Book reviews

Jörg Stelkens and Hans G. Tillmann (eds.), *Klangforschung '98: Symposium zur elektronischen Musik*. Saarbrücken: PFAU-Verlag, 1999. 201 pp. Softback, ISBN 3-89727-086-2

A review of a proceedings booklet is never an easy task given the breadth of such publications. I am nevertheless pleased to introduce the reader to this volume, if for no other reason than to describe this annual event and the types of papers which have been offered. According to one of the two conference organisers, Jörg Stelkens, 'Klangforschung' (Sound Research) is a conference series that attempts to reach a public beyond the 'scientific ghetto'. Instead, specialists from a variety of areas, in particular the arts and sciences, meet to share their experiments and experiences leading towards greater mutual understanding. This means that scientific entries have not been filled with equations many other constituents of the community would not be able to understand. The speakers, including many well-known figures, came from the German-speaking countries with the exception of Klarenz Barlow who currently resides primarily in the Netherlands.

The 1998 conference in Munich was split into five daily themes: History of Electronic Music, Sound Production and Manipulation. Three-dimensional Music and Sound Sculpture, Sound and the Networked Computer World, and Soundscape and Sound Design. To the readers who attend conferences such as the annual International Computer Music Conference, this list may seem restricted. Nevertheless, after reading the volume, one has the feeling of having been taken on a voyage stopping at eighteen diverse locations within the world of electronic music. (It is noteworthy to see how the terms computer music and electroacoustic music seem to be absent in many of these papers.)

Paper subject areas covered a broad scope of subjects ranging from new products on the market to electronic music in higher education, physical modelling in a musical context, free jazz and interactivity, algorithmic composition and net-based as well as site-specific art. Still, the one reservation I had with this very worthwhile collection is that there was little interpretation of the historical subjects focused upon and that many entries were more descriptive than challenging. In a sense, the reader assumes that the debates at this symposium were where true interdisciplinarity took place as opposed to the presentations themselves. Still, it was welcome to see experimental artists sharing their wares with developers and musicologists. It is a bit of a shame that the third category of specialist seemed somewhat underrepresented; they might just have drawn the links that *Klangsforschung* aims to achieve.

Leigh Landy

Alan P. KeFauver, *Fundamentals of Digital Audio*. The Computer Music and Digital Audio Series, Vol. 14. Milison, WI: A-R Editions, Inc. 1999. 174 pp. Softback, ISBN 0-89579-405-5. RRP $39.95.

Fundamentals of Digital Audio is a new book in the well-established Computer Music and Digital Audio Series, originally founded by John Strawn, from the US publisher A-R Editions. The series editor is currently Christopher Yavelow. In this book, Alan Kefauver, course coordinator for the Recording Arts and Sciences Program at the Peabody Institute of The John Hopkins University (USA), sets out a series of explanations of digital audio principles designed to introduce the subject to relative novices in the field. The book opens with a possibly unavoidable overview of basic principles of sound waves, the decibel and timecode. It then proceeds to deal with components of the digital signal chain, recording and editing systems, editing and mastering, and finally signal interconnection and transmission.

One of the most difficult aspects of writing about complicated technical subjects at an introductory level is finding ways of expressing complicated issues in an easy-to-understand manner without actually saying anything wrong. This book is clearly based on a wide-ranging collection of information amassed by the author, and the majority of common system components and principles are at least touched upon. There are quite a lot of useful facts and adequate explanations in the book, and for this reason it is hard to know how to deal with the problems contained therein. It would be unfair to dismiss the book out of hand, but the overall impression gained by this reviewer is unfortunately a negative one. Possibly the bad points stand out so much that the good points of the book are obscured, but it is quite unsatisfactory in many ways and could possibly mislead or confuse the student who has little knowledge of the field. Unfortunately, there are numerous instances where explanations are only partially correct, and while there

https://doi.org/10.1017/S1355771800221079 Published online by Cambridge University Press

are a number of correct facts mixed in, there is also much half truth or confusion. When a book contains some useful explanations interspersed with quite a lot of incorrect material and misleading explanations, it is very hard for the novice reader to know which is which. Not knowing the subject, the novice has no means of knowing whether to trust a book.

It would take quite some time to comment on all of the specific points leading to the conclusion that this book cannot be recommended, but it would be unfair to make such a statement without justifying it by highlighting some of the more serious examples.

Chapter 1 in the section on timecode basics, in explaining the reason for drop frame timecode, states that 'black and white TV has a carrier frequency of 3.6 MHz whereas colour uses 3.58 MHz'. Carrier frequency is not in fact the issue here, and in any case TV signals are generally modulated onto much higher frequency carriers (VHF and UHF) than this for transmission. The figure of approximately 3.58 MHz relates to the subcarrier frequency that was added to the monochrome baseband spectrum in the NTSC system to carry the colour information. It is not clear where the author gets the 3.6 MHz frequency from. He also compares the timecode signal to a '400 Hz square wave with many odd harmonics', but it is not clear where the 400 Hz comes from or what a square wave without odd harmonics would be (a sine wave, presumably). Time code, if it can be likened to basic square waves, tends to look more like a combination of 1 and 2 kHz rectangular waves (or 1.2 and 2.4 kHz) depending on the frame rate.

Chapter 2 describes the digital encoding process, and in a valiant attempt to make the explanations simple, some unusual problems have crept in. The explanations of sampling and aliasing are hard to follow, especially since the figure (2.1) showing 'waveform sampling and the Nyquist frequency' shows samples occurring at the critical point where the sample instants coincide with the zero crossing points of the wave, thereby leading to samples with zero amplitude and no output from the sampler. Quite what the author is trying to show here is not clear, as he simply refers to it in the text as illustrating 'a waveform that has been sampled twice', and later 'two sample periods' (even though more are shown). On p. 26 it is stated that 'when a signal that is above the Nyquist frequency is sampled, alias frequencies are created. That is, the system would define the time between the two samples as a frequency.' The meaning of the second sentence is hard to fathom as an expert, so the novice would presumably be left in the dark.

On p. 28, when discussing the quantising process, it is stated that 'because it is easy to store pulses of different amplitudes on magnetic tape, a base 2 numbering system is ideal for codifying this data'. This is an odd sentence as it does not really explain the multiple reasons for the use of binary data. Possibly the author means 'two amplitudes', but the sentence implies multiple amplitudes. Furthermore, it rather confuses the arguments from the previous page about it not being possible to store 'specific numbers in the magnetic domain'. If, as the author states, it is 'easy to store pulses of different amplitudes on magnetic tape', why is binary the ideal coding scheme? One main reason a binary data form is useful is that it enables one to use a channel with a poor signal-to-noise ratio and still retrieve the data, since only two states ever have to be recognised. The more intermediate levels adopted in any modulation scheme, the less resilient the data signal is to noise and other distortions.

The description of dither on p. 31 has some useful truth in it, but again the elements of truth are compromised by confusion. In stating that dither noise helps to *mask* the quantising error, the author is fundamentally wrong (but this is a common misconception perpetuated in some other places as well). Dither noise is intended simply to decorrelate the quantising error from the signal, thereby *converting the error signal into noise*. Implemented correctly, quantising distortion is *removed* by dither, leaving no distortion artefacts to be 'masked'. There are hints that the author appreciates this fact, but in referring to a masking effect he is misrepresenting the basis of the dithering principle. It is just possible that a masking effect would be useful in the case of insufficient dither amplitude, giving rise to a situation where the quantising distortion components had not been completely removed by full decorrelation, but this situation would not normally arise in a correctly engineered system.

A lack of clarity regarding the role of pre-emphasis is evident on p. 43, and it is not clear at what stage in the record signal chain the author believes different types of pre-emphasis are to be used. The section is headed 'record modulation', and the author states 'you may also find the pre-emphasis circuit here'. It is possible that audio pre-emphasis is being confused with magnetic recording pre-emphasis such as might be used to counter recording losses or shape the data signal for optimum recovery, but this type of data recording equalisation does not affect audio signal-to-noise ratio. The type of audio pre-emphasis sometimes used before A/D conversion to reduce HF noise after replay de-emphasis, often using the time constants of 50 and 15 µs, is not used in the record modulation process at this stage. Some digital audio systems *did* use optional audio pre-emphasis with the time constants indicated, but this was normally applied in the analogue domain *prior* to quantisation, not 'after digitisation but prior to record modulation', as stated.

In Chapter 3, describing the digital decoding process, some straightforward explanations of basic error correction are offered, and there is little to complain about here except the limited depth. In the description of D/A conversion on p. 57, we begin to get into deep water again, and it is stated that 'because the sample pulse is

https://doi.org/10.1017/S1355771800221079 Published online by Cambridge University Press

not infinitely short, side bands equal in size to the base (or audio) band are created above and below the sample frequency'. Again, this is fundamentally incorrect: the side bands are indeed created but not for the reason stated. The sidebands are the result of the original pulse-amplitude modulation process (not the finite width of the sample pulse), and the repetitions of these spectra are a feature of the harmonic spectrum of a pulse wave. There is a side-effect of the pulse width used in D/A conversion, but it is manifested in a phenomenon known as 'aperture effect' which is described reasonably clearly by the author a few pages earlier.

Chapter 4 describes tape-based storage and retrieval, summarising rotary and stationary head recording systems. The basic facts on different common formats are available here, along with errors such as the statement that the DAT long play moe of 32 kHz and 12 bits non-linear coding conforms to the DAB (digital audio broadcasting) standard (it doesn't, at least not in Europe). Also one cannot begin to unravel the statement that the bandwidth reduction of the data signal provided by channel coding 'is accomplished by sacrificing word length, but the density increase to the storage media is an order of magnitude greater because of the decrease in data bandwidth. Remember that the digital bit stream density is the product of quantisation (bit density) times sample rate (frequency bandwidth).' There are multiple layers of apparent confusion here involving original sample rate and quantising resolution, channel coded representations of the signal, audio bandwidth, coded and uncoded signal bandwidth and 'bit density'.

On p. 83 it is stated that there is a high resolution (HR) version of the multitrack DASH tape format that runs the tape at double speed and uses 24 bit resolution at 96 kHz sampling rate. This is not a standard implementation of DASH HR, and not one of which this reviewer is aware. The DASH HR, as implemented in the Sony PCM-3348HR machine, for example, runs at 1.5 times normal speed in HR mode, and uses 24 bit resolution at 48 kHz (Sony has not used 96 kHz PCM in its Pro-Audio products as it is moving towards the 1-bit DSD system at the present time). It is just possible that there is a third-party product available that does some of these things, but it is not part of the standard DASH HR implementation.

Chapter 5 is concerned with disk-based storage, and covers various optical and magnetic storage media and formats. There are a number of occasions here and in other places where, either for typographical reasons or whatever, 'megabits per second' is confused with 'mega*bytes* per second' (the abbreviation MB is often used, apparently interchangeably). For example, on p. 102 it is stated that the data rate of DVD is up to 10 MB per second (this is approximate and should mean mega*bits*) and then later that the capacity of CD is 780 MB (this refers to mega*bytes*). The author also states that MPEG-2 compression is used to accomplish audio *and* video data reduction on DVD, but then goes on to mention Dolby AC-3 coding in passing when discussing multichannel surround sound on DVD. It is not clear how the two relate to each other or whether they would both be used at the same time. MPEG and Dolby Digital (AC-3) are two alternative options for audio coding on DVD-Video, and in fact there are few commercial DVDs around with MPEG-coded autio. It was originally intended that MPEG would be used in Europe and Dolby in USA and Japan, but commercial reality has led to Dolby Digital being adopted almost universally in all regions. On p. 103 the CD pickup laser wavelength is quoted as 750 nm, whereas elsewhere it is stated as 780 nm (780 is correct).

In discussing the ATRAC data reduction system used on MiniDisc, it is stated that when the data reduced bit stream is played back, the 'missing data' are 'interpolated'. This is not the case – it is not an interpolation process but a requantisation process whereby frequency domain samples quantised at a lower resolution than the original PCM are restored to the original resolution and returned to the time domain. No interpolation takes place, at least not in the way this reviewer understands interpolation. It is also stated that each generation of copy reduces the data by a further factor of 5:1 . . . 'each generation has less data than the preceding one . . .'. This is not the case either, as the data is returned to the original PCM rate before it can be copied. Successive generations of low bit-rate coding, passing through the PCM domain on each occasion, simply serve to reduce the sound quality on each pass (the loss of quality is acknowledged in the text). The data rate does not keep on getting lower on each occasion. Even if the coded data could be copied without going back via PCM, the data rate would not keep on getting lower as there would be no need to recode the signal.

It would be inappropriate to continue this catalogue. It would also be wrong to say that the book has no value, since there are some reasonable sections. Nevertheless, the problems exemplified above give too much cause for concern over the security of this book as a serious teaching resource. A few errors can normally be overlooked or dealt with, rather as in digital audio, but there comes a point when the evidence stacks up a little too heavily in the negative direction and the human error correction system becomes overwhelmed.

Francis Rumsey

Todd Winkler, *Composing Interactive Music – Techniques and Ideas Using Max*. The MIT Press, Cambridge, MA/London, England, 1998. 350 pp. + CD-ROM with programming examples from the text and additional material (requires an Apple MacIntosh computer with MAX software installed). ISBN 0-262-23193-X.

https://doi.org/10.1017/S1355771800221079 Published online by Cambridge University Press

The title alone of this book raises some important, even controversial, issues in contemporary music making. What, to begin with, is meant by 'interactive music'? There is a clear sense in which all music can be said to be interactive. Whether it is a solo pianist interacting with Chopin's score, an acousmatic composer interacting with sonic material in the studio, or an improvising ensemble interacting with each other, there is always some form of interaction involved in any musical activity. But 'interactive' has become one of those words that many people use, but not always to mean the same thing. Fortunately, Winkler tackles this question at the outset. Chapter 1 begins by defining his terms; by paragraph four he states, 'Interactive music is defined here as a music composition or improvisation where software interprets a live performance to affect music generated by computers.' In Chapter 2, 'Interaction: Defining Relationships between Computers and Performers', he cites four models of how a computer might interact with performers: 'The Conductor Model – Symphony Orchestra', 'The Chamber Music Model – String Quartet', 'The Improvisational Model – Jazz Combo' and 'Free Improvisation'. Winkler uses these models to refine further his definition of interactive music, making it clear that his discussion is about interaction between performer(s) and computer. The whole area of direct interaction with the public is left to some intriguing references to multimedia and installation works in the final chapter. While one might wish for a broader definition of interaction, the more limited scope of this book seems logical and reflects a widely held use of the term.

The presence of the word 'composing' in the title raises other interesting questions. Is this book really about composing at all? The MAX software, which is used for the programming examples, is not a composition environment, but an instrument building environment. Indeed, the main body of this book is about designing and building interactive computer instruments, and it is not until Chapter 9 that Winkler gets to discussing 'Compositional Strategies', although there is some mention of composition earlier. Electroacoustic performance is an area, however, where composition and instrument design are frequently closely intertwined, even symbiotic. There are even cases where the instrument design is the composition. While there are examples of more generalised instruments designed to serve a range of situations, very often the instrument is designed to suit the requirements of one piece, so instrument and composition are one concept. Equally, there are examples of instruments designed by one person for another's composition, but the professional computer instrument designer has not yet become a norm. Typically, most composers in this field, unless they are content to work entirely with commercially available devices, or are able to work in institutions which can provide assistants, will need to design their own instruments, and MAX is one of the best environments for doing this. Chapter 2 also attempts to deal with the relationship between instrument design and composition, but, perhaps inevitably, leaves it less well defined. Whereas Winkler's definition of interactive music has clear limits, the compositional possibilities of interactive pieces are left wide open. These reservations aside, this is a very useful book, and the author's discussion of the elements of interactive instruments themselves is thorough.

The book's ten chapters are organised into four sections: Introduction, History and Theory; Programming Foundation; Core Components; Advanced Techniques and Concepts. Section I sets out clearly where he is starting from Chapter 1 gives an overview of the background, culminating in a brief history of interactive composition. One does not expect any kind of completeness from such a history, but rather an orientation of the reader to the author's main points of reference. Chapter 2 starts with the discussion of Performance Models mentioned above, follows with a discussion of Musical Form and Structure as applied to interactive composition, culminating in a brief section on Instrument Design. These chapters set the context for the main part of the book.

Section II consists of three chapters: first a clear and well-written overview and introduction to MAX, which will be useful to MAX beginners when used in conjunction with the Max Tutorial. Winkler manages to cover the range of possibilities well in only thirty pages. The ensuing chapters on Program Structure and Design, and Interface Design are equally clear guides to good programming practice. I have a small wish that the 'spontaneous experimentation' side of MAX was not dismissed quite so curtly. All of us have, lurking somewhere on our hard disks, some 'quick and dirty' improvisations, sometimes thrown together in desperation, that just seem to work! I am reminded of the wonderfully anarchic creations of some of the great analogue electronic instrument designers, and would not like to see such freedom of expression banished entirely from our computers. One of the joys of MAX is that you can improvise solutions, experiment and explore. Acknowledgement of this aspect, of the role of serendipity, and how to turn the results into something that can be relied on in a concert far from home, is an important element in MAX programming, as it is in most music making.

Section III has chapters on the computer as a listener and composer. 'Listening' in this case means receiving and analysing MIDI data on pitch, loudness and time in order to extract the relevant musical information. Again, Winkler's discussion is clear; he introduces special MAX objects which assist in handling and analysing complex data, deals with problems created by erroneous data and moves on to analysis of higher concepts, such as melodic and harmonic analysis, and tracking musical changes over time. In the next chapter he shows how the computer can respond to the analysed material through the use of composition algorithms. Here we have another definition of composition; composition by the computer.

https://doi.org/10.1017/S1355771800221079 Published online by Cambridge University Press

To be fair, Winkler cannot be blamed for any confusion that may arise here. Precisely how much composition is being done by the computer, and how much by the composer? Is the computer really composing, or performing a variable score? Is the computer even performing at all, or is it a special kind of instrument that is really being played by the human performer? These are very much open questions, and in a rapidly developing field you will get very different answers from, for example, George Lewis, Cort Lippe, Robert Rowe, or even myself, not to mention many others. And once again Winkler's discussion is good; he introduces a number of useful concepts for transformative, generative and sequencing responses. Most important in these two chapters, as well as in other parts of the book, is the wealth of examples, including highly useful modules created by Winkler and others. Not only do these give excellent examples of how to create useful MAX modules, but many of them are very usable straight out of the box. I have not examined all of the nearly 200 programming examples, but those I have tested are well designed, well documented and do exactly what they say they do.

Section IV comprises the final three chapters. Sound Design addresses the limitations of the MIDI protocol and introduces some techniques for overcoming them. This is, necessarily, a more generalised discussion, as precise solutions will depend on the capabilities of the MIDI synthesis and processing modules available, but it shows how understanding and ingenuity can produce subtle and interesting results. A very important question is how a computer can keep track of where it is in the performance, and this receives a whole chapter to itself. A number of approaches are discussed culminating in an extended introduction to computer score following techniques. Once again the extensive and detailed examples provide much useful material. The final chapter on Interactive Media and New Controllers gives a broad overview of possibilities that go beyond the scope of the book – some intriguing hints of trends for future development. The book concludes with an appendix providing a detailed cross reference of the printed figures, the examples on CD-ROM and the MAX objects they illustrate, followed by a six-page list of references and a reasonably comprehensive index.

Winkler has provided a broad overview of his subject with much useful detail. It is inevitable that a book such as this will be out of date before it is even printed, but this does not diminish its usefulness. Most of the examples he has provided will continue to find a role in the more advanced versions of MAX now emerging. They will also provide useful material for anyone using computers in performance, even if their work is not interactive in precisely the way he defines it.

Finally, it is unfortunate, perhaps, that this book was published exactly at the time when the possibilities of signal processing with MAX were becoming more widely available. Winkler can only say of the IRCAM Signal Processing Workstation, which pioneered these possibilities, 'The ISPW pointed to the future, proving the viability of a single, integrated, interactive system.' That future is now here. David Zicarelli's msp extensions to MAX for MacIntosh computers (soon to be available for Windows as well), Miller Puckette's pd and other MAX developments are bringing the integrated control and signal processing environment pioneered by the ISPW to more widely available platforms. It is this integration of signal processing and control that really allows the interactive techniques Winkler presents to come into their own. Maybe we can hope for a sequel.

Lawrence Casserley

https://doi.org/10.1017/S1355771800221079 Published online by Cambridge University Press

https://doi.org/10.1017/S1355771800221079 Published online by Cambridge University Press