Chroma

Newsletter of the Australian Computer Music Association, Inc.
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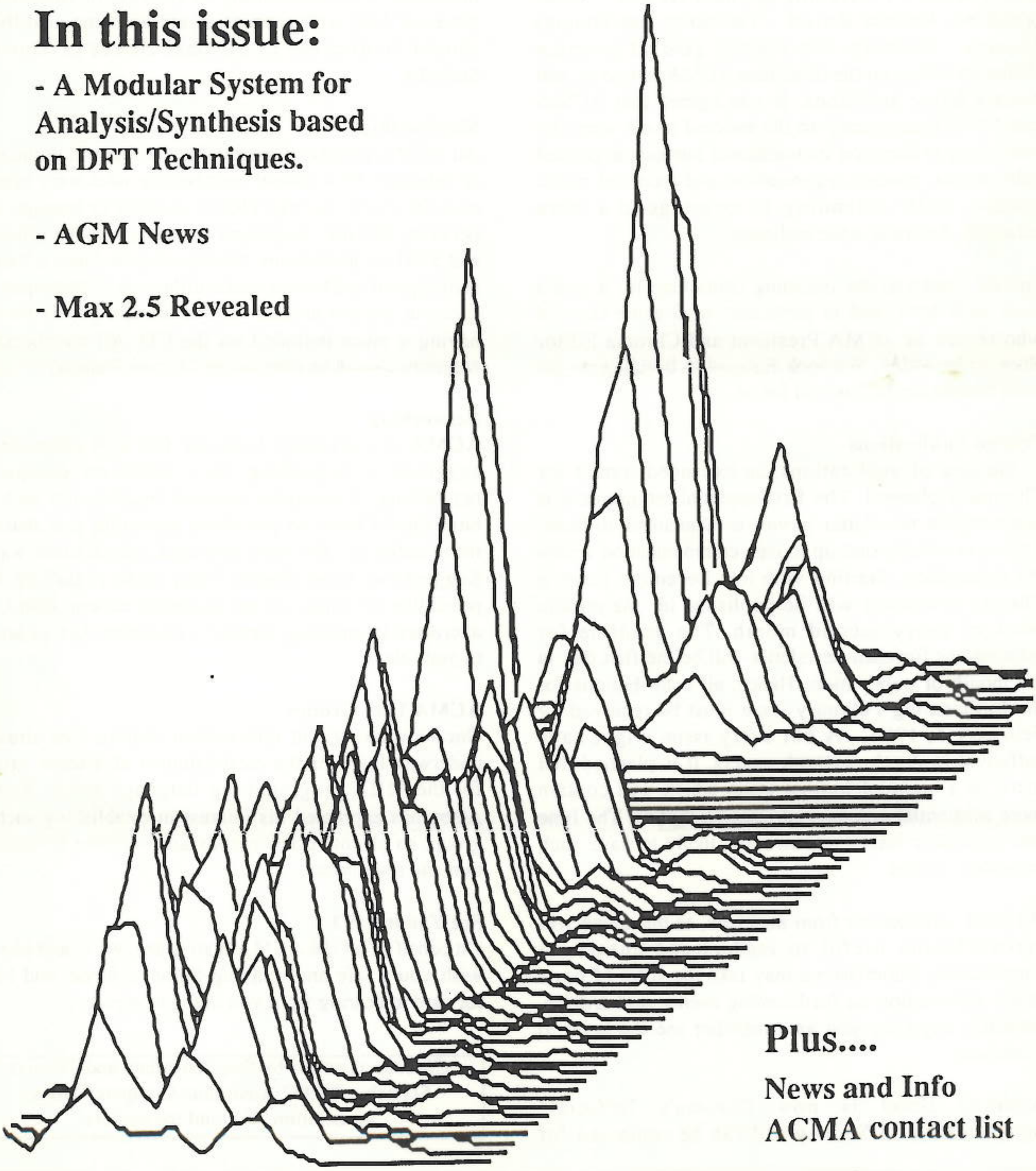

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AGM News - New Committee

Welcome to the final issue of Chroma for 1992. The coming year promises to be a big one for ACMA, with many activities planned. At our AGM, held in the Tallis wing at Melbourne University on the 9th November, a new committee was elected with plenty of ideas and fresh enthusiasm. The office bearers are now David Hirst - President, Michael Hewes - Vice-president, Andrew Brown - Treasurer, and Thomas Stainsby - Secretary. The meeting gave all present a chance to reflect on the directions ACMA can go in, and discuss future ambitions. It was agreed that ACMA could contribute greatly to the national music scene by continuing to focus on its traditional interests in printed publications, concert organisation, and recorded music release, while extending its coverage to a more balanced, Australia-wide audience.

Special thanks to the outgoing committee for a year's work well done, and in particular, to Graeme Gerrard who retires as ACMA President and Chroma Editor since its inception. We look forward to building on the solid foundation he has laid for us.

Printed Publications

In the area of publications, an expanded format for Chroma is planned. The first essential requirement is that a regular newsletter service is maintained, to foster a sense of reliable and up to date communication within the association. Starting with this December issue, a Chroma newsletter will be published in the middle week of every second month. The deadline for submissions for each newsletter will be the first day in the month of publication. Hence, all contributions for the forthcoming February issue must be received by February 1, 1993. As not every issue might have sufficient in-depth research papers, it is planned that there be a biennial journal issue which will contain more substantial articles and research reports. The June and December issues appear most likely to have such expanded content.

We need submissions from members, as this alone will make Chroma useful to readers and ensure its continuation. Submissions may take the form of news items, information on forthcoming events, and articles, basically anything you as a member see the need to contribute.

Anthony Hood is now Chroma's "official" representative in Sydney and can be contacted for submissions and local concert information. See the back page for a list of ACMA contacts.

Concerts

More concerts are planned for 1993, and we hope that for the first time we will be able to have concerts outside of Melbourne. It would be great if the momentum created by the six concerts held in 1992 can be continued to produce a varied and challenging year's programme. Concert organisation this year is handled by Jason Hellwege and Michael Hewes.

Recorded Music Release

Mid-1993 should see the release of the second CD in the ACMA computer music series. This series is intended to showcase new electroacoustic works by ACMA members. All expressions of interest in submitting pieces or help with production are welcome, and these should be directed to Michael Hewes or Thomas Stainsby.

Membership

All ACMA memberships are due for renewal at the end of February. New annual membership rates were agreed to at the AGM, namely $10 for students or unemployed persons, $20 for employed persons, $30 for schools, and $100 for institutions. Membership includes a Year's subscription to Chroma, and entitles one to participate in concerts, publish articles in Chroma, and be eligible for having a piece included on the CD. All membership enquiries should be directed to Thomas Stainsby.

Networking

ACMA is currently looking for any memebers' suggestions regarding their ideas on computer networking. A computer oriented organisation such as ours should have no problems accessing and sharing information in the most practical and efficient ways. Suggestions have already been made regarding the possibility of setting up an electronic bulletin board, or alternatively emailing members en-masse. Let us know of your ideas.

ACMA User Groups

One networking and information sharing idea already underway involves the establishment of a users' group for the MIDI programming language MAX. Roger Alsop has expressed his interest in establishing such a group, so all interested parties are welcome to contact him and share ideas.

See You in 1993

On behalf of all the ACMA committee, we would like to wish you a safe and relaxing Summer break, and look forward to hearing from you in the new year.

Chroma is edited by Stephen Adam and Thomas Stainsby.

A Modular System for Analysis/Synthesis Based on DFT Techniques.

by Chris Scallan.

1.0 Introduction.

The Twentieth Century has witnessed an unprecedented expansion in the range of sonic possibilities from which composers may choose. No longer are they confined to the organisation of pitch in order to convey musical intent. The incorporation of new sonic elements means that the analysis of music can no longer be one based purely on the unravelling of the pitch structures. An approach that explains the musical phenomena in terms of some physical correlate must be found.

A greater understanding of the physical correlates of sound allows for precise specification in the synthesis of sound. Synthesis goes hand in hand with analysis in music as we need to be able to verify any findings aurally (Cann 1988, 47).

One approach to a quantitative analysis of music is to analyse each sound element in terms of its frequency spectrum. Every sound has its own unique "spectral fingerprint" within the frequency spectrum. That is to say that different sounds can be fully represented by a particular series of weighted frequency components. An analogy can be drawn between the audio frequency spectrum and the passing of light through a prism and its separation into its colour spectrum. Similarly sound can be separated into frequency components. At a very specific level the information generated from an analysis of the frequency spectrum can be used to investigate the "tonal microstructure" of a particular sound - how different frequency components vary over time, or from one sound to the next. Thus, we can build up specific models of spectral behaviour that may equate to changes in the timbre, for example. At a more overarching level the spectral information may be used to detect more general structures such as definite pitch in sequential sound events. Once the spectral correlates of pitch are understood the information gathered from the spectrum can be used as a tool in the investigation of pitch organisation (Smalley 1986) (Risset 1985). These are just two examples of how spectral information can be used to investigate music. Ultimately it may be possible to provide the researcher with the tools that will enable the quantitative rather than qualitative analysis of musical elements.

A large part of my work, in collaboration with Thomas Stainsby, over the last six months has been the development of a computer environment to carry out investigations into the frequency domain representation of sound. This is a continuation of the work carried out by David Hirst and Thomas Stainsby in 1991. The aim has been to provide a unified approach to the analysis and representation of spectral information. The result has been the development of the AnnaLies3.0 program, an application written for the Apple Macintosh computer. The operation of this program was described in a previous edition of Chroma by Thomas Stainsby. Digital signal processing techniques are used to produce a frequency domain representation of sound.

2.0 Frequency Domain Representations.

Digital signal processing (DSP) is a relatively new tool, its increasing accessibility through the rapid development of less expensive and more efficient computer hardware in the past ten years has led many to investigate its possibilities. The fact that through software one can modify the signal processing function of hardware makes DSP extremely flexible in the investigation of sound - a versatility not available in analog processing.

One of the most important tools DSP has given us is a means to represent time domain signals in terms of the frequency domain. A frequency domain representation allows for the investigation of sound purely in terms of frequency content.

The Fourier Transform is the mathematical link between time domain and the frequency domain representations of a signal. This process is directly implemented on digitally sampled sound using the Discrete Fourier Transform. In order for the Discrete Fourier Transform to be useful to the user an efficient, fast algorithm must be designed. Fortunately a series of algorithms exist for the calculation of the frequency spectrum in an extremely efficient manner. These algorithms are grouped together under the title of 'Fast Fourier Transforms' (FFT) and take advantage of elements of symmetry and periodicity inherent in the Discrete Fourier Transform. For a full description refer to Moore (1985), Oppenheim and Schafer (1975), and Proakis and Manolakis(1989) for excellent discussions on the DFT and DSP in general.

The AnnaLies3.0 program provides the user with the means to carry out frequency domain analysis of a signal using the FFT. A Radix 2 decimation in time algorithm has been implemented and yields acceptable speed results on >=68030 Macintoshes. Sounds recorded digitally are analysed by the computer and quickly and accurately yield specific information about the spectral behaviour of the sound. The program takes full advantage of the Macintosh graphic interface in order to make it more useful to a larger range of users. On the Macintosh, the operations are invisible to the user and no specific knowledge of the mathematical process is required.

At this stage of development the system offers different ways of acquiring and representing this spectral information. Ultimately the system will offer varied strategies for the analysis and representation of sound.

The emphasis of the project has not been purely on the precise calculation of the spectrum but equally upon presenting this data in a clear and understandable manner. A graphical representation of the numerical data provides the user with a clearer picture of trends present within the data than would a simple numerical listing.

The most musically interesting possibility for this program is the temporal representation of spectral information. There are three ways in which the program is able to present this data.

Figure 1 is a is a spectral snap-shot, representing the average spectral content of a sound. It provides specific information about the spectrum at a particular point in time. The resolution of this information is dependent on the analysis parameters set up at the beginning of the program. The length of the transform determines the frequency resolution of the snap-shot. There is, however, a trade off. In order to obtain higher frequency resolution a larger portion of sound needs to be analysed, thus, the temporal resolution of the spectral components is reduced. This graph does not allow us to look at the sound in its entirety but only a portion of it. It is, however, possible to look at a series of these snapshots in order to generate an image of the whole sound. A preferable means of presenting this information is, however, available within the program.

Figure 2 is a three dimensional representation of the frequency spectrum. It is built up by the combination of two dimensional spectral plots. This diagram allows for the identification of more general features than the previous two dimensional representation. The program allows this graph to be viewed from several different perspectives in order that different spectral components may be traced visually. Unlike the previous example we are now able to see how partials behave over time and also observe more general features that may be of interest. Where more specific information is required it is possible to return to the two-dimensional plot.

While this representation summarises the general spectral activity over time, it is of little use in the specific analysis of sound. Of greater use would be a method that enables individual partials to be graphed over time. In essence this would require a combination of the above two illustrations - the specificity of the two-dimensional plot with the time range of the three dimensional representation. This feature has been built into the program.

Figure 3 shows a diagram uses an intelligent algorithm to track partials within a sound. Partials are prominent frequency components within a sound, they are related to the physical process of sound production in acoustic signals. The partial tracks are represented as lines on this diagram. Although this graphic projection does not provide detailed amplitude information, it does provide specific information about the temporal organisation of spectral components. It is possible to derive information about the relative ratio of harmonics. In terms of pitched instruments we are then able to derive pitch. It is therefore feasible to identify specific sounds of known

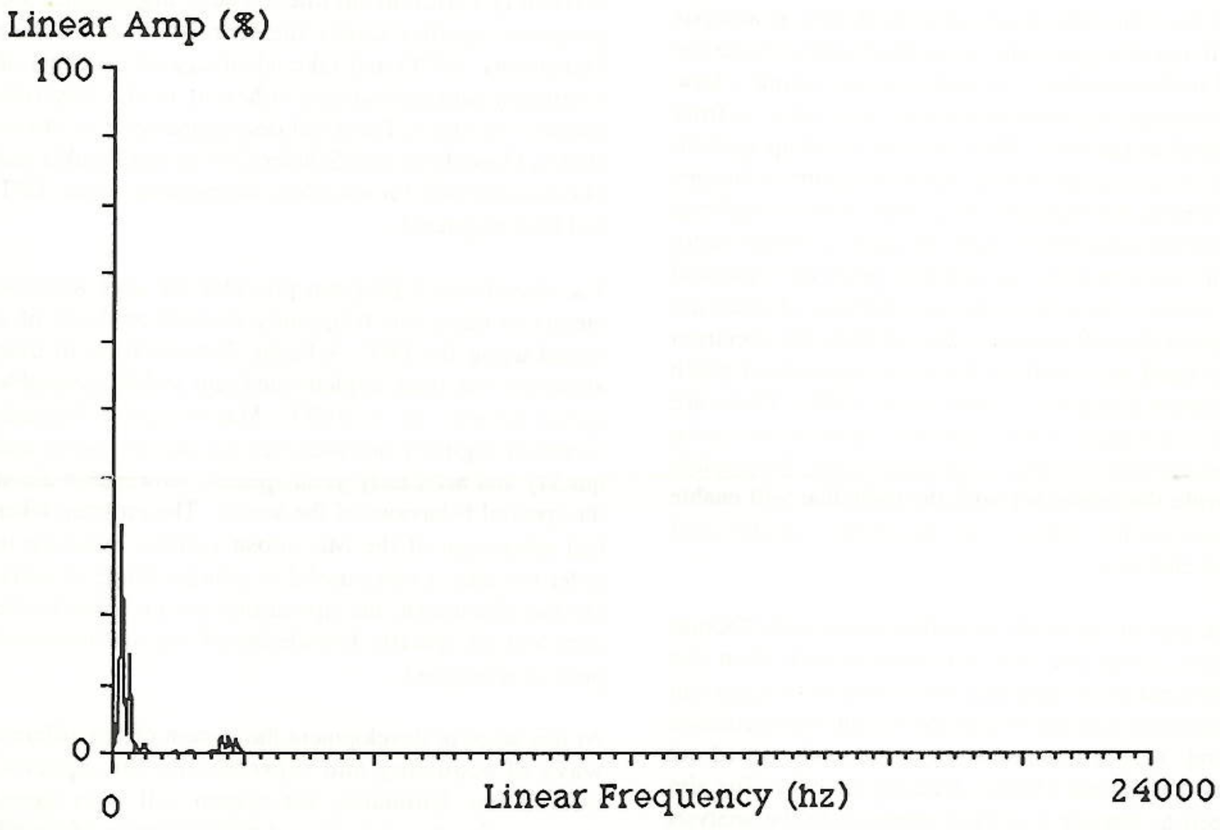

Figure 1 Two Dimensional Amplitude Representation of FFT of one frame. (Amp vs Freq)

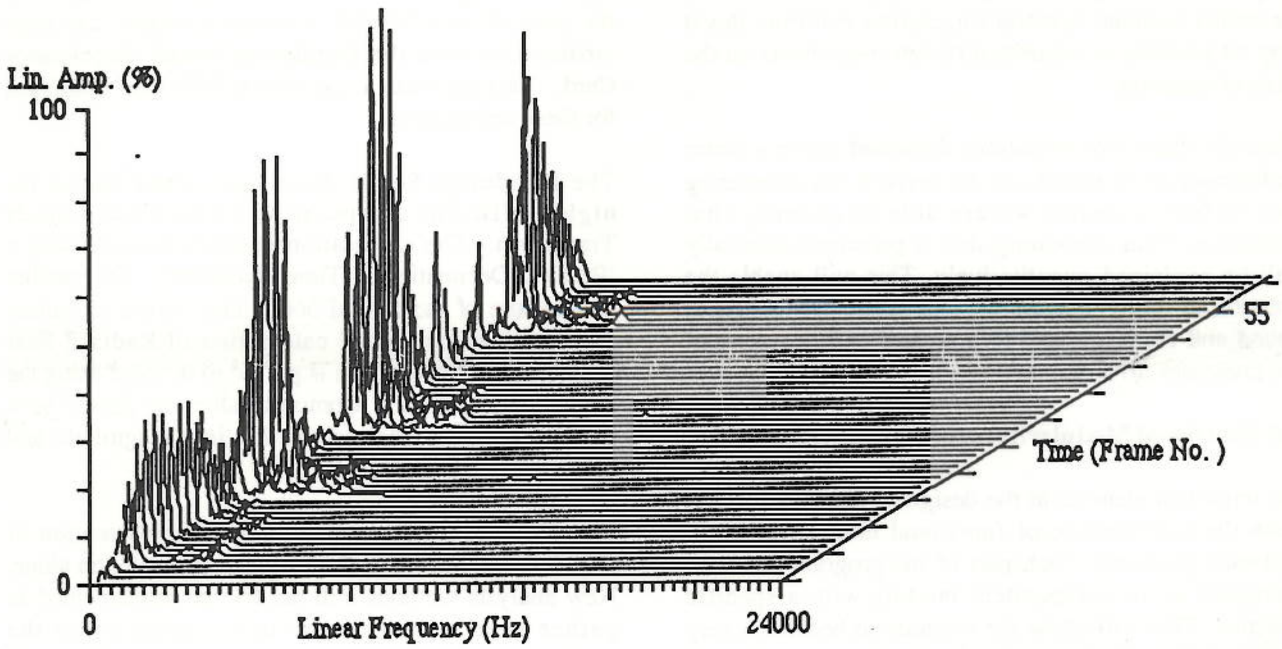

Figure 2 Three Dimensional Amplitude Representation of Successive FFT frames. (Amp vs Freq vs Time)

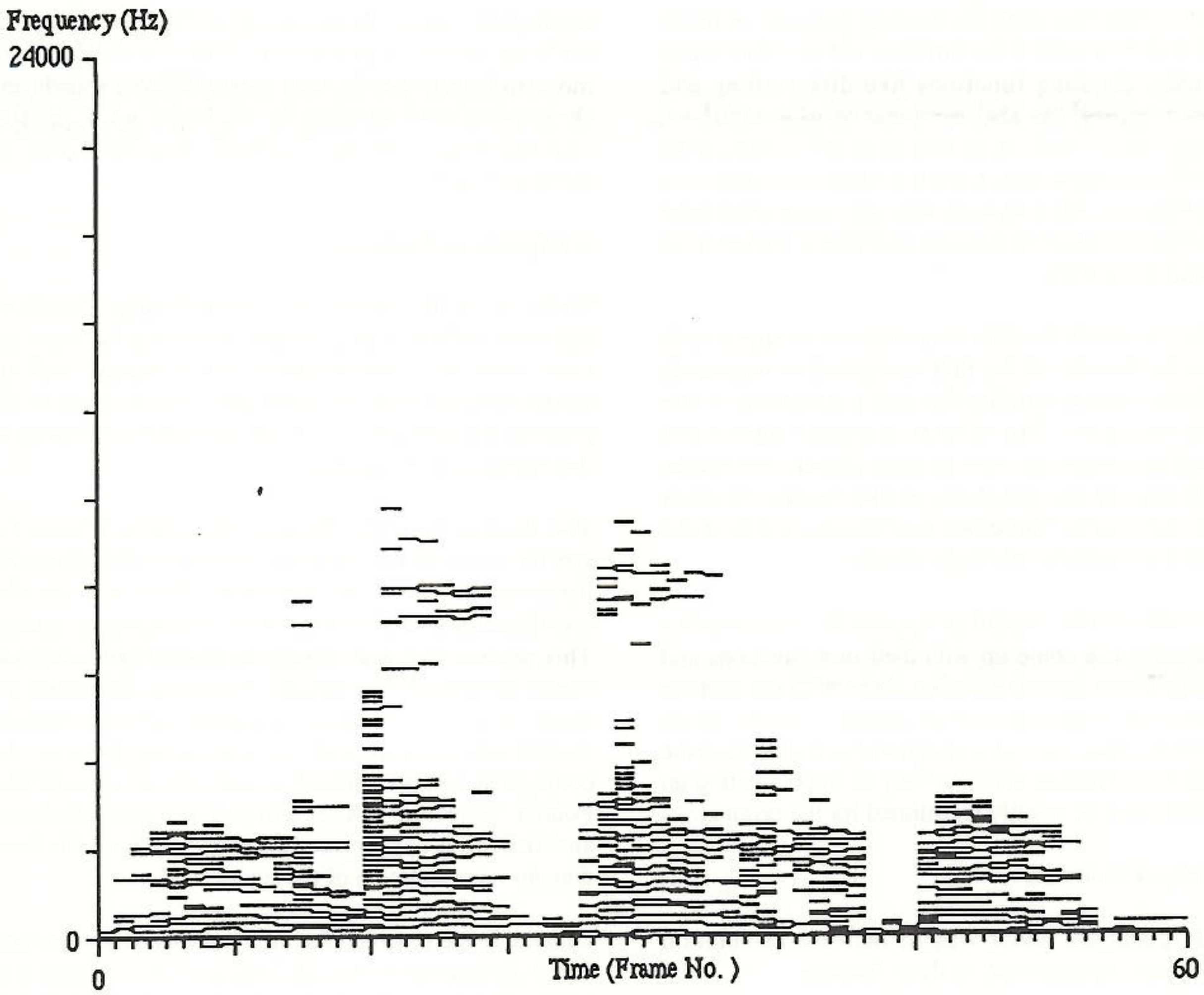

Figure 3 Two Dimensional Representaion Frequency vs Time

frequency make-up. One important application of this is the ability to create spectral fingerprints and from this it may be possible to separate different instruments on the basis of spectrum.

From the three representations discussed above a better understanding of sound can be derived. By comparing two different sounds we are able to quantify that difference. Thus something that is perceived musically will be explained quantitatively. This will enable the creation of more accurate models of the production of sound and therefore lead to a greater understanding of the processes involved.

3.0 Functional Modularity.

An important element in the design of AnnaLies3.0 has been the maintenance of functional modularity of all software processes. Each part of the program has been designed as an independent module with a specific function. This will allow the program to become a very useful platform for future development.

The advantage of modular functionality is that new functions can be added as menu items without any need for a major rewriting of the program. A new function is written as an independent procedure that has a software interface compatible with the existing program. A menu item is added to invoke the function and it is then ready to be used. Existing functions like disk reading and window handling are also available to new additions. These have been made as generic as possible and can be accessed in the same way a software function call calls a library function. This style of software design has been embraced by what is known as Object Orientated Programming (OOP).

The way in which the data structures are designed aids modularity. Results of the FFT are stored as magnitude and phase values suitable for quick graphing in the display modules. The display modules have been designed to graph any two or three dimensional data, this allows for the graphing of the results of other analysis strategies. These are just two examples of the modular approach this package adopts.

The result of the modular approach is that other programmers can come up with their own functions and need only know how to interface them with the graphic interface of AnnaLies3.0 in order to make them available to users. Also if a programmer is able to come up with a better more efficient way of implementing an algorithm it can be readily substituted for the original.

4.0 Analysis Modules.

Already plans are under way to increase calculation performance and add more analysis features.

In order to further optimise the frequency analysis of a signal more specialist hardware can be incorporated. In the case of AnnaLies3.0 a compute engine has been written that uses the Digidesign Sound Accelerator Card. This card has a specialised DSP chip optimised for these calculations.

The Digidesign Sound Accelerator Card allows for highly efficient computation of the Fast Fourier Transform. The calculation is implemented using a 'Radix 2 Decimation in Time algorithm'. The parallel architecture of the central 56001 chip further enhances an extremely optimised calculation of Radix 2 Fast Fourier Transforms. Data is passed to the card using the Apple's NuBuss port. Complex values are passed back to the Apple for final calculation into magnitude and phase ready for graphing.

The potential for frequency domain representation of sounds is not limited to the Fourier representation alone. New analysis methods will need to be implemented to gather even more specific information about the temporal organisation of frequency components. Analysis modules on the drawing board include wavelet and Hartley Transforms. Many of these techniques are mathematically intensive but with the falling price of computer hardware and the ever increasing performance of machines they become closer to a reality. More intelligent feature detection algorithms such as pitch tracking become a possibility. With the collection of more and more specific data about different sounds and their physical correlation we build up a greater understanding of sound in general thus improving our analytical tools.

4.1 Synthesis Modules.

So far in the discussion the Discrete Fourier Transform has been looked at as a means of moving between the time domain representation of a signal and its corresponding frequency spectrum. This is a reversible process and provides one of the potential extensions to the AnnaLies3.0 program.

The Inverse Discrete Fourier Transform allows for specifications in the frequency domain to be translated directly in the time domain. This allows the specification of a sound in terms of frequency content. This process can, once again, be carried out using our highly optimised Fast Fourier Transform. By using the mathematical symmetry property of the Fourier Transform we can find its inverse by passing the conjugated complex frequency data to the normal Fast Fourier Transform. After carrying out this calculation and scaling it appropriately we end up with the time domain representation of the signal.

Ultimately the potential for the synthesis of new sounds, based on spectral models, is unlimited. A composer will have complete control over the musical attributes of these sounds provided the frequency domain correlates are fully understood (Wessel 1988). In the ultimate system, specification of sound will be in terms of musical parameters rather than numerical ones.

5.0 Conclusions.

In conclusion, the heart of any quantitative process is the ability to explain perceptual phenomena in terms of a physical correlate. The program under discussion attempts to approach this in a number of ways. The use of a variety of different representations of elements of the same spectral information helps us build up a visual picture of the sound under examination. The modularity of the program allows for future expansion both in software and hardware.

While still in the preliminary phases of investigation the indications so far are that the AnnaLies3.0 program will provide invaluable tools for the investigation of sound in terms of frequency and ultimately the modification or synthesis of sound specified purely in terms of spectrum.

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If you are interested in obtaining a copy of AnnaLies3.0, contact Chris Scallan or Thomas Stainsby at the addresses shown on the back page of this issue of Chroma.

Gleaned from the Net:

DIVIDED
P.O.Box 8302
chicago, IL 60680
(312) 489-3951
mabed@robbie.acns.nwu.edu

Greetings:

DIVIDED is presently accepting submissions for a series of several compilations to be released beginning in the spring and summer of next year. Any composer working in the electronic / electroacoustic / Computer music fields are welcome to submit compositions for consideration. DIVIDED was founded a year ago to pursue lesser known composers worldwide and give greater exposure to this genre of music. Recently we released "Scend" by Jim O'Rourke and are expecting the release of new works by Jean-Francois Denis early in 1993.

PRESENCE is DIVIDEDs ongoing compilation project. Each compact disc released will contain several composers. All compositions submitted should be on DAT and may not be longer than fifteen minutes each. The deadline to submit entries for the initial CD is January 31st 1993. A properly self-addressed stamped envelope should be included to retrieve one's master tape.

We believe the copyrights should remain with the composers. The production rights shall be given to DIVIDED while the CD is in print and a contract will be drawn allowing composers fifteen percent of the wholesale price of each CD.

By assembling these compilations we wish to bring to listeners new works by international composers which are artistically innovative and dynamic as well as emotionally engaging. DIVIDED wishes to encourage all composers working in these fields to submit compositions knowing that their work will be seriously considered.

Please direct all correspodence to the above address.

Mohammad Abed — Ken Hoenigman
P.O.Box 8302
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(312) 489-3951
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TONALITY SYSTEMS

PRESS RELEASE
21 OCTOBER 1992

Tonality Systems has announced SYMBOLIC COMPOSER, Expert System for Music Composition on the Macintosh and Atari. The software is based on the 6th Generation Common Music Language (CML).

The SYMBOLIC COMPOSER is designed for composers and producers working with TV, film, ambient and electroacoustics, giving them the ability to realise large-scale and complex projects and develop new composition strategies using a set of high-level fractal routines, generators, processors, visualizers and neural networks.

Mr. Peter Stone, Founder of Tonality Systems and designer of the 6th Generation Common Music Language, commented "This system is a breakthrough in MIDI production technology, since it unifies the traditional composition techniques with the latest discoveries of Artificial Intelligence, and is compatible with all current state-of-the-art MIDI sequencers and scorewriters, like Mac's Finale, Performer, Vision, Professional Composer, Encore, Cubase, Mastertracks, Beyond and all the others. In Atari it is compatible with Cubase, C-Lab Notator and Dr.T's KCS."

Mr. Nigel Morgan, composer and music educator commented "SYMBOLIC COMPOSER is a truly remarkable and challenging system. It catalyses you into discovering new ways to think, imagine and interact with your music. Its detailed and authorative manual provides a way into serious programming without having to leave your music behind for a moment".

The composition is written in the 6th Generation Common Music Language (CML) using fractal generators, vector and symbol processors, digital synthesizers, library managers and neural networks to describe musical parameters. The language is very economical and elegant, and even the most complex compositions take only a few kilobytes when written in CML. When the CML file is converted in a MIDI file, the corresponding MIDI file size is in the range 100-1000 kilobytes, which indicates the efficiency of the CML language. The MIDI file can be played with all MIDI-compatible sequencers and scorewriters in full multitasking environment, and improvised MIDI files can also be loaded into SYMBOLIC COMPOSER for further processing. CML is the most powerful programming language for music yet developed on a computer. It contains a full range of high-level composition tools that are effective for any type of music. As all data is processed symbolically the same operations can be applied to different musical parameters, making the language extremely powerful and composer-friendly. The system is easily extendible using programmable neural networks or the underlying Lisp language, allowing the composer to explore the musical possibilities of stochastics, astronomy, physics, mathematics, genetics or other areas of his/hers special interest. All compositional data can also be easily visualized before mapping it onto musical parameters. The built-in generators include among the others Lindenmayer L-system, a system developed to model the growth of plants, Brownian Generator, nature's very own randomizer, Lorenz Attractor, the basic chaotic equation of Gaia, and Associative Generators&Neural Experts that model the natural associative/logic thinking process of the human mind. SYMBOLIC COMPOSER has already generated a lively interest among experimental composers working with MIDI systems and is currently being evaluated by the universities and studios worldwide. Its open architecture and multitasking capabilities make it an attractive tool for composers exploring the far horizons of multimedia and virtual reality. SYMBOLIC COMPOSER is available in USA, UK and GERMANY through dealers specialised in MIDI equipments and software.

For further information, please contact:
Peter Stone, Designer:
+31-20-6757 993 (NL)
Nigel Morgan, Composer:
+44-924-383 017 (UK)

Tonality Systems
Tel&fax: +31-20-6757 993 Veerstraat 55/1
Internet: 100021,2510@compuserve.com

ICMC 1993 Submissions

The 1993 ICMC will be held at Waseda University, in Tokyo Japan, September 10 - 15. The deadline for abstracts of papers, poster sessions and for music submissions is 30th January 1993. For non ICMA members, the fee for music submissions is 3000 Yen (currently about $35 Australian).

Conference information and submission kits are available by contacting:

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Babashita-cho 9, Shinjuku-ku,
Tokyo 162, Japan

Tel: +81-3-3202-7521
Fax: +81-3-3202-7523
Email: icmc93@waseda.ac.jp

MAX 2.5 for the Mac

Since its initial release, Max for the Macintosh has grown to become one of the most elaborate and widely used musical programming applications around. The release of version 2.5 in October this year has seen the addition of a number of new objects, modifications of some earlier objects and further refinement of the Max "shell". For those familiar with Max, the following is a list of newly added Objects, with basic functional descriptions:
bpatcher: Allows subpatchers to be visible and operable inside a patcher window. Useful for grouping collections of user interface objects. Displayed portions can be controlled by the use of offset messages.
cycle: Directs a stream of numbers to successive outlets
env and **envi:** A script-configurable user interface for function editing, oriented toward the task of editing data in synth patches, with graphic representation(envi).
fpic: Displays a picture from a PICT file.
funnel: Maps a number to a list which identifies its inlet.
glove: works with a Mattel Power Glove and reports coordinates and finger positions. Requires a GoldBrick™ interface.
libbang: Sends a bang after all librarian objects have initialised.
libnames: Outputs a librarian's patch names for a menu
movie and imovie: Requires Quicktime init; enable Quicktime movies to be played from within a patcher.
menubar: Allows the creation and modification of existing Macintosh menus by the creation of menu scripts.
playbar: Implements the standard Quicktime movie controller; for use with the movie and imovie objects.
pv: Functions as a localised version of the value object; in other words, this object will only share received values within the same patcher and its subpatches.
spray: Distributes an integer (received as a 2 or more element list) to a numbered outlet.
spell: Converts incoming data such as Symbols numbers and lists to their corresponding ASCII representations.
thispatcher: Enables the control of patches in which the object exists. Examples are updating patcher files, disposing of the patcher, and creating new objects.
urn: Genrates random numbers without repeats.

Additional (Unsupported) objects include:
aebang: Receive an Apple Event.
andGate: Logical and bitwise and of multiple inputs (by David Roach)
orGate: Logical and bitwise or of multiple inputs (by David Roach)
lcd: Draw text and lines in a patcher window (by Michael Lee)

Functional changes and improvements have also been made to a number of objects including Bucket, coll, the libXXX collection and librarian Scripts, menu, message, msd, omsinfo, pcontrol, snd and vdp. Minor changes have been made to the buddy, led, preset, print, seq, serial and ubutton objects.

There is also a number of experimental and unsupported objects from Miller Puckette, David Zicarreli, David Roach and others. Some are quite involved and can take a considerable amount of time and effort to understand. Two large applications have been added to the previously distributed applications bundled with Max. These are an Editor/Librarian for the Emu Procussion drum module which makes use of the newly added librarian features, and MacSoutiLs from INA and GRM; a suite of Composition and MIDI sound effect generation programs.

Changes to the Max Program include a new file format, which resulsts in a 40-50% reduction in file size and faster loading of patchers. Files can optionally be saved in the new, old (ver 2.1 and earlier) or text format. It is also possible to save files in a way that prevents them from being unlocked by anyone (however it seems quite easy to get around this feature by simply copying the locked file).

Improvements have been made to the file search system (File preferences) with subdirectories being searched within specified folders. The order of file search has been re-organised. File aliases with System 7 are supported. This is particularly useful for multi-volume systems.

Briefly, other changes include a menu option to hide or show the Max window at startup, a Max object list with new categories, the ability to save to the clipboard in both PICT and binary formats, Patcher window zoom box, Colour (Color) dialog for objects whose colour can be set and finally, a listing of subpatchers and objects used in a patcher if Get Info is selected with no objects highlighted.

Max now contains over 130 objects. More if you count those considered to be "experimental" and "unsupported"(theres that word again) or those not widely distributed. This suggests a multiplicity of ways to achieve certain tasks and at least one way of doing virtually any. While not adding any groundbreaking new features to Max, version 2.5 has ensured that it is an indispensible component of any Macintosh based music software suite, especially where experimentation is a priority.

The Australian distributor for Max and other Opcode Products is:

Music Technology,
9 Edgecliff Rd., Bondi Junction, 2022.
(02) 3893601

Employment opportunity

Dear Friends,

Faculty of Environmental Information at Keio University in Japan announcesa position opening in Computer Music, which may extend to include Multi-Media Arts. Qualified person is invited to pursue his/her own area of interest. At the same time, s/he is expected to offer technical support to faculty members and students. Interested persons are encouraged to send(no e-mails at this moment please!)

(1) a letter of application,
(2) at least 2 letters of recommendation,
(3) certificate of degrees,
(4) personal history/resume
(5) works recorded on DAT and/or video,
(5) works recorded on DAT and/or video, to

Dr. Hideo Aiso
Dean
Faculty of Environmental Information
Keio University
5322 Endoh, Fujisawa
Kanagawa 252, JAPAN

Following is the description of the position.
Position: Research Associate in Computer Music/Multi-Media Arts Qualification: Higher degrees in computer music and its related areas. Training in computer science /engineering is also welcome.

Responsibility: To maintain the computer music/multi-media lab. and to develop creative environment, which may include developing software and system administration. The person may teach a few courses. Starting salary: Depends on the person's qualification.For example, annual income of a 27 years old person with a Ph.D. would be about $41,600.00 for the first year, $43,500.00 for the second year, and so on.

Starting Date: April 1, 1993
Period of employment:Contract is made annually up to 3 consecutive years.

Deadline of Application: January 31, 1993
Housing: All furnished apartment is provided by the university with charge.

Our on campus facilities include hundreds of computers ranging from mainframes to various kinds of workstations which are connected through optical fiber LAN. In the computer music lab., our current facilities include Mac, NeXT, Sparc, and various MIDI machines.

Keio is Japan's oldest private university founded in 1858. among its 8 faculties, ours is the newest just opened in 1990. The working environment is very friendly and open to any eligible person regardless of religion, age sex, and national origin.

City University Science and Music Conference 1993: ~Timbre Composition in Electroacoustic Music~

Keynote Speakers: Considerable interest has been generated following the first call for papers and we are happy to announce that two keynote speakers have agreed to participate: Steve McAdams (IRCAM and University of Paris V): ~Psychological research on perception and memory of relations among musical timbres~ Denis Smalley (University of East Anglia, UK): UDefining Timbre - Refining Timbre~

The Sessions: Sessions will run daily from 9am- ca 7.00pm with dinner and bar facilities following. In addition to 27 paper sessions (no parallels!) there will be plenary discussions at the conclusion of each day and tape playback sessions (see below). The conference will take place within a performance space with high quality playback available.

Tape playback sessions: On each day there will be a programmed session for tape playback. While primarily to allow delegates to hear complete works referred to in papers, there may be additional time available for other pieces.

Accommodation and subsistence: Accommodation will be in the University~s Finsbury and Heyworth Halls of Residence about 5 minutes walk from the main University precinct and Music Department. Accommodation will be available for the nights of Tuesday 13th of April - Friday 16th April inclusive. The cost of a single room will be £(sterling)17.00 and of a twin £(sterling)28.00 per night (tax included) including breakfast at the Hall. Lunch will be available in the University's self-service refectory. The conference has organised an optional evening meal in the University on the first and second evenings (see booking sheet below).

There will be a Conference Reception hosted by Gordon and Breach Publishers .

Fees: Conference registration: £(sterling)40 Student concession rate: £(sterling)20 Proof of student status will be asked for at registration. The cost of meals is not included in the conference fee. See booking sheet below. Day registration will be available on the days of the Conference only: £(sterling)15 (students £(sterling)8)

All correspondence:
Conference 93,
Music Department,
City University, Northampton Square,
London ECIV 0HB.
fax: 071-477 8576
e-mail: s.emmerson@city.ac.uk
or w.l.windsor@city.ac.uk

CALL FOR PAPERS

City University Music Department's third Science and Music Conference will be held from Wednesday 14th to Friday 16th of April 1993 with the title:
~Timbre composition in electroacoustic music~

Timbre composition in electroacoustic music: papers are invited on all aspects of composition and analysis, of pre-recorded, real-time and live electroacoustic music concerned with an interest in timbre articulation or the relation of sound materials to real world referents. Proposals for Papers adopting cross-disciplinary approaches will be welcomed.

Proceedings of the Conference will constitute a forthcoming issue of Contemporary Music Review (Harwood Academic Press/Gordon and Breach). Conference Director will be Dr Simon Emmerson assisted by Luke Windsor.

Abstract submission deadline: January 1st 1993
Notification of acceptances will be mailed on January 10th 1993.

Submission of Paper (30m presentation):

Please supply a paper title and abstract of no more than 150 words. Please head clearly ~Paper proposal: City Music Conference 1993'. Please include on the same sheet full name, address, telephone number and e-mail number (if available). Please indicate any special equipment needs.

ACCOMMODATION BOOKING FORM

NAME:

ADDRESS:

Telephone/e-mail:

I wish to book accommodation in Finsbury Hall for Conference 93:

Delete one: Single/Double

Tick once for each night required:

Tuesday 13th April
Wednesday 14th April
Thursday 15th April
Friday 16th April

I wish to book meals during the Conference
(please tick): Evening meal Wednesday 14th April (£(sterling)7.50) Vegetarian?
Evening meal Thursday 15th April (£(sterling)7.50) Vegetarian?

This booking form must be accompanied by a deposit of £(sterling)20
- cheque or money order (£ sterling) payable to 'City University':

The balance will be payable at conference registration.

ACMA Contact List

To contact any of the persons mentioned in this issue or for any other information, feel free to write to:

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PO Box 4136
Melbourne University
3052

Electronic mail can be sent to :
mussa@lure.latrobe.edu.au or...
stainsby@klang.latrobe.edu.au

Alternately, you may wish to contact us individually on the following phone numbers:

Steve Adam	(03) 457 655(AH) email: mussa@lure.latrobe.edu.au
Andrew Brown	(03) 344 8721 (BH)
Jason Hellwege	(03) 479 1390 (BH and AH)
Michael Hewes	(03) 457 6551 (BH andAH)
David Hirst	(03) 479 1502 (BH)
Anthony Hood	(02) 882 8343 (BH) (02)428 1734 (fax) email: ianf@extro.ucc.su.oz.au
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