseconds later seems to arrive as part of and from the same direction as the original sound. This effect, he says, eliminates confusion and restores intelligibility. Although working some fifteen years later than Baker, Haas was using mono, single source sound, and it is not known to the writer if he was aware of Baker's earlier work.

These two effects, together with the generally-accepted influence of amplitude as an aural substitute for proximity, go a long way towards explaining the problems encountered in attempting to create a stereophonic background music system which works. The two effects also explain why, in a mono system, the ear is drawn virtually exclusively to the nearest speaker. This dominance by the nearest speaker is clearly undesirable in terms of stereo reproduction, and that it is also regarded as undesirable by the listener can be readily judged from the public response to traditional Muzak-type systems. It is unfortunate that a major brand name should have acquired pejorative status along with terms like wallpaper music. In most cases, switching such a conventionally designed system from mono to stereo operation will not help much unless the listener is lucky enough to find himself in a 'hot spot' between two speaker channels. To make stereo possible across a wide area of the venue or listening room thus will clearly require something extra from the loudspeaker and the source.

It is at this point that one of the characteristics of digital sound reproduction enters the picture, in the shape of channel separation. Because, at up to 90dB, this is so vastly superior to the 30-55dB analogue norm, a digital source will yield a much richer crop of acoustical cues, separating the component sounds in the composite stereo sound 'stage' with great accuracy. So it proved in the testing room, but in larger venues the sound was little different when comparing analogue and digital sources while using conventional, direct-radiating speakers.

With multi-directional speaker enclosures, the difference was dramatic, suggesting that, in order to achieve a worthwhile and convincing stereo effect, it is necessary to use both a digital source and multi-directional loudspeakers. This implies a need to diminish or eliminate the Precedence Effect, and this would also explain why, under these conditions, the improvement given

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by the digital source is so noticeable, since switching back into mono, the effect is restored, the Haas effect still applies and no loss of intelligibility is found during, for example, paging announcements.

It is probably fair to say therefore, that to achieve an evenly diffused stereo sound field around a venue, it is necessary to use a digital front end, a multi-directional type of loudspeaker and of course to position the speakers in such a way and in such numbers that the listener can always hear some sound from a right and some from a left channel-assigned unit. The effectiveness of the system is either much reduced or absent altogether if all of the factors desribed are not brought to bear in the final design or layout.

While touching earlier on subjective, or properly, qualitative issues, I could have mentioned another undesirable, compression. Whether deliberately introduced to even out the differences in loud and soft musical passages, or occurring as a consequence of the shortcomings of the front end equipment, this needs to be carefully controlled, if not eliminated since it will adversely affect the number of acoustical cues the system can deliver. The further down the reproduction chain towards to speaker that this compression occurs, the greater will be the damage. Much of the directional information needed by the ear and representing the crucial part of the composite stereo recorded signal is present at the higher frequencies, and at relatively low amplitude levels. Analogue reproduction equipment will tend to eliminate, reduce or distort this part of the signal first and most, and in this regard can be considered manifestly inferior to the digital alternative.

The playing apparatus which has evolved through the development phase of this project is as follows, and there are now about 100 installed systems operating this way within the U.K.

High quality (mainly digital) source material, taken from CD's and studio master tapes is recorded on to 2-hour beta-cassettes using 16-bit professional replay equipment, passing through a temporary (ultimately un-necessary) analogue phase, and PCM analogue-to-digital conversion. These cassettes are then replayed in the system by two beta-VCR's, the output from which is led through a PCM digital-to-analogue converter and then into the analogue audio chain, at which point any necessary equalisation is applied before amplifying the sound. All the components are housed in the same unit and there is no user access to any of them. Since the use of this particular digital format leaves extra space on the tape, which is normally taken up by analogue stereo or mono sound tracks in the normal video or beta-hi-fi operating mode, that space can be used to carry control command and other information which is sensed, monitored and reacted to by a microprocessor control unit, thus allowing a high degree

of automation for the system, which is as a consequence very easy to use. It can also carry other programmed information to enable control of lighting, generate computer graphics or a multitude of other functions. The provision of these facilities using analogue equipments has traditionally been attended by reduced sound quality or considerable extra expense.

The analogue sound output from the system is fed via a high quality cable to a configuration of several pairs of multi-directional loudspeakers, wired series-parallel for constant impedance. The placement of speakers is a skill in itself, and as part of our attempt to make the skill more scientific, one of my colleagues will be showing later in this conference a new, computer-aidedtechnique for doing this.

Among the problems encountered during the field trial stages of the system, I would highlight a couple. Both concern software. The tendency of some video tapes to stretch, especially in the extended lengths is disastrous in the digital domain. The tapes will corrupt rapidly, failing to track after only a few passes. If the tape is sufficiently inelastic, we find no problems in achieving the 300 or so passes required in this application. The point here is that the video and audio 'picture quality' of the tape is less important than its mechanical properties. Since there is no analogue stage which uses the tape, the magnetic characteristics are not as significant in PCM digital use.

Another problem has been the time lag between the release dates in CD compared to black vinyl formats, but this is expected to shrink dramatically as the CD medium grows in public use. As all the key elements in the system are in essence domestic, the availability etc. is guaranteed by popular demand, but there are some small sacrifices to be made.

On the plus side of this, the impact of head wear on the system's VCR's has turned out to be even lower than anticipated, and analysis of machines which have been in service continuously since May 1984 show that the initial estimates of 5000 hours of head life are likely to be exceeded.

Initial experiments are now under way using the new 8mm DAV format, and while the results are not quite in the same class as the 16-bit large format system described here, the difference is sufficiently small to provide the effects referred to earlier, and we can therefore predict a bright future for 8-bit, 8mm, 33kHz sampling systems in the future, to the point whereby they may replace the ubiquitous compact cassette in this arena.

In summary, what we have been able to produce through digital sound technology and the application of some basic acoustic theory is the possibility to greatly enhance listener appeal,

improve intelligibility, and move background music out of the dark ages and into the 'lifestyle' oriented marketplace of today in which the consumer demands more accurate sound. By accurate, I do not mean spectrum, colouration or any of the other largely subjective issues which excite the hi-fi industry from time to time. What I mean is the ability of a system to deliver a sufficient number of acoustic cues into the right places to enable the listener to experience a sense of authenticity, and to escape from the nearest loudspeaker unit into the atmosphere of the music and the venue.

REFERENCES

(1) H. M. Tremaine, The Audio Cyclopedia, p.1135, 20.135.
(2) Ditto p.803.
Publ. Howard W. Sams & Co. Inc., The Bobbs Merrill Co. Inc.