Electronic music and the computer: a critical study of the development of electronic music systems and the introduction of computer-based technology, with particular reference to the interface problems encountered in composer/machine communications

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ELECTRONIC MUSIC AND THE COMPUTER

A Critical Study of the Development of Electronic Music Systems and the Introduction of Computer-Based Technology, with particular reference to the Interface Problems encountered in Composer/Machine Communications

Submitted for the degree of Ph.D. in the University of Durham by

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ABSTRACT

This thesis examines the development of electronic synthesis systems and their associated philosophies from the antecedents of the first half of the twentieth century to the early part of the current decade, viewed in the first instance from a musical rather than a technical standpoint. Such a task is in itself an exercise in communications, for it is necessary to evaluate the artistic characteristics of a rapidly expanding area of interdisciplinary activity which has been largely dominated by technological interests, not always to the best advantage of the art and practice of music.

Although the development of the medium has led to the publication of numerous books and articles, ranging from general descriptions to detailed scientific accounts, the former for the most part are uncritical in their approach, and the latter are rarely presented in a form which may be readily appreciated and commented upon by musicians in general. This account thus attempts to fill a major gap in the literature of the subject by presenting a broadly based critique of its most important historical features. After an introductory perspective of the background to electronic music systems, the first volume continues with a critical study of the main developments during the period 1948-1964, concluding with an examination of the characteristics of both voltage and digitally based control technology and their impact on studio designs during the remainder of the latter decade.

The second volume is primarily concerned with the application of the digital computer, both as a means of directly generating sound information and also as a control device for analogue studios, particular attention being paid to the systems MUSIC 360 and MUSYS.

The overriding objective of this study is to establish a sound and coherent basis for determining the artistic criteria which must be applied if future technical advances are to benefit the creative uses of the medium.

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A considerable quantity of the information supplied by the above has not previously been made available for publication, and it must be stated that copyright in the first instance lies with the originators.

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CHAPTER ONE

The Antecedents of Electronic Sound Synthesis Systems: a Perspective

Until the present century the composition and performance of music has relied almost exclusively on the natural acoustical properties of a variety of instruments, each offering a unique and relatively limited range of functional characteristics. The composer Carlos Chavez, writing in 1937, noted that 'Musical instruments have not changed substantially in seven thousand years. In the course of this long period of time there has been a great improvement in the construction and playing of these instruments ... But during seventy centuries there did not appear a single musical instrument containing a new sound agent, or showing a new procedure of vibrating its agent. We received our present sound material complete from pre-history.'[1]

The reasons for this lack of innovation lay not in any deficiency of inventiveness on the part of instrumental designers but more basically with the existence of distinct limitations concerning the techniques available for the generation and modification of sound information. Strings, skins, reeds or air columns acted as the prime vibratory sources for all the pitched instruments developed during this period, and our whole heritage of music had been based on an exploitation of their vibrational characteristics.

The invention of the direct current arc oscillator in 1900,

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[1] Carlos Chavez, 'Towards a New Music' (N.Y., 1937), extract reproduced in Composers on Music, Ed. Samuel Morgenstern (Faber, London, 1950), p.523

followed six years later by the first electronic triode valve,[2] heralded a significant change in this situation, associated with the birth of a totally new field of applied science; electronic engineering. During the first half of the twentieth century advances in this area of technology were primarily concerned with the development of communication systems which were no longer fully dependent on the limited characteristics of direct acoustical or visual transmission. The discoveries of Edison and Berliner[3] during the latter part of the nineteenth century, regarding the storage and retrieval of sound information in a mechanical form as vibratory patterns cut into a revolving wax cylinder or disc, led to a parallel succession of important technological developments, including in particular the acoustic, and later the electro-mechanical gramophone.

The primary purpose of these advances was thus the development and perfection of essentially passive systems for the detection, transmission, storage and retrieval of acoustically generated source material, preserving as far as possible the original quality of the latter. The possibilities of active sound generation and processing techniques were not entirely neglected during this period. The technological products of ventures in this field were nevertheless

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[2] W. Duddell developed the first direct current arc oscillator in the U.S.A. in 1900. The first vacuum tube triode valve was developed by Lee de Forest (b.1873 Iowa, d. 1961 California) in 1906.

[3] During the period 1877-1896 Thomas Edison (b.1847, d.1931) and Emile Berliner (b.1851, d.1929) independently developed and patented the cylindrical and disc phonograph systems.

relatively few in number, principally taking the form of devices intended to function as synthetic counterparts to the standard range of orchestral instruments. These included the *Dynamophone*,[4] the *Sphärophon*,[5] the *Theremin*,[6] the *Ondes Martenot*,[7] the *Trautonium*[8] and the *Hammond Organ.*[9]

[4] The *Dynamophone*, also known as the *Telharmonium*, invented in America by Thaddeus Cahill (b.1867, d.1934). Patent registered 1897 U.S.A. no. 580.033. Prototype completed 1906. Expanded model completed 1914. This instrument was based on a sine wave generation system consisting of a series of toothed wheels which were rotated near the poles of matching electromagnets to act as inductors. Multiple keyboards could be attached to provide simultaneous performances from different players. A complex bank of switches controlled the timbral colouring of the pitches, adding harmonic generators to the fundamental note. The device was essentially a specially modified dynamo weighing in the order of 200 tons. The output was transmitted over telephone wires to receivers in an adjoining room. See Thaddeus Cahill, 'The Generating and Distributing of Music by Means of Alternators', *Electrical World*, XLVII, (McGraw Publishing Co., 1906), p.519. Also references in Feruccio Busoni, *Entwurf einer Neuen Aesthetik der Tonkunst* (Trieste, 1907, 2nd enlarged edn., Leipzig, 1910); English edn. Feruccio Busoni, *Sketch of a New Esthetic of Music*, tr. Theodore Baker (G. Schirmer, N.Y., 1911), p.33, and Joseph Schillinger, 'Electricity, a Musical Liberator', *Modern Music (Journal of the League of Composers)*, VIII, 3 (N.Y., 1931), pp.26-31.

[5] The *Sphärophon*, also known as the *Electrophone*, invented in 1927 by Jörg Mager (b.1880, d.1939) primarily as an aid to the performance of quartertone music. A monophonic instrument generating tones produced by the beat frequency principle where two high frequency generators are tuned and modulated to produce the desired audio frequency as a difference tone. A facility to alter the generator waveforms permits some modification of the tonal quality.

[6] The *Theremin*, also known as the *Thereminovox* or *Aetherophone*, invented by Léon Theremine (b.1896, d.1939) in Russia 1919-1920, presented in an improved version 1924, and marketed commercially in association with the Radio Corporation of America, New Jersey, 1929. This is a monophonic instrument, again generating tones by means of the beat frequency principle. The design is unique for its method of pitch control which is derived from the variable capacitance set up between a projecting metal rod and

one hand of a performer moved towards and away from it. Amplitude control is provided in a similar fashion by the movement of the performer's other hand in relation to a horizontal loop. A foot switch is also available to cut off any note and stop the instrument sounding on through rests.

7 The Ondes Martenot, invented in 1928 by Maurice Martenot (b.1898). A monophonic instrument employing beat frequency oscillators and playable either discretely by means of a keyboard or as a continuously variable pitch instrument using a metal ring worn on one finger of the right hand and slid up and down a carbon track. The latter is normally mounted just in front of the keyboard permitting easy interchange between the two systems. The left hand is employed both to operate a set of switches, which alter the timbre of the output by controlling a series of filter circuits and additive generators tuned at fifths and octaves, and also to control the intensity of output by means of a spring lever which may be depressed to varying degrees. In some versions amplitude may also be controlled by the depth to which the keys on the keyboard are depressed.

8 The Trautonium. A monophonic instrument invented in 1930 by Friedrich Trautwein (b.1886, d.1956) which produces sounds by the direct application of the output of a sweep generator to a thyratron, generating sawtooth waves. Pitch is continuously variable, controlled by depressing a stretched resistance wire at appropriate points along its length with a steel bar. This instrument was specially expanded as the Electronic Monochord of the Cologne Studio für Elektronische Musik, Norwestdeutschen Rundfunks in the early 1950s and this will be described in detail in the next chapter.

9 The Hammond Organ, invented in 1935 by the Hammond Electric Company of Chicago, U.S.A. This instrument generates tones by employing electromagnetic induction techniques similar to those used by the Dynamophone. In the original model a synchronous motor drives a bank of 91 polygonal discs mounted on a common shaft. Each disc passes in front of a magnet around which is coiled a wire. As each highpoint of the disc passes in front of the magnet an alternating current is induced in the coil. The periodic waveshape thus produced is amplified and passed to a loudspeaker. Different timbres are produced by providing discs with different patterns, and these function like drawstops on a conventional organ. This instrument is not an electronic organ in its true sense since it does not rely wholly on the response of all-electronic circuits.

Such instruments, despite their innovations regarding the synthetic production of new timbres, were restricted by their design specifications, which were modelled closely on the characteristics of traditional acoustical instruments. The development of these extensions to standard instrumentation was influenced by two important factors. Firstly, the birth of a totally new means for generating sound information had opened up a field for investigation offering seemingly unlimited possibilities, and it was perhaps inevitable that the initial steps taken towards acquiring a technical understanding of such an unknown medium should be based upon the established range of instrumental characteristics. Secondly, the opening up of new horizons for the processes of sound generation and manipulation necessitated a major evaluation of their uses as part of the language of music, and during the first part of the century such concepts were essentially in a state of gestation.

The history of those compositional developments which created the background for the birth of the first sound processing studios is an area of research which has already been undertaken by David Piper as the major part of his doctoral thesis Electronic Music, completed in 1968.[10] It is nevertheless relevant to highlight some of the major areas of interest which arose during the first half of the twentieth century, for they throw useful light on the artistic

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[10] David Piper, Electronic Music, subtitled Music Concrète and Elektronische Musik: A Comparative Study of the Two Media During their Early Formative Period, and an Attempt to Trace the Sources of their Perspective Development in the Musical Thought and Practice of the Twentieth Century, Ph.D. thesis (Victoria University of Manchester, 1968).

climate which led to the development of the first sound generation and processing systems after the second world war.

One of the earliest attempts to employ non-traditional sound generation techniques as part of a communicative art form arose as part of the activities of the Futurist movement, pioneered by the Italian poet, Filippo Tommaso Marinetti, with the publication of his Manifesto of Futurist Poetry'in the newspaper Le Figaro, Paris on 12 February 1909. This document called for the establishment of a new poetry reflecting the age of machinery and the rejection of traditional forms of poetic expression. The musical side of the movement was launched on 11 October 1910 by Balilla Pratella with his Manifesto of Futurist Musicians[11] calling for a rejection of traditional musical principles and methods of teaching. The manifesto included, interestingly, a declaration that one should consider 'the reign of the singer as finished and demand that the vocal part in a work of art should correspond to that of an orchestral instrument', and another stating that 'we must liberate the individual musical consciousness from all imitation of the past, we must feel and sing with one's soul directed towards the future, receiving inspiration from nature in all its manifestations, human and extra human'.[12]

Five months later, on 11 March 1911, Pratella published his

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[11] Balilla Pratella, 'Manifesto of Futurist Musicians', open letter (Milan, 11 Oct. 1910), reproduced in Music since 1900, Ed. and tr. Nicholas Slonimsky (Cassell, London, 4th edn., 1971), pp.1294-1296.

[12] Ibid., p.1296.

Technical Manifesto of Futurist Music[13] which contained more definitive conclusions as regards the form of Futurist music. It included, for example, a call to 'master all expressive technical and dynamic elements of instrumentation and regard the orchestra as a sonorous universe in a state of constant mobility, integrated by an effective fusion of all its constituent parts', and a declaration that 'All forces of nature tamed by man through his continued scientific discoveries, must find reflection in composition - the musical soul of crowds, of great industrial plants, of trains, of transatlantic liners, of armoured warships, of automobiles, of aeroplanes. This will unite the great central motives of a musical poem with the power of a machine and the victorious reign of electricity.'[14]

Two years later to the day another futurist, Luigi Russolo, published a related manifesto entitled The Art of Noises[15] as an open letter to Pratella which proposed the composition of works based entirely on the use of sound sources, such as those suggested above. 'Musical sound is too limited in qualitative variety of timbre. The most complicated of orchestras reduce themselves to

[13] Balilla Pratella, 'Technical Manifesto of Futurist Music', open letter (Milan, 11 March 1911), reproduced in Music since 1900, Ed. and tr. Nicholas Slonimsky (Cassell, London, 4th edn., 1971), pp.1296-1298.

[14] Ibid., p.1297 and 1298.

[15] Luigi Russolo, 'The Art of Noises', open letter to Balilla Pratella (Milan, 11 March 1913), reproduced in Music since 1900, Ed. Nicholas Slonimsky, tr. Stephen Somervell (Cassell, London, 4th edn., 1971), pp.1298-1302.

four or five classes of instruments differing in timbre: instruments played with the bow, plucked instruments, brass winds, woodwinds and percussion instruments ... We must break out of this narrow circle of pure musical sounds and conquer the infinite variety of noise sounds.[16] This document is notable for its appreciation of the relevance of acoustic laws to the generation of musical structures from noise sources:

> We must fix the pitch and regulate the harmonics and rhythms of these extraordinarily varied sounds. To fix the pitch of noises does not mean to take away from them all the irregularity of tempo and intensity that characterises their vibrations, but rather to give definite gradation of pitch to the stronger and more predominant of these vibrations. Indeed noise is differentiated from musical sound merely in that the vibrations that produce it are confused and irregular, both in tempo and intensity. Every noise has a note - sometimes even a chord - that predominates in the ensemble of its irregular vibrations. Because of this characteristic pitch it becomes possible to fix the pitch of a given noise, that is, to give it not a single pitch but a variety of pitches without losing its characteristic quality - its distinguishing timbre. Thus certain noises produced by rotary motion may offer a complete ascending or descending chromatic scale by merely increasing or decreasing the speed of motion.[17]

The practical manifestations of his proposal involved the construction of specially designed noise instruments: Intonorume, in collaboration with the percussionist Ugo Piatti. The first public performance of the Art of Noises took place on 2 June 1913 at the Teatro Storchi, Milan, barely three months after the publication of the manifesto and with only some of the Intonorume completed. A

[16] Russolo, 'The Art of Noises', op.cit., p.1299

[17] Ibid., pp.1300-1301

second altogether more successful performance using the full complement of instruments was given as part of a concert of Futuristic music, presented by Marinetti and Russolo on 21 April 1914 at the Teatro dal Verne, Milan.

The historical interest in this venture lies not so much in the acoustical design features of the Intonorume themselves, which in any event have been long since destroyed, but more in the motivation which led to their construction; the desire to use sound in its most liberal context as a creative art form. The Futurist movement did not succeed in its attempt to produce a major revolution in the path of new music, but its challenging of traditionally accepted relationships between the science of acoustics and the art of producing musical sounds was to prove singularly prophetic.

Feruccio Busoni (b.1866, d.1924) had already attacked traditional nineteenth-century musical practices in his Entwurf einer Neuen Aesthetik der Tonkunst, first published in 1907,[18] advocating a re-appraisal of the whole language of music 'free ... from architectonic, acoustic and esthetic dogmas'.[19] In championing the use of Cahill's Dynamophone[20] he became possibly the first major composer to suggest that electronic technology might play an important part in shaping the music of the future.

This book caught the attention of the French composer, Edgard

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[18] Busoni, Sketch of a New Esthetic of Music, op.cit.

[19] Ibid., p.34

[20] Ibid., p.33

Varèse (b.1883, d.1965), who just prior to its publication had abandoned Paris for Berlin, temporarily, in an attempt to rationalise his personal struggle with new concepts of musical expression away from the restriction of the traditional doctrines associated with the teaching of the Conservatoire. Varèse, more than any other composer of his time, pioneered in his instrumental music the aesthetics which were necessary for the acceptance of electronic sound processing techniques as part of the language of musical composition. It is thus particularly tragic that it was not until the 1950s, towards the very end of his life, that he gained access to the technological facilities he so fervently desired. As early as 1916 he was quoted in the New York Telegraph as saying 'Our musical alphabet must be enriched ... We also need new instruments very badly ... In my own works I have always felt the need for new mediums of expression.'[21] He was quick, however, to refute suggestions that his efforts were direct by-products of the Futurist movement:

> The Futurists (Marinetti and his noise artists) have made a serious mistake ... Instruments, after all, must only be a temporary means of expression. Musicians should take up this question in deep earnest with the help of machinery specialists ... What I am looking for are new technical mediums which can lend themselves to

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[21] New York Telegraph, March 1916, quoted in Fernand Ouellette, Edgard Varèse, English edn. tr. Derek Coltman (Caldar and Boyars, London, 1973), p.46. Also quoted in Chou Wen Chung, 'Varèse, a Sketch of the Man and his Music', Musical Quarterly, Vol.LII,No.2 (Schirmer, N.Y., April 1966), p.165.

every expression of thought.[22]

Varèse had become acquainted with the electronic designer René Bertrand[23] in May 1913, and this marked the start of a long and lasting friendship. In 1922 during the composer's first stay in America he declared in an interview for the Christian Science Monitor that 'What we want is an instrument that will give us continuous sound at any pitch. The composer and the electrician will have to labor together to get it ... Speed and synthesis are characteristics of our own epoch.'[24]

His ideas at this stage were clearly still based on the concept of an electronic performance instrument rather than a sound manipulation system. The 1920s, however, saw the birth of the commercial 78 r.p.m. gramophone record and the development of electrical recording systems,[25] making more generally available a technique for

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[22] Quoted in Chou Wen Chung, op.cit., p.151. Busoni also refuted a suggestion that his theories were part of the Futurist movement, alleged by Hans Pfitzner in a pamphlet entitled Futuristengefahr, published in the Suddeutsche Monatshefte, retorting angrily 'The word "Futurism" is not used on any page of my little book. I have never attached myself to a sect. Futurism, a movement of the present time could have no connection with my argument.' Quoted in F. Busoni, The Essence of Music and Other Papers, tr. Rosamond Ley (Rockliff Publishing Corporation, London, 1957), p.18.

[23] René Bertrand invented the Dynaphone (not to be confused with the Dynamophone, see note 5) during 1927-28. This instrument is keyboard controlled, generating pitches by means of the beat frequency principle. Timbre may be modified additively by switching in extra generators tuned to the fifth and octave and subtractively by applying filters to the output.

[24] Christian Science Monitor, 8 July 1922, quoted in Fernand Ouellette, Edgard Varèse, op.cit., p.76

[25] Electrical recording, developed by Harris and Maxwell, was introduced in 1926. In 1927 the gramophone industry was given a marked impetus by the issue of a comprehensive series of records by several companies to mark the 100th anniversary of Beethoven's death.

the storage of sound information and also a means for effecting certain acoustical transformations. Darius Milhaud (b.1892, d.1974) realised that changing the speed of a record varies not only the pitch but also the intrinsic acoustical characteristics of the material, and during the period 1922 to 1927 carried out several experiments to create vocal transformations. Percy Grainger (b.1882, d.1961) performed similar experiments during the 1930s, paying particular attention to the use of piano sounds as source material. In both cases, however, their interest was only of a temporary nature, and electronic sound transformation techniques never became a significant part of their musical languages.

During the 1920s Varèse continued his search for new sound textures, but without the assistance of the technological facilities he so much desired. His work with natural instrumental resources in his first published compositions[26] was nevertheless singularly prophetic, for he was concerned to achieve effects which were to become primary characteristics of electronic sound processing; analysis and re-synthesis. He experimented, for example, with the simulation of reversed attack characteristics on individual instrumental notes (easily achieved today by playing tape recordings of naturally produced notes backwards), and the use of instrumental sounds as materials which may be montaged and moulded in terms of

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[26] *Amériques*, 1918-22, first performed 9 April 1926, Philadelphia
Offrandes, 1921 , first performed 23 April 1922, New York
Hyperprism, 1922, first performed 4 March 1923, New York
Octandre, 1923, first performed 13 January 1924, New York
Intégrales, 1924, first performed 1 March 1925, New York
Arcania, 1925-27, first performed 8 April 1927, Philadelphia

textures rather than individual melodic lines: 'Taking the place of the old fixed linear counterpoint you will find in my works the movement of masses varying in radiance, and of different densities and volumes.'[27]

Percussion instruments figured prominently in his works; Ionisation, for example, being scored entirely for instruments of this family.[28] With the aid of such less common effects such as sirens, whips, a lion's roar and sleigh bells he struggled to develop a compositional art which integrated the natural sources of the environment with more traditional means of musical expression to form a continuum. This was not the somewhat crude Futurist Art of Noises exploring the exotic, but an attempt to extract an artistic perspective from the sound world which surrounds any artist of this century who does not seek to detach himself totally from the reality of an increasingly technological environment.

Varèse was not without his imitators; for example, the American composer George Antheil (b.1900, d.1959) whose Ballet Méchanique, first performed in Paris, 1926, and again in New York, 1927, required car horns, aeroplane propellors, saws and anvils. The work of Joseph Schillinger (b.1895, d.1943) which has remained largely unrecognised, is also of interest in this context. Schillinger, a Russian by birth, travelled to America in 1928 in response to an invitation from the American Society for Cultural Relations with Russia, remaining there

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[27] Quoted in Fernand Ouellette, Edgard Varèse, op.cit., p.84

[28] Ionisation, for percussion ensemble of 13 players, 1930-31. First performed 6 March 1933, New York.

until his premature death fifteen years later. He quickly entered into a collaboration with his fellow ex-patriate Leon Theremin over the applications of electronic sound generation[29] and in 1929 composed and conducted the first performance of his First Airphonic Suite for R.C.A. Theremin and Orchestra with Theremin as soloist.[30] During the succeeding years he became deeply preoccupied with the theoretical aspects of musical composition seen within an increasingly scientific environment, and finally produced a rather curious set of twelve books in 1940 entitled Kaleidophone: The Schillinger System of Musical Composition[31] which made a considerable impact on younger composers of the time. Despite some rather curious features, including statistical data supposedly indicating

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[29] See note 6. The Theremin had already been developed five years earlier in 1924. Subsequent activities were mainly concerned with perfecting and modifying versions of the instrument including the commercial R.C.A. Theremin. The lack of a keyboard facility resulted in many compositional problems and involved frequent criticisms.

[30] Cleveland, Ohio, 28 November 1929.

[31] Joseph Schillinger, The Schillinger System of Musical Composition: A Scientific Technique of Composing Music (Carl Fisher, N.Y., 1946). Twelve books in two volumes:

Vol.1	Book 1	Theory of Rhythm
	2	Theory of Pitch Scales
	3	Variations of Music by Means of Geometrical Progressions
	4	Theory of Melody
	5	Special Theory of Harmony
	6	Correlation of Harmony and Melody
	7	Theory of Counterpoint
Vol.2	8	Instrumental Forms
	9	General Theory of Harmony (Strata Harmony)
	10	Evolution of Pitch Families (Style)
	11	Theory of Composition
	12	Theory of Orchestration

as a percentage the degree of stylistic consistency displayed by some of the major classical composers in their works, and the formulation of a set of compositional rules based on empirical analyses of musical structures, his theories do contain some features of interest. He attempted, for example, to analyse musical sounds in terms of musico-acoustic parameters such as melody, rhythm, timbre, harmony, dynamics, density, and in so doing displayed at least an understanding of the specification problems which were to become major stumbling blocks in the later developments of electronic and computer sound generation systems. His theories were further consolidated in a monumental work, The Mathematical Basis of the Arts which was completed two years later. Neither of these works, unfortunately, was published until after his death. His interest in fostering the use of scientific techniques in the composition and execution of music, however, had been illustrated in an article entitled Electricity, A Musical Liberator which appeared in the periodical Modern Music during 1931.[32]

> The growth of musical art in any age is determined by the technological progress which parallels it. Neither composer nor performer can transcend the limits of the instruments of his time. On the other hand technical developments stimulate the creation of certain forms of composition and performance. Although it is true that musicians may have ideas which hurdle these technical barriers, yet, being forced to use existing instruments, their intentions remain unrealised until scientific progress comes to the rescue ... If we

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[32] Joseph Schillinger, 'Electricity, A Musical Liberator', Modern Music, VIII (Journal of the League of Composers, N.Y., March 1931), pp.26-31.

> admit that the creative imagination of the composer may form musical ideas which, under the specific conditions of a given epoch, cannot be translated into sounds, we acknowledge a great dependence of the artist upon the technical position of his era, for music attains reality only through the process of sound.[33]

Towards the end of 1927 Varèse became restless to learn more about the possibilities of electronic instruments. His biographer, Fernand Ouellette notes:

> There was still no inventor creating an instrument which would correspond to Varèse's needs. Jorg Mäger who had introduced the Sphärophon seemed to have escaped [his] attention. Lèon Theremine had not yet arrived in New York: Bertrand and Martenot were working in Paris. Trautwein had not yet demonstrated his Trautonium.[34]

During the same year Varèse contacted Harvey Fletcher, the director of the acoustical research division of Bell Telephone Laboratories, with the object of obtaining access to facilities suitable for the development of new electronic instruments. Fletcher took an interest in his proposals but could not offer the necessary extra funds for such work. Varèse accordingly departed for Paris in the autumn of 1928 both to present his work to Parisian audiences and also to ascertain from Rene Bertrand at first hand what technological developments had taken place in his absence. The most interesting result of this visit, as far as this present study is concerned, was the formulation of a project to develop

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33 Joseph Schillinger, 'Electricity, A Musical Liberator', *op.cit.*, p.26

34 Fernand Ouellette, *Edgard Varèse*, *op.cit.*, p.96

what might have become the first sound processing studio and associated school of composition. Although the details were never officially published, his biographer, Fernand Ouellette, managed to obtain a copy of this document from Ernst Schoen, Varèse's first pupil. The proposal ran as follows:

> Only students already in possession of their technical means will be accepted in the composition class.
> In this department, studies will concentrate upon all forms required by the new conceptions existing today, as well as the new techniques and new acoustical factors which impose themselves as the logical means of realising those conceptions.
> Also under Varèse's direction with the assistance of a physicist, there will be a working laboratory in which sound will be studied scientifically, and in which the laws permitting the development of innumerable new means of expression will be corroborated outside all empirical rules. All new discoveries and all inventions of instruments and their uses will be demonstrated and studied.
> The laboratory will possess as complete a collection of phonographic records as possible, including examples of the music of all races, all cultures, all periods and all tendencies.[35]

The scheme never materialised, but its philosophy even in such a general expression of intent shows an understanding of the methods of approach essential to any use of technological processes for artistic ends. The composer must establish a system of communication with the technologist and his facilities which will foster a rationalisation of the very different semantic conventions associated with the two disciplines of music and science. Accordingly the musician and the physicist must work together from

[35] Fernand Ouellette, Edgard Varèse, op.cit., pp.102-103

the very outset, achieving a mutual degree of understanding through the processes of pedagogic investigation. In such an environment the physicist may develop an understanding of the creative and thus essentially subjective characteristics of musical composition, and the musician may similarly learn to appreciate the objective limitations as well as the possibilities of technological systems.

The main reason for the failure of Varèse's project to come to fruition was a lack of adequate funds. On 1 December 1932, whilst still in Paris, he wrote again to Harvey Fletcher requesting access to the facilities of the Bell Telephone Laboratories in return for his services to the company: 'I am looking to find a situation where my collaboration would have value and be worth pecuniary return.'[36] Varèse was so desperate for laboratory facilities that he was even prepared to sacrifice, at least for a time, his career as a composer. He also applied to the John Simon Guggenheim Memorial Foundation for a grant towards his work. In response to a request for more details he wrote again to the Foundation on 6 February 1933 offering the following proposal:

> The acoustical work which I have undertaken and which I hope to continue in collaboration with René Bertrand consists of experiments which I have suggested on his invention, the Dynaphone. The Dynaphone (invented 1927-28) is a musical instrument of electrical oscillations somewhat similar to the Theremin, Givelet[37] and Martenot electrical instruments. But its principle and operation are entirely different, the resemblance being only

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[36] Fernand Ouellette, Edgard Varèse, op.cit., p.129

[37] An electronic organ using valve oscillators, developed in 1930 by two French inventors, Coupleux and Givelet.

> superficial. The technical results I look for are as follows:
>
> 1 To obtain absolutely pure fundamentals.
> 2 By means of loading the fundamentals with certain series of harmonics to obtain timbres which will produce new sounds.
> 3 To speculate on the new sounds that the combination of two or more interfering Dynaphones would give if combined in a single instrument.
> 4 To increase the range of the instrument so as to obtain high frequencies which no other instrument can give, together with adequate intensity.
>
> The practical result of our work will be a new instrument which will be adequate to the needs of the creative musician and the musicologist. I have conceived a system by which the instrument may be used not only for the tempered and natural scales, but one which also allows for the accurate production of any number of frequencies and consequently is able to produce any interval or any subdivision required by the ancient or exotic modes.[38]

This application, unlike his previous proposal, laid down for the first time the acoustical principles which would serve as the basis for research into the musical applications of sound technology. His allegiance to the Dynaphone must be attributed to his close friendship with its inventor, René Bertrand. This instrument, despite his assertions, did not differ significantly from its relatives, for example, its ability to generate additive timbres using fifths and octaves, is matched by a similar facility in the Ondes Martenot. A more positive observation, however, must be that Varèse knew the designer, and as a result of their extensive discussions both were aware of the potential of expanding the circuit designs to produce not merely a more versatile electronic

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[38] Fernand Ouellette, Edgard Varèse, op.cit., p.129

instrument, but what might have been more accurately described as a sound synthesis system, the heart of an electronic music studio.

The proposal is remarkable for its time, for it shows that Varèse had developed an advanced level of understanding regarding the procedures involved in the production of electronic sound complexes, over a decade and a half before the birth of the first synthesis studios. His first objective, the production of pure fundamental tones, involved the isolation of the most basic periodic wave shape: the sine wave. His second was concerned with the combination of sine waves tuned to the harmonic series of a given fundamental to give distinctive timbres in accordance with Fourier's theories of wave form synthesis.[39] This technique has subsequently become of major importance for both electronic and computer sound synthesis. His third objective, using timbres available from the Dynaphone, would have led directly to an examination of amplitude and frequency modulation techniques as a means for generating sound complexes, and the fourth would have explored the ear's response to frequency regions higher than those normally generated by conventional instruments.

As will be seen in the next chapter, such a line of investigation could well have led directly to the work of one of the principal pioneering groups which emerged during the late 1940s and 50s: the Cologne Studio für Elektronische Musik, concerned initially with the use of all-electronic sound generation techniques.

[39] The principles and application of Fourier techniques will be discussed in subsequent chapters.

The work of this group contrasted sharply with that of the pioneers of Musique Concrète in Paris, who, under the leadership of Pierre Schaeffer, developed studio facilities for the manipulation and transformation of sound material derived almost exclusively from natural sources. It has already been seen, however, that antecedents of the latter techniques may also be traced in the compositional writing of Varèse at this time[40] indicating an equally important link to this second group of pioneers.

The far-sightedness of Varèse's ideas, however, extends far beyond the birth and early development of these two schools of thought, for a dogmatic rift quickly arose between the two centres which continued well into the 1950s. He was thus able, in 1933, to foresee a time more than twenty-two years later when the rigid distinctions between Concrète and Elektronische Musik techniques began to dissolve, providing a more common basis for a wide range of sound generation and processing systems, and their associated philosophies. His only piece entirely for electronic tape, Poème Electronique, written twenty-five years later for the Brussels World Fair 1958, realised many of the concepts of sound structure and organisation he had established during the 1930s. This piece also provided a major turning point for the composer Karlheinz Stockhausen in his own use of technology in music, demonstrated by the differences between the closely defined structures of Gesang der Jünglinge (1955-56) and Kontakte (1959-60), and the universality of Telemusik (1960) and Hymnen (1966-67):

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[40] See above, p.11.

> Varèse is alone in his generation in having composed a work of electronic music and furthermore in having heralded in this Poème a modern formulation of compositional relationships whose true significance can only today be recognised: namely the sequential presentation and superimposition - even though sometimes abrupt and unmediated - of events of a heterogeneous nature (for instance, extremely realistic events, events resembling musical hoardings, and freely invented events)... Anyone living today - Varèse was at the time living in New York - is confronted daily with the hurtling together of all races, all religions, all philosophies, all ways of life ... of all nations. In works by the musician Varèse this bubbling of the cauldron is aesthetically portrayed ... New York, that prime blueprint for a world society, is without question an indispensable experience for the contemporary artist. Ideas one might have about possible integration, about a coherent unification, or about possible syntheses of the influences issuing from all parts of the globe, all these must be tested against living experience if they are to claim any truth.[41]

The Guggenheim Foundation, unfortunately, did not understand the purpose of Varèse's proposal, and despite repeated attempts Varèse failed to win their support. Similarly, despite a certain degree of interest and willingness to support his Guggenheim applications, Harvey Fletcher was unable to grant Varèse access to the Bell Telephone Laboratories. It is perhaps ironic to note that the latter organisation, two decades later, was to be instrumental in developing research into an important new area of sound generation: computer synthesis.

Despite these setbacks progress was still being made in other quarters. During 1929-30 Paul Hindemith (b.1895, d.1963) and Ernst

[41] Extract from one of thirteen public radio lectures given 1964-66 in Cologne, reproduced in Karl H. Worner, Stockhausen, Life and Work (revised edn., Faber and Faber, 1973), p.139.

Toch (b.1887, d.1964) carried out experiments with recorded sounds on phonographs at the Rundfunk-Versuchsstelle Hochschule für Musik in Berlin. Hindemith was primarily interested in the use of such facilities for testing out his theories of acoustics and the analysis of harmonic structures, outlined in his treatise, The Craft of Musical Composition.[42] A by-product of this period of scientific investigation was a collaboration with the scientist Friedrich Trautwein which led to the invention of the Trautonium[43] and also the composition of Hindemith's Concerto for Solo Trautonium and Orchestra (1931). The demonstration of the Ondes Martenot[44] in Paris, 1928, by Maurice Martenot had attracted further interest in the possibilities of electronic instruments, leading to a succession of compositions from composers such as Jacques Charpentier, Olivier Messiaen, Darius Milhaud, Charles Koechlin and Arthur Honegger.

The invention of these and other similar instruments during the 1930s, although stimulating the use of such devices as additions to the traditional range of orchestral instruments, did not advance significantly the establishment and development of sound generation and transformation systems. The time was still not ripe for the general acceptance of such new processes of musical composition and the all-important provision of the funds necessary for the construction and purchase of suitable equipment.

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[42] Paul Hindemith, Unterweisung im Tonsatz (Associated Music Publishers Inc., 1937), English version, The Craft of Musical Composition, 2 vols., tr. Arthur Mendel and Otto Ortmann (Schott and Co., Ltd., London, 1945).

[43] See note 8

[44] See note 7

Varèse, however, was not alone in his endeavours, and the tide of musical opinion slowly began to change. A prophetic address was given extemporaneously by the conductor Leopold Stokowski to a meeting of the Acoustical Society of America on 2 May 1932, entitled 'New Horizons in Music'.[45] Stokowski, as a keen conductor of contemporary music, devoted much effort to bringing young composers into contact with as wide a public as possible, and he realised fully the importance of establishing, even on a general plane, a continuous dialogue between scientists and artists in what was becoming an increasingly technological society. His address included not only a discussion of the artistic implications of the use of technology as an aid to communication through the mediums of the radio and the phonograph, but also some interesting predictions regarding the active use of electronic generators and treatment devices as compositional tools, accurately pinpointing some of the major problems still encountered in systems today:

> Another vista that is opening out is for the composer, for the creator in music ... Our musical notation is utterly inadequate. It cannot by any means express all the possibilities of sound, not half of them, not a quarter of them, not a tenth of them. We have possibilities in sound which no man knows how to write on paper. If we take an orchestral score and reproduce it, just mechanically perfect, it will sound mechanical. It won't have the human element in it. Also there would be so much that the composer was trying to express, that he conceived but couldn't write down because of the limitations of notation ... One can see coming ahead a time when a musician who is a creator can create

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[45] Leopold Stokowski, 'New Horizons in Music', Journal of the Acoustical Society of America, Vol.4 (Lancaster, Pennsylvania, U.S.A., 1932-1933), pp.11-19.

> directly into TONE, not on paper. This is quite within the realm of possibility. That will come. Any frequency, any duration, any intensity he wants, any combinations of counterpoint, of harmony, of rhythm, - anything can be done by that means and will be done.[46]

Over the four decades which have passed since this address the science of electronics has developed across boundaries which could scarcely have been envisaged at that time, helped to an unprecedented extent by the invention and development of the magnetic tape recorder, the transistor and integrated circuit, and the digital computer, the latter offering facilities for the ordered processing of logical instructions. Notwithstanding this technical revolution which has opened up powerful facilities for establishing and developing man/machine communications the aspirations of musicians such as Stokowski and Varèse have not yet been fully realised, and it is pertinent to observe that some of the major artistic problems which still face studio designers today were clearly recognised during this inter-war period of gestation.

Stokowski's predictions were based at least in part on a knowledge of some interesting technical developments which were taking place at that time. Hindemith's experiments with phonograph records had caught the attention of several artists in the Bauhaus movement, including László Moholy-Nagy (b.1895, d.1940), Oskar Fischinger and Paul Arma (b.1905) who became absorbed with the possibility of being able to draw sound wave patterns and then realise them acoustically, and they accordingly carried out their own investigations

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[46] Leopold Stokowski, 'New Horizons in Music', <u>op.cit.</u>, pp.14-15.

during the period 1930-1932. Using records as sound sources they developed techniques for altering their acoustical content by running them backwards against a stylus and scratching new patterns. They soon turned their attention towards the more interesting possibilities of manipulating optical sound tracks which had been developed for use with talking movies.

The recording process for this system involves the transfer of sound information onto film in the form of patterns of varying densities, which may subsequently be detected and reproduced acoustically via a photocell detector. Physical alterations to the shaded contours thus affect the sound reproduction. Research in this area was pioneered by the German inventor Rudolf Pfenninger who discovered, in 1932, that analysis of the shapes on an optical sound track elicited sufficient information for the synthesis of a wide range of musical timbres in terms of hand-drawn patterns.

This work was important, for despite its practical limitations it resulted in the first really flexible means for direct communication between the composer and a sound generation system. Investigations continued in Leningrad, Russia where Vevgeny Sholpo (d.1951) developed four versions of his <u>Variophone</u> for graphically encoding sound information,[47] and in Ottawa, Canada where Norman McLaren (b.1914) completed a series of films employing 'drawn' sound tracks.[48]

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[47] These machines acted as models for the ANS (photoelectric sound synthesiser) developed at the Moscow Experimental Studio, later expanded into the Scriabin Museum Laboratory, 1961.

[48] See also: John Whitney, 'Moving Pictures and Electronic Music', <u>Die Reihe</u>, Vol.7 (Universal Edition; English version: Theodore Presser Co., Pennsylvania, 1965), pp.61-71.

By 1940 a sophisticated sound analysis laboratory had been developed at King's College, Newcastle-upon-Tyne which employed an optical recording system. This involved the production of a line trace of a sample of the source wave characteristics recorded via the light beam of a cathode ray oscilloscope, energised by a ribbon microphone and amplifier to vibrate in a plane at right angles to a moving film. Short lengths of this film containing two or three complete wave traces of the sound were then wrapped around a drum and rotated in front of a photocell analyser to produce an acoustic spectrogram. In a paper to the Musical Association,[49] Dr. E. G. Richardson outlined the principles of this system and discussed how the results of such analyses might be used for the synthetic generation of sound.

> When the intensity of a musical instrument is raised or when we are dealing with complex systems like a bell or the human voice the components are no longer all harmonic to the fundamental ... By re-combining the components in their true magnitude and frequency we can, of course, remake the original sound of the source, or we may experiment with other combinations to initiate new tone qualities ... I should point out that I have spoken up to the present as though the quality of a note on a musical instrument remained unchanged as long as it is elicited, but this is not really true of any instrument, particularly, not of the strings and brass. We ought to include in our study the 'starting and stopping' noises of orchestral instruments and imitate them in duration and characteristics if we desire a true copy of their respective musical functions.[50]

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[49] E. G. Richardson, 'Electrical Tone Production and Analysis', Proceedings of the Musical Association, Vol.66 (London, 1939-40),pp.53-65.

[50] *Ibid.*, p.57

The last observation, concerning the characteristics of the initial transients and decay patterns of naturally generated notes, pinpointed a major obstacle, for it is this feature of musical quality which is lacking to some degree from even the most sophisticated generators of synthetic sound. The nature of these problems will be discussed subsequently within the context of modern electronic and computer music systems.

The use of optical sound recording as a means for storing, transforming and retrieving sound information was soon overtaken by the commercial development of the magnetic tape recorder. Magnetic recording systems had been in existence since 1896 when the Danish scientist Valdemar Poulsen invented his Telegraphone, a machine employing a steel wire which could be permanently magnetised by an electromagnet.

The quality of reproduction, however, was distinctly inferior and the system as a whole decidedly cumbersome. Poulsen made several improvements to the Telegraphone during the early 1900s and launched a series of companies to market the device. These, unfortunately, ran into financial difficulties, and his commerical enterprises had all failed by the end of the first world war. Magnetic recording remained virtually stagnant until a German, Dr. Kurt Stille, began filing patents during the early 1920s. His work led to the development of a synchronised sound system for films using magnetised steel tape. Stille sold the rights of his machine to Louis Blattner who marketed his first commercial machine, the Blattnerphone, in 1929. A model was bought by the British Broadcasting Corporation in 1931 and installed at the Savoy Hill studio. During the early 1930s the firm of Marconi bought the manufacturing rights and began marketing a less cumbersome machine: the Marconi-Stille recorder. Steel tape, however, was still employed as the recording medium and this created many practical disadvantages. Erasure of previously recorded signals was now possible, but the tape was difficult to splice, requiring welded joints. It was also extremely heavy and liable to sheer dangerously when spooled at high speed. A major breakthrough occurred in Germany in 1935 when the firm of A.E.G. produced the Magnetophon, a machine which employed a plastic tape coated with fine ferrous particles. The latter had been developed by Fritz Pfleumer, an engineer from Dresden, and marketed by the firm of I. G. Faben.

This invention marked the demise of the steel tape recorder and the start of a series of technological developments which led, by the end of the second world war, to a compact and versatile recording system, soon to rival the direct disc cutting techniques employed in the inter-war period.[51] The primary advantages of the new medium were the facility to re-use the recording tape, the ease of editing, and the ability to record two or more discrete tracks of recorded information simultaneously on the same piece of tape. Magnetic tape recording rapidly displaced optical sound systems mainly as a result of its superior quality of reproduction. This

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51 A detailed account of the development of magnetic tape recording may be found in: Basil Lane, '75 Years of Magnetic Recording', Wireless World, Vol.81, Nos.1471, 1472, 1473 (London, March, April, May, 1975), pp.102-105, 161-164, 222-225. See also Werner Kaegi, 'Music and Technology in the Europe of 1970', La Revue Musicale, Unesco (Paris, 1971), pp.11-32.

process of change was inevitably self-perpetuating, for engineers were diverted from the task of making improvements to the characteristics of optical sound transfer, and as a result one important recording technique of considerable interest to electronic sound synthesis lost the support of commercial development.

Magnetic tape systems supply no direct means of contact between a composer and the characteristics of recorded sound material, for neither are the wave patterns visible to the naked eye, nor may they be meaningfully altered by any physical action.[52] The information may thus only be identified and altered in content via an electronic/electromechanical system. The merits of the direct communication facilities offered by optical sound systems were widely recognised by the principal advocates of electronic sound generation during the 1930s and 1940s, and it should be noted here that a renewed interest in visual methods for controlling and manipulating sound information has developed over recent years, particularly in connection with the use of computer graphics. This phenomenon has demonstrated that effective solutions are still required for artistic problems of composer/machine interaction recognised during the inter-war period, but to a considerable extent overlooked during the subsequent development of electronic sound generation and transformation systems.

Varèse, despite repeated failure in his attempts to obtain

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[52] Except by splicing to juxtapose complete event sequences or by cutting into the tape along its length which provides some measure of amplitude attenuation control by reducing the track width for playback.

funds for a properly equipped electroacoustic laboratory, continued to lobby support for his cause. His difficulties were not just financial, however, for during the late 1930s he entered a period of deep personal crisis regarding his whole language of composition. Having spent a considerable amount of time conducting experiments with phonograph records during 1936, becoming in the process increasingly dissatisfied with the limitations of his equipment, he spent the next three years attempting a rationalisation of his concepts for a new sound world. As a result of this effort he delivered one of his most important lectures to the University of Southern California during 1939. This included the following observations:

> When you listen to music do you ever stop to realise that you are being subjected to a physical phenomenon? Not until the air between the listener's ear and the instrument has been disturbed does music occur ... In order to anticipate the result, a composer must understand the mechanics of the instruments and must know just as much as possible about acoustics ... We composers are forced to use, in the realisation of our works, instruments that have not changed for two centuries ... Personally, for my conceptions, I need an entirely new medium of expression: a sound producing machine (not a sound re-producing one) ... Whatever I write, whatever my message, it will reach the listener unadulterated by 'interpretation'. It will work something like this: After a composer has set down his score on paper by means of a new graphic, similar in principle to a seismographic or oscillographic notation, he will then, with the collaboration of a sound engineer, transfer the score directly to this electric machine. After that anyone will be able to press a button to release the music exactly as the composer wrote it ... And here are the advantages I anticipate from such a machine. Liberation from the arbitrary, paralysing tempered system; the possibility of obtaining any number of cycles or if still desired subdivisions of the octave, consequently the formation of any desired scale; unsuspected range in low and high registers, new harmonic splendors obtainable

> from the use of sub-harmonic combinations now impossible, new dynamics far beyond the present human power orchestra; a sense of sound projection in space by means of the emission of sound in any part or in many parts of the hall as may be required by the score.[53]

His understanding of the possibilities of sound processing systems based on visual specification techniques at that time is illustrated further by an article written one year later, entitled Organised Sound for the Sound Film:

> Being master of the greatest ranges of sensations and emotions from the most physical reactions to the most abstract conceptions, organised sound may be called on to intervene at the point where the spoken word has reached the limit of its efficacy, and where the precision of the image only tends to limit the flight of the imagination ... We are now in possession of scientific means not merely of realistic reproduction of sounds but of production of entirely new combinations of sound ... Any possible sound we can imagine can be produced with perfect control of its quality, intensity and pitch, opening up entirely new auditory perspectives - And these sounds must not be speculated upon as separate entities for sporadic, atmospheric effects but taken as thematic material and organised into a score standing on its own merit.[54]

Such ambitious predictions inevitably contained elements of speculation regarding the precise technical means by which the desired results could be achieved. Composers then, as is still the case today, faced major problems of specification, particularly in equating the subjectively based sound world of the creative musician

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[53] Quoted in Fernand Ouellette, Edgard Varèse, op.cit., pp.146-7

[54] Edgard Varèse, 'Organised Sound for the Sound Film', The Common Weal (22 April, 1940), quoted in Fernand Ouellette, Edgard Varèse, ibid., pp.150-2

to the current states of technology. By the end of the 1930s scientific investigation nevertheless had produced the basic theories necessary for establishing the feasibility of sound generation and manipulation systems as part of the functional process of musical composition, and the imaginative musician was thus in a position to make definite predictions regarding the outline specifications to be explored and realised by design engineers. The degree of perception shown in the writings of Varèse at this time is remarkable for its appreciation of the need to resolve the problems of composer/machine communication before any system, no matter how sophisticated its functions, may be creatively exploited to its fullest extent.

The writings of both Stokowski and Varèse on the potential applications of electronics in music at that time were endorsed by John Cage, a composer who in most other respects subscribed to a totally different school of aesthetics. Speaking to a meeting of a Seattle Arts Society in 1937 he postulated:

> I believe that the use of noise ... to make noise ... will continue and increase until we reach a music produced through the aid of electrical instruments ... which will make available for musical purposes any and all sounds that can be heard. Photoelectric film and mechanical mediums for the synthetic production of music ... will be explored. Whereas, in the past, the point of disagreement has been between dissonance and consonance, it will be, in the immediate future between noise and so-called musical sounds.
> Wherever we are, what we hear is mostly noise ... We want to capture and control these sounds, to use them not as sound effects but as musical instruments. Every film studio has a library of 'sound effects' recorded on film. With a film phonograph it is now possible to control the amplitude and frequency of any of these sounds and to give it rhythms within or beyond the reach of the imagination ... Many inventions of electrical musical instruments have attempted to imitate eighteenth and nineteenth century instruments

> just as early automobile designers copied the carriage ... When Theremin provided an instrument with genuinely new possibilities Thereministes did their utmost to make the instrument sound like some old instrument, giving it sickeningly sweet vibrato, and performing upon it, with difficulty, masterpieces of the past ... The special function of electrical instruments will be to provide complete control of the overtone structures of tones (as opposed to noises) and to make these tones available in any frequency, amplitude and duration ... The composer (organiser of sound) will be faced not only with the entire field of sound but also with the entire field of time. The 'frame' or fraction of a second, following established film technique will probably be the basic unit in the measurement of time. No rhythm will be beyond the composer's earth.[55]

These commentaries proved to be more than mere conjecture, for they contained remarkably accurate details of many features which were to become of primary importance in the new post war world of electronic and computer music systems. Collectively they established artistic principles which were well in advance of their practical realities, and it may thus be seen that the subsequent birth of such systems took place against a background in which many of the problems regarding the relationship of such techniques to the language of music had been clearly defined, if not satisfactorily resolved. On 9 December 1939 John Cage performed his Imaginary Landscape No.1 in Seattle, employing a muted piano, cymbal and two variable speed turntables playing R.C.A. Victor test recordings of fixed and variable frequencies. In 1942 he produced Imaginary Landscape No.2 for percussion quintet and amplified coil of wire, and Imaginary Landscape No.3 for percussion, tin cans, muted gongs, audio frequency

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55 John Cage, 'The Future of Music: Credo', 1937, reproduced in his own collection: Silence (Calder and Boyars, London, 1968), pp.3-6.

oscillators, variable speed turntables, frequency test recordings, buzzer, amplified coil of wire, and marimba, amplified by a contact microphone. The birth of live electronics had thus clearly taken place before the end of the second world war, and the stage was set for the first properly equipped sound processing studios, and their associated schools of composition.

CHAPTER TWO

The Development of Electronic Music Systems, 1948-1964

The development of the first electronic music systems during the late 1940s and early 1950s centred primarily around the work of two separate groups of pioneers in Europe, one furthering the philosophy of musique concrète in Paris, and the other that of elektronische Musik in Cologne.

The importance of these two centres in shaping the subsequent paths of electronic music demands a detailed study of their individual activities. To present a balanced perspective, however, account must be taken of secondary lines of development in America, involving initially music for tape and subsequently tape music. The latter movement grew slowly from tentative beginnings to become a highly significant force during the late 1950s, acting as a catylist for a variety of developments, involving both analogue and digital technology.

Reference has already been made in the previous chapter[1] to the distinct polemical differences which arose in the early 1950s between the schools of musique concrète and elektronische Musik. The reasons why these movements pursued such different courses at the outset are directly associated with the very different compositional outlooks of the pioneering groups of composers concerned. Despite some degree of consolidation and cross-fertilisation of ideas during their more mature periods of development, the diversification of applications in the late 1950s, which resulted from the interest of ever increasing numbers of composers and system designers, served to expand the range

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[1] See page 20.

of activities to such an extent that it is difficult to make general observations regarding the musical characteristics of this phenomenon.

Many facets of system designs, however, have not progressed along such diverse paths, retarded to a considerable degree by major problems of composer/machine communication. During the thirty years since the end of the second world war major advances have been made in the fields of science and technology. Such a pace of development has no direct parallel in the evolution of any creative medium, and there is no historical precedent for the progressive integration of technology and music. Consequently, many of the operational techniques to which composers have become accustomed over the years tend to reflect the technology of the 1950s rather than the technology of the day, for there has been a general reluctance by both designers and composers to explore new approaches. The main features associated with both electronic, and more recently, computer sound processing systems may thus be clearly traced back to the activities of the early pioneering groups, and it is important to identify the considerable influences which their work has played on the subsequent course of studio design and development.

1 Musique Concrète

Pierre Schaeffer (b. 14 August 1910, Nancy), the founder of the school of *musique concrète*, served his apprenticeship as an electronic engineer, entering the service of the *Radiodiffusion-*

Télévision Française as a trainee sound technician after a period of study at the Paris Polytechnic. His technical skills led to rapid promotion within the R.T.F., and in 1942 at the age of only thirty-two he persuaded the company to initiate research into the science of musical acoustics with himself as director. This led to the founding of the Studio d'Essai, expanded in 1946 into the Club d'Essai. In 1948 he turned his attention towards the isolation and analysis of sound events using the processes of sound recording, and began to conceive the idea of using material prepared in this manner as the basis for a form of musical composition.

Over the next four years Schaeffer developed the primary phases of what he came to describe as his art of musique concrète, recording details of his day-to-day observations in diaries which were to provide an invaluable source of reference for his subsequent writings on the subject. In 1951 the R.T.F. recognised the potential of musique concrète as a unique area of research and development, and agreed to provide him with a special studio for further investigations. His first publication of significance on the subject, Introduction à la Musique Concrète,[2] appeared in 1950, describing the course of events during the first year of research. The text suffers unfortunately from several chronological errors which were corrected in a far more substantial work, A la Recherche d'une Musique Concrète,[3] 1952, which covered his progress over the whole four year

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[2] Pierre Schaeffer, 'Introduction à la Musique Concrète', Polyphonie, Revue d'Esthetic Musicale, 6 (Paris Editions Richard Masse, 1950)

[3] Pierre Schaeffer, A la Recherche d'une Musique Concrète (Editions du Seuil, 1952)

period. This latter account is divided into four sections: 'Premier Journal de la Musique Concrète, 1948-49', 'Deuxième Journal de la Musique Concrète, 1950-51', 'L'Expérience Concrète en Musique, 1952', concluding with his important 'Esquisse d'un Solfège Concrèt' which presented a concise set of definitions regarding the language and syntax of musique concrète.

The musical value of the compositions produced during this period is rather variable, and these pieces are best considered as being the results of a period of experimental research into a new field of musical communication. It must nevertheless be observed that the depth of thought applied by Schaeffer and his colleagues towards the formulation of a 'solfège' for their musique concrète was not convincingly matched by the musical results at this time. This unfortunate shortcoming served to fuel the criticisms of opponents from the school of elektronische Musik who were, perhaps understandably, unwilling to investigate further and evaluate the merits of their rivals' underlying philosophy.

Schaeffer's investigations commenced with his own examination of the natural sound world of the Futurists, experimenting with the possibilities offered by the use of montage and elementary disc transformation techniques. Although his discoveries during this early period would appear to have much in common with the work of his predecessors, the potential application of recording technology opened up a far more flexible and creative world of sound. 'In fact the Italians in the shape of Marinetti have been pioneers [over] twenty years previously. But he was concerned with concerts of directly produced sounds, leading, as one can see, to an impasse.'[4]

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[4] Pierre Schaeffer, A la Recherche, op.cit., footnote 1, page 31

Most of the first four months of 1948 were spent in preliminary investigations into the effects of striking percussion instruments, such as bells, in different ways, leading to the important discovery that any single musical event is characterised not only by the timbre of the main body of the sound but also by the characteristics of both the initial attack and the subsequent decay. On 21 April he carried out experiments recording bell tones onto disc, where, by fitting a manually operated attenuator between the microphone and disc cutter, he was able to remove the natural attack of each note and substitute an artificial attack pattern, radically altering the character of each event. Two days later he speculated whether an instrument could be constructed which might reproduce to order the sounds of an orchestral instrument by means of a bank of previously recorded pitch events. This idea was singularly prophetic, for today digitally encoded note information is used to provide the sound patterns for so-called 'computer' organs which are able to imitate realistically the tonal range of any selected conventional pipe organ.

Schaeffer's dedication to the use of only naturally generated sound sources led him quickly to attack the work of the German electronic instrument designers, well before the Cologne studio for _elektronische Musik_ had been founded.[5]

> I am mistrustful of the new instruments; ondes and ondolines, of what the Germans pompously call 'elektronische Musik'. I hold the same attitudes towards all electric music ... We are interpreters ...

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5 Pierre Schaeffer, _A la Recherche_, _ibid._, p.15

> I seek direct contact with sound material without the interference of electrons.

Having made an initial study of the attack, steady state and decay characteristics of isolated sound events and also the effects of playing recordings forwards and backwards, Schaeffer turned his attention towards the possibilities of composing pieces from such material. His first work, <u>Etude aux Chemins de Fer</u>, included the sounds of six locomotives whistling at the depot for the Gare des Batignolles, Paris, trains accelerating, and the sounds of wagons passing over joints in the rails. Such material involved not only single events but also multiple events with their own rhythmic characteristics, and he quickly encountered the problems of association which arise from the use of sound sources which retain a significant proportion of their identifying characteristics subsequent to treatment.

In an attempt to overcome this problem Schaeffer reverted to the more normally accepted range of musical sound sources and investigated the results of playing recordings at different speeds, noting that such alterations affected not only the pitch and time scale of events but also the timbre. This led him to make an important observation regarding the nature of variable speed processing.[6]

> I conclude that musique concrète differs ... from classical music in one important aspect. For classical music a <u>do</u> is a <u>do</u> no matter what its placing in the tessiture. For musique concrète a sound, generally 'complex', is inseparable from its situation in the

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[6] Pierre Schaeffer, <u>A la Recherche</u>, <u>ibid.</u>, p.22

> sound spectrum. Nothing is superimposable, divisible, transposable.

A further examination of the relationships between pitch, rhythmic, timbre and envelope characteristics led to a series of studies during the early summer of 1948, including Le Diapason Concerto (Etude pour Piano et Orchestre), Etude aux Tourniquets, Etude au Piano 1, dite Etude Violette, Etude au Piano II, dite Etude Noire, Etude Pathétique, dite Etude aux Casseroles.

The Etude pour Piano et Orchestre endeavoured to combine the sounds of an amateur orchestra tuning up with a spontaneous piano improvisation, played by Jean-Jacques Grunewald. The result was largely unsatisfactory, for there was no coherent dialogue between the two areas of sound material, creating the impression that two apparently unconnected pieces had been roughly mixed together. Tape montage was to become a major feature in many applications of sound processing, and this early discovery of the existence of problems concerning the integration of dissimilar sound material was an important one.

Schaeffer had considered the possibility of a piano à bruits from a very early stage in his investigations and gave some thought to the potential of the prepared piano,[7] unaware at this time of similar investigations being carried out in America by John Cage.[8]

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[7] 'There is no instrument to play musique concrète.' Pierre Schaeffer, A la Recherche, ibid., p.22

[8] 'The words 'prepared piano' have been employed in a systematic manner by the American John Cage, whose work I did not know at this time. The use of the piano is similar in both cases, but differs in that the music of Cage remains fairly abstract in both conception and execution.' Ibid., footnote, pp.26-27

He concluded, however, that live performance on a modified piano constituted little more than a mere extension of the functional characteristics of the normal instrument, and turned his attention towards the possibility of manipulating naturally produced sonorities on tape. Both Etude au Piano I, dite Etude Violette and Etude au Piano II, dite Etude Noire, were based on a series of textures prepared by Pierre Boulez, characterising different musical styles, for instance, classical, romantic, impressionistic, atonal. These were manipulated to produce two pieces, each endeavouring to preserve some degree of continuity between elements of the same texture whilst creating an overall structure concerned with the juxtaposition of stylistically contrasted events. L'Etude Violette is more angular in construction, exploring the use of widely separated pitch areas of relatively short duration, whilst L'Etude Noire is more concerned with the structuring of melodic phrases.

This first phase of study concluded with an experiment with speech, using the voice of the actor Sacha Guitry, montaged with recordings of a Balinese priest, a barge moving upriver, and an American accordion, to produce Etude Pathétique, dite Etude aux Casseroles. The significant characteristics of Etude aux Chemins de Fer which had initiated this first period of musique concrète have been succinctly summarised by Fritz Wieland and Gottfried Koenig in their introductory notes for the electronic music course at the Institute of Sonology, Utrecht.[9]

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[9] Gottfried Koenig and Fritz Wieland, Summary Introduction (Institute of Sonology, Utrecht), p.26

1 The act of composition was accomplished by a technological process.

2 The work could be replayed innumerable times in precisely the same manner.

3 The replaying was not dependent upon a human performer.

4 The basic elements were 'concrète', thereby offering the listener a mode of audition quite different from that of perceiving 'abstract' music.

These characteristics were not to remain entirely consistent, for Schaeffer became increasingly conscious of the performance problems of a piece which was entirely pre-recorded, and played without human interpretation to an audience via a set of loudspeakers.

The first presentation of musique concrète to the general public took the form of a broadcast recital of all these early studies with the exception of Le Diapason Concerto (Etude pour Piano et Orchestre) in a programme entitled Concert à Bruits , transmitted by the R.T.F. on 5 October 1948. A spirited controversy ensued, inspiring Schaeffer to commence work on his article 'Introduction à la Musique Concrète', intended to provide an explanation of the motivations behind his work.[10] A pause in his investigations followed, brought about as the result of commitments to attend conferences abroad on behalf of the R.T.F. during the winter and spring of 1948-49.

So far, Schaeffer had worked almost entirely on his own, helped only occasionally by musical friends such as Grunewald and Boulez, and the sound engineer Jacques Poullin. During the summer of 1949 the composer Pierre Henry joined him as a regular co-researcher, and

[10] See pages 36-7 and note 2.

Poullin was permitted by the R.T.F. to become involved with studio engineering tasks on a permanent basis. Schaeffer began to reappraise the role of natural instruments as sources of sound, carrying out investigations which retraced much of the ground covered by Varèse, twenty years previously. His next work, *Suite pour Quatorze Instruments* is significant for it provided the starting point for his preliminary definitions of a syntax for *musique concrète*. His main preoccupation at this time concerned the parallels which could be drawn between the processes of conventional instrumental composition; *Musique Habituelle* and the processes of *musique concrète*; *Musique Novelle*, identified as follows:[11]

	MUSIQUE HABITUELLE (dite abstraite)		MUSIQUE NOUVELLE (dite concrète)
PHASE I	Conception (mentale)	PHASE III	Composition (matérielle)
PHASE II	Expression (chiffrée)	PHASE II	Esquisses (expérimentation)
PHASE III	Execution (instrumentale) (de l'abstrait au concret)	PHASE I	Matériaux (fabrication) (du concret à l'abstrait)

Previous to his *Suite pour Quatorze Instruments* his compositional approach had been from a *musique concrète* to a *musique abstraite.* Now as he became increasingly concerned with the live aspect of musical performance, he began to develop the reverse approach, achieved via three stages:[12] i) A point of departure; the natural

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[11] Pierre Schaeffer, 'Introduction', *op.cit.*, p.86. Also Pierre Schaeffer, *A la Recherche*, *op.cit.*, p.35.

[12] Pierre Schaeffer, *A la Recherche*, *ibid.*, p.37

sounds of musical instruments, ii) A process of transition involving the application of <u>concrète</u> procedures to the recorded sources, leading to iii) A point of arrival - a new music. This change in philosophy may not appear to have been dramatically significant, but it raised an important artistic distinction regarding the processes of composition with technology which is relevant today. On the one hand, a composer may choose to start by developing conceptually a clear picture of the sonological structures he wishes to achieve. Such a picture would then have to be rationalised and probably modified in terms of the technical facilities available, leading to a set of operating instructions which could then be executed. On the other hand, a composer may wish to start with a selection of potential sound sources, offering a range of characteristics with which he may experiment, building up from the results of such investigations the elements for a complete composition. The former approach is associated with a system of sound generation which is capable of offering the composer a sophisticated specification language, capable of interpreting a wide range of suitably expressed musical ideas as an equivalent set of studio procedures. The latter approach involves a less complicated dialogue between the composer and the system working in terms of the specifications offered by the generators and processors themselves. These may thus be treated as an 'orchestra' with defined functional characteristics.

The merits and practicalities of designing a modern computer-aided system offering one or a combination of these approaches are matters of the utmost importance and will be examined in due course. In the present context, however, the relatively simple nature of the technology involved focuses attention clearly on the operational techniques applied in fulfilment of the musical specifications.

The *Suite pour Quatorze Instruments* is divided into five movements, each highlighting a different aspect of *musique concrète* as follows:

1	Prologue	Original instrumental recordings left virtually intact except for the judicious addition of occasional reverberation, echo and overlay, permitting the introduction of additional simple rhythmic counterpoint.
2	Courante	A monody by way of a contrast assembled from extracts drawn from the entire range of musical instruments and treated to simple juxtaposition and montage.
3	Rigardon	An intensely rhythmic jovial movement, highlighted by interjections from trumpets and drums. (Considerable use is made of recordings played in reverse, creating symmetrical envelope structures with their sources.)
4	Gavotte	Material derived from a single short phrase to construct a set of variations, each concerned with the interpretation of the phrase by a group of three instruments, the recorded extracts being subjected to transposition by playing the recordings at different speeds.
5	Sphorade	In contrast to the four preceding movements, a freely expansive quasi-improvisatory piece constructed without the constraints of any set technical procedures.

The musical quality of this work is not entirely satisfactory. The Gavotte, for example, by Schaeffer's own admission,[13] failed by virtue of its reliance throughout on a single musical phrase which,

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[13] Pierre Schaeffer, *A la Recherche*, *ibid.*, p.63

despite its many interpretations and subsequent transpositions, retained recognisably many of its original characteristics, thus generating a repetitive and highly monotonous structure. These difficulties provoked him to carry out closer analyses of the nature of sound sources, leading to a preliminary definition of what he called his objet sonore: a fundamental sound event, isolated from its original context and examined for its innate characteristics, free from any association with other sounds or groups of sounds. Schaeffer asserted[14] that the abstraction of such events from a series of source sound complexes to provide elements for the regeneration of musical material involved processes compatible with the principles of post-Webernian serialism.

As a basis for his concept of an objet sonore, Schaeffer tried to establish why his transformation procedures failed to remove or alter many of the distinctive characteristics of source events. He concluded that techniques such as speed and pitch transposition, simple montage or the playing of recordings in reverse did not produce anything essentially new. The use of musical instruments, musical habits and musical structures had conditioned the way in which he had carried out his processes of analysis and synthesis, and it thus seemed appropriate to return to his starting point, the world of noises, as a more basic source of sound information. Such a move, however, did not remove the problems of association, as he had already discovered in preparing Etude aux Chemins de Fer, and it proved necessary not only to examine the nature of sound events themselves in more detail but also to

[14] Pierre Schaeffer, A la Recherche, ibid., p.42

develop new technological procedures which expanded the range of available transformations.

Taking sections of sound material of varying lengths as sources, Schaeffer began to examine the extracts on both a macro level, considering the structure as a whole; and also on a micro level, considering the inner ordering of events. Each source sound event displayed a natural macrostructure assembled from the superimposition of a variety of fundamental characteristics. By recording such events onto disc, it became possible to divide the progression of events into micro time intervals each containing a selected portion of the whole event, for example the attack, steady state or decay of a simple envelope, or the initial, middle or final stages of an essentially dynamic sound; for example a glissando or a progressively changing timbre. Alterations could then be made to the macrostructure of the whole by juxtaposing or removing some of these constituent elements.

Such exercises, however, did not offer a complete solution to the problems of association, for the selected micro elements might be of sufficient duration for their own internal characteristics to provide a macro event, retaining distinctive features when subjected to juxtaposition or transformation. At the other extreme, the temporal division of sound events into increasingly smaller quantities leads eventually to a loss of any meaningful response by the human ear and brain as to the nature of the extract. Time division also does not constitute a thorough analysis of the fundamental characteristics of a sound event, for the technique provides no information regarding the frequency spectrum associated with each element. If Schaeffer had carried his isolation principles a stage further and devised techniques for identifying the individual harmonic components of each sample, he would have established a link between the structures of naturally generated sounds and those of electronic origin.

Schaeffer decided that his work had reached a stage where he could embark on a major piece of musique concrète, and in collaboration with Pierre Henry commenced work on a composition entitled Symphonie pour un Homme Seul, 1949-50.[15] During the early stages of formulation, Schaeffer had considerable difficulty in selecting a suitable source of sound material. Two lines of development were uppermost in his mind at the time: i) the extension of the possibilities of instrumental sources by means of new technical aids, and ii) the development of his principles of objets sonores and their rules of composition.

His quest for an area of sound material which would prove sufficiently rich to sustain a major composition led him to select a source which in many aspects offers connections with both instrumental material and noises; the sounds of a man. His initial idea was to select sound material from noises produced naturally just by a man alone; for example breathing, walking or the sounds of a heart beating, but this selection became extended to include sounds drawn from the man's communication with the world outside. The information was accordingly divided into two groups reflecting internal and external attributes:[16]

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15 The definitive version was not realised until 1966.

16 Pierre Schaeffer, A la Recherche, op.cit., p.64

Intérieur à l'Homme	Extérieur à l'Homme
Elements de souffle	Pas ou analogues
Fragments de voix	Frappements de porte
Cris	Percussions
Voix fredonnées	Piano préparé
Airs sifflés	Instruments d'orchestre

The inclusion of a prepared piano as an external sound source contradicted his earlier views regarding the work of John Cage,[17] and this element of ambivalence shows that Schaeffer was still far from achieving a thorough consolidation of his ideas at this time. In this case the influence of Henry served to widen his outlook on techniques of composition, resulting in a less dogmatic approach towards the use of technology as part of a creative musical process.

The work is divided into eleven movements: Prosopopée 1, Partita, Valse, Prosopopée II, Collectif, Erotica, Scherzo, Cadence, Eroïca, Apostrophe, Strette. Some of these, as their titles suggest, are modelled loosely on classical structures. The rhythmic pattern of the spoken word or phrase acts as the central theme, highlighted by the use of repeated loops and the juxtaposition of extracts with complementary fragments of instrumental and percussive patterns. The mood is light and humorous, sharply contrasting with the seriousness and tight structures of many of the early works associated with elektronische Musik.

During the winter of 1949-50, Schaeffer and Henry turned their attention towards staging the first public concert of musique concrète,

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[17] Schaeffer now demonstrated his regard for the work of John Cage by using the musical intervals C A G E as a motif.

which was held in the hall of L'Ecole Normale de Musique, Paris on 8 March. Schaeffer was at last able to investigate how the characteristics of a concert auditorium might best be exploited, and with Poullin commenced work designing and building a complete live performance system incorporating several sets of turntables, loudspeakers and mixing units. The concert, unfortunately, was not entirely successful, for the performance routines involved in mixing and projecting the sounds around the auditorium were under-rehearsed, and the complexities of creating live montages from unwieldy turntables proved overwhelming at times. The audience were thus more aware of the antics of the performers than the music, which was not the intention.

The concert, nevertheless, was favourably received by many of those who attended and was followed by further public recitals on a more modest scale in the Club d'Essai, where the equipment of Schaeffer's studio could be employed in situ. In July the critic Roger Richard reported that 'The concerts already given at the Club d'Essai, and also last March at L'Ecole Normale de Musique have proved that a public not especially prepared or warned to be on their guard readily accepts the impact of this extraordinary music. ... Musique concrète is ready to leave the laboratory. It is time musicians exploited it. When musicians and musicologists such as Roland Manuel, Olivier Messiaen and Serge Moreaux express interest in it we can trust in this departure.'[18]

After a short period of absence, Schaeffer returned to the Studio in the Autumn of 1950 to find his colleague, Henry, working on two of his own compositions, Concerto des Ambiguities and a Suite. Henry

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[18] Roger Richard, review in Combat (Le Journal de Paris), 19 July 1950

had encountered considerable difficulties in establishing an acceptable method of notation for his realisation score, and Schaeffer became preoccupied with the problems of creating a specification syntax, using these two works as models. The characteristic source phrases in the *Concerto* had been notated entirely in normal musical notation whilst the material for the *Suite* consisted of a series of graphic drawings. The structure of the *Concerto*, however, rapidly rendered the use of conventional notation unsatisfactory, for the principal sound source was a prepared piano, producing acoustic results which differed significantly from the original score.

After much thought he concluded that it was necessary to assemble a *solfège* for his *objets sonores* which would classify sounds in terms of hierarchies of tessiture, timbre, rhythm and density. He discovered that if the sounds were at all homogenous it became difficult to distinguish between different events within the hierarchy. Schaeffer decided to adopt a provisional system of scoring closely modelled on the classical system. Using conventional five line staves for each sound element, a page of the score would be divided into four areas as follows:[19]

Concrète Score

1 Living elements, voice, etc.

2 Noises

3 Prepared instruments

4 Normal instruments

[19] Pierre Schaeffer, *A la Recherche*, *op.cit.*, pp.83-86

Schaeffer gives an example of the score layout for Henry's <u>Musique sans Titre</u>, a piece written shortly after his Suite:

Group (1)	<u>Voix</u>	VI)	Voix (humaine ou animale) incidentale
		VP)	Voix (élément vocal) périodique
		VC)	Voix (élément vocal) continue
Group (2)	<u>Bruits</u>	BI)	Bruit incidental
		BP)	Bruit périodique
		BC)	Bruit continu
Group (3)	<u>Instruments Préparés</u>	PP)	Piano préparé
Group (4)	<u>Instruments Ordinaires</u>	O)	Instruments d'orchestre

A linear time scale is employed, drawn along the bottom of the score in seconds, with a vertical dashed line drawn every five seconds. For source instrumental and vocal parts, clefs and notational symbols are employed as normal, excepting that the durational values of the individual pitches are rationalised to conform to the time markings. For <u>concrète</u> sounds, elements of standard notation are combined with extra graphical symbols to give an approximate indication of the characteristics of events with respect to time. Since the vertical axis represents pitch the result could only show clearly the movement of pitch areas with respect to time. Schaeffer realised that the system could not indicate changes in timbre, and the use of conventional symbols for dynamics was not wholly satisfactory, but this method of representation was a useful intermediate step.

So far only passing reference has been made to the technical facilities employed by <u>musique concrète</u>. One interesting fact which emerges is that the equipment used was for the most part of a relatively simple specification, for Schaeffer's ideals did not permit the use of many of the generation and processing devices that will be described in connection with elektronische Musik. The first stages of development were thus restricted primarily to the multiplication and improvement of devices already generally available, in particular the facilities for mixing, recording and reproducing acoustic information. These changes did not result in any radical expansion of the functional characteristics of the studio system, with one important exception: the introduction of the tape recorder to the studio in 1951 - well after the composition of Symphonie pour un Homme Seul had been completed. All the recordings up until this date had been made directly onto discs, producing considerable practical difficulties for subsequent operations on the encoded material. For example, the isolation of sound events could only be achieved by employing a manually operated volume control or switch to select the required extract whilst re-recording the passage concerned. Further loops of sound could only be created by scratching the record to make a particular groove repeat, where the speed of the record and the circumference of the selected groove would determine the duration of the loop. The introduction of magnetic tape recording revolutionised the techniques of musique concrète, for it became possible for the first time to cut and splice sound extracts together to any desired durational specification with the minimum of difficulty.

The year 1951 proved to be extremely important for another reason, for the R.T.F. officially recognised the significance of his work and commissioned the first studio to be constructed expressly for the realisation of musique concrète. The studio was designed by Poullin and Schaeffer, and featured the following equipment: a set of tape recorders including one three-track model; a Morphophone, two Phonogènes, and a mixing desk, incorporating a Pupitre de Relief Spatial, otherwise known as a Potentiometer d'Espace; a special device for creating illusions of sounds moving in space, which will be described in more detail later.

The Morphophone consisted of a twelve channel magnetic tape loop system with adjustable filters for producing variable iteration or artificial reverberation. The two Phonogènes were specially converted tape recorders capable of playing tape loops at different speeds. One was designed to function over a continuously variable range, whereas the other was controlled by a twelve note keyboard with a two-position transposition switch, providing twenty-four discrete pitch transformations.

It is interesting to note that the introduction of tape facilities to the studio was not received with much enthusiasm initially. An entry in Schaeffer's diary for 30 April 1951 declares, 'The studio is a battlefield: Everybody is fighting.'[20] Henry and even Poullin himself felt that the tape recorder destroyed many of the features characteristic of the old disk recording system. This instinctive resistance to change demonstrates that a long and close association with a particular sound processing system will often lead to a relationship with the equipment such that its very limitations become essential characteristics of the musical principles developed around it. In the instance in question such prejudices were dispelled in time, but it

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[20] Pierre Schaeffer, A la Recherche, op.cit., p.96

is relevant to note from a more general point of view that it is reactions such as these which have encouraged the retention of many features characteristic of studios in their early stages of development without a continuous critical assessment of their usefulness to the needs of the day.

The new studio led to a large expansion of activities, and extra taping equipment was soon added, including for the first time a five-track tape recorder. Schaeffer and his associates adopted the title Groupe de Recherches Musicales, and this organisation was formally adopted by the R.T.F. as part of what is known today as Services de la Recherche de l'O.R.T.F., responsible for all sound and visual experiments required by their radio and television channels. Several young composers came to work at the studio, including Jean Barraqué, Pierre Boulez, Michel Philippot, Hermann Scherchen and, fleetingly, Karlheinz Stockhausen. Visits were also made by Yves Baudrier, Marcel Delannoy, Henri Dutilleux, Jean-Jacques Grunewald, André Jolivet and Olivier Messiaen.[21]

Many new compositional ideas were exploited, particularly in connection with the use of serial procedures employed to define details of duration, pitch, dynamics and timbre, notated in the realisation score. Boulez, for example, employed a precise plan for both duration and pitch in his Etude I sur un Son, 1952, using the Phonogène to give chromatic intervals and octave transpositions of

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[21] Several of the pieces from this period, including Boulez, Etude I; Messiaen, Timbres Durées, and Henry, Antiphone are discussed in Antoine Golea, 'Tendances de la Musique Concrète', Vers une Musique Expérimentale (La Revue Musicale, Paris, 1953), pp.36-49.

a single sound source. Schaeffer and Henry worked intensively on the first opèra concrète: Orphée, first performed in Paris 1951, a revised and considerably expanded version being presented at Donaueschingen in October 1953.[22] Many practical problems arose in the construction of a score, and Schaeffer found his visions of a grand opera greatly tempered. The challenge forced him to develop further his ideas regarding a solfège for musique concrète. He formulated the idea of an orchestre concrète, based on the observation that certain sounds would inevitably continue to display specific characteristics whatever the degree of transformation effected within the perceptive limitations of the human ear. The persistence of these characteristics resulted in these elements being treated as 'pseudo' instruments, notated in the realisation score in much the same manner as conventional instruments. Schaeffer postulated three rules concerning their use in musique concrète:[23]

> 1 An orchestral element (pseudo instrument) is identified by the permanence of one characteristic in all its various transformations.
>
> 2 Musical forms, notated or not, are created by the evolution within a period of time or the superimposition at a given time of pseudo instruments. This evolution may effect all the characteristic elements of a given material (tessiture, dynamics, timbre, note structure, criterea of the same structure), except one at least which remains invariable and as a consequence characterises the identity of the orchestral element to which the form applies.

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[22] Henry produced a new version of the work: Le Voile d'Orphée in 1966.

[23] Pierre Schaeffer, A la Recherche, op.cit., p.95

3 The instrumental techniques or methods of execution, consist of a collection of methods, recognisable or not to the ear in which the causal character must be clearly distinguished. To the ear, the affect alone is important.

Schaeffer also felt it necessary to prepare two different types of score for musique concrète: 1) The operational score concerned with notating the technical procedures involved in realisation within the studio; 2) The effects score concerned with indicating the development of the musical forms in terms of parallel staves, each concerned with an element of the orchestra. An idea of the structure of a work could only clearly be given by the second type of score, which would normally take the form of the provisional score discussed earlier. He was also interested in the possibility of devising a system of exact notation for the parameters involved, but he came to realise that this was a task of immense proportions. Schaeffer was anxious to explore new methods of score notation for use in Orphée, but problems of sound classification and pressure of time prevented him from making any major advances in this area at this time, and his frustration led to a personal crisis involving a period of close self-examination.

In an entry in his diary for 2 May 1951[24] Schaeffer speculated whether he might have embarked on the development of a musique concrète as a reaction to some form of musical repression. From discussions with his colleagues he discovered that they were more concerned with developing musical ideas around a sound system which they knew and trusted, whereas he realised that a useful key to new and better

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[24] Pierre Schaeffer, A la Recherche, ibid., p.103

compositions lay in an unceasing search for new sounds and techniques. This tendency of musicians to accept or reject technological systems uncritically disturbed him and he began to appreciate fully that significant differences existed in outlook between musicians and scientists over the use of technology in musical composition.

During the early summer the scientist André Moles joined the team in a research capacity and this provided a valuable stimulus for Schaeffer's work on a *solfège*, for Moles's interests included the study of perception and psychoacoustics. Moles had already written a thesis entitled *La Structure Physique du Signal en Musique Microphonique*, which provided an important contribution to the study of aesthetics, including some interesting comments on the properties of recorded musical sounds. He observed that the techniques of recording a musical work corresponded in effect to the projection of time into space, governed by the following spatial properties.[25]

1	*Permanence*	The musical work is defined by its registration in terms of all the elements indicated in the conductor's score.
2	*Reproducibility*	Which makes it an object of scientific investigation, whereas all that was possible previously was a comparison of points of view.
3	*Reversability*	A recorded track may be as clearly recognisable in one direction as in the other. This is its most important characteristic since time is irreversable and its passage is

[25] Quoted in Pierre Schaeffer, *A la Recherche*, *ibid.*, p.118

> never simply defined. ... This is not true with space, and the superimposition of space on time allows one to measure the latter objectively.

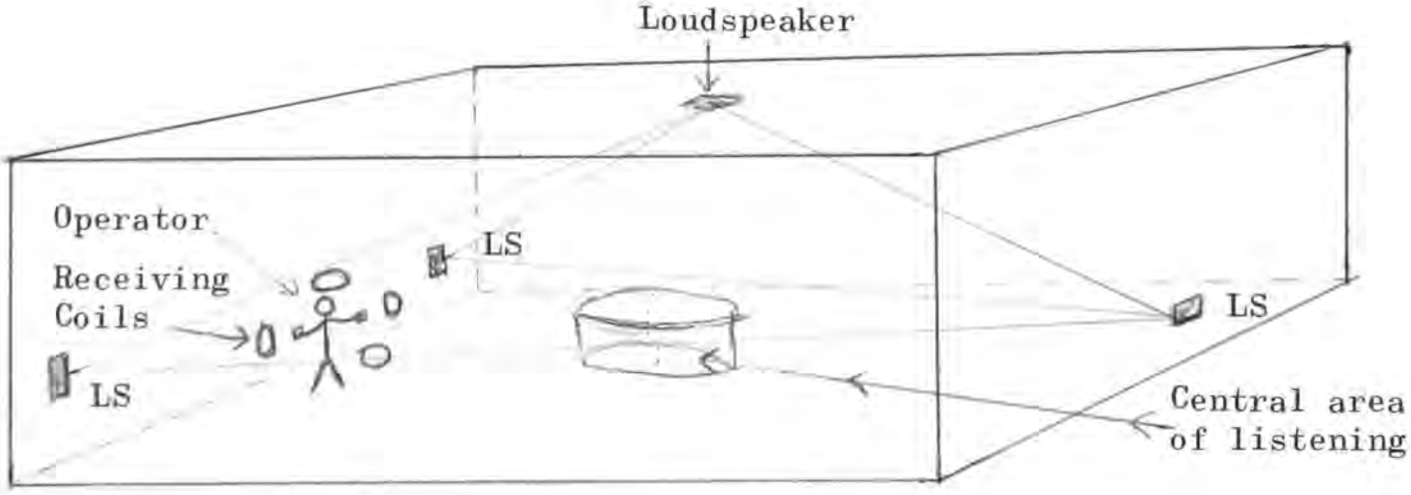
Moles, like Schaeffer, proposed the use of a three-dimensional model for representing individual sound elements using frequency, duration and loudness for the co-ordinates. He also carried out detailed analyses of musical sounds and postulated that up to thirteen million individual characteristics making up the 'atomic' structure of music could be isolated. The problems of communication between the investigator and his sound world became a major preoccupation and he advocated the design and development of machines which could record and display acoustical features in a graphical form.

Poullin's contribution to research as studio engineer should not be overlooked at this point, for ever since the first public concert of musique concrète he had been particularly concerned with the development of techniques which would improve the manner in which sounds could be projected into an auditorium. His preliminary investigations were published in the form of an article entitled 'Son et Espace'[26] and a more detailed account of the device he devised as a sound projection aid, the Potentiometer d'Espace was included in a paper describing the use of recording techniques in the production of material for musique concrète.[27]

[26] Jacques Poullin, 'Son et Espace', Vers une Musique Expérimentale, op.cit., pp.105-114

[27] Jacques Poullin, 'L'Apport des Techniques d'Enregistrement dans la Fabrication de Matières des Formes Musicales Nouvelles. Applications à la Musique Concrète', L'Onde Electrique, Vol.34 (324) (1954), pp.282-291. Tr. D. A. Sinclair, 'The Application of Recording Techniques to the Production of New Musical Materials and Forms. Application to Musique Concrète', National Research Council of Canada, Technical Translation No. TT 646 (Ottawa, 1957)

It is important to realise that in the early 1950s very little was known about the use of multi-channel sound recording techniques. The monophonic long playing record was only just beginning to replace the old 78 and the principle of the stereophonic groove was yet to be perfected in the research laboratory. Poullin's development of a special four-channel reproduction system for the concert hall presentation of <u>musique concrète</u> was thus remarkable for its time, giving composers the opportunity to explore the potential of spatial projection as a musical parameter in their works. The four loudspeaker groups were arranged at the corners of a tetrahedral, two situated at the front of the auditorium, one in the roof and one at the back, focusing sound images into the central area:



The area of listening was ideally to be kept as small as possible for deviations from the central position of focus resulted in distortions in the relative strengths of signals from each loudspeaker, changing the locations of the perceived images. Practical concert hall conditions involved audiences spread over rather a large area, so the effects of mis-positioning were reduced as far as practically possible by using specially designed loudspeaker units which concentrated their radiated energy in a 60° cone. The dimensions of the hall itself and its reverberation time added further complications, requiring adjustments to be made *in situ* to the levels of amplification supplied to each loudspeaker unit.

Composers could thus prepare discretely encoded four-channel tapes using four of the five tracks available on the studio multi-channel recorder. The fifth track could, optionally, be employed to supply a further channel of information, distributed between the four primary output channels via the *Potentiometer d'Espace*. This device consisted of a small hand-held transmitting coil, and four wire receiving loops arranged in a tetrahedral around the performer to represent in miniature the location of the four main loudspeakers in the auditorium. Moving the coil about within the receiving area induced signals of varying strengths in the loops, and this information was used to alter attenuators controlling the superimposition of the signal onto the main loudspeaker lines. The location of the resultant sound information could thus be directly controlled as a live performance feature by a controller, seated in front of the audience.

It was during 1951 that the previously mentioned disagreements between the proponents of *musique concrète* and *elektronische Musik* began in earnest. Schaeffer's *Symphonie pour une Homme Seul* was broadcast on radios Cologne, Hamburg, Baden-Baden and Munich, and was received with considerable hostility by those favouring the new German school. The Summer School at Darmstadt, the *Internationale*

Ferienkurs für Neue Musik, took up the debate that year by organising a symposium on the subject of sound technology and music, which proved to be extremely controversial. The French and the Germans disagreed violently and the Swiss criticised both for describing their work as 'music'.

Schaeffer returned to his studio to spend several months in a further period of consolidation, determined to defend and expand the aesthetic principles in which he believed. His diary at this time reflects the conflicts which arose at Darmstadt.[28] In particular he criticised the concepts of elektronische Musik for providing no obvious key to the solution of the communication problems associated with contemporary music. He also refuted the suggestion that musique concrète had no connection with the language of either Schönberg or Stravinsky, voicing an opinion that it had a middle role to play, polarised by the work of both musicians. In support of this view Schaeffer equated techniques of montage and tape looping with the polytonal and polyrhythmic structures of Stravinsky and suggested that the objet sonore provided a basis for an extension of Schönberg's Klangfarbenmelodie, reaching beyond the concept of a melody of timbres derived from a series of pitches to include more comprehensive structures derived from other acoustical parameters. For example, 'The density of a note complexe permits not only treatment of the timbre but also a correlation between a melody of pitch and a melody of timbre.'[29]

Schaeffer realised that the principal points of disagreement lay

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[28] Pierre Schaeffer, A la Recherche, op.cit., pp.123-141

[29] Ibid., p.141

more in the language of music employed rather than the techniques of studio operation, and accordingly attempted to clarify his principles of sound reorganisation by preparing a syntax for musique concrète to serve as a description for 'the enlargement of sonorities involving the use of an ever expanding register of timbres, where the timbre itself in its development becomes an expressional value which is concerned with the combinations of sounds'.[30]

Esquisse d'un Solfège Concret was finally published in 1952.[31] This treatise is divided into two main sections: the first, entitled Vingt-Cinq Premiers Mots d'un Vocabulaire, establishes a set of twenty-five definitions for the description of complex sounds and the technical procedures involved in their manipulation and transformation whilst the second is concerned with the application of these definitions to create an operational language for the processes of musique concrète. Koenig and Wieland have summarised the objectives of the Solfège as follows:[32]

> Three music-theoretical problems existed for the Groupe de Recherches Musicales:
>
> a) The reciprocal effect between sound as the physical corner [manifestation] of music and the listener's perception, proceeding from the sound object: l'objet sonore.
>
> b) The selection of certain objects which can be used for music because of perceptive criteria:

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30 Pierre Schaeffer, A la Recherche, op.cit., p.131

31 Pierre Schaeffer and A. Moles, Esquisse d'un Solfège Concret, quoted in Pierre Schaeffer, A la Recherche, ibid., pp.203-228

32 Gottfried Koenig and Fritz Wieland, Summary Introduction, op.cit., p.39

this leads to a sound morphology and a music typology.

c) The value of certain objects in the sphere of musical composition, or vice versa the influence on the music by the selection of certain musical objects - objets musicaux.

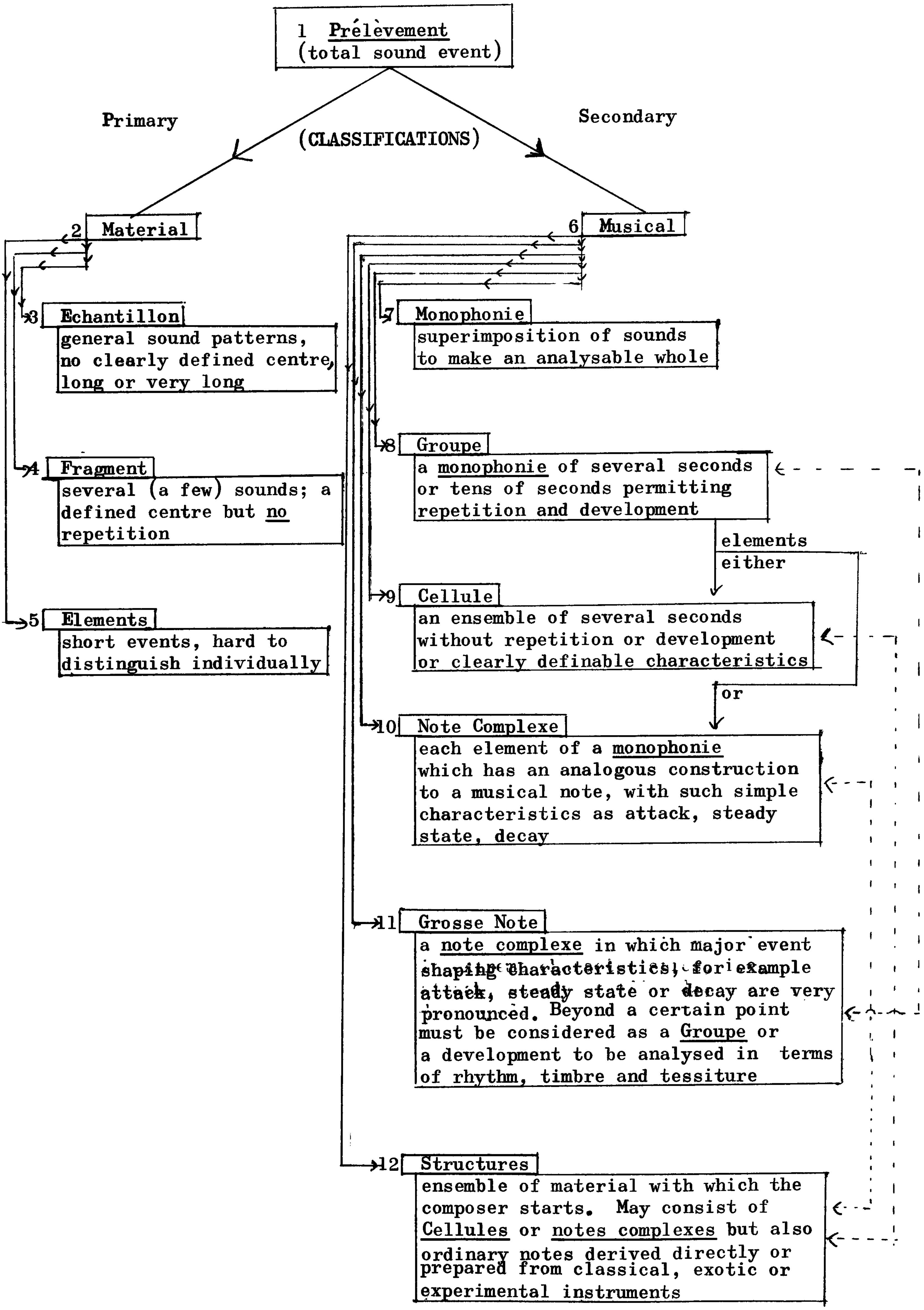
Koenig and Wieland also draw attention to the similarities between Schaeffer's work with musical sounds and the study of linguistics. In the latter, distinctions are made between phonetics (the study of vocal sounds) and phonology (the study of the structure of language systems), and lexicology (word or text explanation) and syntaxis (the study of sentence construction).

In musique concrète it is possible to draw similar parallels between acoustics and acoulogy (solfège), and music theory and the study of composition.

From a practical point of view the solfège was intended to help composers select and manipulate sound material in a more effective manner by providing them with a better understanding of their basic characteristics. The whole classification centred around Schaeffer's basic element, the objet sonore, now after four years' work defined as a sound event which may be extracted from its natural context and studied objectively outside its time continuum.

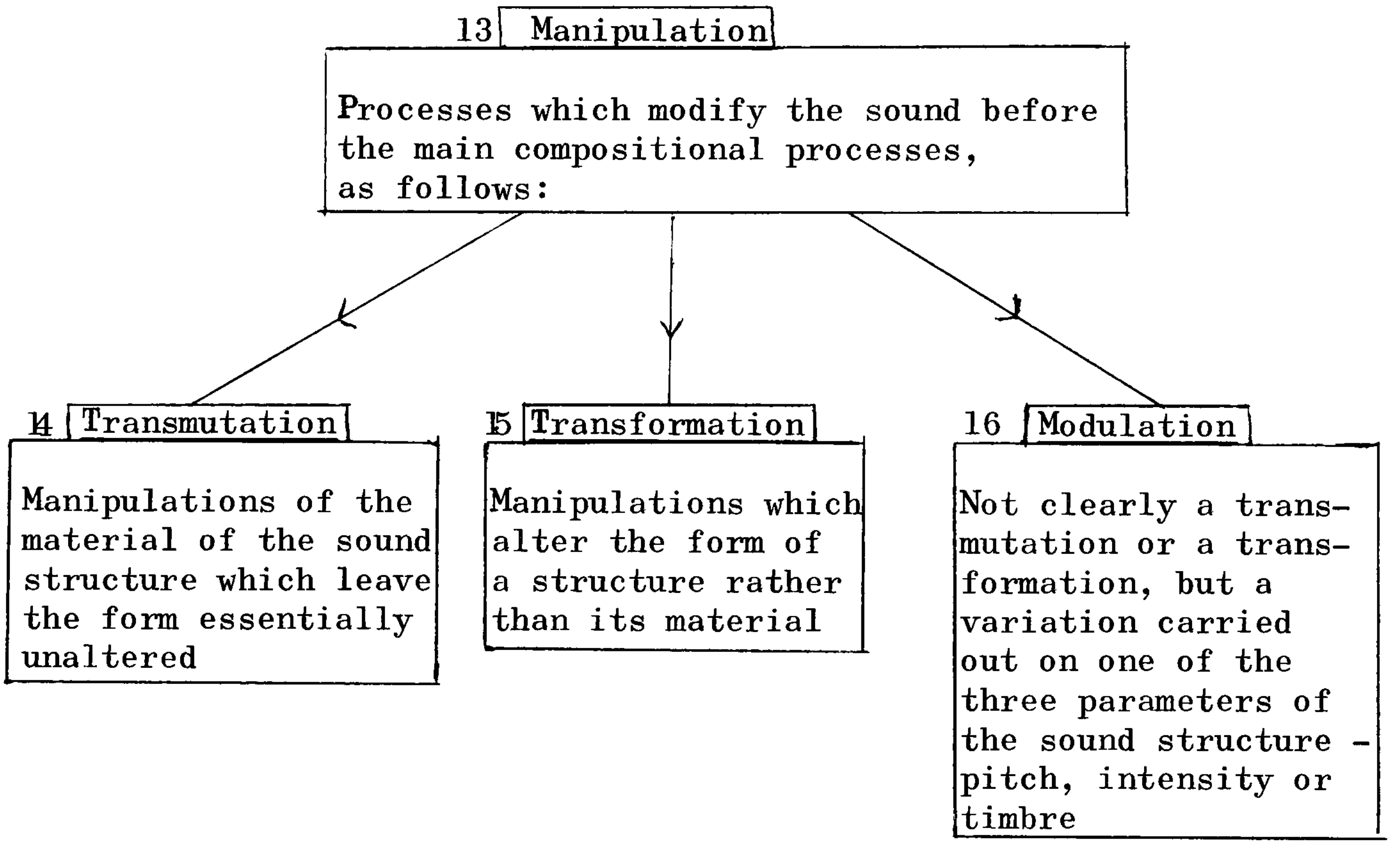
The twenty-five definitions are derived as follows, identified by number: 1) the Prélèvement is identified as the source sound event once isolated on recording tape or disk, ready for analysis. Any Prélèvement may be examined from two viewpoints - i) in terms of its sound material and ii) in terms of its musical significance. The first approach; 2) Classification Matérielle des Objets Sonores is to be considered a necessary prerequisite to any technical or aesthetic analysis, and is based on the temporal length of a sound object, defined around any identifiable centre of interest. Three classifications are made in this context; a) 3) Echantillon, b) 4) Fragment and c) 5) Elements. The second approach; 6) Classification Musicale des Objets Sonores takes into account the need to make value judgements concerning the content of a sound object, in particular whether it tends to be more simple or more complex. Four distinctions are made; a) 7) La Monophonie, b) 8) Groupe, c) 9) Cellule, and d) 10) Note Complexe. These may be further qualified by the use of two additional terms; 11) Grosse Note and 12) Structures. The organisation and definition of these twelve elements may best be illustrated in the form of a diagram:

Objet Sonore



The operations involved in processing the <u>objets sonores</u> are divided into three stages:

1 Initial processing of the sound



13 **Manipulation**

Processes which modify the sound before the main compositional processes, as follows:

14 **Transmutation**

Manipulations of the material of the sound structure which leave the form essentially unaltered

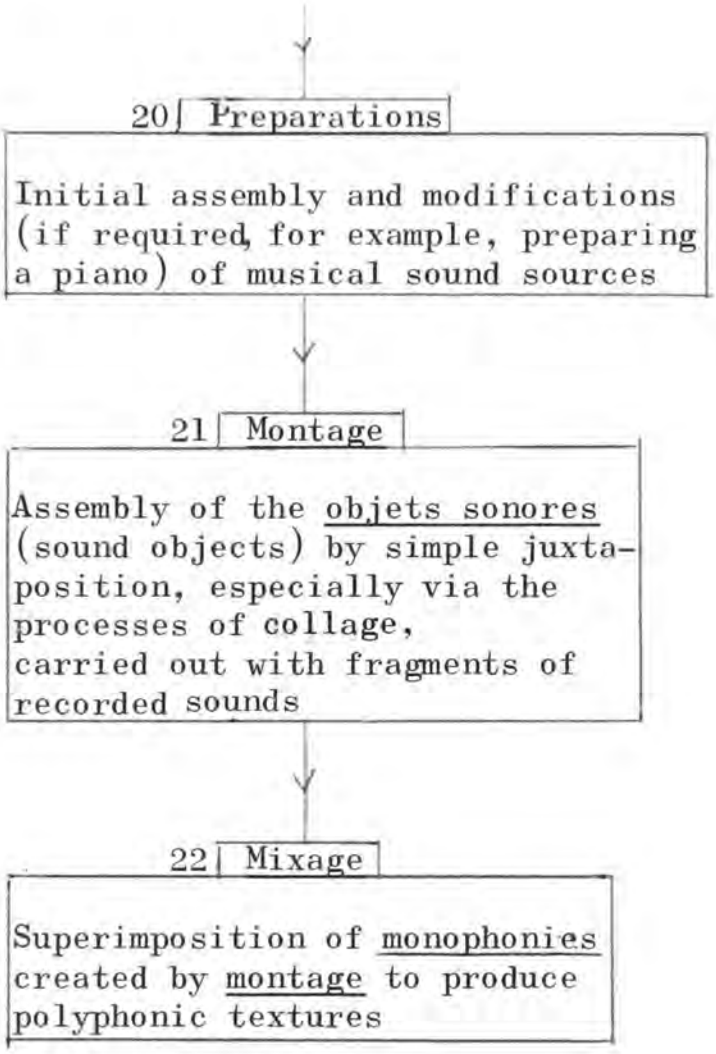
15 **Transformation**

Manipulations which alter the form of a structure rather than its material

16 **Modulation**

Not clearly a transmutation or a transformation, but a variation carried out on one of the three parameters of the sound structure - pitch, intensity or timbre

2 Processes involved in the main compositional procedures in the following stages



20 Preparations

Initial assembly and modifications (if required, for example, preparing a piano) of musical sound sources

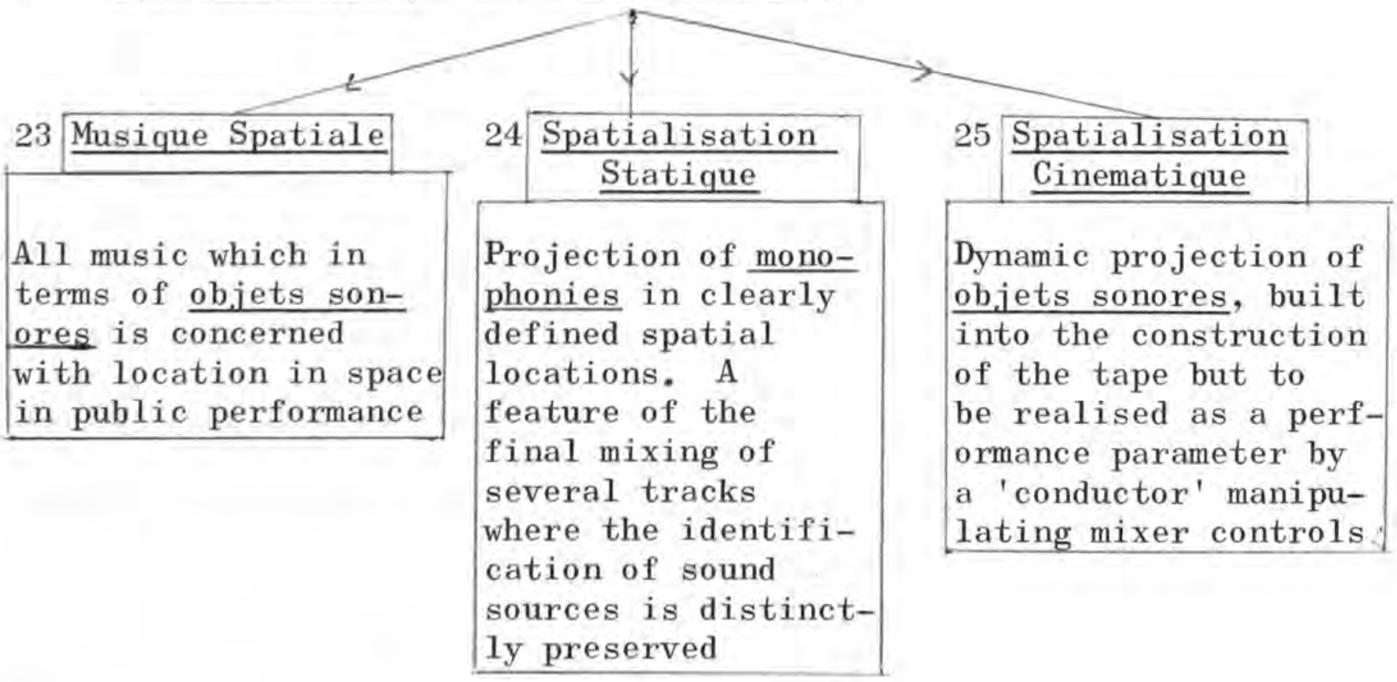
21 Montage

Assembly of the *objets sonores* (sound objects) by simple juxtaposition, especially via the processes of collage, carried out with fragments of recorded sounds

22 Mixage

Superimposition of *monophonies* created by *montage* to produce polyphonic textures

3 Definition of spatial characteristics



23 Musique Spatiale

All music which in terms of *objets sonores* is concerned with location in space in public performance

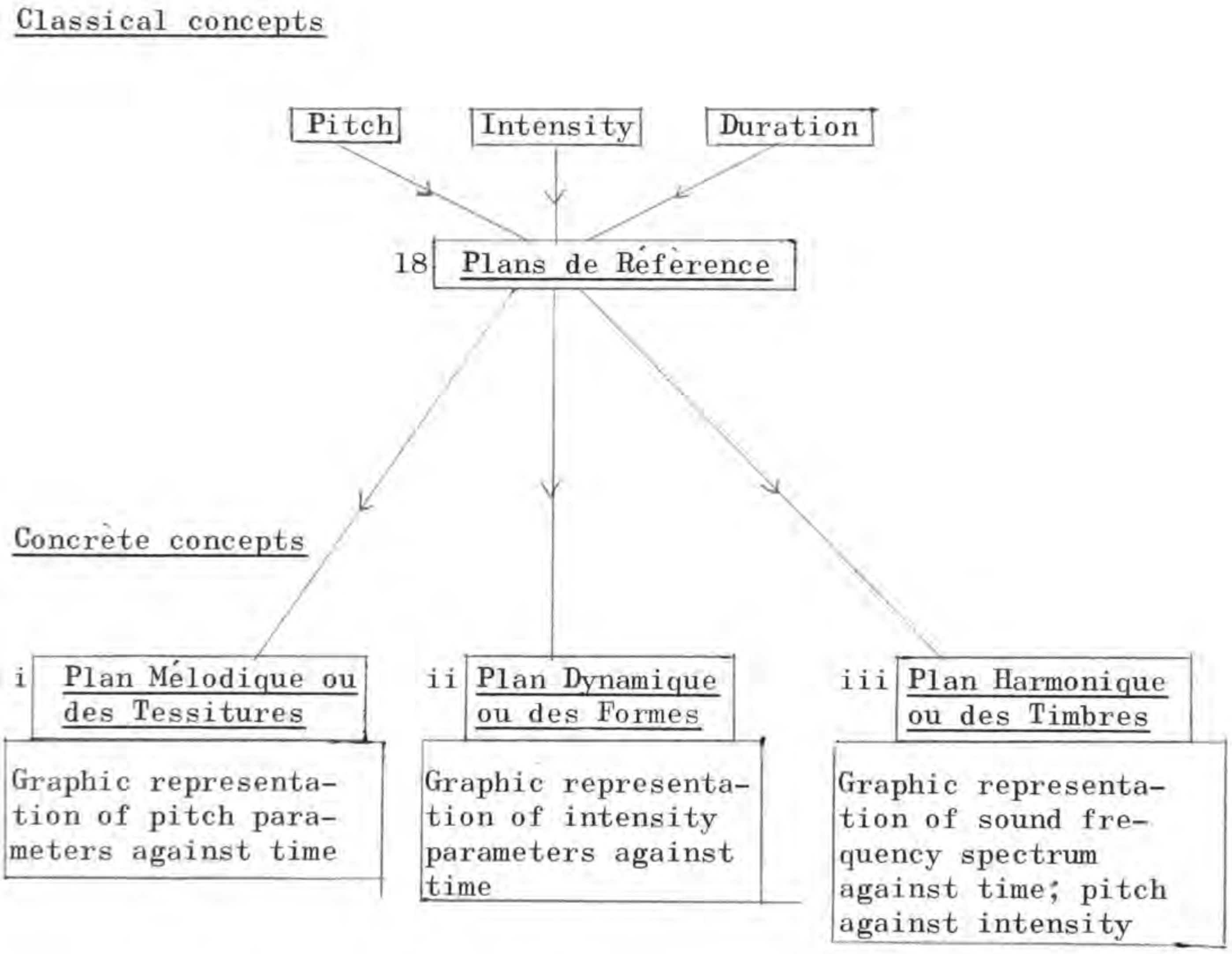
24 Spatialisation Statique

Projection of *monophonies* in clearly defined spatial locations. A feature of the final mixing of several tracks where the identification of sound sources is distinctly preserved

25 Spatialisation Cinematique

Dynamic projection of *objets sonores*, built into the construction of the tape but to be realised as a performance parameter by a 'conductor' manipulating mixer controls.

Finally three *Plans de Référence* are defined for the analysis of *objets sonores* in terms of the fundamental parameters of pitch, intensity and duration, combined to form a single three-dimensional plan.



The second section of this Treatise is concerned primarily with an explanation of the function of the three *Plans de Référence*. After a short description of the basic principles of sound wave generation and transmission, and the loudness response of the human ear, Schaeffer establishes his criteria for examining the characteristics of sounds in this manner, starting with the observation that in nature nothing exists which is entirely pure. In the case of sound, it is said to be musical in the classical sense if a 'fundamental' predominates sufficiently for it to be identified as a note belonging within the normal instrumental pitch range. Such a sound is far more complex than most musicians imagine, being composed not only of a collection of harmonics (which are not always stable in their composition but varying as a function of the note) but also of an equally important 'noise' element. The piano, violin, or the human voice produces notes which in common with the majority of musical sounds contain elements of noise, that is to say complex 'transitory phenomena'. Musicians have no need to define these for they are entirely implicit in the description 'violin', 'piano' or 'voice', and are inherent in the production and application of the instrumental sound.

Duration is in itself an abstract parameter for in reality the sound of a note does not correspond exactly to its notated representation, for the fluctuations in intensity associated with its growth and decay create an important pattern which gives the sound a form. Similarly the classical musician is not directly aware of this feature, because he is not able to modify and use it as a means of expression or execution.[33]

The musical features which implicitly determine the nature of a

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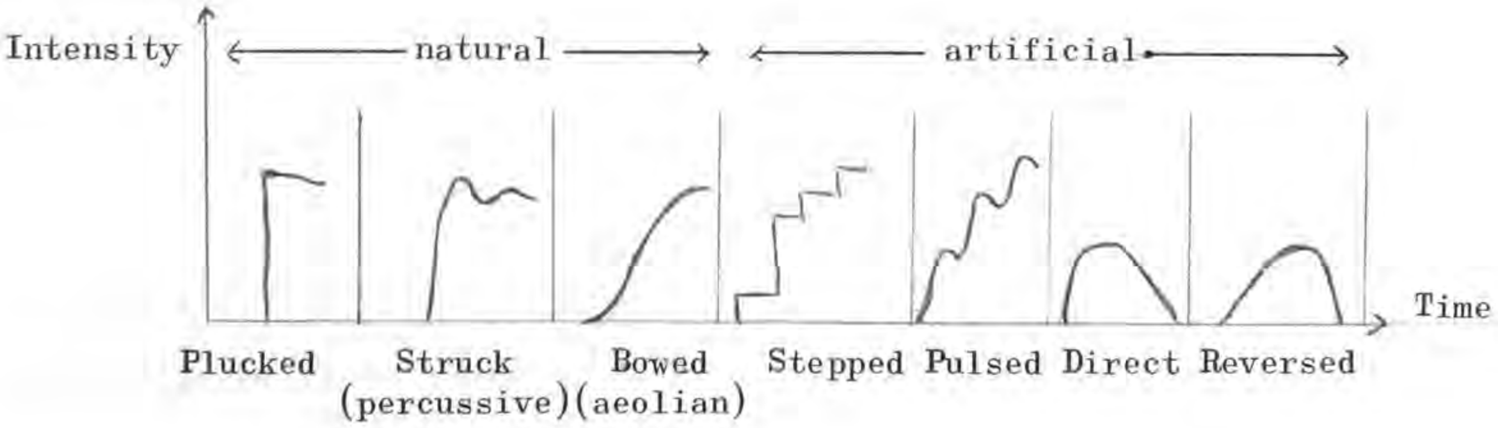
[33] This is not strictly true. Brass players in particular are able to exercise a considerable degree of control over the shape of both their attack and decay characteristics. See the reference to Varèse's experiments with simulated reverse envelope shapes, Chapter 1, page 11.

musical note may be reduced into three dimensions concerned with the features of duration, intensity and pitch. If, however, one wishes to be not only more thorough but also closer to musical reality, one must substitute the idea of *Plans de Réfèrence* which describe the evolution of the note itself by relating the development of each parameter as a function of each of the others.

The *Plan Dynamique* is concerned with a graphical representation of the intensity/time characteristic of a musical sound complex, divided into three parts: i) the *attack* of a note, ii) the *body* of the note, during which time the intensity remains more or less steady, iii) the *decay* of a note.

i) The *attack* of a note, when played on traditional instruments, obeys the natural laws of acoustics, and may be identified as being one of three distinct types: a) *plucked*, b) *percussive* (for example, the piano) or c) *aeolian* (a progressive rise in intensity from zero to a maximum, such as may be achieved, for example, with a bowed violin or a reed wind instrument,) With *musique concrète*, however, far more complex types of attack are possible, as illustrated in the following diagram:



Schaeffer draws attention to the importance of the attack in any musical sound, and the significant alteration to the character of a note which results from the substitution of an artificial characteristic. Particular reference is made to the use of a decay characteristic in reverse to provide long attack times.

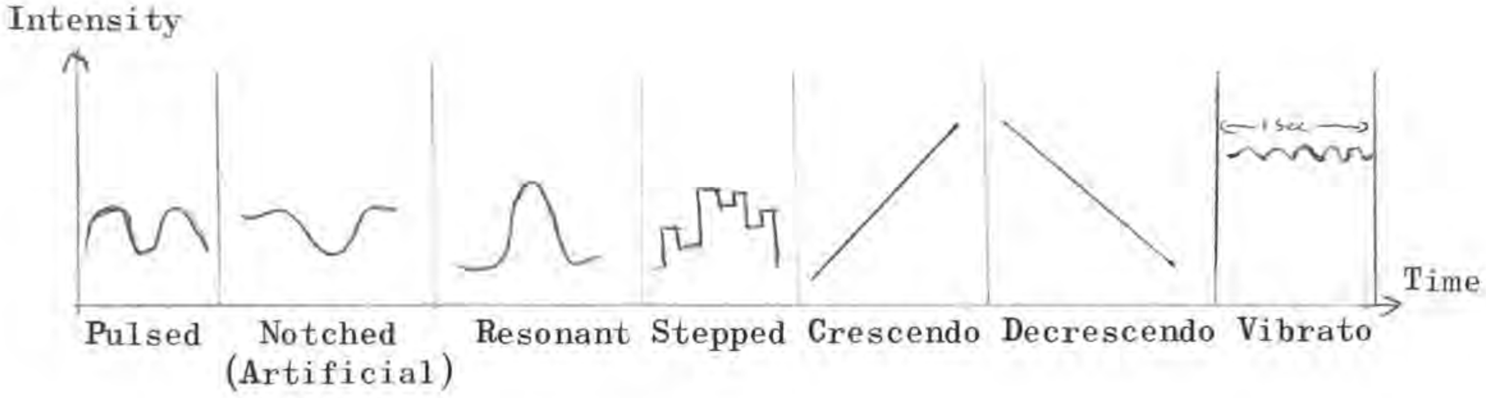
ii) The <u>body</u> of a note. As in the previous case the rules of physics determine and limit the characteristics of the body of notes produced by traditional instruments. A completely steady note of constant amplitude and indefinite duration may only be obtained from an electronic laboratory oscillator. All natural instruments, including the human voice, can produce only pseudo steady state sound levels for it is impossible to eradicate completely all minor fluctuations. Frequently the executant deliberately adds an element of amplitude vibrato, generally involving a change in the order of ten to fifteen per cent at a periodic rate of five to eight vibrations per second. These minor alterations to the body of a note are identified by the listener as being an intrinsic characteristic of the note. Finally the character of a note is determined <u>a priori</u> by the dynamic level with which it is played, in accordance with the instructions of the composer. The accepted musical signals:

ppp pp p mf f ff fff

according to Stokowski (as quoted by Schaeffer) correspond to the following levels of loudness:

+20 +40 +50 +60 +75 +85 +95 decibels.

The use of musique concrète techniques greatly extends the range of possibilities: for example, the dynamic level may be reduced below that of both the end of the attack and the beginning of the decay, or an artificial resonance point may be created in the body of the note. Additional characteristics which may be encountered or created include stepped dynamic levels, a general crescendo or decrescendo or an induced vibrato.[34]



iii) The decay of a note. This feature is characterised more by its duration than by its shape, and it is a matter of some delicacy to determine precisely when a sound has disappeared completely. In the case of traditional instruments such as the violin, piano, voice, etc., it is convenient to define the time of extinction as the moment when the note ceases to be generated; that is to say, no more energy of vibration is being produced. These instruments lose their energy progressively, following an exponential curve in accordance with the

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[34] It should be noted that Schaeffer does not state clearly what is the maximum degree of deviation from a basic dynamic level to be considered acceptable within his definition of a steady state sound.

general laws of acoustics. This decay time may vary in length from a few tenths of a second to several seconds, influenced to a large extent by the reverberation factor induced by the response of the listening room. A distinction is made between continuous reverberation which simply prolongs the decay time curve, and vibratory reverberation which superimposes characteristics after a short delay as a vibrato or a pulsation on the decaying sound.[35]

In conclusion, it is observed that the theoretical analysis of natural sound events into these three parts is a very artificial affair, for it is often not possible to distinguish accurately between the sections. For example, all the notes of a piano, when played sharply reach their maximum intensity almost instantaneously and do not display a body since they begin to decay immediately, whereas a violin note is a source of great richness, offering an aeolian type of attack which may vary in characteristic (for example, staccato, legato), a central region employing a clearly defined vibrato, and a very quick decay, heavily susceptible to modification by the response of the listening room. With musique concrète, on the other hand, it is possible to isolate each of these parts, labelled with a definite characteristic, and also to reassemble them in a variety of combinations, giving rise to an almost indefinite number of pseudo-instruments.

The Plan Harmonique is concerned with a graphical representation

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35 Although not specifically mentioned, the use of tape head echo, where during a recording sequence, sounds just recorded are detected by the playback head and fed back into the recording chain, was a feature commonly used in musique concrète to produce a pulsating decay. Electronic or electromechanical reverberation and echo devices were not employed at this time, for in common with filters, ring modulators and other such treatments, such techniques of sound processing were alien to Schaeffer's principles.

of the sinusoidal components of a complex sound, known generally by acousticians as the harmonic spectrum. Amplitude as a function of frequency depicts timbre, a feature normally associated by musicians with instruments which generate stable (that is, clearly pitched) sounds, where all the individual notes obtainable from the source display common characteristics, creating a general instrumental timbre. Strictly speaking, this notion of timbre cannot be maintained for the duration of a note complexe, for the strength and frequency of each harmonic component change independently with respect to time. In practical terms the human ear is normally able to record the instantaneous nature of a note if it persists for more than $\frac{1}{20}$th of a second, during which time sufficient of the spectrum has been established for the timbre to be considered representative.

One of the more consistent characteristics of musical instruments is their use of simple materials (strings, membranes, columns of air) whose vibrating properties obey a numerical ordering according to a principle known as the harmonic law. This states that the frequencies of the many components of a note of an instrument are in simple multiples of each other with a lowest common factor which is defined as the fundamental. Also the spectrum produced from an instrument is formed from a definite number of harmonics, rarely more than twenty,[36] in which the amplitude generally decreases with ascending overtones.

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[36] This is rather an oversimplification, for in many instrumental sounds a considerably larger number of harmonic components may be detected. There is a widely accepted psychological theory which postulates that it is the ordering of these higher partials, known as formants, which accounts for the differences in quality between one instrument and another, particularly with regard to the violin family. Fourier's theories regarding the analysis and synthesis of sounds in terms of individual sinusoids are discussed later in this chapter. See footnote 62 and accompanying text, page 101.

With *musique concrète*, free from the limits imposed by traditional pitched instruments, one is able to consider the structure of all sounds, both natural and artificial, as having a musical character. An analysis of timbre may thus include elements which are far more complex and not necessarily related harmonically. In some cases these elements may lose their discrete values and form a continuous band spectrum rather than a clearly ordered line spectrum.

Schaeffer classifies the timbres of sounds into two separate groups, depending on whether the analysis is quantitative or qualitative:

Quantitative

i) *poor* - a sound whose spectrum contains only one or a very small number of components of significant amplitude

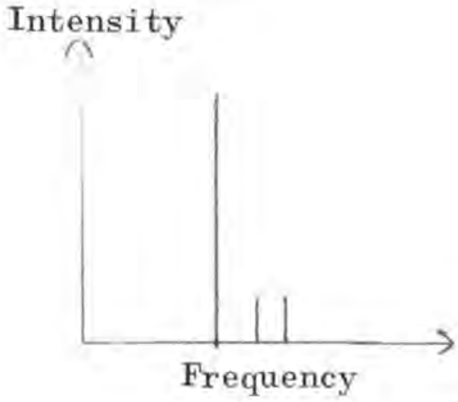
ii) *rich* - a sound containing numerous overtones of significant amplitude.

Qualitative

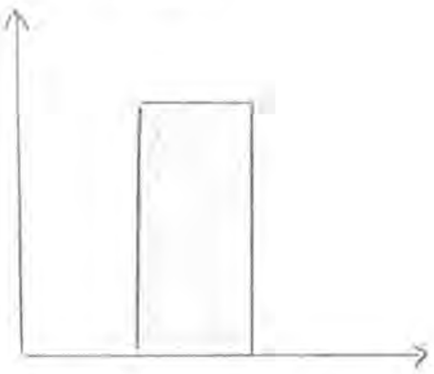
i) *brilliant* - sounds consisting of a large number of overtones in which the amplitudes do not decrease rapidly in proportion to frequency

ii) *clear* - sounds of the same family, but consisting of only a very limited number of overtones

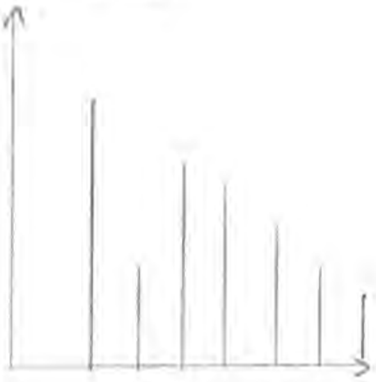
iii) *dull* - sounds with only a few overtones, whose amplitudes decrease rapidly in proportion to their frequency.



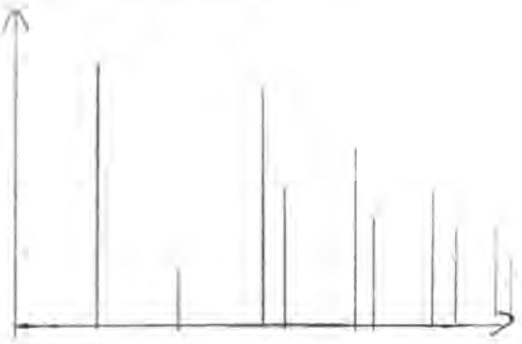
Poor



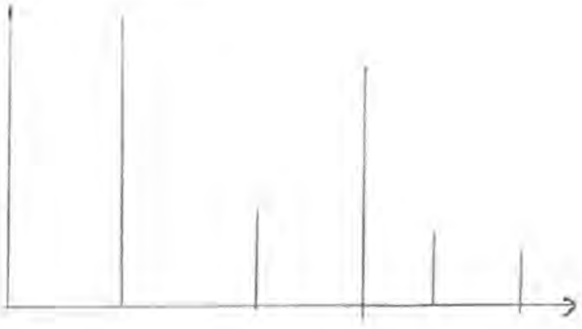
Thick



Rich



Brilliant



Clear

The Plan Mélodique (or Plan des Tessitures). In the strictest sense it is impossible to describe simply the evolution of pitch as a function of time; that is to say, the development of the whole harmonic spectrum of an event. The problem, however, may be simplified by an important principle employed in the psychology of perception. This is concerned with the instantaneous quality of a sound at a selected instant, for one may consider that all the acoustic phenomena occurring just before this instant will be perceived by a listener as if they have occurred simultaneously. Sufficient information is contained in an extract of the order of $\frac{1}{30}$th to $\frac{1}{20}$th of a second long, and an analysis of the events may be represented on a graph of intensity plotted against frequency.

In practical terms the general laws of information theory in acoustics permit the detection of two very different types of events.

1) Very short events are associated generally with periods of attack or sharp changes to the form of a *note complexe* in the *Plan Dynamique*. During these events the spectrum table is exceedingly complex, containing a great number of spectral elements and creating what may be seen as a continuous spectrum - white noise. These elements do not display any simple numerical relationships with each other. They follow scarcely or not at all the rules of selection put forward by way of a plan of timbres. They develop at random, in complete disorder during these short events. One relates thus this transitory occurrence to the property of *noise* or disorder, and this creates a fundamental characteristic from the viewpoint of the aesthetic of perception.

2) In other cases, corresponding to the greater part of the duration of a *note complexe* and separated clearly from the preceding events, the spectrum table is much more simple. With a reduced number of events involved, the elements evolve slowly during the time span, retaining a sort of memory and approximating to a permanence which gives the ear time to appreciate fully their structure. One is thus able to define an average tessiture corresponding to a perceptibly constant spectrum.

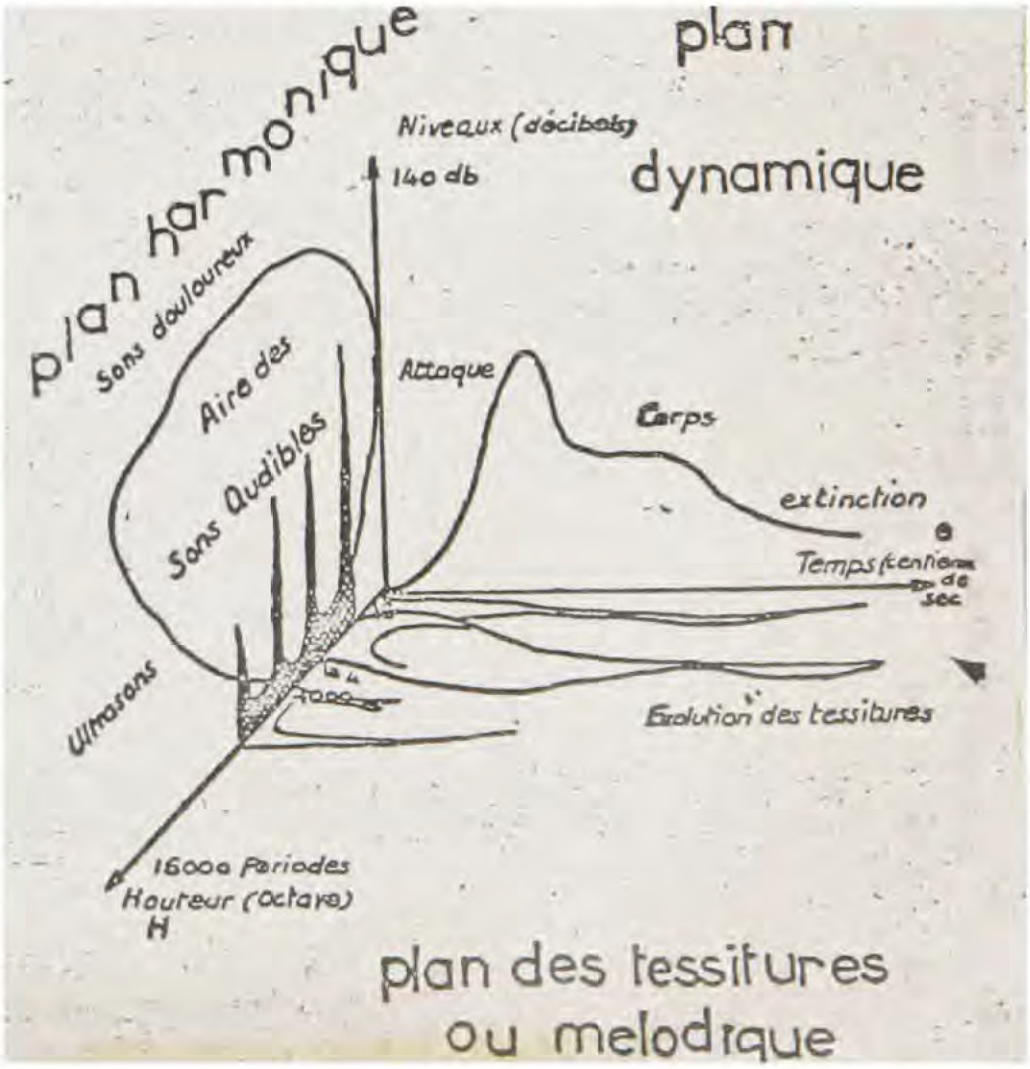
Pitch characteristics may be distinguished as follows:

a) *stable* or *unstable*	according to whether the lowest pitch is constant or not during the time span
b) *rising* or *descending*	according to the progression of the lowest zone of the spectrum during the time span
c) *extended* or *restricted*	according to the size of the musical interval (if it is discernible) within which the pitches evolve.

On a more detailed scale, distinctions may be made between

a) tessitures vibrées	in which the nominal pitch undergoes periodic fluctuations, generally at a speed of 5 to 6 times a second, with a deviation of between 1 and 5 per cent, which corresponds, moreover, to the effects of vibrato already considered in the Plan Dynamique
b) tessitures filées	in which the pitch of the sound complex fluctuates very quickly within a very restrained margin during the course of a note, especially at the beginning or the end. (This may in fact be identified in traditional instruments such as the ukelele and the balalaika.)
c) tessitures scintillantes	in which the rapid sequence of perceptible pitches, considerably disordered, prevents an easy location of pitch
d) tessitures indistinctes	white noise.

The three Plans de Référence are combined to create a three dimensional model, using the parameters of intensity, time and pitch. For example:[37]



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[37] Pierre Schaeffer, A la Recherche, op.cit., p.207

Schaeffer consolidates these analyses by defining a set of *Critères de Caractérologie Sonore*, identifying the principal characteristics which are uniquely associated with each *Plan de Référence*:

In the *Plan Dynamique*

3	Criteria of attack	*plucked* *struck* *aeolian*
6	Criteria of sustaining	no sustaining at all; *shock* sustaining by the means of *resonance*, *natural* or *artificial* sustaining in a manner similar to the attack: sustaining by repetition of attack; *pulsation* *artificial* sustaining by means of montage
3	Criteria characterising the variation in intensity of the note	*stable* (constant intensity) *cycle* *continuous* variation (crescendo or decrescendo)
5	Criteria for the decay of a note	*no* reverberation (*cut dead*) *normal* reverberation (reverberant) *artificial* reverberation: *continuous*, *discontinuous* or *cyclic* .

In the *Plan Harmonique*

4	Criteria for the density (or purity) of a sound	*pure* (a single fundamental) *thin* *thick* *white*
2	Criteria for the importance of the timbre	*poor* *rich*
3	Criteria for the colour of the timbre	*brilliant* *clear* *dull*

In the Plan Mélodique (concerned as in the Plan Dynamique with variations in the body of the note)

5 Criteria

stable tessiture (fixed pitch)
cyclic (vibrato)
continuous (climbing or descending)
discontinuous (flickering)

The graphical representations of these criteria are related to the three-dimensional model as follows:[38]

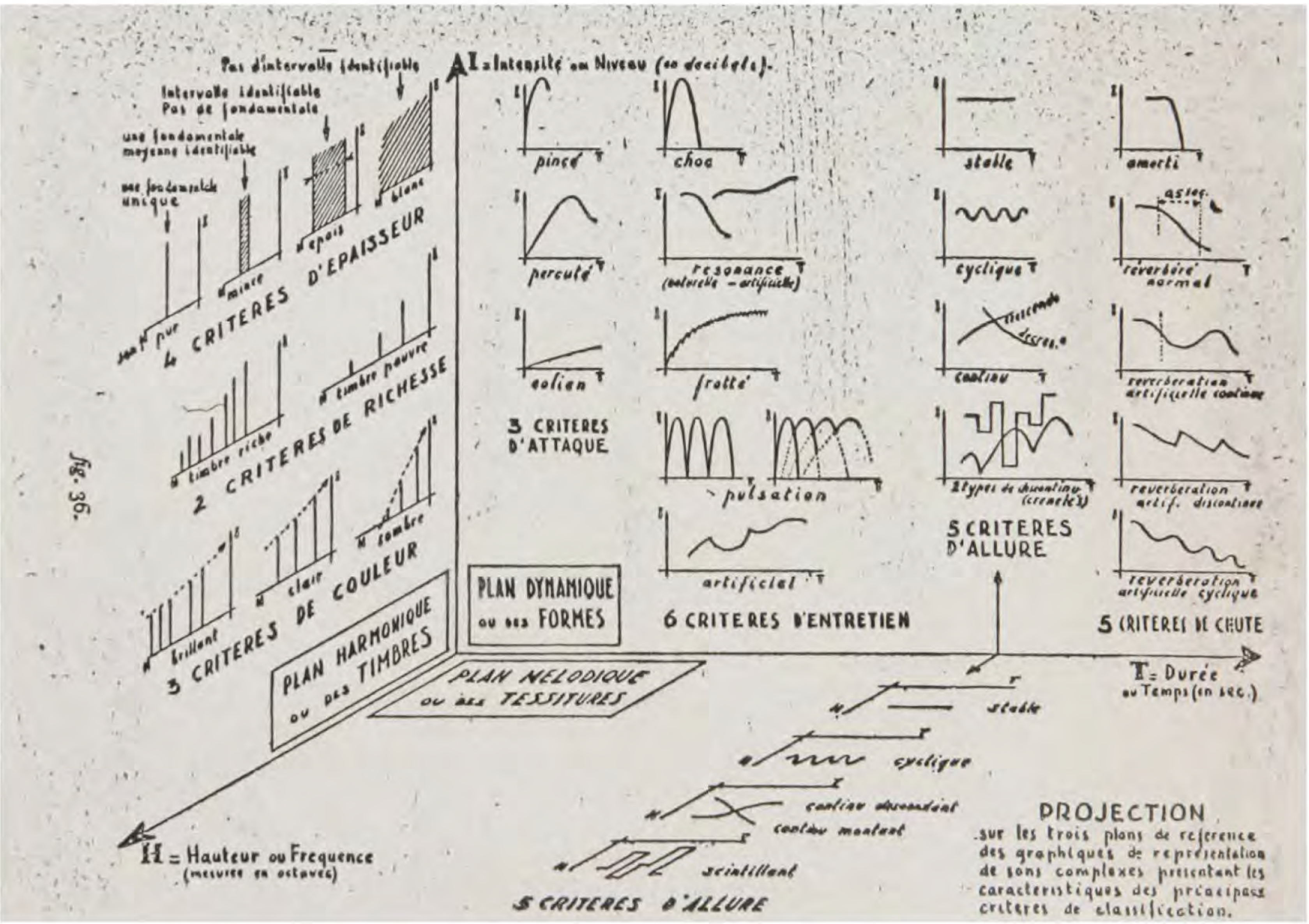

fig. 36.

PROJECTION
sur les trois plans de reference des graphiques de representation de sons complexes presentant les caracteristiques des principaux criteres de classification.

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[38] Pierre Schaeffer, A la Recherche, op.cit., p.225

The *Esquisse pour un Solfège Concret* was only intended to be a provisional treatise, designed to serve the needs of a particular school of composition which grew around the study of *musique concrète* in the late 1940s and early 1950s. This work, nevertheless, is of general importance not only for its position chronologically in the history of electronic music but also for its insight into fundamental problems of sound identification and specification still encountered by composers using present day systems. Objective analyses of sound phenomena in terms of the scientifically measurable quantities of pitch, amplitude and time, interrelated to provide details of both envelope and timbre, are of little value to the creative musician unless the information thus extracted may be usefully related to the subjective world of aural perception. It is the latter feature which is of primary importance to the majority of composers in their quest to employ sound as an artistic means of communication, whether through the traditional systems of instrumental realisation or through the direct generation, manipulation and organisation of sounds within a synthesis system. Some composers, it is true, seek to produce works which are entirely conceived in terms of mathematical principles which are readily translatable into the objective quantities outlined above. The final products of such ventures are, nevertheless, transmitted as a pattern of aural events, and will thus be evaluated subjectively by those who choose to perceive them.

The problems encountered in overcoming the communication barrier which exists between these spheres of subjectivity and objectivity have proved fundamental stumbling blocks in the establishment and development of creatively useful sound generation and processing systems. Schaeffer not only appreciated the significance of such barriers in the development of his own musique concrète but also realised at a very early stage in his investigations that their removal required the formulation of a linguistically based interface system, demanding in turn a suitable syntax. Over the years technological considerations have played an increasingly important role in shaping the functional characteristics of electronic music systems. Man/machine communications have thus become an integral part of the relationship between a composer and his sound world, and it will be seen later in this account how many of the shortcomings encountered in system designs may be traced to an insufficient regard for the fundamental problems of language so carefully studied by Schaeffer over twenty-five years ago.

Schaeffer's work, far from remaining a phenomenon to be associated with the early period of electronic music, has continued to develop to the present day. The output of compositions from the studio at the Club d'Essai during the mid 1950s was considerable, particularly from Henry and Arthuys. In 1954 Milhaud produced a work entitled La Rivière Endormie, and Varèse visited the studio to produce the first version of the tape for Déserts, an event which will be discussed more fully later in connection with developments in America.

Towards the latter part of the decade the principles of musique concrète became far more diffuse, and gradually electronic sound generation and processing techniques became acceptable studio practices. During 1957-58 the Greek-born composer Iannis Xenakis produced his first major studio work, Diamorphoses, in which he applied principles of mathematical organisation to the processes of composition with concrète sounds. Over the same period Schaeffer began a major reorganisation of both his methods of composition and also the studio facilities, and the term musique concrète was replaced by a far more general description: expériences musicales. Henry departed to found his own studio, Apsome. Luc Ferrari and François Bernard Mache joined Schaeffer's studio staff, and the group renamed themselves Groupe de Recherches Musicales. Xenakis continued his work with sounds of both electronic and natural origin and began to experiment in earnest with computers, initially as normal data processing machines but later as direct sound synthesis systems in their own right.

By the end of the period currently under discussion (1964) the rigid distinctions between the philosophies of musique concrète and elektronische Musik (to be discussed in the next section) had long since been replaced by far more universal attitudes towards composition with sounds. Developments in France, however, remained largely polarised around the work of the Groupe de Recherches Musicales who have exerted a dominating influence to the present day, still under the leadership of Schaeffer.[39]

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[39] In 1966 Schaeffer published an extensive treatise in nine sections consolidating his pedagogical work up until that date and outlining the aims of the Groupe de Recherches Musicales: Pierre Schaeffer, Traité des Objets Musicaux, (Editions du Seuil, Paris, 1966).

2 Elektronische Musik

Unlike musique concrète, the birth and early development of elektronische Musik was brought about not through the endeavours of a single individual but as the result of a collaboration between several interested parties drawn from both musical and technological backgrounds.

During 1948 Dr. Werner Meyer-Eppler (b.1913, Antwerp; d. 1960, Bonn), an eminent physicist who was at that time director of Phonetics at Bonn University, received a visit from Homer W. Dudley, a research physicist at Bell Telephone Laboratories, New Jersey, U.S.A. One of the purposes of his visit was to demonstrate a machine called a Vocoder (A Voice Operated Recorder) which could function both as a speech analyser and also as an artificial talker. The instrument operated in the former mode by detecting the energy levels of successive sound samples measured over the audio frequency spectrum via a series of band pass filters, the results being displayed graphically as a function of frequency against time. In the latter mode the procedures were reversed, where by scanning a graph containing a shaded representation of a suitably drawn intensity spectrum over a suitable time interval, feedback networks to the filters could be energised via a noise generator to produce synthesised sound information.

Although the fidelity of the machine was distinctly limited, its purpose being primarily the processing for speech rather than music, the characteristics of the instrument made a strong impression on Meyer-Eppler. During 1949 he published an important treatise on the history and design of electronic musical instruments entitled Elektrische Klangerzeugung which included a description of the Vocoder,[40] and used tape recordings of sounds produced via the instrument to illustrate a lecture on the development possibilities of sound generation techniques, given at the North-West German Music Academy at Detmold. The audience included Robert Beyer of North-West German Radio, who was already known for his interest in the uses of electronic technology in music.[41]

The Detmold meeting attracted so much interest that it was decided to include lectures on 'The Sound World of Electronic Music' during the 1950 International Summer School for New Music at Darmstadt. Both Beyer and Meyer-Eppler presented papers, concerned respectively with the design and functional characteristics of electronic musical instruments, and the state of research into techniques of speech synthesis. Edgard Varèse lectured at Darmstadt that year, and it is strange that no lasting mutual lines of communication were established at this time.[42]

The composer Herbert Eimert was also at the Summer School, and as a result of this meeting Eimert, Beyer and Meyer-Eppler agreed to enter into an informal association concerned with investigating the musical possibilities of electronic sound generation. During the Autumn of 1950, the electronic

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40 Werner Meyer-Eppler, Elektrische Klangerzeugung, subtitled Elektronische Musik und Synthetische Sprache (Dümmler, Bonn, 1949)

41 Robert Beyer had written an article entitled 'Das Problem der "kommenden Musik"' in Die Musik, Vol.XX, 12 (1928), pp.861-866 which included a discussion on the use of electronic instruments in musical composition.

42 Varèse had just started work on Déserts (see Developments in America later in this chapter.

design engineer Harald Bode delivered one of his Melochords[43] to Meyer-Eppler at Bonn, and the latter used this instrument to prepare a number of Klangmodelle; simple studies in the production of sound events created from sine wave generator tones used in combinations. These provided the basis for subsequent discussions between the three parties. The results of these preliminary investigations were presented by Meyer-Eppler at the Darmstadt Summer School in July 1951 in the form of a lecture entitled 'The Possibilities of Electronic Sound Production'. Beyer contributed a paper on 'Music and Technology', and Eimert discussed 'Music on the Borderline'. Schaeffer attended the meeting, and, as already noted earlier in this chapter, this confrontation provided a sharp impetus to the already growing disagreement between the French and German philosophies of electronic music.

On 18 October 1951 the Cologne radio station broadcast an evening programme entitled 'The Sound World of Electronic Music' which consisted of a forum held between Eimert, Meyer-Eppler and Beyer, using Meyer-Eppler's Klangmodelle as illustrations. On the same day a committee consisting of the three forum participants, joined by Fritz Enkel, technical director of Radio Cologne, and a few of his assistants, decided to establish an electronic music studio 'to follow the process suggested by Dr. Meyer-Eppler to compose directly onto magnetic tape'.[44]

Work began on the studio that Autumn but it was not until nearly two years later that the system became fully operational, with Herbert Eimert

[43] The characteristics of this instrument will be discussed shortly.

[44] Quoted in Otto Luening, 'An Unfinished History of Electronic Music', Music Education Journal, LV, 3 (Nov. 1968), p.46

appointed as artistic director.

Throughout the intervening period interest continued to grow. In December 1951 Meyer-Eppler lectured on 'New Methods of Electronic Tone Generation' to an audience of nearly a thousand at a meeting of technologists in Bonn. During the first half of 1952 the composer Bruno Maderna produced a piece entitled Musica su due Dimensione in association with Meyer-Eppler at the Institute of Phonetics, Bonn, and presented the work at the Darmstadt Summer School that year to an audience which included Karel Goeyvaerts, Bengt Hambraeus, Gisetler Klebe, Gottfried Michael Koenig and Karlheinz Stockhausen, all of whom were to become involved subsequently with the composition of elektronische Musik at Cologne. Viewed in retrospect, the scoring of Maderna's piece for flute, percussion and a loudspeaker reproducing a tape of electronically generated material, is of some interest: this integration of natural and electronic sound worlds conformed neither to the principles of musique concrète nor to the early manifestations of elektronische Musik, anticipating instead a time when the uses of electronic sound generation and manipulation in musical composition would cease to be subjected to such rigid doctrines.

Beyer and Eimert, with technical assistance from Enkel, composed their first all-electronic compositions while the studio was still under construction: Klang im Unbegrenzten Raum (3 movements, 1951/52), Klangstudie I (1952) and Klangstudie II (1952/53). The studio gradually came into commission towards the end of 1952, permitting work to be transferred in stages from the laboratory bench to the more congenial surroundings of a specially constructed system. During the first half of 1953 Beyer and Eimert composed Ostinate Figuren und Rhythmen, and Eimert on his own composed Struktur 8 and Glockenspiel.

In June of the same year these first complete pieces of elektronische Musik received their first public performance in Paris during the Festival of New Music organised by the Centre de Documentation de Musique Internationale in association with North-West German Radio,[45] and a few weeks later extracts from this programme were presented by Eimert and Meyer-Eppler at the 1953 International Summer School, Darmstadt. During the Autumn developments at the studio gathered momentum. Goeyvaerts composed Compositie nr. 5 met Zuivere Tonen and Stockhausen his Studie I (Komposition 1953 nr. 2)[46] followed by Studie II, completed in 1954.

As the system and its associated school of composition developed, many of its exponents publicised their work not only in compositions but also in writings which collectively were to exercise a considerable influence on the design and function of the systems which were subsequently developed elsewhere in Europe. By far the most widely circulated group of articles to appear during the early 1950s comprised the first volume of Die Reihe, 1955, devoted entirely to the subject of electronic music.[47]

[45] Described in Herbert Eimert, 'Musique Electronique', Vers une Musique Expérimentale, op.cit., pp.45-49. Pierre Schaeffer presented a concert of musique concrète during the same festival.

[46] Karlheinz Stockhausen (b.22 August 1928, Mödrath, Cologne) had become acquainted with Eimert during 1951, and on his suggestion paid his first visit to the Darmstadt Summer School that year. From January 1952 until April 1953 he studied in Paris with Messiaen and Milhaud, and it was during this period that Goeyvaerts introduced him to the properties of electronically generated sound waves. Before returning to Cologne he paid a short visit to Schaeffer's studio, composing a short piece of musique concrète entitled Etude.

[47] Die Reihe, Ed. Herbert Eimert and Karlheinz Stockhausen, I (Universal Edition, A. G. Wein, 1955), English translation (Theodore Presser Co., Bryn Mawr, Pennsylvania, 2nd Impression, 1959)

This publication, however, is lacking in objective detail upon which to base a critique, and it is necessary to turn in the first instance to a slightly earlier but more comprehensive group of twelve papers published by the North-West German Broadcasting Corporation in their series Technische Hausmitteilungen des Nordwestdeutschen Rundfunks, 1954.[48]

The technical facilities of the Cologne studio are described by Fritz Enkel in the third of these papers and this account provides a useful starting point for a study of the primary features of elektronische Musik.

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48 Technische Hausmitteilungen des Nordwestdeutschen Rundfunks, Vol.6 (1954), pp.4–54, divided into twelve papers, English translation, National Research Council of Canada, Technical Translations, TT 601-TT 612 (Ottawa, 1956):

TT 601, Herbert Eimert, 'Electronic Music', tr. D. A. Sinclair
TT 602, Werner Meyer-Eppler, 'The Terminology of Electronic Music', tr. D. A. Sinclair
TT 603, Fritz Enkel, 'The Technical Facilities of the Electronic Music Studio (of the Cologne Broadcasting Station)', tr. D. A. Sinclair
TT 604, Fritz Enkel and Heinz Schütz, 'Magnetic Tape Technique', tr. D. A. Sinclair
TT 605, Karl-Heinz Adams, 'Filter Circuits for Electronic Sound Production', tr. D. A. Sinclair
TT 606, Friedrich Trautwein, 'The Electronic Monochord', tr. H. A. G. Nathan
TT 607, Harald Bode, 'The Melochord of the Cologne Studio for Electronic Music', tr. H. A. G. Nathan
TT 608, Werner Meyer-Eppler, 'The Mathematic-Acoustical Fundamentals of Electronic Sound Composition', tr. H. A. G. Nathan
TT 609, Fritz Enkel and Heinz Schütz, 'The Production of Sound Effects for Radio Dramas', tr. D. A. Sinclair
TT 610, Herbert Eimert, 'The Place of Electronic Music in the Musical Situation', tr. D. A. Sinclair
TT 611, Karlheinz Stockhausen, 'Electronic Musical Composition No.2, 1953', tr. D. A. Sinclair
TT 612, Herbert Eimert, Fritz Enkel and Karlheinz Stockhausen, 'Problems of Electronic Music Notation', tr. D. A. Sinclair

Enkel, examining the characteristics of sound processing from an engineering standpoint, identifies three fields of activity in which electronics have been actively applied in the production of music.[49]

1 Electronic imitation of acoustic musical instruments (electrical imitation)

2 The production of acoustic effects by electronic denaturation of the acoustic phenomena (concrète music)

3 The production by electronic means, in accordance with a preconceived plan, of musical sounds which cannot be produced mechanically (electronic music)

He goes on to state, interestingly, that 'the work of the Cologne studio is devoted primarily to the fields of electronic and concrète music'.

This declaration of a dual purpose for the studio facilities at this time is of some significance, for despite the further qualification that the <u>concrète</u> function was primarily concerned with the production of sound effects for radio drama productions[50] it would seem that the studio was used on occasions for processing naturally generated sound materials using procedures similar in nature to those carried out in Schaeffer's studio. Enkel clearly understood the close technical similarities between many of the processes of <u>elektronische Musik</u> and <u>musique concrète</u> from a very early stage in the development of the studio. In contrast, Eimert, the artistic director of the studio, was ready to reject out of hand any suggestion that

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[49] <u>Technische ...</u>, <u>op.cit.</u>, paper 3 (TT 603), p.2

[50] The ninth paper (TT 609) by Enkel and Schütz would appear from its title to offer an account of the use of the studio for these <u>concrète</u> functions. Unfortunately, the article proves to be only a superficial description of two sets of simple processing operations related to radio drama extracts.

that concrète techniques were of relevance to the production of elektronische Musik, defensively dismissing the comparisons made by others as the products of their technical ignorance:

> Along with electrical music appearing in the form of hit tunes, fugues and concertos, we have the unmonitored electric sounds, the decorative sound phenomena which can be used for illustrative purposes. Such naturalistic adjuncts are as far removed from music as nature from art. The extent of the general confusion prevailing in this regard can be estimated from the performance of 'musique concrète' at the Donaueschingen Music Festival of 1953. Many critics referred to these performances - a double confusion! as 'electronic music'. It must be admitted, however, that the novelty of the phenomena combined with the general unfamiliarity with the acoustical effects and ignorance concerning the technical methods, to some extent excuse the error.[51]

The distinctly varied quality of the early musique concrète pieces justified criticism on musical grounds, but such a vehement attack on their use of naturally generated sound material suggests an exceedingly limited perspective on the part of Eimert at this time.[52]

According to Enkel, the agreed brief for the studio included provision of the following equipment:[53]

a) Electronic sound and noise-producing sources to provide the raw material for further processing

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[51] Technische ..., op.cit., paper 10 (TT 610), p.3

[52] As will be seen later in this chapter, his reluctance to modify his ideas in the later 1950s despite pressures from more perceptive composers such as Stockhausen led eventually to his resignation as director of the studio.

[53] Technische ..., op.cit., paper 3 (TT 603), p.3

> b) Electro-acoustic shaping means for the purpose of influencing the tone frequency spectra and the transient processes of sound phenomena. The methods used here are taken from communications technique and yield sound phenomena which cannot be procured by mechanical methods
>
> c) Magnetic sound recording apparatuses for further processing of the material obtained with devices a) and b).
>
> These means make it possible for a composer to set down his works exactly in accordance with his conceptions and to have it heard without the aid of interpreters (director and orchestra)!

The last statement above is open to question, for it assumes that the functional characteristics of the system would readily fulfil the practical requirements of any composer wishing to incorporate the processes of electronic sound generation and treatment as part of his or her compositional language. No system, even today, has yet provided the range of facilities and techniques necessary to achieve such an objective, and it is the functional limitations of each studio which restrict the musical operations which may be performed. It must nevertheless be remembered that this account was written at a time when electronics were only just beginning to be explored in musical composition, and then, as is still true today, expectations frequently exceeded the achieved results.

The system diagram for the early Cologne studio is given as follows:[54]

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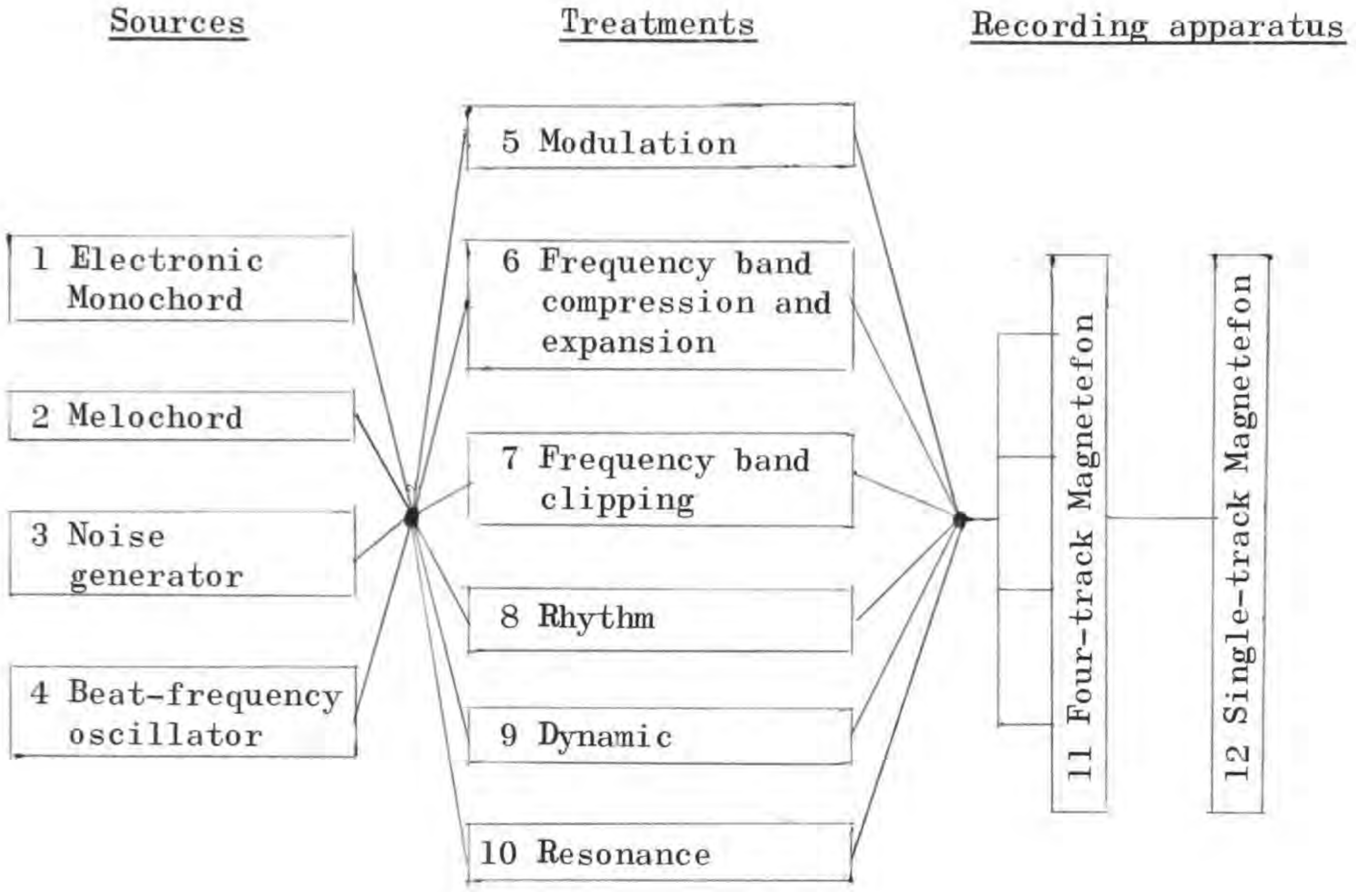
[54] Technische ..., op.cit., paper 3 (TT 603), p.15



The designs of the Electronic Monochord and the Melochord are discussed in detail in two of the papers later in the series.[55] The Monochord was ordered from Trautwein, and was intended to be a development of his concert Trautonium designed especially for the Studio. Trautwein, however, failed to finish the commission himself and Enkel completed the model in the studio workshop. The delays in construction and shortage of funds resulted in the omission of several features available on the concert version, for example, no suppressors were provided for the removal of spurious noise components which sometimes resulted from rapid changes in timbre or pitch register, and no independent subharmonic generators were available to enhance the range of tonal sources. The later economy was justified on the

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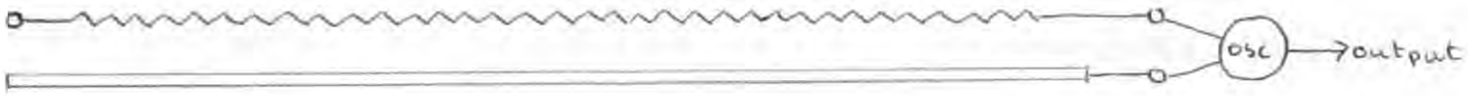
[55] <u>Technische</u> ..., <u>op.cit.</u>, papers 6 (TT 606) and 7 (TT 607) respectively

grounds that in electronic composition 'superimposing recordings of several voices played separately on a single sound track permits such effects, although correspondingly, more time is required'.[56] In its place a frequency division facility was provided which enabled the composer to draw upon a range of subharmonic tones from a single pitch generator.

The playing device in the concert Trautonium consists of a resistance wire stretched over a metal rail, each of these elements being connected to either side of a bipolar oscillator control circuit.



By means of a series of press keys positioned over the wire, the latter may be forced into contact with the rail at equally spaced points along its length to complete the oscillator control circuit, the frequency of the note thus generated being determined by the resistance of the selected length of wire. It is desirable that a normal electronic keyboard instrument should generate a suitable range of equally tempered pitches based on the basic musical interval of a semitone, and this requires conversion of the control function from a linear to an exponential characteristic. In the Trautonium the resistance wire is suitably tapered towards one end so that equal increases or decreases in the tapped length produce the required constant percentage change in its effective resistance.

A similar variable resistance system was adopted for the Electronic

[56] Technische ..., op.cit., paper 6 (TT 606), p.4

Monochord, modified, however, to allow both the pitch range and the interval division of the notes produced from the keyboard to be continuously variable. A resistance wire of constant cross-section is employed, wound around an elliptical drum at a winding pitch arranged so that a series of equal musical pitch intervals may be generated from equally spaced key positions. The return connection to the oscillator control circuit is provided by a piece of electrically conducting, brocaded material, enveloping, but not quite in contact with, the resistance wire. Depression of a key forces the material into contact with the wire at that point, completing the circuit.

The Monochord is equipped with two keyboards each serviced by its own source oscillator. The outputs from the latter are monophonic, the pitch of the highest key selected being generated if two or more notes are depressed simultaneously. The size of the pitch interval factor may be controlled by altering the sensitivity of the oscillator control circuit, and the range of pitches generated varied by rotating the drum to alter the positioning of the resistance wire under the keys. Two methods of amplitude control are provided for each keyboard; a pivoted pedal and a pressure-dependent resistance placed underneath the elastically suspended keyboard. For the latter a stroke of a few millimeters after initial contact has been made by the key increases pressure on a liquid resistance tube which in turn controls the output of the generator over a restricted range of about fifty decibels. The range of intensities detectable by the human ear between the extreme limits of the threshold of hearing and the threshold of pain extends to more than one hundred decibels in the middle frequency ranges. A touch-keyboard variation of fifty decibels nevertheless proved quite sufficient for Cologne's purposes, for wider variations in in attenuation could be achieved by additionally employing the footpedal or by altering level controls elsewhere in the studio system.

The wave generators produce a sawtooth wave which is gently filtered and attenuated according to the selected frequency. Lower notes have richer overtone structures than higher ones, and are also of a greater overall amplitude. In addition to the sub-harmonic frequency division circuit referred to earlier, the tone colours may be altered further after generation by switching on additional series of filter networks.

The Melochord, first developed in 1949 by Harald Bode, working near Munich, is in some respects similar to the Electronic Monochord, being an electronic keyed instrument with two independent monophonic keyboards. There are, however, important differences in its functional characteristics which merit examination. The instrument was designed in the first instance to serve the requirements of the German Broadcasting System, its primary purpose being the interpretation of conventional musical material for use in the preparation of light music and drama productions. The establishment of the Cologne electronic studio, however, opened up new possibilities for such an instrument, and Bode accordingly developed a special experimental version.

Bode outlines his design philosophy for the modified instrument as follows:[57]

> The basic idea in designing this instrument was to have outside the instrument the controlling apparatus for all the sound parameters which may be represented by known

[57] <u>Technische</u> ..., <u>op.cit.</u>, paper 7 (TT 607), pp.2-3

> aids, while the latter should contain only the elements which are characteristic of it ... each playing range of the melochord contains a separate tonal generator assigned to a separate keyboard, or part thereof, a separator stage for preventing reactive effects from the consecutive sound determining devices on the generators ... a control device for the artificial simulation of known build-up and fall-off processes (sounds produced by wind and pluck-string instruments) and ... crescendo pedals for controlling the volume.
>
> A particularly interesting feature of the melochord is the fact that it is equipped with a step-by-step filter, which can be tuned with pitch. For example, when the pitch of the fundamental tone is varied the frequency of the formant may vary with it. Additional improvements in the musical tones may be obtained in a two-tone instrument, if the alternating voltages from the two sound channels are conducted into a modulator and are mixed in multiplication.[58] Furthermore, it is also possible to modulate the tonal generator with white noise or with any arbitrary portion from the spectrum of a noise generator, or, with the generators turned off, to transmit white noise to the other parts determining the sound. ... Additional aids ... include above all the utilisation of magnetic tape techniques, both old and new, and the inclusion of a resonant volume.[59]

The envelope shaping facilities controlling the attack and decay of individual notes and step-by-sound filters which track the selected keys are both useful aids; the latter enabling a selected overtone structure to be repeated at different pitches and the former providing a consistency of dynamic shaping. Provision is also made for frequency modulation of the generators themselves; in the first instance for use at relatively low speeds, for example, 6 to 8 Hertz, to give vibrato effects, though this option, however, was rarely employed in elektronische Musik

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[58] For example, via a ring modulator, a device to be discussed shortly.

[59] By the use of some form of reverberation unit.

compositions in view of its considerable musical banality.

The keyboards are tempered in semitones, each covering a range of 37 notes. Release keys on the left-hand side of the instrument permit octave transposition of the keyboard in the ratio 1 : 2 : 8 providing a total pitch range of seven octaves. No precise information is given in Bode's article regarding the nature of the basic generator waveforms, excepting that they are rich in overtones and thus highly suited to the processes of filtering. Meyer-Eppler's book Elektrische Klangerzeugung, however, indicates that the 1949 prototype of the Melochord employed a sawtooth generator, similar to the Electronic Monochord.[60]

Comprehensive switching arrangements for the keyboards and their associated electronic circuits provide a flexible range of operational characteristics. For example, whilst the upper keyboard controls the frequency of a source generator, it is possible to arrange for the lower keyboard to control its associated timbre shaping network independently. Alternatively, the generators may be connected to an external filter bank,[61] inserted prior to the envelope shaping and general crescendo control circuitry.

Despite the flexible range of characteristics offered by the Melochord and the Electronic Monochord for the generation and sequencing of pitch events, these instruments were not extensively used in the Cologne studio,

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[60] Meyer-Eppler, Elektrische Klangerzeugung, op.cit., p.105

[61] On the original Melochord a bank of filters was included as an integral part of the instrumental design. For the Cologne studio, however, it was felt that the flexibility of externally connected networks was more desirable.

composers generally preferring to work directly with tones obtained from accurate laboratory oscillators which would cover the entire audible frequency range in discrete steps. It has been determined by Fourier[62] that any periodic wave form however produced may be analysed mathematically into sinusoidal components derived from an identifiable fundamental frequency. By applying this process in reverse it is thus theoretically possible to synthesise any desired timbre by adding together sinusoidal audio signals of a suitable frequency, amplitude and phase lag. To the pioneers of elektronische Musik the sine wave generator provided the basic structural element upon which to build electronic manifestations of post-Webernian serial techniques. Stockhausen himself constructed his first piece of elektronische Musik, Studie I (Electronic Composition No.2), 1953 entirely from sine tones which were subjected to a total organisation of pitch, intensity and duration.

> After a period of preparatory listening and testing I decided not to use any electronic sound sources which produce ready-composed sound spectra (Melochord, Trautonium), but only to employ sinusoidal tones from a frequency generator ('pure' tones without overtones). Sinusoidal tones differ from each other only in frequency and amplitude. Thus, when I used simple frequency and amplitude conditions for the musical construction the sound will be the result of these combinations.[63]

At the other end of the information spectrum lies the random distribution

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[62] Fourier's theorem states that any motion which recurs at a definite frequency can be built up from a number of simple vibrations whose frequencies are integral multiples of the fundamental recurrence frequency.

[63] Technische ..., op.cit., paper 11 (TT 611), p.2

of frequency components associated with the generation of noise. The device commonly employed for the production of the latter is known as a white noise generator whose output must satisfy the condition that one of the roots of its voltage must be proportional to the band width over the complete audio spectrum.[64] The Cologne studio was equipped with such a generator, and this was extensively used as a source for subtractive processing, where, with the aid of filter banks, specific frequency areas of noise could be isolated. Such procedures complemented the additive techniques associated with Fourier synthesis.

The Cologne studio of the early 1950s was thus equipped with four primary sources of sound material: the Electronic Monochord, the Trautonium, a noise generator and a sine wave oscillator.[65]

Enkel's description of the facilities of the Cologne studio continues with an explanation of the characteristics associated with the treatment and recording processes (items five to twelve on the studio diagram given on page 95).[66] Firstly an account is given of the use of modulation techniques, whereby a 'multiple mixture of sounds, new phenomena are

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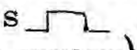
[64] Another device commonly to be found in larger studios is the pink noise generator which satisfies the condition of constant energy per musical octave rather than constant energy per absolute bandwidth.

[65] Additional generators were added later providing sound sources in the form of ramp/sawtooth waves;  or  , square waves  , triangle waves  , and impulses (a derivative of square waves). The mathematical characteristics of these sources are listed in the Appendix , and will be discussed later in this chapter.

[66] Technische ..., op.cit., paper 3 (TT 603), pp.5-10

produced which are no longer related acoustically with the raw material'.[67]

67 This is only true of the fundamental frequencies if the two input signals are in a complex ratio, one to another. If the ratios are simple, harmonics of the sources are generated. Meyer-Eppler in his article 'The Mathematical-Acoustical Fundamentals of Electronic Sound Composition' (*Technische* ..., *op.cit.*, paper 8 (TT 608), p.24) lists the harmonics occurring when a simple tone and a sound containing harmonics are ring modulated together in simple frequency ratios:

Frequency ratio	Musical interval starting from a simple tone	Only harmonics of the following ordinal numbers occur														
Frequency of the simple tone lower than the fundamental frequency of sound.																
1 : 2	octave up	1		3		5		7		9		11		13		15
1 : 3	duodecimo up		2		4	5		7	8		10	11		13	14	
2 : 3	fifth up		2		4	5		7	8		10	11		13	14	
1 : 4	double-octave up			3		5		7		9		11		13		15
3 : 4	fourth up	1				5		7		9		11		13		15
2 : 5	tenth up			3				7	8				12	13		
3 : 5	major sixth up		2					7	8				12	13		
4 : 5	major third up	1					6			9		11			14	
Frequency of the simple tone higher than the fundamental frequency of sound.																
2 : 1	octave down	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
3 : 1	duodecimo down	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
3 : 2	fifth down	1		3		5		7		9						
4 : 3	fourth down	1	2			5		7	8		10	11		13	14	

Two devices are described in this context: 1) the ring modulator, and 2) the quadripole modulator. During this early era of electronic music systems the use of the ring modulator is of particular interest, for in the absence of voltage controlled devices this process provided the principal means for deriving modulated information, including envelope shaping, from electronic sources.

The primary characteristic associated with this device is its ability to generate sum and difference products from the components of two independent signal sources, applied to either side of a bi-polar input circuit. If two sinusoidal waves having frequency values at a particular instant of 75 Hertz and 500 Hertz respectively are employed, for example, the output from the modulator will consist of a summation tone of 575 Hertz and a difference tone of 425 Hertz, both source frequencies being heavily suppressed.

If one or both of the waves contain overtones, however, the results are more complex. If a wave containing first, second, third and fourth order harmonics giving components at 75, 150, 225 and 300 Hertz is modulated against a carrier sine wave tuned to 500 Hertz, summation tones of 575, 650, 725 and 800 Hertz, and difference tones of 424, 350, 275 and 200 Hertz would be produced.

Enkel observes that 'the ring modulator is used primarily for frequency transposition, i.e. the displacement of frequency spectra to other frequency ranges while retaining all the frequency intervals. In this process the carrier and signal frequencies are suppressed without special means.'[68]

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[68] <u>Technische</u> ..., <u>op.cit.</u>, paper 3 (TT 603), p.5

This comment is of some significance, for the use of the terms 'carrier' and signal suggest a technical rather than a musical appreciation of the device's potential. A true carrier in its accepted electronic sense would consist of a sine wave of fixed frequency and amplitude modulating a source to produce one set of summation tones and one set of difference tones. If, however, the frequency setting of the carrier is treated as a variable, it becomes possible to alter dynamically the frequency transpositions of a source while it is being modulated. Further, replacement of the carrier with a signal which varies not only in fundamental pitch but also in timbre permits the generation of a complex series of frequency transpositions from a source, regulated by the frequency and strength of the signal's partials.

Enkel's slight lack of understanding regarding the musical applications of sound processing becomes more apparent with his description of the quadripole modulator, a device which he maintains, on technical grounds offers features significantly different from those associated with the ring modulator. He states that from two source signals 'four different frequency spectra of the sounds ... can be obtained by means of different switch positions ... frequency band transposition, frequency band transposition and one of the output sounds, amplitude modulation, amplitude modulation and one of the output sounds'.[69] These differentiations assume operating conditions which are more applicable to processes of telecommunications than the creative uses of sound transformations, being based on mathematical rather than perceptual observations. It is assumed that not only will a steady sinusoidal carrier be always

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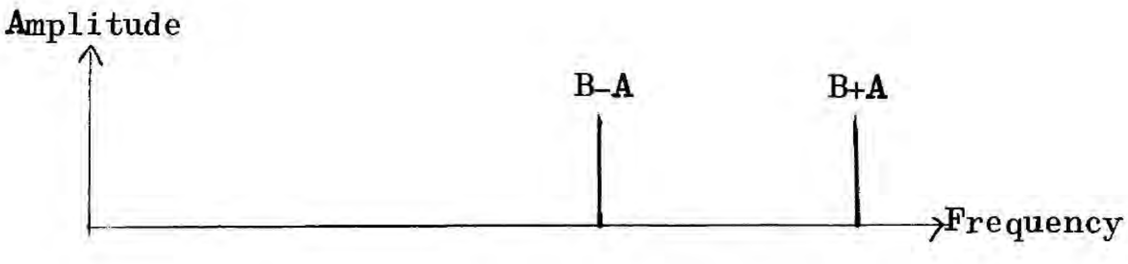
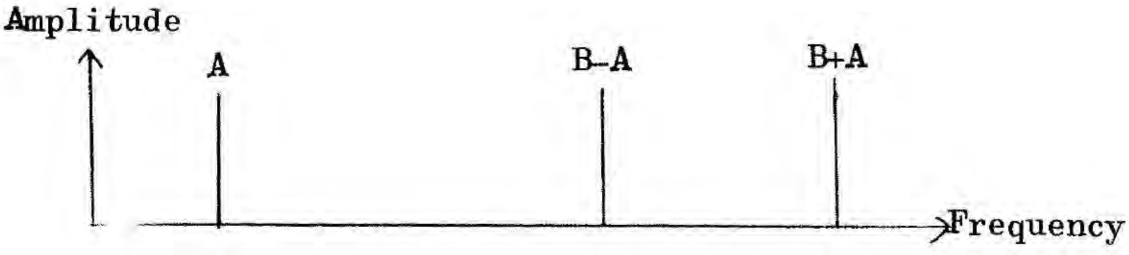
[69] <u>Technische</u> ..., <u>op.cit</u>., paper 3 (TT 603), pp.5-6

employed as one of the input sources, but that its frequency will lie in an area totally removed from the frequency ranges of the source signal and its principal harmonic components. Even if it is accepted that such conditions might be demanded in particular compositional circumstances, the differences in the functional characteristics of the four output options as asserted by Enkel are of limited musical value.

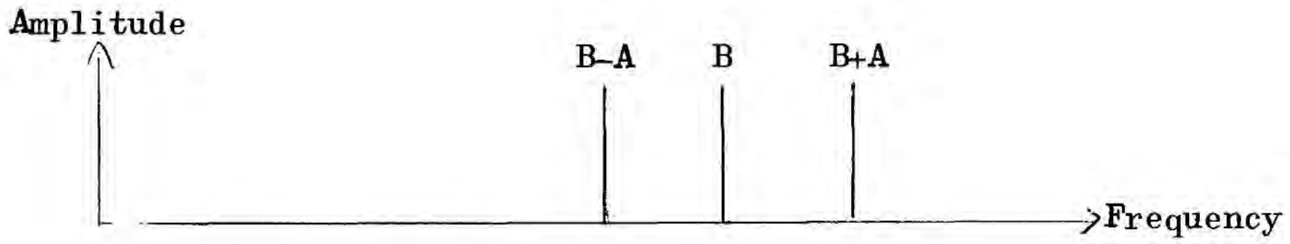
The first option, frequency band transposition, involves the use of the device as a simple ring modulator, as described above. If the products of the fundamental only of an input signal A and a sinusoidal carrier B are examined at a selected instant, and the frequencies of the two sources satisfy the conditions that A is of a relatively low value and B of an intermediate or high value, sum and difference tones at frequencies of B + A and B - A will be obtained.



The second option, frequency band transposition and one of the output sounds, mixes in source signal A with the ring modulator products.

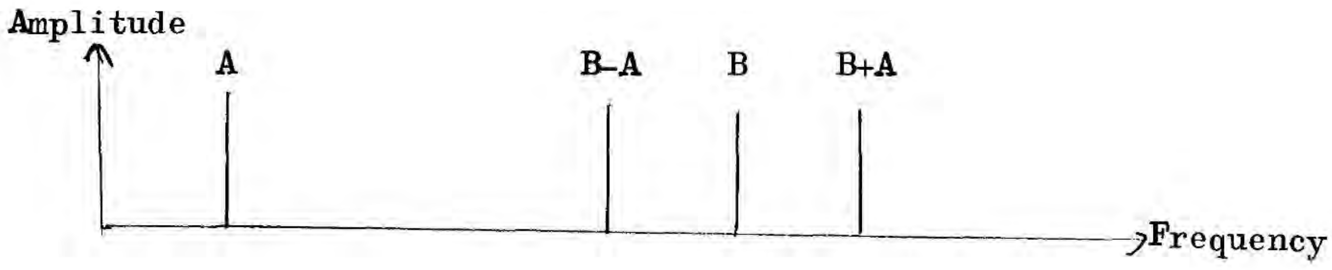


The third option, amplitude modulation, would appear from its description to involve the production of a significantly different type of modulated output. In practical terms, however, the device merely mixes in the carrier signal B instead of A with the ring modulated products.



As will be shown in the next chapter, in a discussion of voltage controlled modulation techniques the level of B may be adjusted to simulate the timbre complex which would result if the output amplitude of B was being directly modulated by A. The use of the term, however, can only act as a source of confusion for the composer, and presupposes that the ring modulator will always be operated with the above-mentioned artificial restriction on the frequency settings of both A and B. If this latter constraint is removed and the ring modulation of a carrier and source both derived from similar frequency regions is contemplated, the functional distinction made between the type of outputs associated with the second and third switch options becomes both mathematically and musically meaningless.

The fourth option, described as amplitude modulation and one of the output sounds provides the last combination; the ring modulated products mixed directly with both of the source signals.



These examples serve as illustrations of the risks involved in presenting technical concepts of device functions to composers without careful regard for their musical relevance. To a musician any restrictions on the frequencies of ring modulator inputs will prove detrimental to any creative use of the technique, and there is thus no purpose in suggesting that there is a major functional difference between switch options two or three above. The quadripole modulator may far more profitably be seen as a convenient method of mixing either or both of the source signals to a ring modulator with its products.

The second area of sound processing discussed is concerned with the techniques of frequency band compression and expansion, better known today under the heading of variable speed tape techniques.

> If a sound record is played at a speed other than standard, then depending on the extent of the change of speed, the reproduction may be changed from the subjective point of view so as to be unrecognisable. A curtailment of the time in such a case is acoustically equivalent to an expansion of the frequency band, while a lengthening of the time constitutes a contraction of the frequency band. Frequency band compression and expansion can be used extensively for influencing sound records. The intervals, in this case, remain unchanged while the transient processes are subjected to a very far reaching transformation.[70]

The use of the terms frequency band compression and expansion is again to be associated with a technically biased outlook regarding the processes involved. To a musician it is important to know that changing the speed of playback of a previously recorded tape involves a frequency transposition

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70 Technische ..., op.cit., paper 3 (TT 603), pp.6-7

of the pitch components of the source sounds, preserving the relative disposition of the constituent harmonics and effecting an associated proportional change on the speed at which the events occur. A recording of a work played at half its original speed, for example, would lower the pitch information by an octave, the changes in information content occupying twice their original time spans. This transformation changes the subjective nature of the sound material considerably, for all the constituent time-dependent features such as vibrato, and attack and decay times are proportionally affected.

It should be noted that in the case of acoustically generated sound material, the musical quality of different pitches obtained naturally from a selected instrumental source will invariably involve a timbre continuum which varies with respect to frequency, and this cannot be copied by recording information at one pitch setting and replying it at different speeds. A continuous clarinet note played at A = 220 Hertz, for example, is significantly different in harmonic structure to one played at A = 880 Hertz. Recording a note at 220 Hertz and increasing the speed of reproduction by a factor of four will generate a note pitched at 880 Hertz, but this will retain the harmonic proportions of the former and will thus sound artificial in quality.

With electronically generated sounds many instances will occur when the timbre continuum of the source will remain constant. Such material, derived directly from standard laboratory generators may be readily transposed by variable speed tape techniques to simulate the range of tones obtainable directly from the device. In the case of an electronic instrument source such as the Monochord, however, the overtone structure of the generated pitches is made to vary with respect to frequency as an inherent design feature, and thus the nature of recorded and transposed notes will differ from those directly generated from the keyboard.

The machine employed at Cologne for variable speed operations during the early 1950s was an old A.E.G. tape recorder with a specially modified drive mechanism, arranged to be continuously variable over a tape speed range of between 9 cm/sec. (about $3\frac{1}{3}$ inches per second) and 120 cm/sec. (about 48 inches per second), giving a pitch transposition range of about $3\frac{3}{4}$ octaves.

The third area of sound processing concerns the isolation of frequency bands or areas obtained from broad source spectra via the processes of low pass, high pass and band pass filtering. The design of filter circuits for electronic music systems is extensively discussed by Karl-Heinz Adams in the fifth paper of the series,[71] and consequently Enkel makes only a superficial reference to the characteristics associated with this major area of processing. He specifically cites the use of filter techniques for shaping the harmonic spectra produced by sawtooth oscillations (encountered, for example, in the Melochord and the Electric Monochord) and for converting white noise into coloured noise.

Adams commences his article by making an important distinction between two basic methods of preparing sound material electronically.[72]

> Two fundamental methods are employed in the technique of shaping for the production of sound in electronic music:

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[71] Technische ..., op.cit., paper 5 (TT 605)

[72] Ibid., p.2

> firstly, the so-called synthetic method, which composes the sound spectra from discrete sinusoidal oscillations of given mathematical proportions and secondly, the analytical method which employs very wide time spectra as its raw material. One of the chief requirements of the analytical method is the analysis of the raw material into spectra of various types for the production of timbre. All known types of filter circuit, namely, low pass, high pass and band pass[73] are used for the technical realisation of this task.

From a compositional point of view the differences between these two approaches is important. Synthesis of sound material from sine waves produced by electronic generation is an additive or multiplicative operation. This may loosely be compared with the processes of conventional instrumental composition, with the qualification that in the latter the sound sources consist of a differentiated set of instruments, each associated with a particular range of functional characteristics realised according to established performance practices. The second approach, however, involves procedures which have no direct parallel in the field of conventional instrumentation, involving the removal of frequency components from a rich source signal to provide material which is simpler in its acoustical construction and which may be perceptually of a very different nature.

The wide range of possible filter designs and networks has created a situation where no one device may be practically constructed capable of providing all the most useful features of subtractive processing as a consolidated set of functions. In consequence it is common to find several

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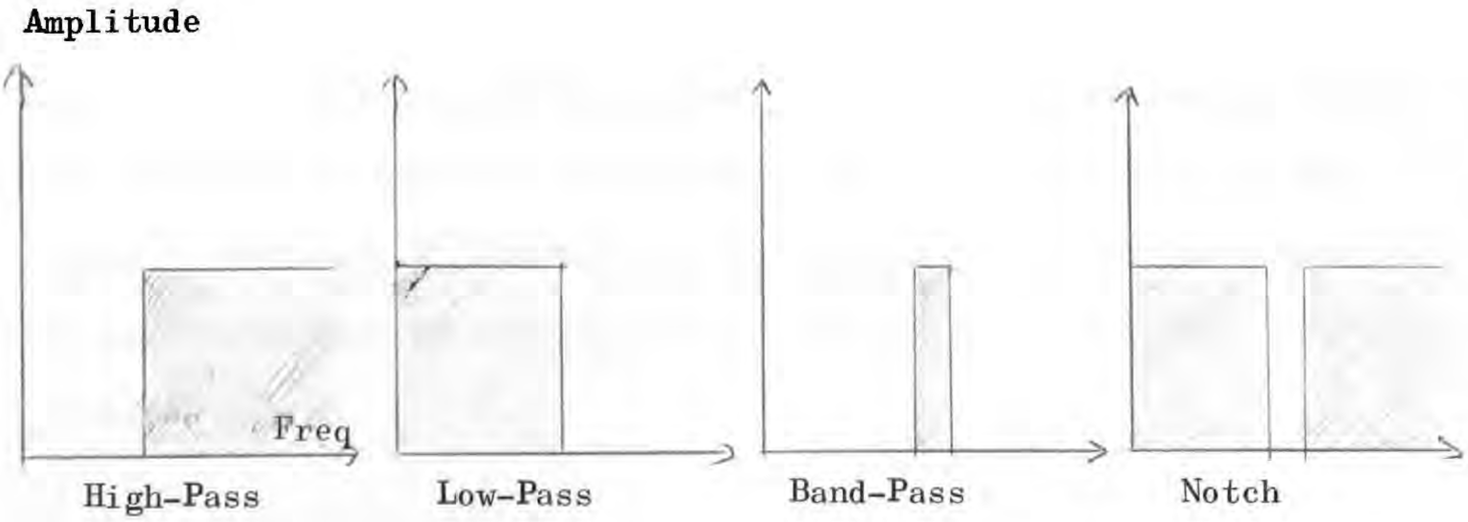
[73] Adams omits to mention the use of notch (or band reject) techniques where a small selected band of frequencies may be removed from an otherwise unaltered wide band spectrum. This function is an exact mirror of band pass processing.

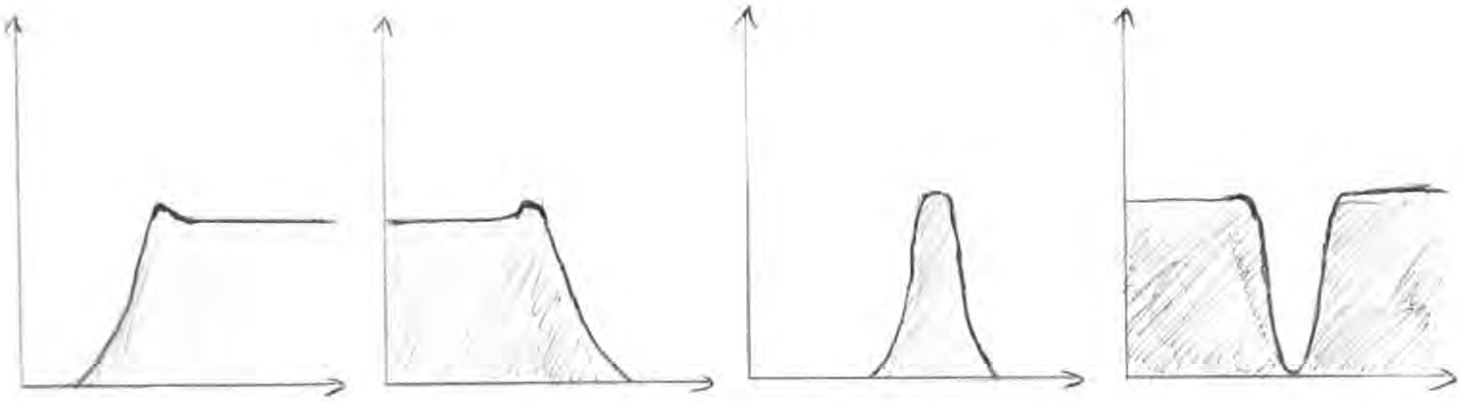
different filters included in a studio system. Before studying Adam's description of the facilities provided by the Cologne studio in the early years, it is necessary to examine, from a musical standpoint, the basic characteristics associated with filter processing.

Filters are devices which may be employed to remove certain areas of the audio frequency spectrum, allowing others to pass. High-pass filters act to remove all spectral elements below a particular frequency setting, and low-pass filters act to remove all spectral elements above. Band-pass filters remove all but a selected band of frequencies and notch filters remove only a narrow band of frequencies. To the electronic engineer ideal filters should be capable, at an extreme setting, of providing a sharp transition between the two states of signal pass and signal reject:

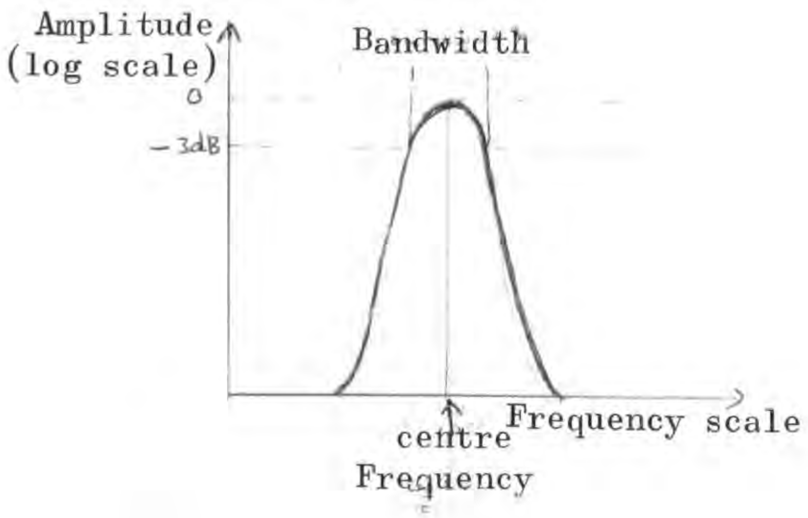
Shaded area = Frequency regions passed by the filter unit.



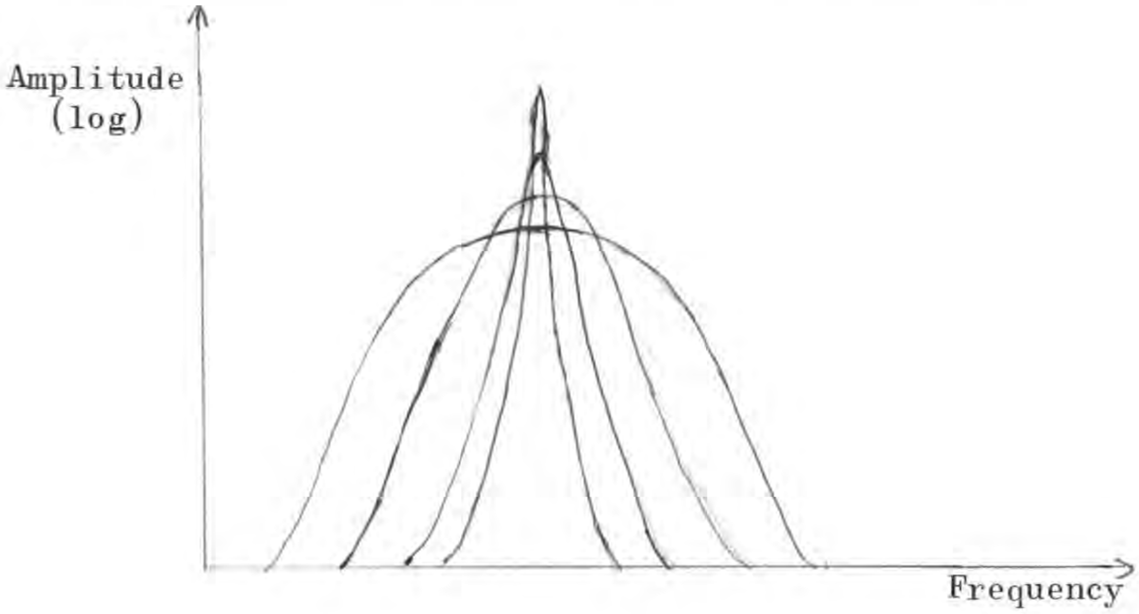
In practice it is not possible to design filters which will change from an infinitely steep flank to a perfectly flat pass band response. The functional characteristics of practical designs thus involve transfer characteristics from the states of signal pass to signal reject which are curves of varying steepness. The following are representative functions for the four basic filter types, discussed above.



The band-pass filter provides the most comprehensive range of characteristics for isolating frequency areas within a signal rich in spectral content. The following features are significant for the musical user: 1) The centre frequency of the pass band; 2) The effective width of the pass band; the limits being defined as the points at which the energy levels have fallen to 70 per cent (3 decibels) of the energy level at the centre of the band; 3) The flatness of response within the pass band and the associated rate of attenuation (or response characteristic) either side of the pass band.



Unless the band-pass filter consists of two independent high and low pass units connected in series (see later) the band width and response characteristics are usually inter-linked: the steeper the response, the smaller the effective band width. The sharpness of a band-pass filter's response is described either in decibels of attenuation per musical octave deviation from the centre frequency, or as a 'Q' factor; a measure of the curve's exponential characteristic with respect to the centre frequency.[74] The following diagram illustrates a representative range of responses from a 'Q' of about 1 to a 'Q' of about 100:



It can be seen from this diagram that an increase in the 'Q' factor produces a gain in the amplitude of the centre frequency. Perceptually, however, this effect is not usually appreciated, for the overall gain of the filter diminishes as the 'Q' increases. To the musician the following representation is more useful, showing the effect of different 'Q's with the change in gain factor removed.

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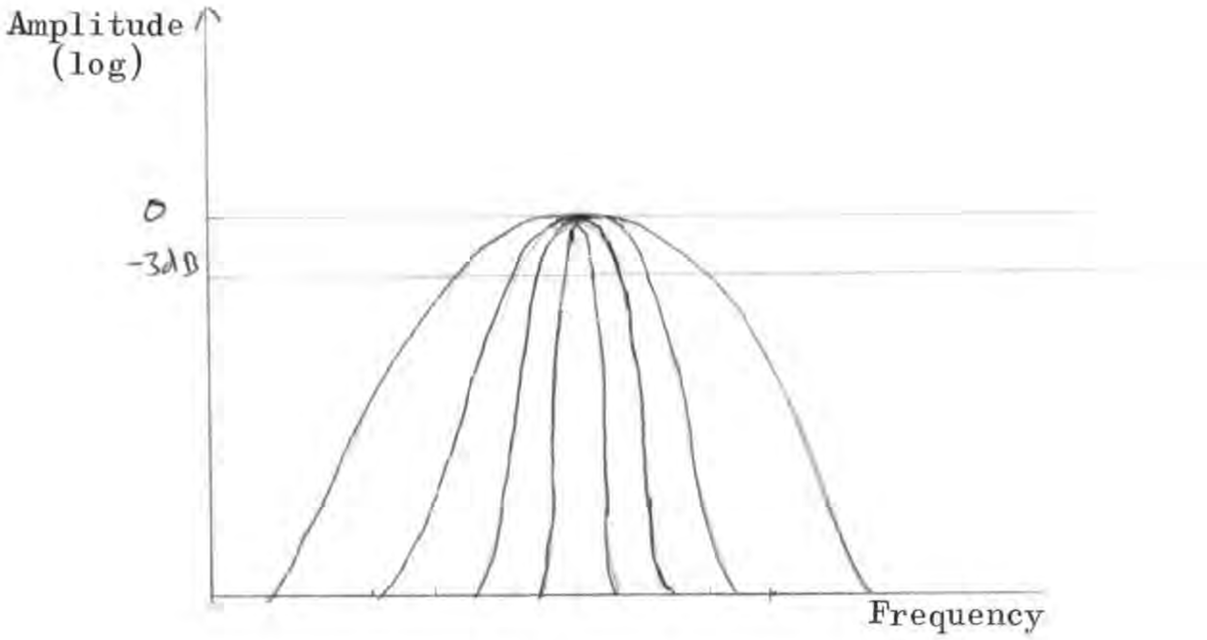
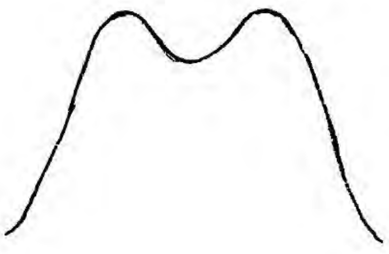
[74] Q = Centre frequency/3 dB band width. In electronic terms the 'Q' of a resonant circuit is also defined as the ratio of inductive reactance: WL, to the circuit resistance; R, i.e. Q = WL/R.



At higher Q values, filters impart a 'ringing' quality to the sounds which are allowed through the pass band. This is due to irregularities in the pass band itself and also the sharpness of the response characteristic itself. Some designs, for example, produce a ripple in the pass band, others produce more pronounced secondary peaks.

i)

ii)



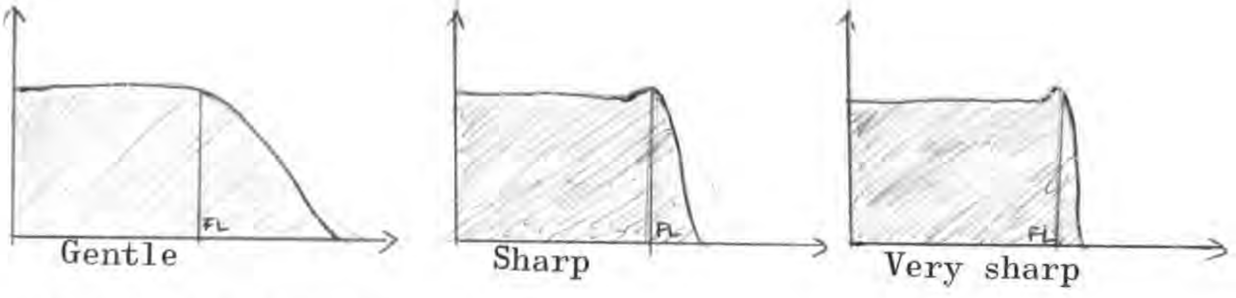
A more pronounced 'ringing' characteristic, accompanied by a greatly increased overall gain may be produced at even quite low 'Q' values by the careful application of positive feedback to the filter unit. This technique, although musically useful in certain situations, is nevertheless difficult to control, for even a small over-stimulation of the feedback loop will cause the filter to resonate as a sine wave oscillator.

A distinction should be made here between two fundamentally different types of filter; those which are passive, and those which are active. Passive filters consist entirely of resistive/capacitive networks powered only by the signals which are applied to them for processing. Active filters, on the other hand, require their own special power sources since they are essentially special types of amplifiers. In the early days of electronic music systems the former type predominated; today, largely due to the development of the transistor and the integrated circuit, the latter type is far more common. Active units offer one particular advantage to the musical user, for they may be designed with internal amplifiers for regulating their overall gain, permitting signals not only to be spectrally shaped but also regulated dynamically.

One of the most useful arrangements of band-pass filter circuits in a studio is in the form of a bank of units connected in parallel, each tuned to a particular centre frequency, covering and dividing the audio spectrum into fixed band widths, typically third, half or single octaves. The tunings for the filters are derived from a suitable reference frequency. An octave filter bank, for example, might commence with a filter tuned to 55 Hertz (= ), forming the basis for an ascending series of units tuned to centre frequencies of 110, 220, 440, 880, 1760, 3520, 7040 and 14080 Hertz respectively and each covering a half octave band width either side of these settings. Each filter is equipped with a gain control permitting the user to select and vary the strengths of the spectral bands derived from a suitable source signal.

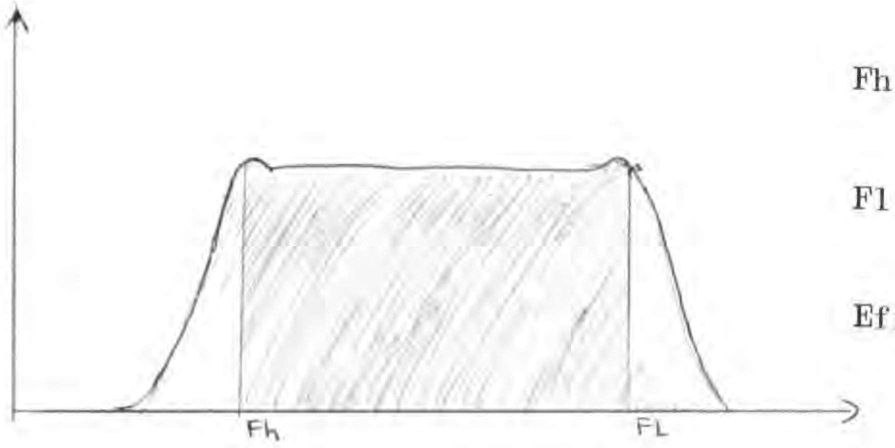
Filter banks, by their very nature, cannot offer all the variable characteristics with which a musician may wish to experiment, for the need to divide the frequency spectrum into rational proportions demands that the centre frequency and response of each unit should be fixed at the design stage. Single band-pass filters offering discrete control over centre frequency, response and overall gain are useful studio devices, and it is common to find both these designs available in a comprehensively equipped system.

High and low-pass filters are useful for the removal of varying proportions of the entire frequency spectrum. A low pass filter, for example, may be adjusted upwards to remove just the highest frequencies of a signal, or downwards to remove all but the lowest frequencies. Varying the response curve affects the pass area of the spectrum only marginally near the frequency of cut-off, but again a very sharp characteristic will result in a 'ringing' quality which may or may not be desirable, depending on the context in which the unit is applied.



FL = frequency of cut-off

High and low-pass filters may be connected in series to provide a particularly flexible form of band pass filter.



Fh = High pass filter cut-off frequency

Fl = Low pass filter cut-off frequency

Effective centre frequency =

$$\frac{Fh + Fl}{2}$$

Since each part of the pass band is controlled by a different circuit, the effective band width may be varied over wide limits independent of the rates of attenuation either side. If Fl is made smaller than Fh all frequencies become completely blocked. From a studio point of view it is thus useful for some high and low pass filters to be wired in pairs so that the above type of band pass shaping may be easily effected.

According to Enkel[75] the Cologne studio was initially equipped with a switched high pass/low pass/band pass filter of variable centre frequency and 'Q', (which had originally been assembled as a radio-drama limiting device) and an eight-element octave filter bank offering centre frequencies of 75, 150, 300, 600, 1200, 2400, 4800 and 9600 Hertz respectively. Adam's description of suitable electronic filter designs includes a discussion regarding the limitations of filter banks which divide up the frequency spectrum in terms of musical pitch intervals. Such an arrangement is non-linear, for the ear detects musical intervals as proportional increases in frequency. It may be seen from the specification quoted above that the frequency settings for an octave filter bank involve a doubling of the pass-band width and a progressive increase in the distance between centre

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75 <u>Technische</u> ..., <u>op.cit.</u>, paper 3 (TT 603), p.7

frequencies, as the spectrum is ascended. If a composer wishes to isolate individual harmonic partials in a selected musical sound with an identifiable fundamental pitch, a bank of band-pass filters would be required, tuned to the natural overtone series of the note. The frequencies of harmonics derived from a selected fundamental, however, are in simple multiples, creating a linear series. The first eight harmonics of a 100 Hertz fundamental, for example, are 100, 200, 300, 400, 500, 600, 700 and 800 Hertz respectively. If each harmonic of this series is isolated and perceived separately the progression would appear to be an ascending sequence of pitches following an interval pattern which becomes progressively smaller. Even a semitone filter bank is thus incapable of isolating higher order harmonics. Adams[76] notes that

> The channel means for separating individual harmonic overtones is determined by the necessity of having a filter with constant absolute band width. ... If it were desired to carry out this method exactly and to separate the 25th and 26th harmonic, for example, of a fundamental in the range of 100 to 200 c.p.s., it would be necessary to have an uneconomically large number of filter channels. In any case, it is musically more interesting to be able to separate the lower harmonics individually, while taking the higher ones together. A compromise solution would therefore be to have a channel scheme in which absolute band widths were provided with the band range position rising by groups ... connecting several channels simultaneously in parallel.

He continues by suggesting the following arrangement:

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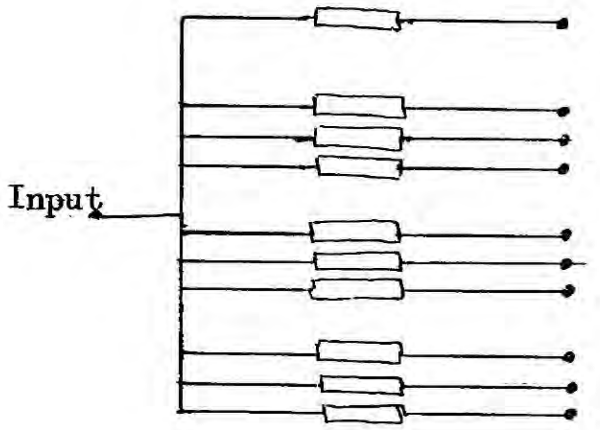
[76] <u>Technische</u> ..., <u>op.cit.</u>, paper 5 (TT 605), p.12



I 1 Filter, 3 channel positions - 25-100 Hertz, Band width = 25 Hz

II 3 Filters, 12 channel positions - 100-1000 Hz, Band width = 75 Hz

III 3 Filters, 13 channel positions - 1000-5000 Hz, Band width = 307 Hz

IV 3 Filters, 10 channel positions - 5000-15000 Hz, Band width = 1000 Hz

Plan for a partials filter bank consisting of 10 parallel separately variable band-pass filters with grouped constant absolute band width in 38 channels.

Such an arrangement would indeed isolate harmonics more selectively than a standard third-octave unit. Studios have nevertheless shown little interest in such designs over the years. The main drawback to a partials filter bank is its inflexibility when it is employed to provide similar spectral shapings to a sequence of different pitches. The fixed centre frequencies of the individual filters is a restricting characteristic in such circumstances, for it becomes inconvenient if not impracticable to reset the bank accurately for each note. Single band-pass filters of variable 'Q', capable of adjusting automatically in centre frequency in relation to the pitch of the selected source, have proved far more versatile in this respect. This latter technique of timbre shaping will be considered further in connection with voltage-controlled processing in the next chapter.

The generation, organisation and modification of electronic sounds involves the structuring of acoustical information as a progression of events over a period of time. As has been shown in the earlier examination of

Schaeffer's work, this linear ordering of material is a characteristic of primary importance in the study of the nature of any sound event. In the context of composition with electronic sound, interest thus centres on the facilities available to aid the dynamic shaping of events as a function of time.

The most basic technique available in every studio involves the manual control of sound intensity by means of a sliding or rotating attenuator, sometimes referred to as a volume control or a potentiometer. In a simple application such a device might be employed to vary the strength of a single sound source such as an oscillator. The processes of composition in a studio, however, involve the superimposition of many such operations, and it is quite common to discover that even a relatively short pattern of events has involved many such attenuation processes. Manually controlled filters are effectively attenuators of a special type; varying the strengths of signals as a function of frequency, according to the response and frequency setting applied.

Over the years electronic techniques for the production of attenuation functions have been introduced to augment the processes of manual control. The envelope shaper, for example, is an electronic device which has been developed to provide an automated means for controlling the growth and decay of musical events. Such innovations, however, have largely been associated with the advent of voltage-controlled systems during the 1960s, a subject to be studied in detail in the next chapter. The more basic technological facilities of the early Cologne studio demanded the development of alternative methods for controlling attenuation. Enkel outlines two such facilities as items eight and nine in the studio description; one concerned with the provision of rhythmic patterns, and the other with the more general shaping of dynamics.

> Rhythm may easily be imposed on the musical sound structures by means of ring modulation and tape loops.
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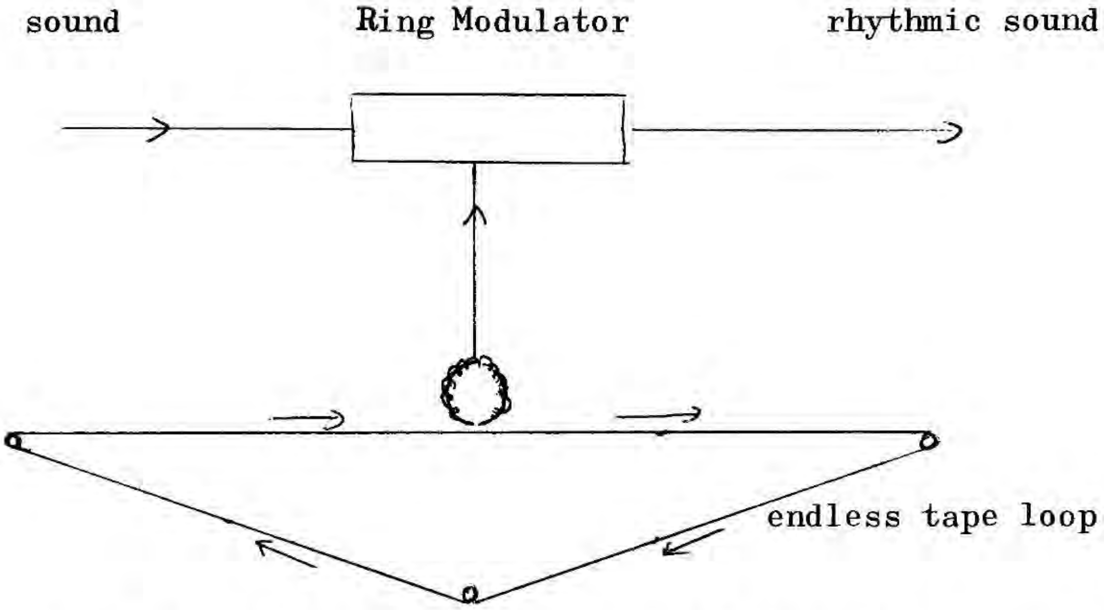
> 
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> This shows how the sequence of sounds and the audio frequency voltages picked up from a scanned tape loop are fed to the input terminals of the modulator. The ring modulator only passes the sound when carrier and signal frequency are applied simultaneously ... if a series of pulses is applied to the tape loop in accordance with the desired rhythmic structure, the modulator will be blocked against the musical sound whenever the tape is free of impulses.[77]

If these impulses lie within the audio spectrum, the output from the system will be 'gated' bursts of sound in the form of sum and difference products generated from the two inputs. If, however, the impulses are in the form of high frequency bursts, well above the audio range, the modulator inputs are extremely unbalanced with respect to frequency. Under such

[77] Technische ..., op.cit., paper 3 (TT 603), pp.7-8

extreme operating conditions not only will the sum and difference products lie outside the audio spectrum but also the modulator will tend to 'leak' the unmodulated inputs, albeit at a reduced strength, whenever both are present. This effect may usefully be harnessed to provide a 'gate' for the audio frequency information without any perceivable by-products.[78]

The Cologne studio used a pulse burst of 30 Kilohertz for this purpose, within the frequency range which could be accurately recorded and reproduced by their tape recorders, but still just high enough for the difference products derived from all but the very highest source frequency elements to be outside the audio range. The speed of a recorded pulse pattern could be varied by altering the speed of the control tape. Unfortunately, such an operation also changes the frequency of the pulse carrier itself, and this severely limits the range over which the technique may be used without undesirable side-effects. This problem may be partly overcome by selecting a different carrier frequency for the original recording, which, on playback at the desired speed, would be reproduced at or near 30 KHz.

The Cologne studio included another system of pulse control which permitted the timbre of a sound source to be varied rhythmically via the

78 Dynamic control of an audio frequency source via a ring modulator may also be achieved by using a slowly changing voltage function, derived perhaps from a sub-audio oscillator acting as the second input. Under these conditions the applied modulation characteristic is one of period rather than frequency. The effect is thus an enveloping of the frequency source with attack and decay functions determined by the characteristics of the applied wave. True 'gating', however, cannot be achieved by using this technique, for this would require a rapid switching between dynamic levels rather than a progressive change. If a sub-audio square wave was applied as a controlling input, for example, the resultant output would be just a series of clicks, for the ring modulator will only respond to voltages which are dynamically changing.

octave filter bank. All the filter units were fitted with 'make-or-break' contacts in their output lines, permitting the operator to connect or disconnect each element by means of a series of relays. The latter could either be activated manually from the main studio console or electronically from a control tape consisting of pulses recorded at different frequencies.

> For this purpose the ... keys are bridged by switching circuits operated at audio frequency. There are eight of these switching circuits present, each operating an octave filter. ... With this arrangement selection can be obtained over a band width of 4000 c.p.s., so that eight switches can be released at frequencies between 1000 and 5000 c.p.s. These frequencies are recorded on a tape loop in accordance with the rhythmic requirements, and on reproduction they control the selective switches.[79]

The switching circuits themselves consist of a special bank of narrow (that is, high 'Q') response band pass filters each connected to a level detector, which in turn activates a corresponding relay. The 'gating' functions for both the filter bank and the ring modulator could be made non-repetitive by replacing the tape loops by a conventional reel to reel recording system, enabling the construction of control sequences of any desired length.

The technique employed for shaping general dynamic levels offered a range of functions which were in many respects far more flexible than those associated with modern voltage-controlled envelope shapers, for the latter all too frequently offer a very limited range of attack and decay characteristics. The basis of the system consisted of a photo-resistor and a source

[79] <u>Technische</u> ..., <u>op.cit.</u>, paper 3 (TT 603), p.8. They do not control the discrete settings of the attenuators, however, which are still set independently by hand.

lamp positioned either side of a film strip which is transported at a steady speed. On this 'white[80] film ... the required variation of the dynamics of any desired balancing processes can be recorded in the form of varying greyness values with the aid of a quick-drying varnish.'[81] The varying density of the film controls the quantity of light falling upon the photo-resistor, and thus the voltage developed across its output. This function in turn is used to control an electronic attenuation circuit which may be inserted at any desired point in the processing chain.

The use of variable density film for the purposes of sound recording has already been discussed in the previous chapter,[82] and it is interesting to note that the Cologne studio only adopted this technique for a single control application. The reasons why optical sound generation and processing methods were not explored more generally are not clear. It might be noted, however, that the potential versatility of such an approach was possibly outweighed by the desire to develop electronic sound processing techniques which were based on the principles of precision and total control; features not associated with optical sound processing technology at that time. The existence of a facility to 'draw' dynamic characteristics of any desired contour nevertheless created a control characteristic which was not the product of a mathematically based function, indicating that the need to provide an element of free creativity in the production of basic electronic sound material was acknowledged, even at the very outset of the venture.

[80] i.e. transparent

[81] <u>Technische</u> ..., <u>op.cit</u>., paper 3 (TT 603), pp.9-10

[82] Page 25.

The tenth item in the studio plan concerns the provision of facilities for the addition of resonance to an electronic sound to enhance its quality. Enkel makes a distinction between three different processes of resonance.

> If the difference in time between the direct and indirect sound is less than 50 millisec. the resonance merely results in an increase in volume. However, if the reverberation occurs later, up to 100 millisec., then a blurring of the original sound becomes evident. All reflections which return after a time greater than 100 millisec. are separated from the original and heard as an echo.[83]

The production of reverberation and echo involves complicated acoustical phenomena generated by systems which frequently offer only very imprecise methods of control over their functional characteristics. Both forms of sound enhancement involve a single or multiple reiteration of material which may be achieved by a variety of acoustical, mechanical and electronic methods. The distinction between the two classifications may be explained as follows: the human ear and brain exhibit a response characteristic which limits the number of individual events they may detect over any given time interval. As Enkel observes, in this process of reiteration the boundary of definition is reached when the interval between events is shortened to about one tenth of a second, which is a time span approximately equal to the persistence factor of our aural response. Any further compression of events causes new information to be received before the previous image has been forgotten, resulting in an overlapping which will be interpreted as an apparently continuous process of sound enhancement.

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[83] *Technische ...*, *op.cit.*, paper 3 (TT 603), p.10

The borderline between these two states of perception is nevertheless rather imprecise, for our aural response will alter significantly according to the nature of the sound events themselves. If, for example, the source sounds are low in pitch, lacking in higher harmonics and free from any sharp attack and decay characteristics, repetitions spaced considerably more than a tenth of a second apart may be sufficiently blurred to appear reverberant. Conversely, the ear can detect details of certain transient phenomena which themselves change significantly in character within a tenth of a second.

The nature of sound enhancement is influenced by several other important factors, in particular the number, the rate of decay, and the acoustical quality of the repetitions. A further variable is created if the system permits feedback of the output sound into the repetition network to create a self-regenerating process. Most studio systems provide a degree of control over some of these parameters depending on the technology involved. The acoustical nature of the repetitions, however, is one quantity which is determined by frequently rigid characteristics fundamental to the design of the particular echo/reverberation device, and it is the limitations associated with the latter which have accounted for the proliferation of techniques for sound enhancement over the years.

Initially, no specially designed reverberation devices were available to composers using the Cologne studio. Instead they relied upon natural reverberation produced by means of a loudspeaker and microphone suitably positioned in an empty room, so that information projected by the former would reach the latter both directly, and also indirectly by means of reflections off the surfaces of the walls, ceiling and floor. Source material could thus be sent to the reverberation room, modified and returned to the studio either to be blended with the original signal or used on its own. There is much to commend this technique even today, for since the system is based on natural acoustic response, it is free from any major undesirable characteristics. The main disadvantage lies in the inability to alter the reverberation time conveniently, for to change the room's response the absorption characteristics of the reflecting surfaces would have to be altered, for example, by removing or adding cushions.

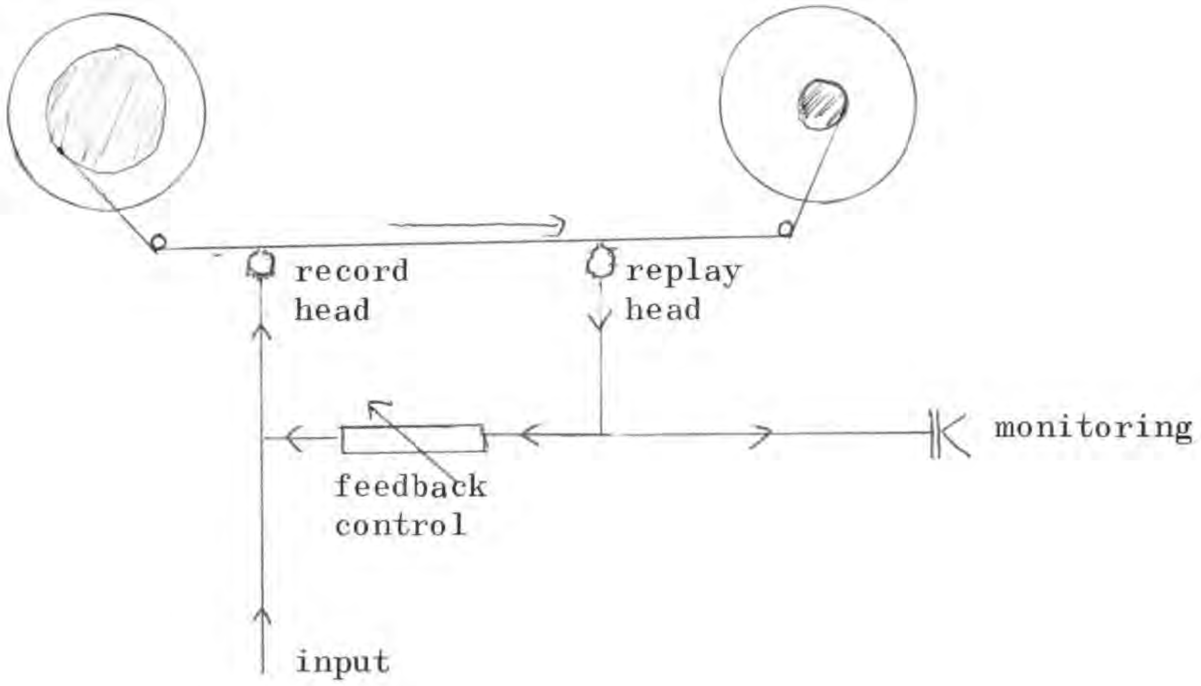
Shortly after the publication of Enkel's technical description, the Cologne studio acquired E.M.T. plate reverberation units, offering a decay time characteristic which could be easily varied from about one to four seconds in duration. The operational convenience and flexibility of such enhancement devices is not, however, without its drawbacks, for they always add an artificial element to the sound quality. This is particularly true of spring reverberation units which will frequently generate undesirable mechanical pulsing effects, derived from the natural bounce of the spring itself.

Both types of reverberation unit are based upon the same electromechanical principle, the conversion of electrical impulses into mechanical vibrations which are transmitted relatively slowly either along a coiled spring or across a metal plate. On reaching the other end of the spring, or the edges of the plate, the waves are reflected, setting up a complex oscillating pattern which slowly fades. These patterns are detected by special contact microphones which convert the information back into electrical impulses. The decay time of a plate may be altered by applying a damping pad. Spring systems may also be damped to a certain extent, but it proves extremely difficult to modify their vibrational response without introducing further spurious characteristics.

Echo facilities were provided by a tape delay system:

> The process depends on the time interval between recording and reproduction, which depends on the distance between the recording head and the pick-up head. The reproduction attenuator is connected via a control to the recording amplifier so that the reproduction voltage is recorded after a certain interval.[84]

This technique may best be illustrated with a diagram:



Providing the feedback control is set to give less than unity gain a series of echoes will be produced through successive recordings, the sound level each time decaying logarithmically according to the percentage of the replay level returned to the record head. (If the feedback control produces a gain greater than unity the result is a positive sound level gain which will rapidly escalate with increasing distortion until the system

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[84] Technische ..., op.cit., paper 3 (TT 603), p.10

becomes fully saturated.) On most standard tape recorders the positioning of the record and replay heads is permanently fixed, thus the only method of varying the echo delay is to alter the speed of the tape. If this characteristic is continuously variable, rather than switchable between standard settings, a discrete range of delays may be produced. Enkel fails to observe that at higher tape speeds the delay time may become shortened to less than 100 milliseconds, creating a means for producing a form of reverberation rather than echo.[85]

Such a technique might appear to be the most versatile of the general enhancement methods so far discussed. It must be realised, however, that the process is entirely electronic, where, allowing for the progressive degradation in quality resulting from successive re-recordings, the images are exact reflections of their sources occurring at regular time intervals. Colouring may be introduced by the use of filter networks in the feedback chain, but the results, when compared with room reverberation are still inevitably artificial by nature, leading all too easily to characteristics which may become undesirable clichés, unless handled intelligently. On a more positive note, however, it may be observed that the technique of tape delay and regeneration, when treated as an electronic processing facility rather than a general method of sound enhancement, has been profitably used by many composers over the years. In particular, the use of relatively long gaps between record and replay functions by passing the tape between two or more separate recorders has become much favoured in many studios as a method

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85 If, for example, the tape is run at 30 inches per second and the record and replay heads are 2 inches apart, the delay time is $\frac{1}{15}$ th of a second, approximately 67 milliseconds.

for reiterating, freezing and progressively transforming extracts of source material.

Enkel concludes his study of the main studio system with a description of the provisions for recording: items eleven and twelve on the studio plan. Only two separate taping facilities were available initially: a single-track recorder, and a four-channel system created by connecting the capstans of two two-track tape recorders to a common drive system. These machines were originally designed for use with film and accordingly employed a sprocketed tape drive, permitting perfect synchronisation between the two pairs of channels.

He notes that 'Using a Magnetophon with four tracks the recording of several layers independently became possible. Each of the four layers can be altered or erased without affecting the others. ... The four-track magnetic tape apparatus can be manipulated so that the material on three recorded tracks is transferred to the fourth, the three tracks then being erased can have new material recorded on them, so that in this way any number of layers can be combined.'[86]

No mention is made in this section of the special A.E.G. variable speed tape recorder described earlier[87] in connection with the processes of frequency band compression and expansion, and it would appear that a distinction was intended between the procedures of sound generation and transformation, and the preparation of a piece from recordings of the latter, mixed together according to a realisation plan. The greater compositional freedom afforded

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[86] Technische ..., op.cit., paper 3 (TT 603), p.11

[87] See page 110

in later years led to a less rigid definition of functions, for it became clear that tape recording facilities should be considered a fully integrated part of the realisation system itself, available for use at any stage of material preparation. The information fed to a ring modulator, for example, need not necessarily be derived directly from a pair of audio generators, for it may prove more appropriate to prepare a sequence of events for one or both inputs on tape first.

Enkel became concerned over the presentation aspect of elektronische Musik at a very early stage in the development of the studio, and he not only provided flexible loudspeaker facilities within the electronic studio itself but also persuaded the broadcasting authorities to provide multi-channel loudspeaker systems for concert hall recitals in the main recording studio of the Cologne radio station, despite the prevailing broadcast restrictions to monophonic transmission.

During the early years three-channel sound projection was employed, using eighteen loudspeakers spaced 'along the sides and ends of the studio ... divided into three separate groups, each being connected to one reproducing channel of the Magnetophon '.[88]

Three tracks of the four-track tape recorder were thus allocated to provide discrete information for a specific group of loudspeakers. The fourth track, if so desired, could be superimposed on any of the main output channels via a set of three frequency-controlled switches and an associated prepared control tape loop or reel to reel recording. In a recital the latter facility could be employed to preserve some degree of active realisation, though it is

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[88] Technische ..., op.cit., paper 3 (TT 603), pp.11-12

to be assumed that the doctrines of post-Webernian serialism demanded total control over its function, unlike Poullin's Potentiometer d'Espace[89] which influenced the location of sound events entirely in response to physical human movements which are not only imprecise by their nature but might also be deliberately improvisatory.

The use of sound in space was to become a major compositional preoccupation during the middle and late 1950s. Pierre Boulez, writing in the first volume of Die Reihe[90] noted that

> A final obstacle linked with 'interpretation' is the continuity of projection of a work in space. ... We are here faced with definite limitations; the attraction of an 'objective' work is speedily dissolved, for psychological reactions of an audience to which the music is fed by loudspeakers can hardly be avoided where the audience is deprived of the possibility of associating a sound with a gesture. Thus the arrangement in space becomes a structural necessity and represents considerably more than an appropriate setting for a more or less spectacular exhibition.

It is thus rather surprising that none of the early pieces realised in the Cologne studio has been retained in a multi-track form, for according to the studio catalogue[91] the first non-monophonic tape listed is Karlheinz Stockhausen's Gesange der Jünglinge, composed 1955-56, available in a four-channel version, mixed down from an original five-track master tape.

[89] See page 62

[90] Pierre Boulez, 'At the ends of fruitful land ...', Die Reihe, Vol.1 (German edition; Universal Edition, Vienna, 1955; English translation, Theodore Presser Co., Pennsylvania, 1959), pp.20-21

[91] Listed in Hugh Davies, International Electronic Music Catalog (M.I.T. Press, Cambridge Massachussetts and London, 1968), p.49

The Cologne Studio für elektronische Musik established the first comprehensive system for the generation and transformation of electronic sounds, attracting an influential group of composers, several of whom in their turn moulded the development of facilities both at Cologne and elsewhere in Europe during the middle and late 1950s, and the first part of the 1960s, when the introduction of commercial synthesisers led to a rapid diffusion of activities and interests. Herman Heiss set up a studio at Darmstadt in 1957, Henri Pousseur was appointed director of the APELAC studio, set up in Brussels during 1958, moving later to develop facilities at Ghent, and Gottfried Michael Koenig, after a long period working at Cologne, moved to become director of the studio at the Institute of Sonology, Utrecht State University. The influence of Cologne is also clearly apparent in the design of the Milan studio established in 1955 under the artistic direction of Luciano Berio by the engineer Dr. Alfredo Lietti, and this will be studied shortly.

The Cologne studio was not just concerned with the production of elektronische Musik, for the advancement of the medium depended on the development of communications between composers and sound synthesis systems and this could only be achieved by investigating the very nature of sound itself. Indeed it was a deep preoccupation with this latter subject which had inspired Dr. Werner Meyer-Eppler to lay the foundations of the studio itself, and he continued to be an active member of the team until his premature death in 1960 at the age of fifty-seven. In his article 'The Mathematic-Acoustical Fundamentals of Electrical Sound Composition', Meyer-Eppler examines the relationships between the physical, psychological and physiological features of electronically generated sounds. In any analytical investigation it is necessary to define the terminology which is employed, and in this highly interdisciplinary context there are many opportunities for conflict and confusion.

The American Standards Association[92] and the German Committee on Acoustics,[93] for example, prepared two sets of definitions which are not precisely complementary, and Meyer Eppler's article commences by a comparison of these:[94]

The German Committee on Acoustics recommends the following definitions:

Simple tone	sound of sinusoidal curve form
Tone mixture	sound composed of tones of arbitrary frequency
Simple musical sound	sound composed of harmonic partial overtones
Musical sound mixture	sound composed of musical sounds with fundamental tones of any frequency
Noise	tone mixture to which a continuous spectrum corresponds, or which is composed of a great number of individual tones whose frequencies are not related to each other as integral numbers
Report	sound impulse, chiefly of great sound intensity

Of course, this list seems strange to a musician since in places it is contradictory to his own idiom. The explanations contain the following sentence: 'It is a known fact that the definition of the terms "tone" and "musical sound" in acoustics differs from those in music. The musician denotes a simple musical sound as "tone" but a musical-sound mixture as "musical sound" (e.g. a triad) ...'

[92] Acoustical Terminology, New York, 1951

[93] DIN 1320

[94] <u>Technische</u> ..., <u>op.cit.</u>, paper 8 (TT 608), pp.3-4

The American Standards Association attempted to supplement the physical definitions by defining the qualities of sensation.

Tone	A tone is a sound sensation having pitch
Simple tone (pure tone)	A simple tone is a sound sensation characterised by singleness of pitch
Complex tone	A complex tone is a sound sensation characterised by more than one tone

Surprisingly enough, the American terminology does not contain a definition of what might be denoted as 'noise sensation'. But a term for timbre which is absent in the German list has been included:

Timbre (musical quality)	Timbre is that attribute of auditory sensation in terms of which a listener can judge that two sounds similarly presented and having the same loudness and pitch are dissimilar

Eimert presents his own set of terms for describing electronic sound in his <u>Die Reihe</u> article[95] entitled 'What is electronic music?' using the descriptions 'tone', 'mixture' and 'complex' for different purposes to those outlined above:

The <u>Tone</u>	is unknown to traditional music, is without overtones, is pure or sinusoidal: all sound phenomena may be reduced to it. No tonal system in the traditional sense may be constructed of sinus tones: they have no traditional place of a system, no tonal 'character'. Thus the sinusoidal tone system can only be a theoretical system of reference; the composer may build structures out of this system by serial organisation.

[95] Herbert Eimert, 'What is electronic music?', <u>Die Reihe</u>, Vol.1, p.4

The *note*	is what every musician knows as a tone. It is built up from a series of harmonic overtones (partials, sinus frequencies). Thus, the 'tone' of an instrument is not the tone but the *note* which is immutable in its components, which determine its timbre. These partial components may only be varied by electronic means.
In the *note mixture*	the frequencies of the partials are not ordered harmonically, i.e. they cannot be expressed in terms of simple numerical proportions. Note mixtures are always sinus tone mixtures and are not the same as 'chords'; they have a higher degree of internal fusion of components and can be regarded as units more similar in category to the single note than to the instrumental chord ...
Noise	defined by specific sound character and approximate 'pitch level'. Only 'blank noise' which fills an acoustic region may be determined in position. Filtered parts of 'blank noise' are called 'coloured noise' or 'noise colour'.
The *chord* (*note complexe*)	is identical acoustically and traditionally. It must be observed that the note and the chord are clearly differentiated in instrumental music; in electronic music the note mixture intervenes between the two with its particular levels of fusion of its constituent parts. Note and tone mixtures are electronically 'composed' not according to a harmonic or natural system but according to a composer's predetermined ordering.
Impulse or *pulsation*	also known as *beats* or *clicks* (regular or statistic) at high dynamic levels corresponds to 'detonation'.

This volume of *Die Reihe* was published in 1955, a year after Meyer Eppler's article and Eimert's own discourse on 'The Place of Electronic Music in the Musical Situation'. In the latter a definition of terms is given for the description of the sounds encountered in *elektronische Musik* which is close to the acoustical definitions of the American and German Committees on Acoustics:[96]

> Electronic acoustical phenomena can be classified as simple tones, complex tones with harmonic partials, tone mixtures, noises and intervals.
>
> 1 Simple tones, in this case, refer to the pure or sinusoidal tone, without overtones, which is not found in traditional music, but which is the basis of all musical sound processes. ... The musical discovery of the sinusoidal tone terminates the historical analytical development which began with twelve-tone music and passed through the stage of Anton Webern's isolated tone to the tone as distinct from the overtones. Sinusoidal tones cannot be used to build up a system of tones in the traditional sense. They have no place in the traditional scheme of things, no 'tonal' character. The sinusoidal tone system can thus be no more than an imagined system of reference from which the composer builds structures in the form of sequences, ratios, series and classes.
>
> 2 The complex tone with harmonic partials is composed of a succession of harmonic partials (sinusoidal oscillations). What we call the 'tone' of an instrument is not a tone at all, but a sound which is unequivocally determined by the components which contribute to timbre. Electronics now, for the first time, makes these components variable.
>
> 3 In the simple tone mixture the frequency of the partials are not harmonically related to the fundamentals; they cannot be expressed in terms of integral ratios. However, mixtures are still mixtures of sinusoidal tones and hence not to be confused with chords ...
>
> 4 Noises are determined by their specific character and their pitch. The pitches of 'coloured' noises, musically speaking, have the value of approximate positional relationships. Only the so-called 'white noise' extending over the entire auditory range, cannot be determined positionally.
>
> 5 Two different sounds given simultaneously produce an interval (more than two sounds; a chord). In instrumental music complex tone and interval are clearly distinguished from one another. In electronic music, however, the tone mixture with its new binding levels forms a bridge between the two ...

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96 Technische ..., op.cit., paper 10 (TT 610), pp.9-11

The differences between these sets of definitions illustrate the problems which arise when employing terms which have different meanings in two or more disciplines as the basis for the study of functions which cross their boundaries. In the light of the above examples, for instance, a composer would have to accept new meanings of the ideas of 'tone', 'tonal' and 'tonality', which if not fully understood could lead to serious breakdowns in communications between the artist and the physicist.

Meyer-Eppler avoided the temptation merely to present his own set of definitions based on borrowed terms, presenting instead a careful study of the connections between the ear's response characteristics and the physical nature of sounds, objectively analysed in terms of their parameters of frequency, intensity and duration. His treatise commences with a firm warning concerning the difficulties encountered in the application of objective scientific analyses of acoustical parameters to the subjective world of aural perception.[97]

> The human ear does not uniformly respond to all acoustical stimuli ... while the realisation of psychological optics that 'colours' are not properties of objects but exclusively qualities of sensation has become general knowledge, little attention was hitherto paid to the respective relationships in the fields of acoustics. With respect to the definitions alone there is widespread lack of competence in making distinctions between the physical acoustic phenomena (and the abstracted representation in the form of written music) and the qualities of acoustical sensation. The pattern presented by notes and sound sensation are often too closely correlated.

It is thus important to qualify any mathematical analysis of a progression of sound events with a study of the associated aural response to such

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97 Technische ..., op.cit., paper 8 (TT 608), p.4

information. As illustrations of this factor Meyer-Eppler points out that a diminished fifth played by two horns in mid-register sounds subjectively quite different to the same interval played two octaves higher on a xylophone, although the relationship between the intervals of the fundamental pitches is functionally the same. Further, the ear's response to a particular sound event is not absolute, for it is influenced by the events that precede, and to a certain extent by the events which follow. In terms of relative amplitude alone, for example, a fortissimo following a piano appears significantly louder than the same fortissimo following a forte. Factors such as these are of primary importance both in the composition of electronic music, where great freedom is afforded over the choice of acoustical material, and also in the design of new systems and new methods of sound specification. Without a working knowledge of the characteristics of aural perception, the designers of sound generation systems run the risk of overlooking some important compositional considerations, which in the sphere of natural instrumental sounds are instinctively understood by the creative musician.

In studying the mathematical methods available of describing the nature of sound events Meyer-Eppler first examines the usefulness of harmonic analysis:-[98]

> The most elementary and best known classification method which is almost exclusively applied in the literature of musical science. In terms of mathematics this means [the] resolving of periodic acoustic pressure distributions into sinusoidal components ('harmonics') whose frequencies are related to one another as integral numbers. The term 'periodic' implies ... that the oscillation phenomenon is completely uniform and has neither a beginning nor an end. Harmonic analysis can only be used as an approximate method of classifying sound phenomena, since it does not explicitly contain the time as the most essential

[98] Technische ..., op.cit., paper 8 (TT 608), p.8

> element of all sound phenomena. This important restriction is frequently overlooked and, as an unavoidable result, the data thus obtained do not show agreement with the psycho-acoustical observations.

The limitations of sound analyses which take no account of the progression of information with respect to time had become clear to Pierre Schaeffer at a very early stage in his study of musique concrète.[99] The identification of his objets sonores involved the isolation of particular events within the time continuum, yet even these extracts were associated with a finite time interval, and the changes in their structure occurring during this period formed an important part of the process of analysis. This accounted for his adoption of three dimensional plans de référence displaying the relationships between the parameters of pitch, intensity and time over the course of the event.

Meyer-Eppler also recognised the necessity of a three-dimensional approach to the study of the nature of sounds:[100]

> Very good agreement between the result of resolving and auditory sensations is obtained from the time-frequency analysis in which the transformation from the Fourier series to the Fourier integral is carried out. If it were possible to express the result of a harmonic analysis by a number of discrete spectral amplitudes independent of time (line spectrum), then the time-frequency spectrum to be calculated by means of the Fourier integral is continuous and variable with time as well. Since time, frequency and spectral amplitudes are independent co-ordinates, two dimensional symbolisation is not possible here, as opposed to the line spectrum of the harmonic resolution. Either a method of perspective representation must be used or the third co-ordinate must be expressed by optical characteristics (density graduations).

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[99] See page 48

[100] Technische ..., op.cit., paper 8 (TT 608), p.8

He continues his treatise with an important observation regarding the representation of non-continuous sounds within a time-frequency spectrum:

> In the deduction of the time-frequency spectrum (by calculation or determination by apparatus) another very essential fact must be taken into account. While only one type of harmonic analysis is admissible for a periodic sound process, any non-periodic process may be represented by a time-frequency spectrum in infinitely many ways. This is due to a freely selectable parameter, which is designated as the 'analysis interval' of spectral analysis. The smaller the analysis interval the finer will be the subdivision of the spectrum with respect to time, and at the same time the rougher will be the spectral analysis.

The dissection of continuous streams of acoustic information into a sequence of time-sliced sections is a function encountered in the processes of digital sound analysis and these will be discussed in a later chapter. In this particular application a high degree of resolution is required within each sample, and the time span is accordingly very short. This results in an approximation nearing to an analysis of the integrated function with respect to time, such as would be displayed on an ordinary oscilloscope.

A wider analysis interval, although supplying a less accurate indication of the location of a sample with respect to time, provides a more comprehensive picture of the spectrum characteristics, which in terms of a graphical model would be more informative. The extreme case of an infinitely wide analysis interval, however, removes the time location characteristic completely and merely presents a harmonic analysis as outlined above.

A restricting factor in the correlation of time-frequency analyses to their perceived characteristics is the response time of the human ear itself, for this imposes a limitation on the minimum duration of a sound sample which may be coherently interpreted. Schaeffer, by deriving his objets sonores in terms of acoustic response characteristics had avoided the construction of analytical models which isolated features clearly beyond the boundaries of perception. Meyer-Eppler also took account of this factor when outlining his more mathematical approach to the problems of sound analysis:[101]

> From numerous investigations it is well known that the order of sound sensations following one another with a period of less than 25 milliseconds is no longer perceived. Whether at an interval length of 25 milliseconds an observer can actually hear all the spectral details that may be determined mathematically depends to a high degree on his acoustic-gnostic capabilities, the direction of his attention and his familiarity with the sound being presented. Therefore, as far as the average listener is concerned, the structure of a sonogram with respect to frequency should be considered an upper limit of the analytical resolution. It would be useless to go beyond this limit.
>
> Since the perception volumes due to a given acoustic pressure distribution are centralised phenomena, they cannot be read from mathematical spectral representations at all. Nevertheless, the opinion of P. Schaeffer that time intervals of less than 0.1 sec. provide the sound material seems justifiable, although sound forms only occur in longer time intervals.

The last statement would also seem to be in agreement with Enkel's observation[102] regarding the speed of event repetition which marks the boundary between the phenomena of echo and reverberation, thus analyses of sound events based on analysis intervals of less than 0.1 seconds (100 milliseconds) will isolate events shorter in duration than the persistence time of the ear. The earlier mentioned boundary of a 25 millisecond time span indicates the length

[101] Technische ..., op.cit., paper 8 (TT 608), pp.10-11

[102] See page 126

of the shortest detectable duration. Such a brief extract, however, cannot convey sufficient information for the ear to identify coherently its pitch or overtone structure, and as Meyer-Eppler observes, the sound length necessary for comprehension of these details varies not only with the frequency of the sound itself but also from one individual to another.

A time-frequency spectrum analysis thus cannot accurately represent the ear's response to a selected pattern of sound events. As a graphical aid, however, it comes far closer to providing a useful interpretation of what is a human and hence highly subjective reaction than either a function-time or a harmonic analysis, and this method of approach has been of considerable use both in the study of the nature of sounds and also in the development of new composer/machine communication techniques for the generation and manipulation of sound material.

Limitations nevertheless exist regarding the type of information which may be conveyed in this manner, and it is important to appreciate that the response characteristics of the ear involve several features which cannot easily be accounted for in scientific terms. The general blurring of simple details, for example, suggest that a truly representative time-frequency analysis should ignore certain sonological characteristics, but the precise nature of such a distortion is hard to interpret, for aural discrimination varies significantly according to context. As a general guideline in a pattern of sound events, the ear will tend to discriminate against regions of low intensity in favour of information of a higher intensity, even if on further investigation, when both are isolated and considered apart, it transpires that the intrinsic interest of the former is considerably greater than that of the latter. Further, given two sounds of equal loudness, but of different frequency regions, the detail of the higher region will frequently attract more attention than that of the lower region.

The nature of transient responses, associated with the attack of sounds, is another acoustical feature warranting careful study, and our understanding even today is far from being complete, for scientists have not yet developed entirely satisfactory methods of analysis for these complex phenomena, which involve the way in which the elements of a sound spectrum build up dynamically over a brief time interval. Many characteristics have nevertheless been identified, in particular the significant difference between the nature of the transient responses associated with naturally generated sound events and those which are produced synthetically. In the latter case it is frequently impossible to avoid the regularity of a fixed sound spectrum during this process, where all the frequency components increase in intensity in precisely the same manner, controlled, for example, by a simple attenuator circuit. This particular shortcoming presents a challenge for the designers of sound generation and processing systems, and will be examined more closely in due course.

A complicated factor stems from the ear's reaction to steady state sound material, for 'the ear has no analysing properties which are invariable with time. For example, if a pure sound is presented, the ability to determine this sound vanishes after a few seconds.[103] This becomes a major problem in composition with electronically generated sound material, for the essentially invariable nature of the raw material produced by electronic sound generators contrasts sharply with the constantly changing sonological features which predominate in the natural sound world of musical communication.

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[103] Technische ..., op.cit., paper 8 (TT 608), p.11

Another feature in this context referred to by Meyer-Eppler concerns the tendency of the ear to assume rhythmic or harmonic patterns within a sound passage which do not show up on any time-frequency analyses. Such areas of auditory reaction are exceedingly complex and, for the most part, beyond the scope of this study. Their undoubted existence, however, points to major areas of psycho-acoustic research which have become of increasing interest over the years, and which may ultimately supply the keys to many of the problems of musical coherence and meaning in information recognition which are at present only partly understood.

Meyer-Eppler's article continues with a more detailed examination of features particularly associated with the processes of electronic sound generation. Despite the basic limitations of the raw material produced by electronic sound generators, the way in which their output may be manipulated and processed opens up areas of acoustical response which are rarely encountered in the instrumental musical sound world.[104] One such feature concerns the ear's response to extremely abrupt changes in amplitude, where the transient time may on occasions be considerably less than 25 milliseconds. The response of the ear in these circumstances is far from simple, and if the duration of the sound is also relatively short, the perceived characteristics will differ considerably from what might be expected from a physical analysis.

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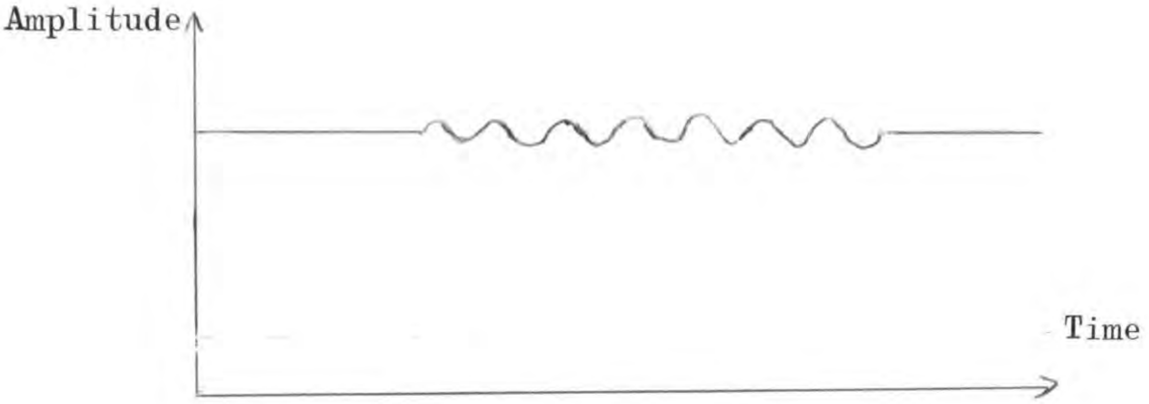
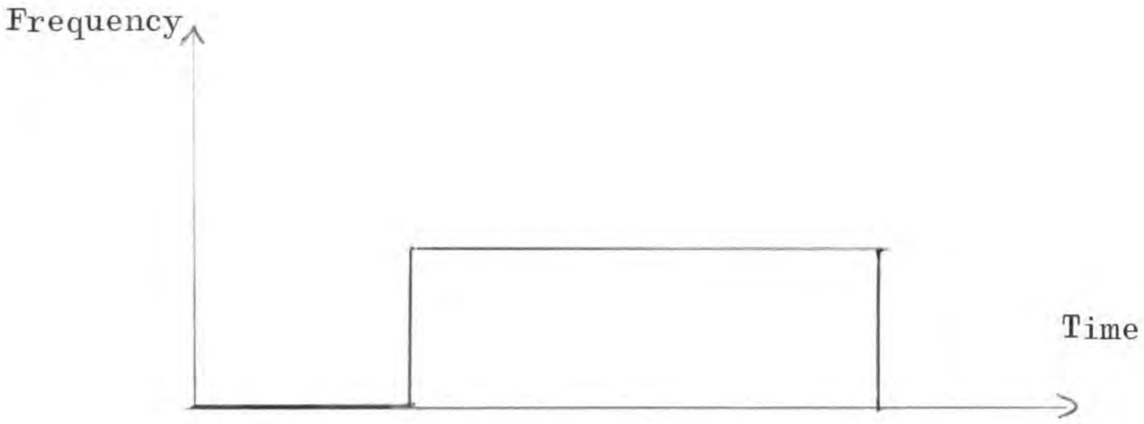
[104] Though it must be realised that some of the features to be described, such as frequency and amplitude modulation, are fundamental to processes discovered in the broader sound world, for example in the production of birdsong.

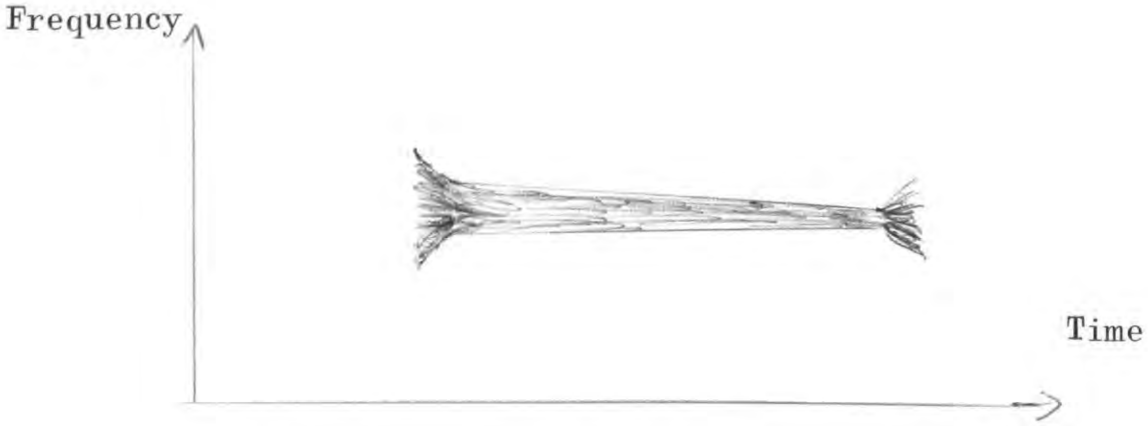
Whereas an oscilloscope display might display[105]



With the following physical time-frequency spectrum



the response time-frequency spectrum would approximate to



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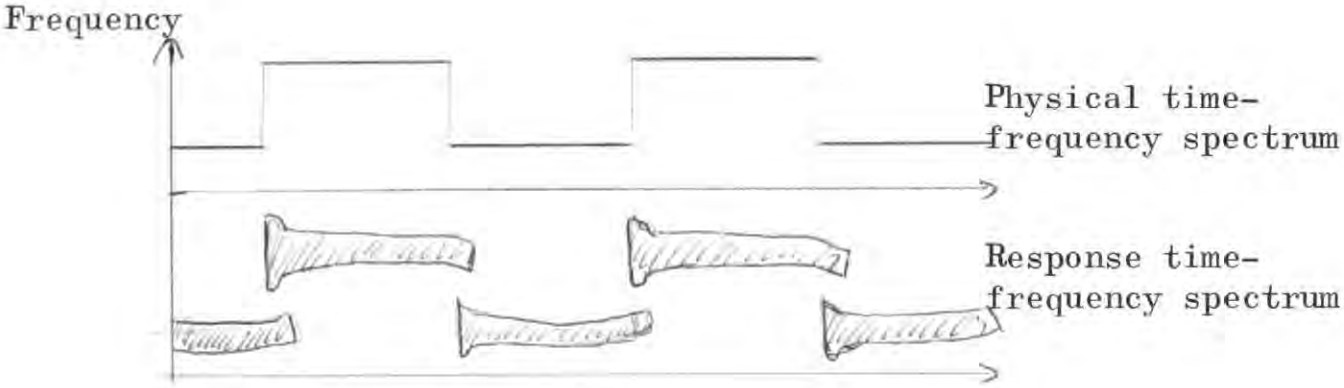
[105] Diagrams reproduced from Technische ..., op.cit., paper 8 (TT 608), p.30

> The sudden start and failure of the oscillation impulse brings about a considerable widening of the time-frequency spectrum, which can be heard as a 'click' (i.e. a physiological transient phenomenon). However, owing to the overriding of the preceding stationary part of the tone and because of the blur caused by the reverberation of the room in which the tone is produced, the final click is heard less intensively than the initial click.
>
> Hence a certain time passes after the initiation of a simple tone before the ear can distinguish the pitch with certainty.[106]

The last statement leads directly back to the earlier discussion regarding the relationship between the ear's response time characteristic and the time intervals selected for analysis samples. The former is now known to require sample intervals for pitch recognition of about 45 milliseconds at 100 Hertz, decreasing steadily to 13 milliseconds at 2000 Hertz, and increasing gently to about 18 milliseconds at 6000 Hertz. Further, the sharper the change from one amplitude level to another, the more marked is the widening of the time-frequency spectrum at this point, and the perceived effect accordingly ranges from a blurring of pitch to a wide band burst of noise. Similar problems occur if a sound is rapidly switched from one frequency to another, for recognition of the two pitches will become blurred. Meyer-Eppler gives the following example of the latter phenomenon:[107]



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[106] Technische ..., op.cit., paper 8 (TT 608), pp.16-17

[107] Diagram reproduced from Technische ... ibid., p.5

There are thus similarities between the perceptual characteristics of frequency and amplitude modulation, both widely used techniques in the processes of electronic sound composition which will be studied more closely in the next chapter. Nevertheless, it should be noted at this point that there is an important difference in terms of the scales of physical measurement between the ear's discrimination of pitch changes, and those of amplitude. Whereas a glissando simulated as a series of discrete pitch settings will demand the use of steps no larger than about 1/96th of an octave in the ear's most sensitive frequency range (about 3 Kilohertz) if it is to appear continuous, the energy level of a pitch in that region may be altered by as much as 1 decibel before any change is detectable. By way of a direct comparison of acoustical quantities, in the above instance each discrete frequency setting would involve a change in the order of 0.7%, contrasting sharply with 25% steps in amplitude levels.

These features become of primary importance when considering the characteristics of digital sound systems where each process has to be divided up into a finite series of steps, recorded as a numerical sequence. Whereas loudness, recorded in terms of 1 decibel steps[108] between the two limits of the threshold of hearing and the threshold of pain, may only require a range of 120 settings, even for the most extreme musical sound levels, division of a ten octave frequency spectrum would demand almost 10,000 settings. Even within the context of all-analogue sound studios this difference in the characteristics of pitch and frequency discrimination is of significant compositional importance, for it may be deduced that pitch structures may be constructed to a far greater degree of

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[108] In practice, as will be seen later, 1 dB steps prove a little crude, particularly at higher amplitudes, and it is quite common to employ 0.25 dB steps or even smaller for the loudest 10 dB of sound levels in a studio. This provision, however, does not radically alter the difference between the sensitivity requirements for pitch and amplitude.

complexity than amplitude structures. It should also be noted that it is only in the realm of electronic synthesis that it becomes possible to isolate these two parameters conveniently.

Meyer-Eppler, referring to the effect of rapid and continuous glissandos, observes that[109]

> The time-frequency spectrum of a sine oscillation, which changes its frequency rapidly but continuously (glissando), by no means faithfully reflects the curve form of the frequency of the oscillation; on the contrary, the result is again a blurred noisy frequency band. If the frequency is permitted to rise or drop rapidly the result will be a crackling noise. The same result is obtained if only a narrow frequency range (approximately up to an octave width) is filtered out from the time-frequency spectrum of the sliding note. If the filter range is low (e.g. between 75 and 150 c.p.s.), a downward glissando sounds like a dull thump.

This observation anticipated a studio technique which was yet to become commonplace, for in a 'classical' electronic music studio such as Cologne, where all the sound generation and processing operations were carried out manually, it was extremely difficult to make rapid alterations to the settings of the devices. Accordingly their processes of frequency modulation, as has already been seen,[110] were based around the techniques of electronic frequency multiplication and division to produce sum and difference tones via ring and quadripole modulator circuits. The characteristics associated with what is generally recognised today as being the more usual process of frequency modulation: the sweeping of the frequency of an audio generator according to

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[109] Technische ..., op.cit., paper 8 (TT 608), p.20

[110] See pages 103-108

a regular cyclic process, were therefore not extensively used, and it was not until the introduction of voltage controlled devices during the 1960s that this technique became a major feature of many studio systems.

Several paragraphs in Meyer-Eppler's treatise are devoted to a discussion of tone mixtures, sound complexes which figured prominently in many of the compositions produced at this time, notably Stockhausen's Studie I and Studie II. The sound continuum, which extends from a single sinusoidal wave to completely random components perceived as noise, may be constructed either additively by combining sinusoidal waves of suitable frequencies, amplitudes and phase or subtractively by filtering progressively narrower bands of noise to produce in the first instance, pitch areas and finally single tones. Neither method used alone provides an entirely satisfactory range of operational characteristics at their limits, but in the areas of transition both methods offer a useful choice of approaches. It is possible to combine these particular options to form a narrower continuum between pitch areas constituted of a relatively small number of frequency elements forming tone mixtures and banded noise spectra within which no frequency area predominates significantly over any other.

The use of additive techniques by its very nature offers a greater degree of flexibility over the content of the resultant frequency complexes, and Meyer-Eppler accordingly examines the response characteristics of the ear when perceiving a mixture of two or more pitched sounds. With reference to tones, sound complexes and noises he notes that these are essentially 'steady processes, i.e. practically speaking their spectral structure does not depend on time when an analysis interval similar to that in a human ear is used'.[111]

[111] Technische ..., op.cit., paper 8 (TT 608), p.13

There is nevertheless a finite, if large, analysis interval and in consequence there are no clearly defined spectral lines such as would be obtained from a simple harmonic analysis:[112]

> This means that certain deviations from the position of these poorly defined spectral lines may be admitted without changing the character of the spectrum, the ear is tolerant. Hence a simple sound with the frequency components 300, 400 and 500 c.p.s. remain a single sound, even if the frequencies are slightly changed, e.g. to 306, 398 and 502 c.p.s. respectively. However, the change is perceived not as impurities of the tone but as beats of timbre.

This observation points to an important difference between note mixtures where each constituent element contains its own overtone structure and simple tone mixtures. If the former had been used for the above experiment the result would have been an extremely rough sound, for in addition to relatively mild beats of timbre, strong beats of amplitude would be generated between the harmonics of the sources and some of these would produce discordant audio frequencies:[113]

> Relative to the cited example this means that the fourth harmonic of 306 c.p.s. (1224 c.p.s.) and the third harmonic of 398 c.p.s. (1194 c.p.s.) carry out beats of amplitude having a frequency of 1224-1194 c.p.s., causing a very perceptible acoustic roughness. For the higher the competing harmonics the more annoying will be the beats of amplitude.

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[112] _Technische_ ..., _op.cit._, paper 8 (TT 608), p.13

[113] _ibid._, p.14

The electronic synthesis of tone mixtures offers thus a far more versatile choice of sonological characteristics than those obtainable from naturally produced note mixtures, for it is possible to regulate the disposition of both harmonic and inharmonic partials individually. If two or more sine tones are tuned to frequencies which are almost identical the ear will perceive a single tone which appears to be changing cyclically in both frequency and amplitude. If the intervals between the frequency settings are then widened slightly and there are sufficient source tones present to provide a discrete range of pitches within this band, the resultant sound will lose pitch definition and appear to be noisy. Pitch identification, however, will still be sufficient for melodies to be created out of the tone mixture band when the latter is transposed complete, and in practice band-widths of up to a third of an octave may be used for this purpose. This provides an important area of interchange between additive and subtractive techniques, for 'physically and psychologically it does not make any difference whether such a noise is composed additively of individual sine oscillations sufficiently close together or whether it is selectively separated by a filter from broad-band noise.'[114]

Noise is not the only source of material suitable for subtractive processing, for square wave, ramp wave and triangle wave generators all supply information rich in harmonic content.[115] Meyer-Eppler highlights an important limitation concerning the degree to which the harmonic components of these sources may be isolated, for no matter how selective a filter network is used,

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114 Technische ..., op.cit., paper 8 (TT 608), p.15

115 See page 65, footnote 65 and tho Appendix

the fundamental tone can never be completely eliminated. If the former elements are significantly stronger than the latter the ear will be confused as to the structure of the sound, for the prominent partials will appear to be pitches in their own right having intervallic relationships both with each other and with the ever present fundamental. Most natural pitched musical sounds, even those of an exceptionally brilliant quality, contain harmonics which in aggregate enhance the quality of the fundamental pitch without establishing other pitch areas in their own right. The organ is the only instrument which may easily be manipulated to provide an exception to this general rule, if mixtures and mutation stops are drawn together and reinforced only by very weak foundation stops. This facility permits the creation of another continuum from notes of distinctive quality to pitch complexes containing individual frequency areas.

Only brief attention is paid to the use of reverberation in the preparation and performance of electronic compositions, a simple distinction being made between the effects of natural room acoustics on the quality of sound reaching an audience from a prepared tape, relayed via a set of loudspeakers, and the use of reverberation techniques as an integral part of studio sound processing routines. His observations nevertheless include an interesting comment concerning not only the acoustical but also the musical significance of such enhancement techniques:[116]

> Natural or artificial reverberation is the correct method of eliminating the technical rigidity of an electronically

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[116] Technische ..., op.cit., paper 8 (TT 608),pp.20-21

> produced sound and of maintaining this sound diffuse in the time co-ordinate. In as much as the processes involved show not only dynamic but also pitch and timbre structure, the temporary blur brings about a blur with respect to frequency and thus noise sensations. Hence reverberation may be utilised for converting sounds into noises.

The use of the word 'correct' raises a query concerning what he regarded as 'incorrect' methods in this context, for these are not explicitly discussed any further. It may be supposed nevertheless that criticism was probably being levelled against procedures which provided inept imitations of certain natural instrumental features such as pitch or amplitude vibrato. Eimert had himself attacked the use of such techniques in electronic musical instruments:[117]

> ... there is a great deal of objectionable 'vibration' in electronic entertainment music. The invention of electrical musical instruments have been in a great hurry to incorporate a vibrato effect, the new electronic qualité de luxe, into their ingenious constructions, to serve, perhaps as a substitute for expression, mood and nineteenth century sentimentality.

Over the years which have intervened since these writings, advancements in electronic technology have facilitated the investigation and development of new sound generation and processing techniques which provide more sophisticated methods of control over the nature of the material produced.

Many of the technological processes currently available in electronic music studios are, however, still far from satisfactory musically, and in due course it will be necessary to examine their shortcomings in more detail.

[117] Technische ..., op.cit., paper 10 (TT 610), pp.3-4

Within the present context, the limited functional characteristics of the equipment associated with the early Cologne studio bring to light some inconsistencies between stated compositional intentions and the actual results obtained.

It may be noted, for example, that Meyer-Eppler, in his commentary quoted above, advocates the use of reverberation as a method of enhancing electronic sounds, where, for reasons associated with their inherent lack of flexibility, they prove to be musically unacceptable. This introduction of an element of acoustical indeterminacy would appear to be in direct conflict with the aims of total control over the organisation of sound events and a lack of any association with the natural sound world, advocated by Eimert as the true calling of elektronische Musik.[118]

> In electronic-serial music ... everything to the last element of the single note is subjected to serial permutation. ... Examination of the material invariably leads one to serially ordered composition. No choice exists but the ordering of sinus tones within a note, and this cannot be done without the determination of the triple unit of the note. A note may be said to 'exist' where elements of time, pitch and intensity meet; this fundamental process repeats itself at every level of the serial network which organises the other partials related to it. ... Today the physical magnification of a sound is known, quite apart from any musical, expressionist psychology, as exact scientific data. It cannot, however, be the function of electronic music to make the sinus tone like the living 'parasite', to feign similarity where disparity exists. Talk of 'humanised' electronic sound may be left to unimaginative instrument makers.

Such an outlook on the use of electronic techniques in music is extremely narrow, for not only does it assert that the principles of total serialism

[118] Herbert Eimert, Die Reihe, op.cit., vol.1, pp.8-9

provide the one and only means of organising synthetically produced sound material in a musical fashion but also it assumes that the sound generation and processing facilities available at the time of writing (1955) were capable of providing every feature necessary for realising such intentions. In his earlier article, 'The Place of Electronic Music in the Musical Situation', he acknowledges that some doubts could be raised regarding the musical use of the medium, stating that[119]

> A far reaching, still unsolved question is whether electronic music as a universal source of all sounds possesses any coherent, form-sustaining force corresponding to tonality - [a feature naturally associated with] a self-sustaining system of timbres.

By the time of writing 'What is Electronic Music?' a year later in Die Reihe, Volume 1,[120] from which the penultimate quotation is taken, the existence of any such misgivings were, however, conveniently overlooked. This action did not pass unnoticed, for in the very next article in the same publication H. H. Stuckenschmidt, writing on the aesthetics of electronic music, observes that[121]

> Eimert has repeatedly drawn attention to the creative possibilities of electronically generated sound, but has disassociated himself from the 'fashionable and surrealistic' musique concrète produced as the Club d'Essai in Paris, and any incidental manipulations or distortions haphazardly put together for radio, film or theatre music. He is opposed to

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[119] Technische ..., op.cit., paper 10 (TT 610), p.7

[120] Die Reihe, op.cit., vol.1, pp.1-10

[121] H. H. Stuckenschmidt, 'The Third Stage', Die Reihe, vol.1, pp.11-12

> all metaphorical synaesthetic interpretation - that he is opposed to the idea of composition and interpretation by association and reference.
>
> Aesthetic understanding of the new art is not facilitated by this attitude. It cannot be denied that this associative effect, which the initiator denies as being of any relevance, has been the principal reaction of the majority of listeners faced for the first time with electronic music. There appears to be a considerable discrepancy between postulation and reception, a discrepancy which must lie at the very nature of the new art form. ...
> In that nothing pertaining to electronic music is analogous to any natural existent phenomenon of traditional music, associations have to be evoked from elsewhere. Instead of being integrated, they remain an ever increasing conglomeration of mentally indigestible matter. Thus the listener's reaction in broad outline, corresponds to his relationship to a humanly transfigured world.

Eimert's writings on the principles of composition with electronic sound took little or no account of the possibility that composers might encounter difficulties in rationalising their creative ideas in terms of totally objective acoustical specifications, or in overcoming the functional limitations of a studio in realising them. In practice, both these features have provided major stumbling blocks for the processes of man/machine communication, and the advancements in the design and specification of studios which have taken place over the years have not provided as many solutions for these problems as might have been expected. Indeed, the proliferation of technological facilities has given rise to a babel of device specifications and studio procedures, which in the case of systems employing computers involves a wide range of operating languages, each offering a different syntactical approach to the problems of sound generation and processing.

In such a climate it has not been easy for composers to establish and develop such an entirely new world of musical communication, and it is thus hardly surprising that audiences have tended to seek associations with their natural environment, both musical and otherwise, sometimes unwittingly encouraged by the composers themselves. To state that all associative characteristics are alien to the concepts of electronic music, however, is to deny composers the possibility of developing the medium as a positive extension of their own natural sound world, where instrumental and electronic material are integrated to create a coherent whole.

These strict doctrines did not remain unchallenged for long, for although many of the Cologne composers, including Stockhausen, had based their early electronic works on principles of strict, post-Webernian serialism, musical considerations concerning the limitations of the medium and changes in their compositional outlook gave rise to a movement directed towards greater flexibility in electronic sound processing: Stockhausen's use of a boy's voice as initial source material, treated and collaged with electronic sounds in his Gesange der Jünglinge (1955-6) was a notable challenge to Eimert's principle that all sound material must be entirely electronic in origin.[122] His Kontakte, written four years later (1959-60) not only replaced the principles of strict serialism with the freer macro/micro structures of

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[122] Stockhausen was concerned to expand the sound world of elektronische Musik, 'bringing together into a single sound both sung notes and electronically produced ones'. (Extract from the composer's own note on Gesange der Jünglinge, quoted in Karl H. Worner, Stockhausen: Life and Work, trans. Bill Hopkins, (Faber and Faber, London, 1973,) p.40.)

moment form[123] but also included instrumental parts for piano and percussion which could be optionally performed with the basic electronic tape.

Whilst Stockhausen was preparing Gesange der Jünglinge, Ernst Krenek (b.1900) was producing a work which in some respects was an even more marked departure from the teachings of Eimert. Krenek had become familiar with the work of Cologne for the first time during 1955 and quickly commenced work on his Whitsun oratorio Spiritus Intelligentiae Sanctus (1955-6). This composition freely combines all-electronic note mixtures and noises with the naturally generated sounds of human voices, his own speaking voice, a soprano and a tenor, combining techniques derived directly from musique concrète with established practices of elektronische Musik. The vocal parts, for example, are transposed over an octave range in thirteen tempered steps, and the studio filter units are used to derive pitched bands from a white noise which are then similarly treated and combined. Particular significance may be attached to the construction of what Stuckenschmidt has described as a 'timbre cadence' [124] at the close of the work, where the effect of a seemingly endless upward glissando is created by bands of sound mixtures which continuously ascend to beyond the upper frequency limit of the ear,

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[123] 'Each movement, whether a state or a process is individual and self-regulated, and able to sustain an independent existence. The musical events do not take a fixed course between a determined beginning and an inevitable ending, and the moments are not merely consequents of what precedes them and antecedents of what follows; rather the concentration on the Now - on every Now - as if it were a vertical slice dominating over any horizontal conceptions of time and reaching into timelessness, which I call eternity: an eternity which does not begin at the end of time, but is attainable at every moment.' (Extract from the composer's own note on Kontakte, quoted in Karl H. Worner, Stockhausen: Life and Work, op.cit., pp.46-47.)

[124] H. H. Stuckenschmidt, Twentieth Century Music (trans. Richard Deveson, World University Library, 1969), p.186

only to be replaced by new sounds entering from the bottom of the spectrum. Stockhausen was to use this technique to great effect in the fourth region of Hymnen, a work published twelve years later (1967), where a continuous downward glissando of pitch complexes lasting several minutes appears to freeze the progression of the event time continuum whilst preserving the incessant movement of the individual pitch elements.

Stockhausen succeeded Eimert as artistic director of the Cologne studio in 1963 and has retained this position ever since. Kontakte, with its use of live natural instrumentation, had widened the rift between the two composers, and pressure from members of an increasingly influential group of studio users, including Pousseur, Koenig, Krenek, Bengt Hambreus, Herbert Grün and Mauricio Kagel, made a change in leadership inevitable. Work began immediately on a major reconstruction of the studio involving, significantly, the expansion of the studio facilities into two inter-linked production rooms, one equipped with the studio sound generation and processing equipment and the other set up with a comprehensive range of recording and playback facilities. This latter provision permitted live material to be recorded for use within the electronic studio without interfering with the normal radio station production facilities. In practical terms the techniques of musique concrète and elektronische Musik became thus commonly accepted facets of electronic music composition.

The disagreements which continued between Stockhausen and Eimert are of some significance to this present study, for their public utterances have raised some queries regarding the precise course of events during the first decade of the studio. In particular an article by Stockhausen which was primarily intended as an attack on those American writers who regard the birth and development of electronic music as a phenomenon entirely attributable to their own country, provoked a reply from Eimert who challenged the accuracy of the information supplied regarding the birth of electronic music in Europe. This article was originally written in response to a request from Douglas M. Davis who was collecting material for inclusion in the American magazine Newsweek. For reasons probably not unconnected with its controversial content a decision was taken at editorial level not to print the article, and it was the Musical Times which finally published 'The Origins of Electronic Music' in July 1971.[125] Eimert's reply, 'How Electronic Music Began', was first published in Melos, January/February 1972, appearing in the Musical Times in April 1972.[126]

Eimert seized upon this opportunity to challenge statements made by Stockhausen not only in this article but also in a previous account of the history of the Cologne studio which had appeared in Die Reihe, volume 5 (1959)[127] in an article entitled 'Electronic and Instrumental Music'. He was clearly unhappy, for example, with the opening sentence of the latter which misleadingly attributed credit for the founding of the Cologne studio to Hanns Hartmann, the former head of the Westdeutscher Rundfunk, Cologne. Hartmann would have indeed been instrumental in persuading the board of directors to allocate resources for an electronic music studio, but it was

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[125] Karlheinz Stockhausen, 'The Origins of Electronic Music', Musical Times (Vol.112 no.1541, July 1971), pp.649-650

[126] Herbert Eimert, 'How Electronic Music Began', Musical Times (Vol.113 no.1550, April 1972), pp.347-349

[127] Karlheinz Stockhausen, 'Electronic and Instrumental Music', Die Reihe, 5, (Universal Edition, 1959, Eng. trans. Ruth Koenig, Theodore Presser Co., Pennsylvania, 1961), pp.59-67

Eimert who shouldered the practical responsibility for the development of facilities. Again, in 'The Origins of Electronic Music', Stockhausen places heavy emphasis on the significance of his first experiments at Pierre Schaeffer's studio in 1952[128] and notes that Eimert visited him there and took an informed interest in both his Etude and also the two composed by Boulez. With reference to the work of Beyer, Eimert and Meyer-Eppler he states that 'stimulated by musique concrète and critically opposed to instrumental "tape music",[129] these three men made some acoustic experiments with sounds produced electrically by the melochord and the trautonium'.[130] Eimert considered this association of musique concrète with the beginnings of elektronische Musik as erroneous, and certainly this would seem to conflict with his own contemporary writings.

Despite these conflicting accounts regarding the musical background to elektronische Musik, the documentation discussed earlier concerning the structure of the technical facilities during the early years of the studio may be considered reliable.

One new device was introduced at Cologne during the late 1950s, and this merits some attention, particularly since its potential as a rich source of sound information has been strangely neglected in the design of many of the commercial systems which are currently marketed. This is the impulse generator, from which Stockhausen derived a considerable proportion of the material employed in Kontakte.

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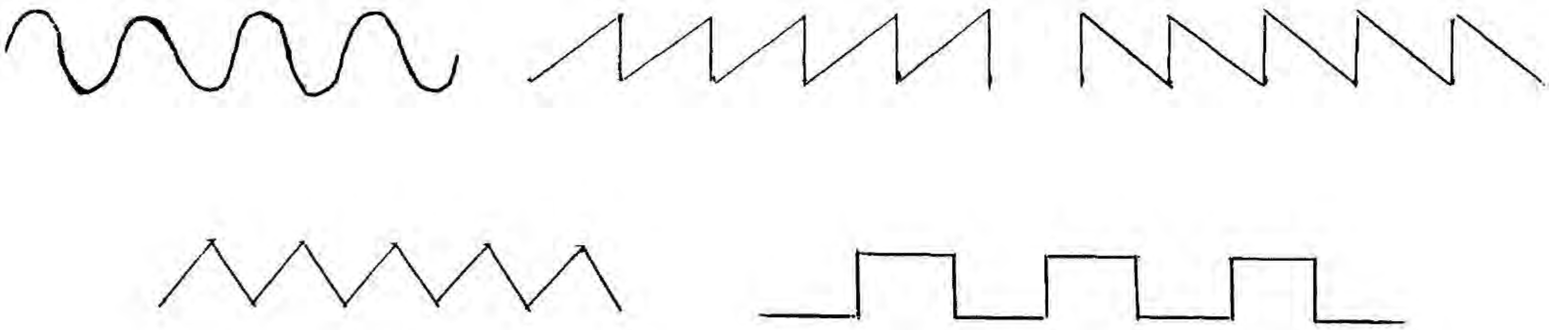
[128] See page 90, note 46.

[129] This American development will be studied shortly.

[130] Stockhausen, 'The Origins of Electronic Music', op.cit., p.649

The primary feature of this generator is the facility to alter the periodic wave pattern it produces and thus vary the quality of the note at source. The main source of pitched sound in the Cologne studio had been the sine wave generator, supplemented initially by the Trautonium and Melochord and later by other standard electronic generators producing square  , ramp (sawtooth)  or  , and triangle  waveforms, each the product of a specific Fourier combination of overtones[131] from a selected fundamental and thus suitable for subtractive processing via a bank of filters. In their normal form all these wave shapes have one feature in common: the patterns they produce are geometrically simple, and their harmonic structures correspondingly regular.[132]



etc.

The sine wave consists of a fundamental component only; the ramp wave is constructed from overtones drawn from the complete harmonic series in linearly decreasing amplitude; the triangle wave is constructed from odd harmonics only decreasing in amplitude according to the square of their

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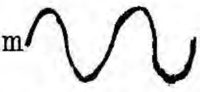
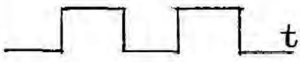
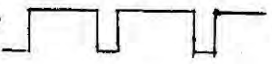
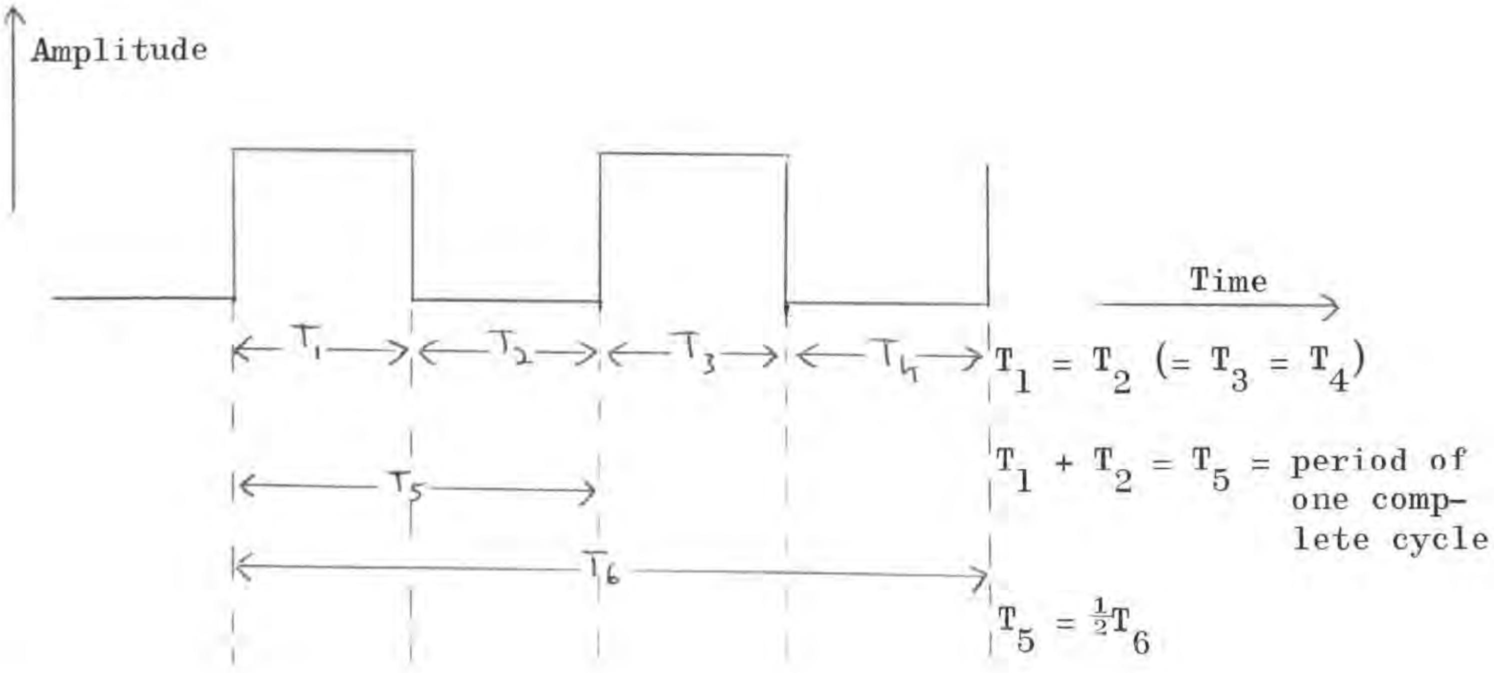
[131] See footnote no.62

[132] The mathematical functions for these waves are listed in the Appendix.

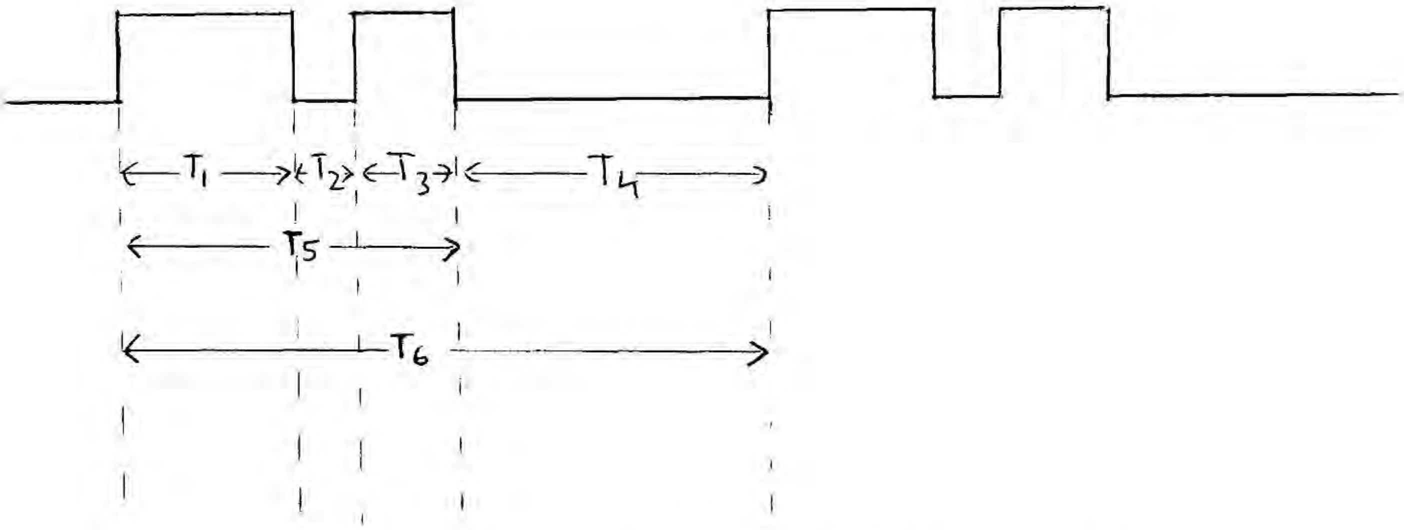
position in the series, and the square wave is constructed also from odd harmonics only decreasing in amplitude as a linear function of their position in the series.

Some modern generators permit simple variations to be made to their basic wave shapes, altering the distribution of harmonics without affecting the fundamental pitch setting. A sine wave, for example, may be electronically distorted from  to  or its complement . A ramp wave may be varied from  to  through the triangle wave , and a square wave may be squeezed from  to  or .

The effects on their perceived timbres are not as dramatic as might first appear, for the changes in the distribution of harmonics are relatively simple. The alterations shown to square wave patterns are nevertheless the most pronounced, and these hold the key to the techniques of impulse generation and the associated generation of complex periodic sound structures. The normal square wave provides the simplest form of impulse pattern, being completely symmetrical in construction.

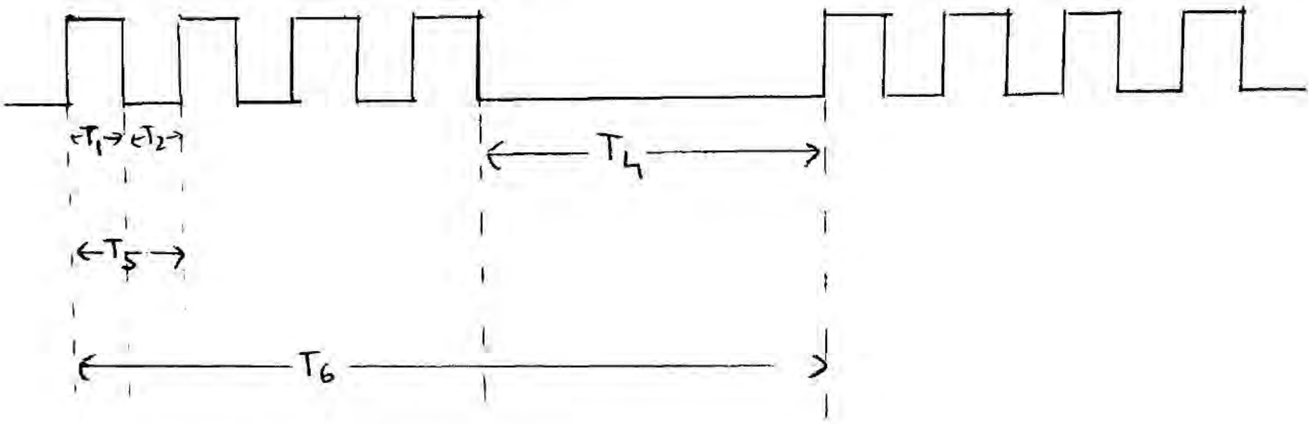


If the proportions $T_1:T_2$ are varied such that $T_1 + T_2$ remains unchanged, the effect is a change in timbre without any alteration to the pitch of the note. The construction of a device which permits T_1 and T_2 to be altered independently creates a totally different situation, for a change to either of these components alters not only the timbre but also the pitch of the note produced. Further, if such a device additionally provides control over consecutive pairs of pulses, i.e. T_1, T_2, T_3 and T_4, it becomes possible to set up complex functional relationships between the four components, providing great subtleties of pitch and timbre.



T_5 no longer determines the fundamental frequency of the wave, providing a secondary centre to the primary pitch region which is determined by T_6. The timbre is influenced by the widths of both T_1 and T_3.

Another useful form of pulse pattern is provided by a burst generator which creates a periodic repetition of a pulse train, the latter consisting of several identical pulses. For example:



Since T_5 and T_6 may be significantly dissimilar, the latter containing several repetitions of the former followed by a gap, further nuances of pitch and timbre become possible.

The addition of an electronic pulse generator to the standard range of audio sources greatly enhanced the sound production facilities of the Cologne studio, providing a powerful complement to the tape-controlled gating facilities already available.[133] In later years, as will be seen in later chapters, pulse techniques were to provide the basis for the construction of digitally orientated devices such as the variable function generator which was directly connected with the birth of the computer-controlled analogue studio. It is therefore surprising to discover that after initial pioneering work, Cologne played no further major role in the development of this or any other major innovation in electronic sound systems, the initiative passing to other centres such as the Institute of Sonology, Utrecht, Stifeltsen Elektronmusikstudion, Stockholm and E.M.S. Putney, London, to name but three European studios who have worked in this area of research.

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133 See pages 121 to 124 concerning the use of recorded frequency pulses in association with the ring modulator and the octave filter bank.

The significance of Stockhausen's Kontakte should not be overlooked at this point, for both musically and technically it remains today one of the major achievements in electronic music. According to Worner[134] the studio impulse generator was operated between about 16 impulses per second and 1 impulse in 16 seconds, providing impulse durations of between $\frac{1}{10000}$ of a second and 1 second. These impulses were then subjected to band-pass filtering to isolate harmonic components and the results recorded onto tape. Complex rhythmic patterns could then be created by preparing tape loops made up from pulse bursts spaced by lengths of blank tape. By means of a series of operations involving recording these patterns at a slow speed and playing back the result several times faster the pulses could be accelerated to become pitches of distinctive timbre, the latter being determined by the inherent rhythmic structure. Further treatments employed included ring modulation and reverberation. Stockhausen extended this continuum from areas of definite pitch to areas of noise by steadily increasing the degree of random distribution in the organisation of the impulse sequences, the band-width being determined by the deviation of the impulse durations from a selected mean.[135] A relationship is thus established between the microstructure of

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[134] Karl H. Worner, Stockhausen: Life and Work, op.cit., pp.136-137. See also Stockhausen's own discourse on 'time' and information theory: 'How Time Passes ...', Die Reihe, 3 (Universal Edition, A. G. Wien, 1959), English trans. (Theodore Presser Co., Pennsylvania), pp10-40.

[135] Stockhausen himself gives a short account of these pulse techniques related to a specific example drawn from the score of Kontakte, pp.19, 20 in Karlheinz Stockhausen, 'The Concept of Unity in Electronic Music', Perspectives on Contemporary Music Theory, trans. Elaine Barkin, ed. Benjamin Boretz and Edward T. Cone (W. W. Norton and Co. Inc., New York, 1972), pp.214-225. The article also appears in Perspectives of New Music (Princeton University Press, Fall 1962 - Spring 1963), pp. 39-48.

a sound concerned with the precisely defined rhythmic organisation of pulse events, and the macrostructure of a sound: the acoustically perceived product.[136] This provides a striking and all too rarely encountered illustration of the degree of integration which can be achieved between creative principles of musical organisation and the objective techniques of electronic sound synthesis, in terms of which they are realised and communicated.

Between 1954 and 1960 several other works were produced at Cologne, including Pousseur, Seismogramme I et II (1954); Koenig, Klangfiguren I (1955), Klangfiguren II (1955-56) and Essay (1957-58); Hambraeus, Doppelröhr II (1955); Heiss, Elektronische Komposition I (1954); Evangelisti, Incontri di fasce sonore (1956-57); Brün Anepigraphe (1958); B. Nilsson, Audiogramme (1957); Kagel, Transición I (1958-60); Ligeti, Glissandi (1957), and Eimert, Fünf Stücke (1955-56), Zu Ehren von Igor Stravinsky (1957) and Variante einer Variation von Anton Webern (1958).[137]

The establishment of an experimental studio for the Italian broadcasting company in Milan, June 1955, was influential in breaking down the dogmatic barriers which had been created between the Cologne and Paris studios, for, despite being clearly influenced in design by the former, the Italian style of composition was far freer and more improvisatory in character than that associated with either of these centres at this time. In consequence it was

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[136] See Jonathan Harvey, The Music of Stockhausen (Faber and Faber, London, 1975), pp. 30 and 88-90.

[137] A complete list is given in Hugh Davies, International Electronic Music Catalog, op.cit., pp.49-50.

immaterial to their philosophy whether or not microphones were employed in the production of material, for their work was more concerned with the perceived characteristics of sound structures rather than the formalistic principles by which they were obtained.

Luciano Berio, writing briefly in Score, March 1956, on the foundations of this Studio di Fonologia Musicale, noted that[138]

> Thus far the pursuit of the other Studios has been classified in terms of musique concrète and 'electronic music' which have become debatable definitions from today's armchair perspective since they seem to have been coined partly from retarded-futuristic pioneerism, partly to be 'dissociated from the rabble' and partly from a simple and legitimate desire to identify the objects of our daily discourse. In the long run, what really counts is the approach itself in its purest conception: it establishes an element of continuity in the general picture of our musical culture and is not to be identified only with its technical means but also with the inner motivation of our musical evolution.

The first pieces to be produced in this studio included Berio, Mutazioni (1955), Perspectives (1957), Thema - Omaggio a Joyce (1958), Différences (1958-60), Momenti (1960); Maderna, Notturno (1956), Syntaxis (1957), Continuo (1958); Nono, Omaggio a Emilio Vedova (1960); Boucourechliev, Etude I (1956), Texte I (1958); Pousseur, Scambi (versions I and II) (1957) and Cage, Fontana Mix (1958-59).[139]

An account of the technical facilities is given by Dr. Alfredo Lietti,

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[138] Luciano Berio, 'The Studio di Fonologia Musicale', Score, 15 (London, March 1956), p.83

[139] A complete list is given in Hugh Davies, International Electronic Music Catalog, op.cit., pp.149-150.

the system engineer, in Elettronica, 1956.[140] The provisions were extensive for their time, and for a short period the studio was to be the most comprehensively equipped in Europe. It was clear to both Lietti and Berio from the outset that commercial electronic equipment was not always ideally suited to musical applications, and many of the devices were built to special specifications. Lietti notes importantly that[141]

> The engineer who is to design the equipment must first of all consult with the musician so as to obtain a clear picture of the various requirements. Here certain characteristics must be overcome which arise from the different trainings received by engineers and musicians respectively, and the different terms they normally employ.
>
> Consider electronic music, for example. The musician may have a clear idea of the sound he desires to obtain, but, of course, it is a musical idea. The engineer, however, is interested in the physical data of the sound and whether it can be produced electronically. Obviously the difficulty can only be resolved by an effort at mutual understanding.

The electronic sound sources consisted of nine high stability sine wave generators, a white noise generator, a pulse generator[142] and a modified onde martenot. The provision of nine good quality sine generators was a considerable improvement over the single master generator facility employed at Cologne, for it permitted live generation and variation of frequency complexes, both features very much to the advantage of the composer. Special fine tuning controls were fitted to facilitate accurate setting, and the amplitude of

[140] Alfredo Lietti, 'Gli implanti technici della studio di fonologia musicale di radio Milano', Elettronica, 5 (3) (1956), trans. D. A. Sinclair, 'The technical equipment of the electronic music studio of radio Milan', National Research Council of Canada, Technical Translation TT 859 (Ottawa, 1957)

[141] Ibid., TT 859, p.4

[142] Described as a 'beat' generator in the studio plan.

each unit could be discretely varied from a central control panel. A cathode ray oscilloscope was also available to check the generator tunings when harmonic overtones for a selected fundamental were required.

Standard recording facilities were provided by a generous array of mono, stereo and four track recorders. The treatment devices consisted of ring modulators, amplitude modulators, a reverberation chamber, a tape head echo unit, two oscillator controlled variable speed tape recorders, various high pass/low pass/band pass filters and an octave filter bank connected to a wave analyser. The latter device could also be employed as an extremely powerful band pass filter reducing octave filtered sound material to band widths as narrow as 2 Hertz.

The two variable speed tape recorders were each fitted with special devices which permitted not only normal frequency compression and expansion but also, within certain limits, variation of the speed of events without altering their pitch. This modification was developed by Springer at Telefonbau and Normalzeit and later became commercially available as the Springer machine or Tempophon.[143] The basis for this system was a rotating system of four playback heads set at 90° to each other, forming the outer periphery of a special capstan:

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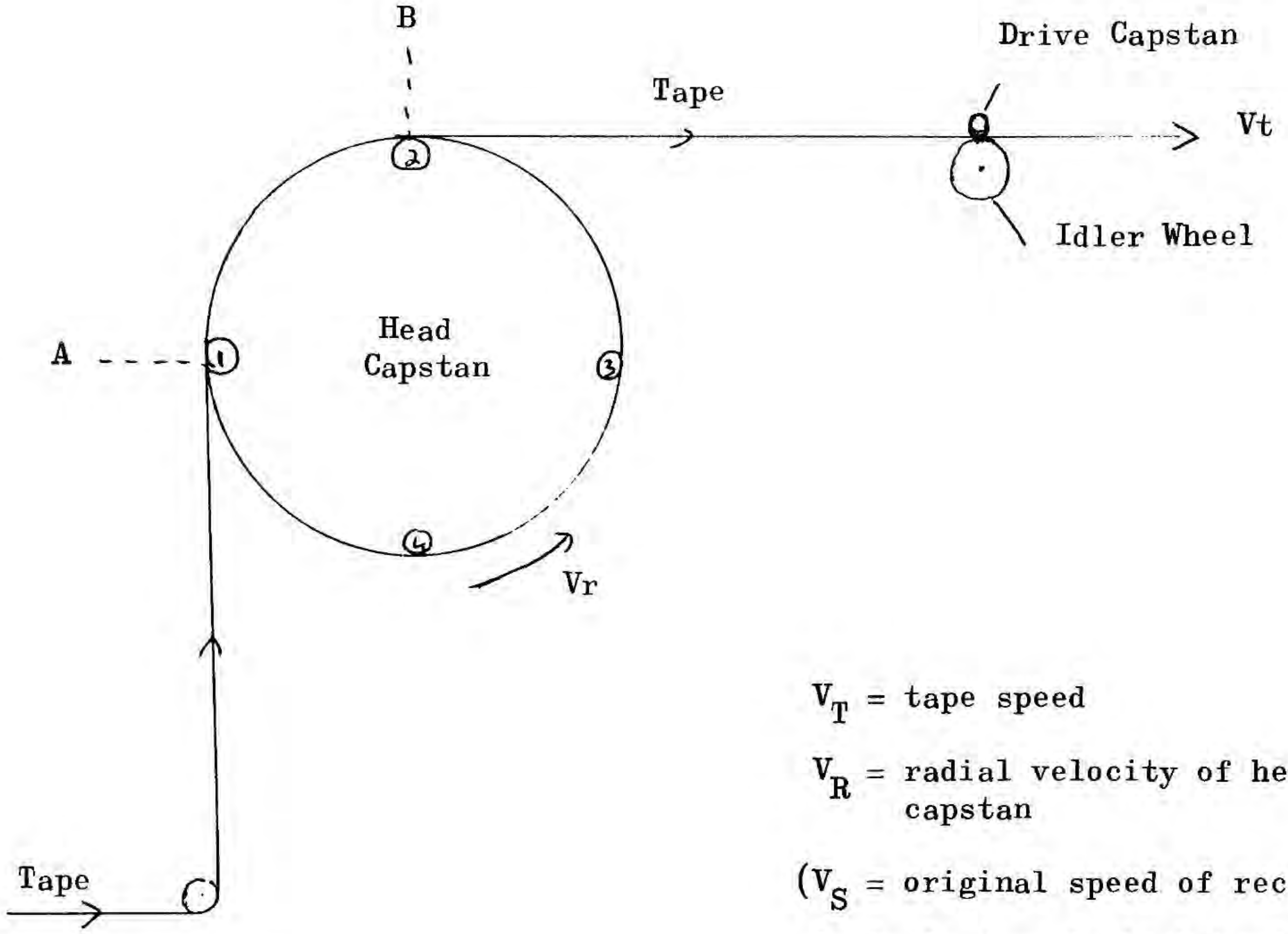
[143] Currently manufactured by Eltro, Heidelberg, Germany.



V_T = tape speed

V_R = radial velocity of head capstan

(V_S = original speed of recording)

1, 2, 3, 4 = playback heads

If the drive capstan and the head capstan are rotated, information from the original material will be reproduced as a series of consecutive samples drawn from each of the playback heads in turn. In order that the material may appear continuous a certain degree of overlapping between samples is required, and as may be seen from the diagram above the transport is arranged so that at the point of changeover the succeeding head makes contact with the tape just before the preceding head drops out.

It is possible to arrange for the head capstan to remain stationary, i.e. $V_R = 0$. If V_T is then set equal to V_S, the pitch and the duration content of the original material will be reproduced exactly. If the drive capstan and the head capstan speeds are varied such that $V_T + V_R = V_S$, the speed of the heads relative to the tape will remain constant whilst the absolute speed of the tape itself, and thus the duration of the recorded material will be changed. The rotating heads thus extract 'snapshot' samples of sound from the travelling tape and reproduce these at their correct pitches. The machine may also be employed in a more extreme fashion by varying the speeds of both drives quite independently, changing both pitch and duration. If the tape is stationary and the heads rotated, for example, the information recorded on a small section of tape - between A and B in the diagram - may be frozen as a continuous sound, varying in pitch as a direct function of the capstan speed.[144]

The fidelity of sound produced from this device is unfortunately rather poor, for not only does the head design limit the frequency response but also the process of breaking up of analogue sound into discrete samples may result in considerable distortion at the point of changeover from one head to the next. The principles involved are nevertheless interesting for they are related to the processes of digital sound sampling, discussed in a later chapter.

Lietti's account concludes with a block diagram of the studio system which is reproduced here:[145]

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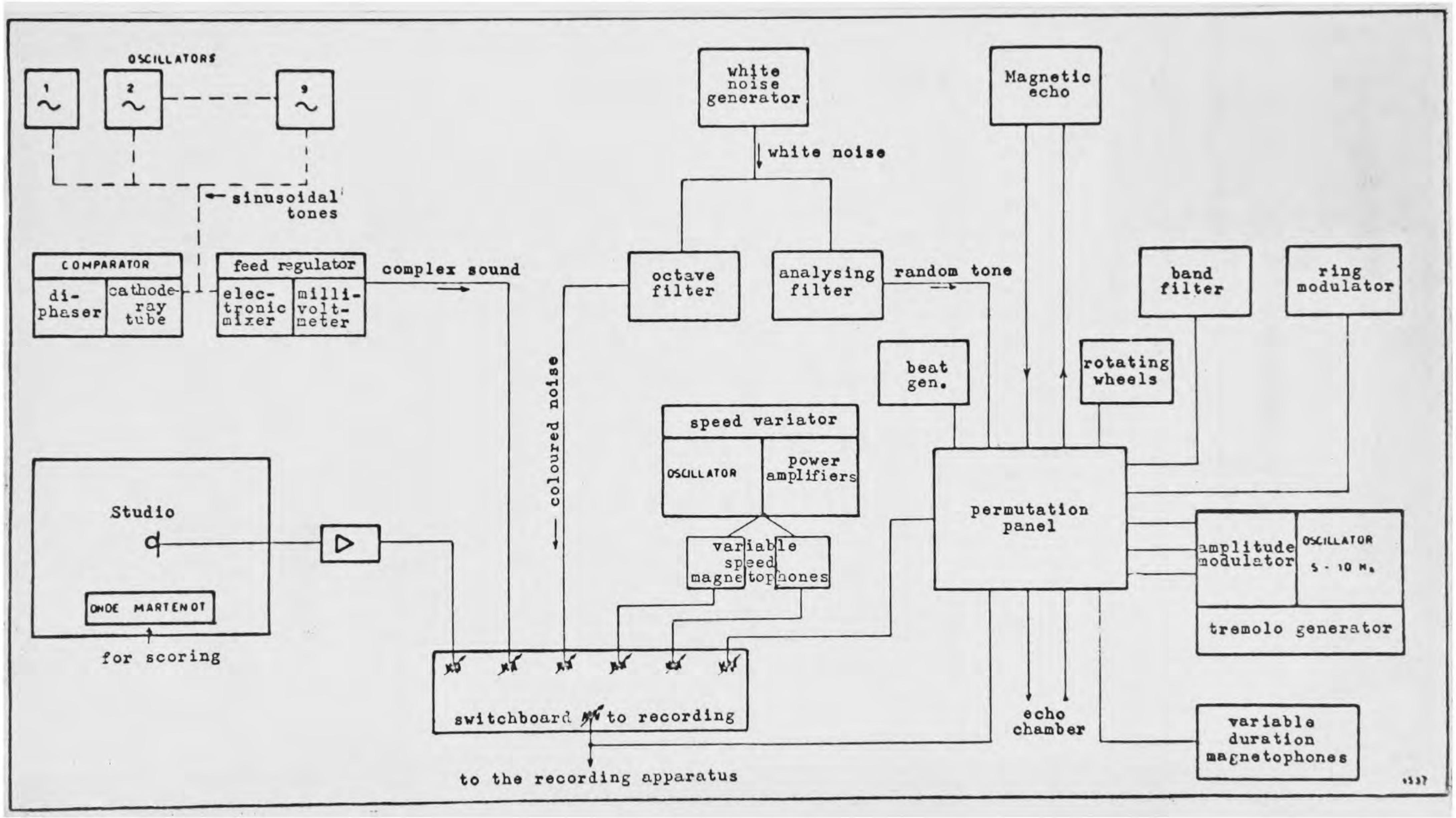
[144] Cologne purchased a Springer machine late in the 1950s, and Eimert used the instrument extensively for his composition Epitaph für Aikichi Kuboyama (1960-62).

[145] Alfredo Lietti, 'The Technical Equipment ... Milan', op.cit., (TT 859), p.15



General block diagram of the component installations of the Musical Phonology Studio

By the early 1960s electronic sound processing and generation systems were being developed at various centres in Europe, and even as far as Moscow and Tokyo. Interest in their facilities within the context of this present study becomes far more general, for the majority were based on the 'classical' analogue principles already described. Voltage and digital control techniques were already being experimented with, however, and a study of these developments will form the basis of the next chapter.

Excluding America and Canada which will be discussed shortly, the following studios are a representative selection of the more important centres which were founded during the period 1948-1964.[146]

PARIS	Groupe de recherches musicales O.R.T.F. Opened 1948 Composers: Schaeffer, Henry, Boulez, Stockhausen, Messiaen, Milhaud, Varèse, Arthuys, Malec, Ferrari, Xenakis, Parmegiani, Bayle
COLOGNE	Studio für elektronische Musik W.D.R. (later N.W.D.R.) Opened 1951 Composers: Eimert, Beyer, Goeyvaerts, Krenek, Stockhausen, Koenig, Evangelisti, Ligeti, Kagel, Boehmer, Von Biel, Fritsch, Johnson
MILAN	Studio di Fonologia Musicale R.A.I. Opened 1955 Composers: Berio, Maderna, Cage, Nono, Castiglioni
TOKYO	Electronic Music Studio N.H.K. Opened 1956 Composers: Mayazumi, Moroi, Yuasa
DARMSTADT	Studio für elektronische Komposition Opened 1957 (closed down 1966) Composers: Heiss

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146 This is a slightly altered version of a list given in Gottfried Koenig & Fritz Wieland, Summary Introduction, op.cit., pp.34-35, augmented by reference to Hugh Davies, International Electronic Music Catalog, op.cit. Only the more important composers using these studios prior to 1965 have been listed.

WARSAW	Studio Eksperymentalne Polskie Radio Opened 1957 Composers: Kotonski, Dobrowolski, Penderecki, Rudnik
EINDHOVEN	Research Laboratories Philips Opened 1957 (closed down 1960) Composers: Badings, Dissevelt, Raaijmakers, Varèse
MUNICH	Studio für elektronische Musik Siemens und Halske Opened 1957 (closed down 1966) Composers: Hambraeus, Kagel, Brün, Riedl
BRUSSELS	Studio A.P.E.L.A.C. Opened 1958 Composers: Pousseur, Küpper
LONDON	Radiophonic Workshop B.B.C. Opened 1958 Composers: Gerhard[147]
PARIS	Studio A.P.S.O.M.E. Opened 1958 Composers: Henry
LONDON (from 1963, Diss, Norfolk)	(Private studio) Opened 1958 (closed down 1975) Composers: Cary
FAIRSEAT/ **WROTHAM,** Kent	Oramics Studio (private) Opened 1959 Composers: Oram
MOSCOW	Scriabin Museum Laboratory Opened 1961 Composers: Artem'ev, Kreichi
UTRECHT	Institute of Sonology, Utrecht State University Opened 1961 Composers: Kagel, Koenig, Badings, Raaimjakers, Weiland, Bruynèl, Boerman
BERLIN	Studio für elektronische Musik Technische Universität Opened 1962 Composers: Blacher, Krenek, Shinohara

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[147] In view of the policies enforced by the B.B.C. virtually all the works produced in the studio during this period were for radio and television programmes and not serious electronic compositions in their own right.

GENT	Instituut voor Psychoakoestiek en Elektronische Muziek Ryksuniversiteit Gent Opened 1962 Composers: Pousseur, Goeyvaerts, de Meester, Goethals
LONDON	Electronic Music Studio, Putney (private studio) Opened 1963 Composers: Zinovieff
STOCKHOLM	Elektronmusikstudion Sveriges Radio Opened 1964 Composers: Blomdahl, Hambaeus, Nilsson

3 Developments in America - Music for Magnetic Tape/Tape Music

Events in America after the Second World War followed quite different paths to those in Europe, due primarily to a lack of institutional support during the early years. Indeed, until the mid-1950s no major systems of interest were constructed, many of the so-called studios consisting merely of a collection of tape recorders and inter-connecting wires assembled in a back room, or at best commercial recording systems leased for experimentation. Despite this lack of dedicated systems several composers managed to investigate the possibilities of manipulating sounds recorded on tape, preparing the way for the properly equipped electronic studios which were eventually to follow.

In 1948 two recording engineers, Louis and Bebe Barron, began to experiment with the musical possibilities of magnetic tape, playing recordings of instruments backwards and forwards and investigating the effects of splicing out selected elements and juxtaposing others. John Cage became interested in their work and in 1951 gathered together a group of musicians and technical advisers for the purpose of making music directly onto tape. The composing members of the group consisted of Cage, the Barrons, Earle Brown, Morton Feldman, David Tudor and Christian Wolff, and for the next two years they worked in the Barrons' studio developing a project which became known as Music for Magnetic Tape.

The only complete pieces produced in the studio during this period, apart from Heavenly Menagerie and some background music for films prepared by the Barrons themselves, were: Cage, Imaginary Landscape no.5 (1951-52), Williams Mix (1952), and Wolff, For Magnetic Tape (1952-53). These compositions explored many of the techniques associated with musique concrète and to a certain extent elektronische Musik, but musically they were motivated by rather different aims. Cage in particular was concerned with exploring principles of indeterminacy: Williams Mix and Imaginary Landscape no.5 were based on 'I-Ching' chance operations involving an elaborate series of tape splicing and looping routines. The source material for the former work consisted of about six hundred recordings prepared from six categories of sounds: basic electronic sounds, manually-produced sounds including instrumental sources, wind produced sounds including singing, city sounds, country sounds and quiet sounds amplified to levels comparable with the rest of the material.

The project terminated in 1953 and the composers went their separate ways. Brown and Feldman continued their investigations for a time at the Rangertone Studios, Newark, New Jersey, producing Octet (for eight loudspeakers) and Intersection respectively in the same year. Only Tudor failed to produce a work at this time. Brown subsequently travelled to Paris and continued his work there. Cage continued to work at various private studios in New York, travelling to Milan to produce Fontana Mix in 1959, returning to work both at Brandeis University, and also extensively at Stony Point, New York, where in collaboration with David Tudor he became preoccupied with the use of electronic equipment in live performance.

While Cage and his associates were experimenting in the Barrons' studio another line of investigation was being pursued by Vladimir Ussachevsky, shortly to be joined in his work by Otto Luening. Superficially their experiments which became generally known as Tape Music, closely related to those of Music for Magnetic Tape, for both approaches were based around the use of the tape recorder as an information storage and transformation device. Closer examination reveals marked differences in musical outlook, for both Luening and Ussachevsky were very much more conservative in their views on musical composition. In consequence they sought to extend traditional principles of tonality and instrumentation by recording notated material and then subjecting it to simple processes of tape transformation, such that the product retained characteristics of its original identity.

During 1951-52 Ussachevsky carried out a series of experiments, and prepared five studies entitled Transposition, Reverberation, Experiment, Composition and Underwater Waltz. These were presented at a Composers' Forum given on 9 May 1952 in the McMillan Theatre, Columbia University.[148] Transposition was simply a study in octave transposition using piano notes as sources, achieved by copying from one two-speed tape recorder to another.

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[148] Although coincidental, it might be noted that this demonstration preceded Meyer-Eppler's series of lectures at Darmstadt by a matter of weeks.

Reverberation was based on the use of tape head echo, which may easily be achieved on any recorder equipped with separate record and replay functions, and the other pieces were simple arrangements of instrumental material subjected to these processes of alteration. Their performance attracted considerable attention, and the composer Henry Cowell wrote an encouraging review in the Musical Quarterly, October 1952.[149]

> People who work experimentally with new sounds seem to have trouble distinguishing between the materials of musical composition and the compositions themselves. They are apt to rush their new sounds prematurely into pieces that are hardly creative work in the generally accepted sense, and that are easily identified as vehicles for new sounds rather than works in which these sounds form an integral part. ... It is therefore refreshing when a composer offers his experiments frankly by that name, without confusion. Vladimir Ussachevsky did just this ... These were not compositions and no attempt was made to call them so. But the sounds are certainly a possible resource for composers.

Luening attended the Forum and invited Ussachevsky to present his experiments at a composers' conference in Bennington, Vermont, during August 1952. Luening had studied with Busoni in Zurich, Switzerland, from 1918 until 1920 and had for many years been interested in the study of musical acoustics and instrumental design. At Bennington, Ussachevsky experimented with violin, clarinet, piano and vocal sounds using an Ampex tape recorder, and Luening began to prepare a tape composition using a flute as source material. News of their work came to the attention of Oliver Daniel who

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[149] Henry Cowell, 'Current Chronicle: New York', Musical Quarterly, XXXVII, 4 (October 1952), p.597

invited them to prepare a group of short compositions for inclusion in a concert to be promoted by the American Composers' Alliance at the Museum of Modern Art, New York, under the direction of Leopold Stokowski. The invitation was accepted, and Luening and Ussachevsky departed for the home of Henry Cowell at Woodstock, New York, where, using borrowed tape recorders, they prepared pieces for the first public concert of Tape Music on 28 October 1952. Four pieces were performed: Ussachevsky, Sonic Contours; Luening, Invention in 12 Notes, Low Speed and Fantasy in Space, the last work being the product of his earlier experiments with flute sounds at Bennington.

Again the critics were cordial in their accounts, and the audience included Luciano Berio who was especially impressed with Sonic Contours. These pieces made considerable use of tape techniques, such as overlays, tape head echo, extreme changes of speed and splicing, but their structures retained many recognisable tonal characteristics such as simple chords, scales and arpeggios. The attention accorded to this recital overshadowed the work of Cage and his associates. Williams Mix, for example, had to wait for two years before receiving its first public performance before an unsympathetic audience at the 1954 Donaueschingen Festival, Germany. Examples of Tape Music had already been presented in Europe alongside musique concrète one year previously during the festival presented by Radiodiffusion Francais in Paris, April 1953. Other potential developments also passed unnoticed at this time: Milton Babbitt at Princeton University, for example, had been interested in electronic sound production for many years, and a series of experiments into the possibilities of hand drawn sound were instigated.

A lack of support, however, led to the initiative in this field passing to Bruce McLaren at Ottawa.[150]

After their early successes Luening and Ussachevsky began to explore the possibilities of using prepared tapes in conjunction with performing instruments. In late 1953 Luening received a commission from the Louisville Orchestra to write a work for them which he accepted on the condition that he could share the venture with Ussachevsky, and this was agreed. Permanent studio facilities were still not available, and the equipment had to be gathered together from a variety of sources. A small grant from the Rockefeller Foundation facilitated the purchase of one tape recorder, but the others had to be borrowed or purchased privately. After a trial performance at the Bennington Composers' Conference the work, entitled Rhapsodie Variations for Tape Recorder, was publicly performed by the Louisville Orchestra on 20 March 1953 under the direction of Robert Whitney. Another commission soon followed, this time for the Los Angeles Orchestra. The work, A Poem of Cycles and Bells, took the form of a paraphrase of their earlier works, Fantasy in Space and Sonic Contours and was altogether more successful, a better balance being achieved between the instrumental parts and the prepared tape.

By the end of 1954 the situation in America was beginning to change, for the publicity accorded to Tape Music had acted as a catalyst for a growing interest in electronic music. Several private studios began experimenting with the medium, but for the most part their aims were directed towards commercial ends, providing background effects for films, radio and

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[150] See chapter 1, page 25

television.

So far no mention has been made of Varèse's role in these developments. After his determined, but sadly unsuccessful agitations for studio facilities in the inter-war period it might have been expected that the changing climate of the 1950s would have placed him at the forefront of advancements in this field. Unfortunately this did not prove to be the case.

The late 1930s and the whole of the 1940s found Varèse involved in the deepest personal struggles with his music. After Densité 21.5 for flute (1936) no works were produced for eighteen years with the single exception of Etude, scored for two pianos, percussion and mixed chorus. This was intended to form part of his Espace, a project destined to remain an uncompleted ambition. What might be described as his period of rebirth and eventual rise to fame occurred well towards the end of his life, for in 1950 when he started composing again in earnest, he was already 67. It is thus hardly surprising that at such a late stage in a career characterised by continual disappointment and disillusionment with the Establishment, he should now leave others to carry on the crusade for studio facilities and concentrate only on the realisation of his own compositional ambitions. In a modest way, nevertheless, his explorations into the possibilities of electronics made a contribution which was far more significant musically than the much publicised early efforts of Luening and Ussachevsky.

During the late 1940s Varèse had been formulating an idea for a piece which would interpolate passages of sound material organised on a magnetic tape with live instrumental performance, and by the beginning of 1950 an outline of the work, Déserts, had been prepared. In the summer he began composing the instrumental parts, completing them towards the end of 1952. During 1953, with the aid of a technical assistant, he began gathering together recordings of iron mills, saw mills and various other factories in Philadelphia with the object of assembling material for use in the taped sections. Gradually he built up a comprehensive library of sound material, in his own private way carrying out investigations just as detailed as those of Schaeffer, working unaided with a very modest collection of taping equipment in his house at Greenwich, New York.

Boulez visited America in the winter of 1952-53, presenting musique concrète to New York for the first time at a special concert for the Composers' Forum, Columbia University. During his stay the two composers met for the first time, and Boulez was thus able to give an informed account of his work on Déserts on return to Paris. Rumours of Varèse's work had spread to Paris sometime previously, and Schaeffer himself had made a rather inaccurate reference to him in A la Recherche d'une Musique Concrète, 1952.[151]

> Varèse has dedicated himself to that poor relation of the orchestra, the percussion section. He has promoted it to orchestral status. He has added to it various effects supplied by American studios. I do not know the details. More or less electronic 'Varinettes', produced I know not how, but occasionally similar to ours.[152] Varèse crosses France without stopping. This Frenchman has not had our misfortune to be a prophet in his own country. He is listened to and revered in Germany. Soon he will return to New York where he is considered Maestro.

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[151] Pierre Schaeffer, A la Recherche d'une Musique Concrète, op.cit., p.80

[152] 'Ours' in this context refers to musique concrète.

The reference to Germany concerns a series of lectures he presented to the 1950 Darmstadt Festival at the invitation of Wolfgang Steinnecke, and _Déserts_ would have naturally been uppermost in his mind at the time. The association of his work with _musique concrète_ is not particularly appropriate, for his approach to organised sound was far more liberal, including elements which might equally be attributed to _elektronische Musik_ or _Music for Tape/Tape Music_; common to all yet restricted by none.

In 1954 Varèse received an invitation from Schaeffer to complete the tape parts for _Déserts_ in his Paris studio. In the absence of any comparable opportunities in America this was accepted and he departed for France in late September. At the Club d'Essai he worked remarkably quickly, completing the work in barely two months. His enlightened approach to principles of sound organisation took Schaeffer by surprise,[153] for Varèse would frequently indulge in elaborate transformations, investigating whatever electronic techniques the engineers could devise. The studio was naturally ill-equipped for such operations; the use of extensive filtering or ring modulation, for example, was foreign to the still strict doctrines of _musique concrète_.

The results were not wholly satisfactory, a combination perhaps of three factors: the relatively short period spent in preparation, the limitations of the equipment, and the immense practical problems which

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[153] Schaeffer was later to remark, 'I didn't like the work ...'. See _Les Lettres Françaises_, 16 June 1965, quoted in Fernand Ouellette, _Edgard Varèse_, _op.cit._, p.185

confront any composer encountering a complex studio system for the first time.[154] The first performance of Déserts took place in the Théâtre des Champs Elysées on 2 December 1954, conducted by Herman Scherchen. Forty years previously this hall had been the venue for the riotous first performance of Stravinsky's Rite of Spring, and the audience for Déserts was quite ready to demonstrate that noisy public disapproval was not quite a phenomenon of the past. Matters were made worse by the fact that the O.R.T.F. were making a live transmission of the concert, which also included works by Mozart and Tchaikowsky. The considerable adverse publicity which followed was positive in at least one respect: the use of electronics in music was attracting the attention of a wide, if often unsympathetic audience.

After two rather more successful performances in Hamburg and Stockholm conducted by Bruno Maderna, Varèse remained in Paris for a while, returning to the United States in the spring of 1955. It is perhaps ironic to note that during his absence the Radio Corporation of America had started preparing the R.C.A. Olson-Belar Sound Synthesiser, Mark I, for demonstration to the public for the first time, an event which as will be seen shortly was to play an important part in shaping the future course of events in America.

During May 1955 Varèse attended an Arts conference at Bennington, Vermont, presenting a lecture on his work on the 16th and supervising the first American performance of Déserts, given in the National Guard Armory the following day. The first major presentation of the work took place on 30

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154 Varèse was to spend the next eight years trying to improve his tapes, creating no less than four different versions. The last, definitive version was produced at the Princeton Columbia Electronic Music Center in 1961, using the Olson-Belar Synthesiser Mark II.

November 1955 in the Town Hall, New York, and on the whole was favourably received. This performance could not have occurred at a more appropriate time, for the interest of institutions in supporting electronic music was just being kindled.

In June 1955 Luening and Ussachevsky had obtained a generous grant of nearly $10,000 from the Rockefeller Foundation, funded through Barnard College, to investigate the state of studio facilities both at home and abroad. During a six week tour of Europe they visited Schaeffer in Paris, Meyer-Eppler in Bonn, Eimert in Cologne and Berio and Maderna in Milan. They were thus able to piece together a very thorough account of these major centres of activity. In Canada they discovered that developments were also well advanced, for Hugh le Caine, working with the support of the Canada Research Council, had established a studio at Ottawa University in 1954, and, as mentioned earlier, Bruce McLaren was making useful advances in the use of optical sound generation techniques.

By comparison only limited progress had been made in America. One or two universities were willing to consider giving some support to suitable ventures, but none had yet committed itself to a major, long-term development programme. At Ilinois, however, they learnt of a project which was to hold a particular significance for the future, for an investigation had been started into the possible uses of computers in musical composition. This research, headed by Lejaren Hiller and Leonard Isaacson, was concerned in the first instance with the use of calculative procedures to generate data for conventionally notated instrumental scores,[155] but this was also to herald the use of the computer itself as a machine for generating sound information. In connection with the latter, Luening and Ussachevsky discovered that one of the research programmes at the Bell Telephone Laboratories, New Jersey, was directed towards the development of techniques for both the analysis and the synthesis of sound information. Their investigations were concerned not only with conventional analogue approaches but also with the possibility of developing computer based methods.[156]

The work at Bell Telephone Laboratories proved to be unique. Apart from Ampex, who were involved in the design and manufacture of recording and amplification equipment, most industrial concerns were not willing to consider supporting basic acoustical research and development unless they could expect to benefit commercially from the results within a matter of months.

[155] Hiller and Isaacson used the computer to write the Illiac Suite for String Quartet (1957). In 1958 Hiller and Robert Baker began a collaboration which produced the musical composition programme MUSICOMP in the early 1960s. A detailed account of these and other related developments is given in Lejaren Hiller, 'Music Composed with Computers - a historical survey', The Computer and Music, Ed. Harry B. Lincoln (Cornell University Press, Ithaca and London, 1970), pp.42-96.
Innanis Xenakis similarly began exploring the uses of computers in musical composition around this time, working in Paris.

[156] This organisation, as will be seen in the fourth chapter, was subsequently to pioneer the use of digital computers as a means of directly generating sound information under the direction of Max Mathews.

Luening and Ussachevsky decided to take the initiative and formally approached the authorities of Columbia University with a view to establishing an electronic music studio within the music department. The idea was favourably received, and they were granted a limited number of technical facilities to set up a small experimental laboratory. By now their approaches to electronic composition were beginning to extend beyond the limitations of the tape recorder as the sole means of processing sound. Ussachevsky's <u>Piece for Tape Recorder</u> produced early in 1956, for example, integrates both electronic and natural sound sources.[157]

In fulfilment of their Rockefeller grant requirements they prepared a report on the state of electronic music in Europe and America, expressing a view that the progress of electronic music in their country could best be assisted by channelling financial support into the universities, where research and development could be fostered in an environment free from commercial pressures. Their recommendation was accepted in principle and protracted discussions regarding the most suitable course of action commenced. The public demonstration of the R.C.A. synthesiser to the American Institute of Electrical Engineers, New York, on 31 January 1956 proved a major influence in the course of these negotiations, for the sudden appearance of a machine capable of generating a wide variety of sound phenomena provided a powerful focal point for the design of a studio. This machine was also quite different to any of the systems they had studied in Europe, for it offered a programmable means of controlling the performance of the various

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[157] A description of the constructional principles employed in this piece is given in Vladimir Ussachevsky, 'Notes on a Piece for Tape Recorder', <u>Musical Quarterly</u>, Vol. XLVl, No.2 (April 1960), pp.202-209.

devices. In an era where voltage control techniques were still unknown, this feature created a major advance in the development of electronic music systems, and the characteristics of the synthesiser will be studied closely in the next chapter.

Ussachevsky in particular was keen to acquire the synthesiser for use as the basis of a studio at Columbia University, and preliminary approaches were made to several R.C.A. executives. It soon transpired that Milton Babbitt, at Princeton University, was also interested in gaining access to the machine, and they agreed to collaborate so that a formal application could be made jointly from both universities. R.C.A. responded by granting access to the machine which was, for Babbitt, conveniently situated at their Princeton Laboratories, and the three composers in turn began experimenting with its facilities.

The stage meanwhile was being set for an electronic composition which was to be of the greatest musical significance: Varèse's Poème Electronique. In 1956 the electronics firm of Philips, based at Eindhoren, Holland, began to consider their plans for the World Fair, to be held in Brussels in 1958. They decided to construct a special pavillion and invited the distinguished architect Le Corbusier to prepare the design. Le Corbusier immediately seized upon the idea of combining technology and the arts by creating not merely a building, but an environment of sound, colour and structure, reflecting the creative role that electronics and associated sciences could play in contemporary society. He collaborated with Xenakis over the preparation of mathematical models for the construction of the building itself and invited Varèse to provide the music in the form of a prepared tape, leaving the composer totally free to approach the world of sound in whatever manner he should choose.

It was thus that Varèse should finally be rewarded for his years of struggle with an opportunity to compose an electronic work which explored extensively the projection of sound in space, an area of investigation which had led him to conceive his ill-fated project Espace twenty years previously.

Like Déserts, the realisation of this piece was to take Varèse to Europe, on this occasion to the Philips Laboratories at Eindhoven, Netherlands. Here he enjoyed a range of facilities which were without precedent at that time, for a complex studio system was especially assembled just for this composition, backed by a team of highly skilled electronic engineers and advisers. The work which resulted reflected a style of electronic composition unique to the composer, tied in no way to any established studio conventions. The source material included machine noises, aircraft noise, bells, electronically generated sounds, singers, and piano and organ sounds, subjected to elaborate processes of pitch transposition, filtering and modification of their attack and decay characteristics. The projection of sound was achieved by employing a three-channel tape system, fed to elaborate arrays of loudspeakers positioned in the ceiling alcoves and the walls. Visual effects, associated with the movement of sounds, were created by means of a comprehensive lighting system which produced changing patterns of coloured images on the walls.

The World Fair opened to the public on 2 May 1958, and by the time it finally closed its doors at the end of the year over two million people had visited it. Poème Electronique was thus accorded a vast audience drawn from all over the world. News of this achievement naturally spread back to the United States, and shortly after his return to New York in the autumn of the same year, a performance of the work was given in his honour in Greenwich Village. This concert unfortunately fell short of expectations, for when relayed in a small theatre over single loudspeakers, devoid of any lighting effects, the work lost a considerable proportion of its original splendour. This occasion was also to result in yet another disappointment. The concert was preceded by a lecture at the end of which Varèse proudly announced that the firm of Bogen-Presto, a division of the Seigler Corporation, had offered to provide him with a studio so that he could continue with his work. News of this received widespread coverage the next day in both the local and the national press, but the firm changed its mind and the offer was withdrawn.

Luening's and Ussachevsky's quest for a proper studio proved, however, more successful. During 1957-58 concrete proposals drawn up jointly with Babbitt were submitted to the Rockefeller Foundation, requesting financial support for the establishment of a permanent centre for electronic music. Initially they suggested that a University Council for Electronic Music should be established consisting of representatives drawn from all the institutions which were already interested or involved in working in this field. The Foundation, however, did not wish to create a situation where it might find itself faced with the prospect of sponsoring a large number of different projects, each consequently being eligible for only a small share of the total grant. The final application thus drew up a plan for a single electronic music studio to be set up between the Universities of Columbia and Princeton only, outlining in detail equipment and staffing requirements for a five year initial period, after which the universities could be expected to provide recurrent expenses.

The proposal was accepted, and a grant of $175,000 was advanced in January 1959 to the Columbia-Princeton Electronic Music Center for a large studio to be established at Columbia University and based on the R.C.A. Synthesiser, which was to be purchased from the manufacturers. Delivery of the Mark I Synthesiser was soon arranged, pending its replacement by a much improved Mark II version, delivered later in the same year.

America at last possessed a fully equipped electronic music system which was to act as a major focal point for interested composers from both home and abroad. The Center gave two inaugural concerts in the McMillin Theatre, Columbia University on 9 and 10 May 1961 before an invited audience in a blaze of publicity. The programmes included Arel, Stereo Electronic Music No.1; Babbitt, Composition for Synthesiser; Davidovsky, Electronic Study I; El-Dabh, Leiyla and the Poet; Luening, Gargoyles for Violin Solo, Synthesised Sound; and Ussachevsky, Creation-Prologue.

By the early 1960s electronic music facilities had been extensively developed in America, reaching an ever widening circle of composers. The following studios are a representative selection of the more important centres which were founded in America and Canada during the period 1948-64:[158]

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[158] This list is a more detailed version of one given in Gottfried Koenig and Fritz Wieland, Summary Introduction, op.cit., pp.35-37

NEW YORK, N.Y.	Louis and Bebe Barron Studio Opened 1948 (closed down 1961) Composers: Barron, Cage, Wolff
OTTAWA	Elmus Lab, Radio and Electronic Engineering Division, National Research Council Opened 1954 Composers: Le Caire, Anhalt
NEW YORK, N.Y.	Columbia-Princeton Electronic Music Center Opened 1955 (works from 1953 now in official archives) Composers: Luening, Ussachevsky, Arel, Babbitt, Davidovsky, Wuorien, El-Dabh, Varèse, Berio, Carlos
ANN ARBOR, Michigan	The Co-operative Studio for Electronic Music Opened 1958 Composers: Ashley, Mumma, Krumm
MURRAY HILL, New Jersey	Bell Telephone Laboratories Opened 1959 (experimental programmes from 1957) Composers: Mathews, Pierce, Tenney, Lewin, Strang, Guttman
TORONTO	Electronic Music Studio, University of Toronto Opened 1959 Composers: M. Schaeffer, Ciamaga, Cross, Aitken, O. Henry
URBANA, Illinois	Experimental Music Studio, University of Illinois Opened 1959 Composers: Hiller, Tenney, Beauchamp, Graburo, Brün, Martirano
SAN FRANCISCO, California	San Francisco Tape Music Center Opened 1960 (closed down 1966) Composers: Subotnick, Oliveros, Riley, Sender
STONY POINT, N.Y.	Opened 1960 Composers: Cage, Tudor (live electronics)
WALTHAM, Massachusetts	Electronic Music Studio, Brandeis University Opened 1961 Composers: Subotnick, Lucier, Cage, Shirley, Lerman, Behrman, Falck, Hughes
NORTHRIDGE, California	Electronic Music Studio, San Fernando Valley State College Opened 1962 Composers: Krenek, de la Vega, Gigsby

The expansion of electronic music facilities throughout the world since the early 1960s has been so extensive that a detailed study of all the developments which have taken place since this date would involve a task of formidable proportions. In 1967 Hugh Davies prepared a Catalogue which attempted to document every electronic composition which had been produced up to that year, classified by studio, composer, title, function, date, duration, number of tracks and details of commercial availability on disk or tape, where applicable. The preface includes the following pertinent observations:[159]

> With the present rapid growth of interest in the medium (witness all the new studios listed as 'under construction', which is certainly not a complete listing, particularly for the U.S.A.), such a complete catalog will never again be possible. Extrapolation of recent development shows that in five years' time electronic music will be much more widespread, and the day when every other music school, college and university music department in the U.S.A. possesses a studio may not be very far off.

The 'recent developments' referred to above concerned the introduction of the commercially manufactured voltage controlled synthesiser in the mid-1960s. This created a situation where for the first time it became possible to purchase completely self-contained systems off a showroom shelf, avoiding the considerable technological problems hitherto encountered in the construction of a studio from individually selected or assembled devices.

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[159] Hugh Davies, International Electronic Music Catalog, op.cit., p.V

The widespread adoption of these synthesisers by centres led to a proliferation of studios each offering broadly similar facilities.

At first sight this general standardisation of studio equipment would appear to be an extremely positive development, for composers were presented for the first time with a common technical basis upon which to develop their own particular techniques of electronic composition. This presupposes, however, that the range of facilities made available were sufficiently comprehensive to satisfy the very varied needs of the enlarged number of users who wished to employ them. Such universality of application unfortunately remains even today a Utopian goal, for, as will be seen in succeeding chapters, fundamental problems of musical specifications and the practicalities of man/machine interfacing have yet to be satisfactorily resolved. Until such time that the influence of a technical operating environment may be diminished to the extent that a composer may work almost exclusively in his own creative terms it must be accepted that the design of any electronic music system determines not only the range of operations which may be performed but also the way in which it may be employed as a compositional tool.

It is thus a matter of some concern that many studio entrepreneurs have been willing to base their systems on these synthesisers without first carefully evaluating the artistic consequences of surrendering all control over the design philosophy to commercial interests, inevitably motivated more by considerations of profit than musical usefulness.

Before such factors can be properly evaluated, however, it is first necessary to study the technology of voltage control operation in some detail, and this subject is considered at length in the next chapter.

CHAPTER THREE

The development of electronic control techniques for analogue studios: Voltage control and the antecedents of computer-assisted systems

The first era of electronic music, studied in the previous chapter (1948-1964), was dominated by one important environmental characteristic. Studios, although relatively few and far between for the first decade, were in general staffed by full-time teams of composers and engineers working together in close cooperation.

This situation was to change drastically during the 1960s, for the introduction of voltage-control techniques and the subsequent large-scale commercial exploitation of the voltage-controlled synthesiser led to a destruction of the natural feedback from composer to designer, with far-reaching consequences. On the one hand, the easy availability of a new and powerful range of sound generation and processing facilities made it possible for a large number of institutes, and even individuals, to establish workable studio systems with a minimum of technical expertise. Opportunities to explore electronic techniques thus became extended to many composers who otherwise would have been denied such facilities. On the other hand, a perhaps inevitable outcome of an ever-increasing mass-production of synthesisers has been an overwhelming pressure on composers to accept these commercial products as ideal means for generating and manipulating sound information.

Voltage-controlled systems, nevertheless, have played a significant role in the development of sound generation and processing facilities, for they provided an important link between the early 'classical' studios and one of the most musically productive techniques for producing electronic material; the computer-controlled analogue or hybrid studio. This progression has been characterised by an increasing use of procedural routines, replacing manual device operations firstly with electronic and subsequently with computer-assisted alternatives. Certain of these developments, as will be seen later, greatly expanded the range of system characteristics, for it became possible to manipulate devices at speeds and in ways which were hitherto hard to achieve. Some of these innovations, viewed in retrospect, have served the medium less well, creating techniques which are easy to implement but yet limited in musical usefulness.

A study of voltage-control techniques, however, does not supply a complete key to the development of hybrid systems, and there are other links which merit study. In particular the background to the R.C.A. synthesisers is of interest, for this concerns the construction of a digitally controlled sound generation system as a product of research started in the late 1940s, at a time when Schaeffer had barely started his first experiments in Paris.

During this period two electronic engineers, Harry F. Olson and Herbert Belar, both employed by the Radio Corporation of America at their laboratories in Princeton, New Jersey, became interested in the possibility of developing technological systems for both the composition and also the realisation of musical works. The company was sufficiently far-sighted to realise that such investigations might lead to useful advances not only in communication theory but also in areas of medical and acoustical research, and accordingly gave them official support for their ventures.

Olson and Belar were inspired by The Mathematical Theory of Communication[1]

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[1] C. E. Shannon and W. Weaver, The Mathematical Theory of Communication (University of Illinois Press, Urbana, Illinois), 1949

published in 1949, which led them to embark upon an elaborate research project concerned with the construction of a machine for musical composition, based on a system of random probability. The machine was completed in late 1950, but a detailed description of the project was not published until nearly eleven years later when an article appeared in the Journal of the Acoustical Society of America, September 1961.[2] Although the acoustical output of the device was limited to the production of a monophonic progression of tones, the operating system is of importance, for it attempted to rationalise some of the creative principles involved in the processes of musical composition into a series of electro-mechanical functions.

A study of the article reveals that Olson and Belar's understanding of the art and practice of music was significantly lacking in several respects, resulting in several unacceptable defects in the claimed characteristics of the system. These defects did not pass unnoticed at the time of publication, and the Journal published a vitriolic attack on their work by J. Murray Barbour in January 1962.[3]

The basis for the composing system lay in a statistical study of the characteristics of twelve Stephen Foster folk-songs. In the first instance the occurrences of di-note and tri-note pitch sequences were studied to provide a probability table. By applying random selection routines, weighted in

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2 Harry F. Olson and Herbert Belar, 'Aid to Musical Composition Employing a Random Probability System', The Journal of the Acoustical Society of America, XXXIII, 9 (Lancaster, Pennsylvania, U.S.A., September 1961), pp.1163-1170.

3 J. Murray Barbour, letter on 'Aid to Musical Composition ...', The Journal of the Acoustical Society of America, XXXIV, 1 (Lancaster, Pennsylvania, January 1962), pp.128-129.

accordance with the probability table, pitch sequences could then be generated as the basis for synthesised tunes. Seemingly unaware of the limitations of their analytical procedures and inadequacies of such a limited sample of data, the authors surmised that 'a composing machine employing a random probability system based on Stephen Foster songs may be developed, which will produce new music similar to that of Stephen Foster songs',[4] and proceeded to build it.

Doubts about the musical validity of their claims are strengthened by the superficial consideration afforded to the rhythmic structure of the songs. The authors declare merely that 'Rhythm must be incorporated into a succession of notes in order to make a satisfactory melody',[5] and supply their own rudimentary probability table for bars in $\frac{4}{4}$ and $\frac{3}{4}$ mètre, containing just seven variations each. This short-sightedness is compounded by serious errors in the published table such that two of the $\frac{4}{4}$ and six of the $\frac{3}{4}$ patterns do not add up correctly to give complete bars.[6]

$\frac{4}{4}$ TIME	$\frac{3}{4}$ TIME	PROBABILITY
𝅝	𝅝 (sic)	$\frac{2}{16}$
𝅗𝅥 𝅗𝅥	♩ ♩ ♩	$\frac{4}{16}$
𝅗𝅥 ♩ ♩	𝅗𝅥 𝅗𝅥 (sic)	$\frac{2}{16}$
♩ 𝅗𝅥 𝅗𝅥 (sic)	𝅗𝅥 𝅗𝅥 (sic)	$\frac{2}{16}$
♩ ♩ 𝅗𝅥	𝅗𝅥 𝅗𝅥 (sic)	$\frac{2}{16}$
♩ ♩ ♩ ♩	𝅗𝅥 𝅗𝅥 (sic)	$\frac{2}{16}$
𝄻 (sic)	𝄻 (sic)	$\frac{2}{16}$

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[4] Harry F. Olson and Herbert Belar, 'Aid to Musical Composition ...', op.cit., pp. 1165-1166

[5] Ibid., p.1165

[6] Ibid., p.1161

Even the di-note and tri-note pitch tables display many errors and omissions when their contents are matched against the twelve named songs upon which they are supposed to be based. These are painstakingly listed by Barbour, who also clearly demonstrates that a sample of 'typical' Stephen Foster music, generated by the machine on 30 June 1951 and quoted in the article, is not only musically unacceptable but also incompatible with the rhythmic probability tables. Having carried out his own analysis of five of the songs, Barbour further observes that[7]

> As to the frequency of the legitimate patterns in $\frac{4}{4}$ meter ... the first two patterns (whole note and two half notes) do not occur at all in the five songs examined, although the authors state that their general probability in a given bar is $\frac{2}{16}$ and $\frac{4}{16}$ respectively.

Overriding these data errors is a far more fundamental criticism of the musicality of the whole approach to the project. Barbour draws attention to simple yet highly essential features in the design of Stephen Foster songs, which were totally overlooked by Olson and Belar.[8]

> Foster melodies are not haphazard assemblages of bars of music. They consist of two bar phrases, combined into four bar phrases. Usually the third phrase is identical with the first, and the fourth is often practically the same as the second except for the cadence. Often the entire four bar phrases are repeated literally to form the 'verse' ... The strongly unifying principle in this somewhat rigid design is the harmony. Foster

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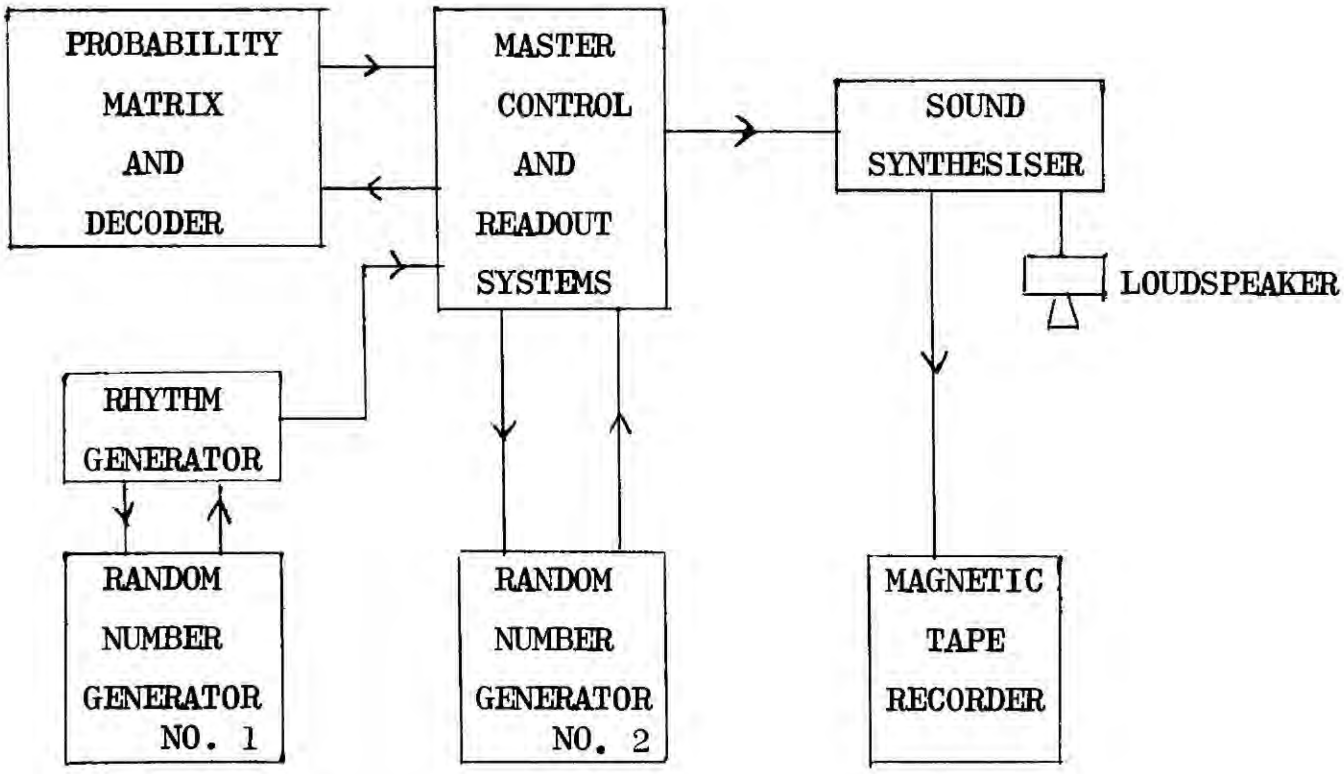
[7] J. Murray Barbour, <u>op.cit.</u>, p.128

[8] <u>Ibid.</u>, p.128

> used certain well-defined harmonic progressions to mark the beginning, middle and ending of his phrases and periods. These patterns repeat themselves more consistently than do the three note melodic sequences, which in large measure they determine.

Olson and Belar's attempt to devise a composing machine for the production of Stephen Foster tunes was thus based on several misconceptions regarding the intuitive processes of musical composition, and the relevance of statistical analyses and probability theory to their simulation. These may be evaluated more fully by first studying the characteristics of the machine itself.

The block diagram of the machine is given as follows:[9]



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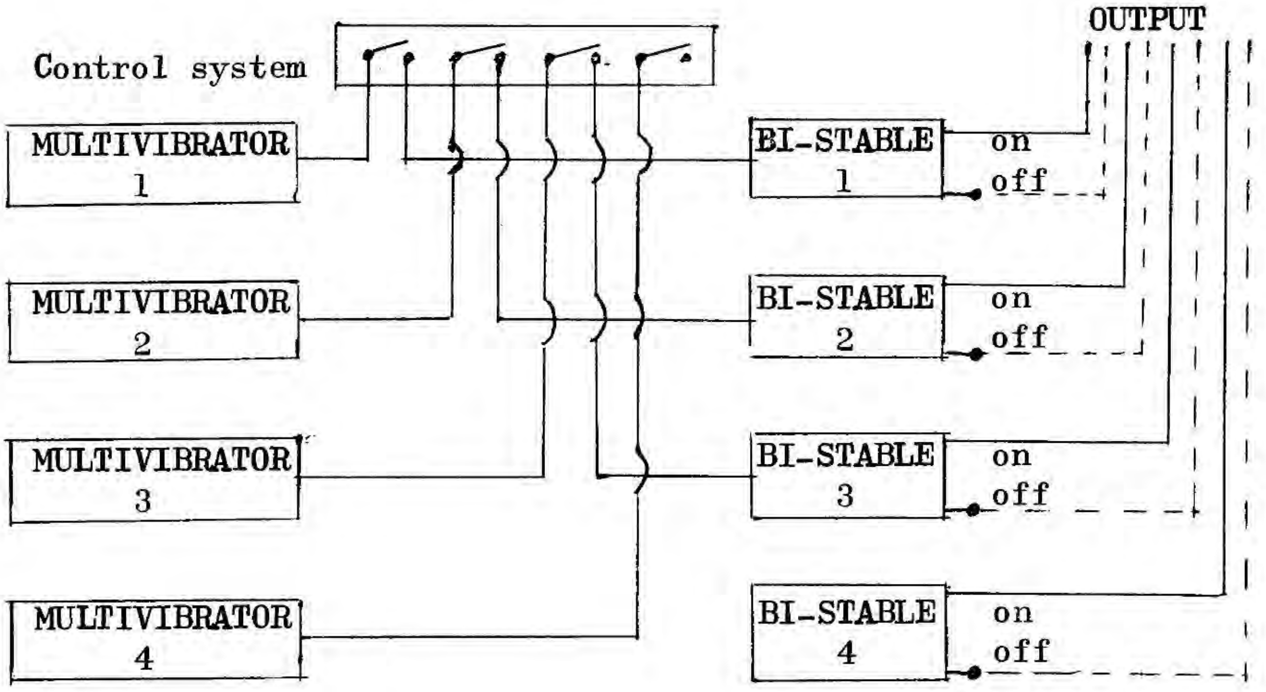
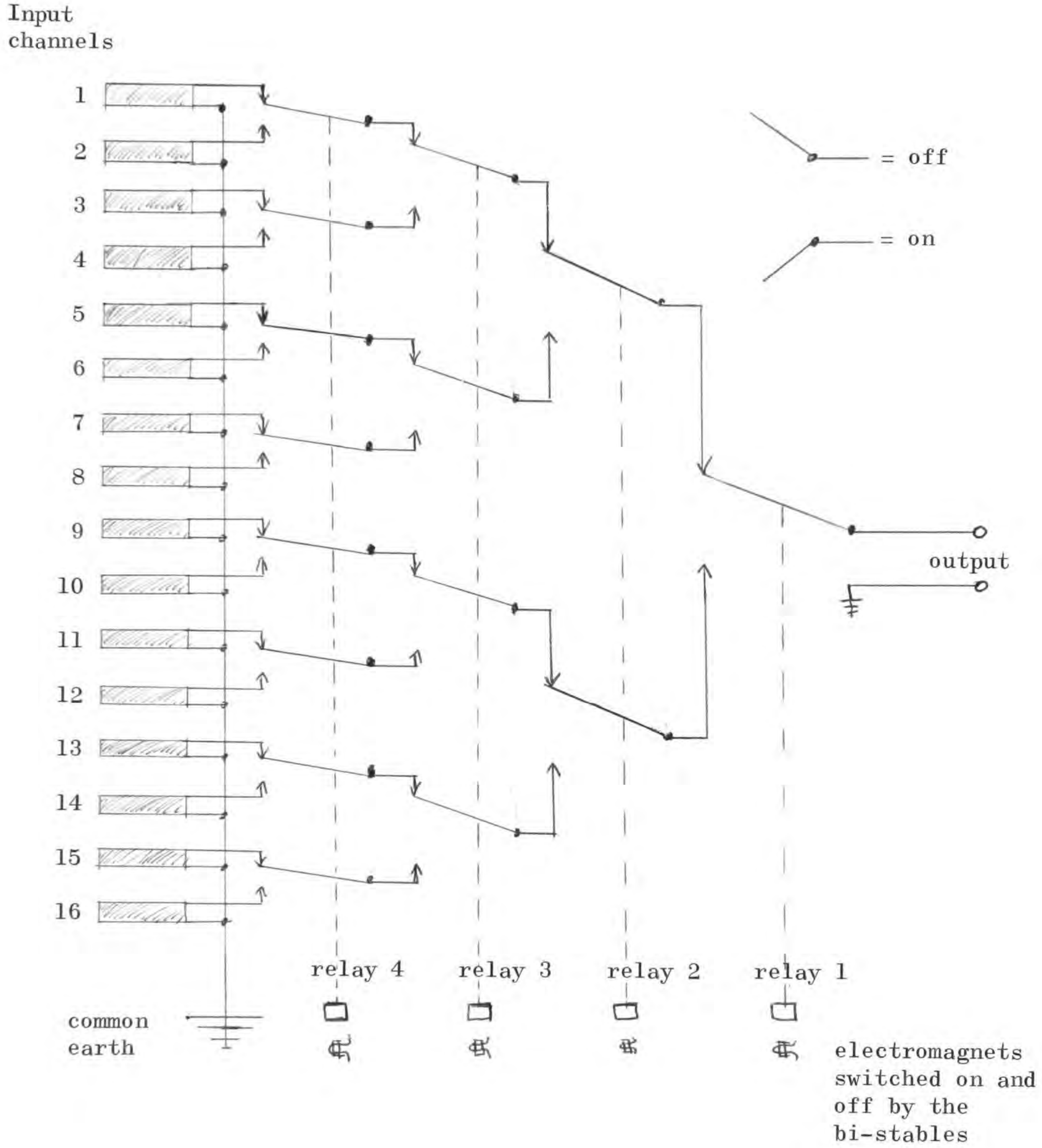
[9] Harry F. Olson and Herbert Belar, 'Aid to Musical Composition', <u>op.cit.</u>, pp.1166, fig.3

Each random number generator consists of a group of four free-running multivibrators set to oscillate at slightly different frequencies within a range limit of 1000 to 1500 Hertz. Each multivibrator is connected via a control system to a complementary bi-stable multivibrator. Whilst a connection is being made, the tubes of the latter are rendered alternately conducting. When the connection is broken the tube activated at that instant is left conducting.



In view of the high frequency of the source multivibrator in relation to the switching speeds employed by the machine, the polarity of each of the bi-stables at the point of selection is effectively random. By treating the outputs of the latter as a series of 'on' or 'off' states, the patterns produced may be employed as a four-bit (or element) digital control code, offering a total of $2^4 = 16$ different combinations, decoded by means of a relay tree to provide 16 separate switched connections.



The above diagram shows all four relays in the 'off' position, providing a connection between input channel 1 and the output. Turning any relay 'on' causes all the two-pole switches on that relay line to change over to their alternate setting. The digital codings associated with each of the input channels are as follows: '0' indicates a relay in an 'off' condition, and '1' indicates a relay in an 'on' condition:

Input channel connected	Relay states			
	No. 1	2	3	4
1	0	0	0	0
2	0	0	0	1
3	0	0	1	0
4	0	0	1	1
5	0	1	0	0
6	0	1	0	1
7	0	1	1	0
8	0	1	1	1
9	1	0	0	0
10	1	0	0	1
11	1	0	1	0
12	1	0	1	1
13	1	1	0	0
14	1	1	0	1
15	1	1	1	0
16	1	1	1	1

In the case of random generator no.1, the relay tree is employed to select one of the rhythmic patterns obtainable from the rhythm generator unit, activated via a series of motor driven switches. If, for the purposes of this functional description, the disputed patterns in the rhythmic probability table quoted earlier are assumed to have been suitably corrected, a total of seven different patterns, either in $\frac{3}{4}$ or $\frac{4}{4}$ time are available for selection. The random number generator and its associated relay tree, however, are designed to operate on a total of sixteen possible source channels. If only seven lines are available for connection to the system, the remaining nine inputs must either be left to record 'null' functions or be paralleled with live lines to provide alternative selection codes for particular functions. The article fails to state clearly which of these alternatives is employed, but it may be deduced from the probability table that the latter technique is adopted, two codes being assigned to six rhythm patterns and four to the seventh, weighting the probability function accordingly.

A series of motor driven rotary switches provides a control system between the random number generator and the rhythm generator unit. These are designed to make and then break the connections between the four multi-vibrators and bi-stables in a quick sequence towards the end of each bar. The digitally coded random number thus created is then fed to the relay tree, which selects a pattern from the rhythm generator at the beginning of the next bar.

The output of the rhythm generator passes to the master control unit where it is used to provide timing pulses for the pitch sequences. Each new pulse activates a bi-polar switch which operates the four internal make-and-break connectors to random generator no.2 sequentially. The latter accordingly produces a random, four-bit output control code which, like generator no.1, is used to activate a 16-pole relay tree.

The logic employed in the next stage of pitch generation is only partially explained, and the article would appear to contain an unfortunate error. According to Olson and Belar[10]

> The 12 outputs of the relay tree are connected to the probability matrix and decode switch. The 12 outputs represent the notes of the scale shown in fig. 1.[11] Over a long period of time the 12 outputs are activated the same number of times but in a random fashion, because the activation depends upon the output of random generator no.2.

Earlier in the article it is stated that 'the relay tree in the random unit of the composing machine has 16 output contacts',[12] translating a four-bit control pattern into a series of discrete connections as described above. If only twelve lines of the relay tree were employed for this processing function, four possible selections by the random number generator would remain unaccounted for. Furthermore, as will be seen shortly, the input to the probability matrix is based on a table expressed in sixteenths, and this demands control from the complete range of relay outputs.

The matrix itself is based around a fifty-position rotary stepper switch,

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[10] Harry F. Olson and Herbert Belar, 'Aid to Musical Composition ...', op.cit., p.1167

[11] The instrument generates the following pitches only; all source tunes for the probability matrix and all output syntheses being handled in D major. Only G♯ was considered a necessary addition to the diatonic scale:

b c♯ d e F♯ G G♯ A B C♯ D E

[12] Olson and Belar, op.cit., p.1164

each position being uniquely assigned to one of the fifty di-notes (two-note pitch sequences) encountered by Olson and Belar in their analyses of the twelve songs. The current di-note at any stage in the synthesis process is supplied by the two notes which have been generated by the two preceding machine cycles.[13] At the beginning of a cycle the stepper switch rotates until it reaches the position allocated to the current di-note, whereupon the associated circuitry routes the sixteen lines of the relay tree to decode switches for one or a selection of the twelve available pitch generators according to a hardwired probability table.[14] On receiving an interrupt from the master control unit, random generator no.2 responds by sending an activating pulse along one of the relay tree lines to the probability matrix. The latter routes the pulse to a decode switch which in turn switches on the appropriate pitch generator. The composition of the new di-note is transmitted to the stepper switch and the process recycles.

This pitch selection technique may be illustrated more clearly by considering a specific example. If the current di-note at a selected instant consists of the pitches <u>b</u> and <u>d</u>, the probability table indicates that there is a 16 in 16 chance, that is a complete certainty, of the next note being a <u>d</u>. The circuitry for stepper switch position <u>bd</u> thus connects all sixteen lines from the relay tree to the same destination: the decode switch for the <u>d</u> pitch generator. When the master generator interrupts the random number generator the matrix wiring guarantees that whichever line is

[13] No information is given concerning the source of the very first di-note in a synthesis operation. It might be presumed that this is either specified manually, or supplied automatically by the last di-note of the piece produced previously on the machine.

[14] The contents of the probability table were derived from the twelve songs by preparing a statistical table of the pitches following every occurrence of each of the fifty di-notes. These ratios were then converted proportionally to serve as probabilities, expressed in sixteenths.

activated, the effect will be to switch the *d* pitch generator on. The new di-note, *dd*, now becomes the current di-note, and the stepper switch automatically repositions itself to the appropriate setting. The probability table now indicates that the following note options and weightings should apply: *c♯*: $\frac{2}{16}$, *d*: $\frac{2}{16}$, *e*: $\frac{9}{16}$, *F♯*: $\frac{2}{16}$, and *G*: $\frac{1}{16}$. The matrix wiring interprets this by directing one line from the relay tree to the *G* decoder switch, two each to the *c♯*, *d* and *F♯* switches, and the remaining nine to the *e* switch. On receiving an interrupt the random generator responds by sending a pulse down one of its lines, to be channelled by the matrix towards one of the five selected tone generators.

Such a machine offered considerable potential as a tool for experimenting with statistically based compositional procedures. Today, the much faster and far more powerful processing facilities of third generation digital computers are generally available for such exercises. Twenty-five years or so ago, the computer revolution had barely begun, and in its time the Olson and Belar machine offered a unique system for creating pitch sequences. The unsoundness of the premises upon which the whole project was based cannot be overlooked, however, for it provides a disturbing example of the consequences of a breakdown in communications between scientists and artists concerning the essential characteristics of each others' disciplines.

Olson and Belar completely misunderstood the processes of musical composition in one particular respect, for they assumed that objectively expressed characteristics derived from an analysis of a selection of a composer's output of scores could supply the information necessary for a simulation of the intuitive processes which inspired them. It would be profitable to refer to the work of Leonard Meyer in this context; in particular to his essay 'On the Nature and Limits of Critical Analysis' which appears in his book Explaining Music.[15] Meyer's following observations reveal all too clearly several vital factors of musicality which Olson and Belar ignored in their attempt to synthesise a composer's style by applying probability theory to a limited data sample of his output:[16]

> To understand a composer's choices is to envisage the psychological-stylistic alternatives open to him at a particular point in the composition. For this reason, particularly in the short run, our guesses about implications and continuations may be partly or wholly mistaken. Ends are generally more accurately envisaged than means. And the predictable route which suggests itself to the critic will not as a rule be one chosen by the composer. Their invention is both more subtle and adventurous than our's - which is why they, and not we are creators ... Even in the long run our most confident surmises about routes and goals may prove wrong. This is because given the particular style within which he works the composer is a free agent. He invents and shapes his initial musical substance - his themes, harmonic progressions, textures and the like . These have implications for subsequent events. But they do not determine them ... Determinism is a mistaken notion applied to works of art not only because implications are plural, but also because, within the style he employs, the composer may at any particular point in a piece be absolutely arbitrary. That is he may invent and use a musical idea or relationship which has nothing to do with - was in no way implied by or dependent upon - preceding events in the piece.

Later in the same essay, Meyer succinctly discounts theories of musical creativity such as those put forward by Olson and Belar:[17]

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[15] Leonard B. Meyer, 'On the Nature and Limits of Critical Analysis', Explaining Music (University of California Press, Berkeley, Los Angeles, London, 1973), pp.3-25

[16] Ibid. pp.19-20

[17] Ibid., pp.22-23

> It is important that 'understanding the choices made by the composers' does not mean knowing what actually went on in the composer's mind when he wrote a particular work. Probably neither he, nor we will ever know his mental processes as they actually occurred. Even when a composer was conscious of making a decision - when his habit of craft was not immediately adequate to the problem in hand - choice may have been largely intuitive. After considerable thought, trial and error, experimentation, and just plain day dreaming, the right solution may have appeared, as it were, out of the blue - often when least expected. The result may seem clear and 'logical', but the route followed in reaching it may well have been veiled and circuitous. In other words, just as there is a difference between the logical steps through which a scientific argument is presented and the act of scientific discovery which may have been the result of unconscious process, owing something to training, to disposition, to the current state of the discipline, and frequently quite a bit to chance; so there is a difference between the coherence and consistency of a completed composition and the composer's creative processes which depend upon a combination of training, tradition, personality, and, again, plain luck.

The above observations are of major relevance to the central theme of this thesis, for it has been the barrier between creative thought and objective interpretation which has proved the greatest stumbling block for musicians directly concerned with the realisation of their own compositional ideas in an electronic music studio. It will be seen in due course how increasing technological sophistication in recent years has not always provided the answers to these major problems of man/machine interfacing.

Certain design features of the random probability machine, in particular the use of relay trees to control device functions, were incorporated in Olson and Belar's next and better-known project: the R.C.A. synthesiser Mark I, first presented to the public in 1956.[18] This event was marked

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[18] See chapter 2, p.190

by the publication of an article by the designers in the May issue of the Journal of the Acoustical Society of America.[19] Work on the project had commenced several years earlier, and the synthesiser was at least partly operational in 1952.[20] In this same year R.C.A. had set up a new, especially equipped research studio for Olson and Belar in their laboratory for Acoustical and Electromechanical research at Princeton, New Jersey. This development, chronologically, just preceded the founding of the Studio für elektronische Musik at Cologne, but it should be noted that the artistic potential of the former had scarcely been explored when, seven years later, the synthesiser at last attracted widespread attention in musical circles concerning the adoption of the Mark 2 version by the Columbia-Princeton Electronic Music Center[21] as the centre piece for their system.

Olson and Belar's account of this machine is more carefully prepared than their description of the random probability machine discussed earlier. It also gives a clearer insight into their outlook on the uses of electronics in music. This feature will require further study in due course, for their philosophy naturally influenced the design of the machine itself.

The overall design of the Mark I synthesiser is as follows:[22]

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[19] Harry F. Olson and Herbert Belar, 'Electronic Music Synthesiser', Journal of the Acoustical Society of America, XXVII, 3 (Lancaster, Pennsylvania, U.S.A., May, 1955), pp.595-612

[20] Olson and Belar's list of completed synthesis exercises dates back to 1 April, 1952, see Olson and Belar, 'Electronic Music Synthesiser', ibid., p.611

[21] See chapter 2, pp.193-194

[22] Olson and Belar, 'Electronic Music Synthesiser', op.cit., p.597, fig.2

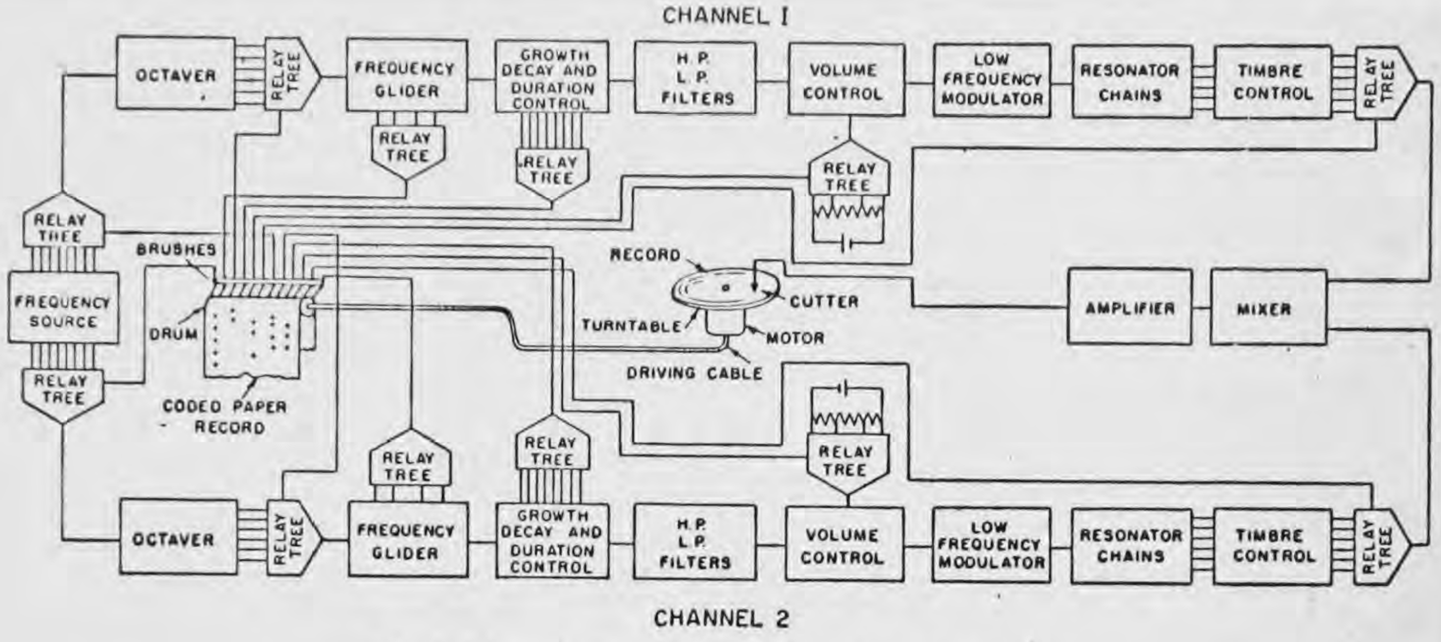
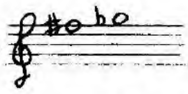
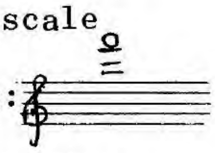
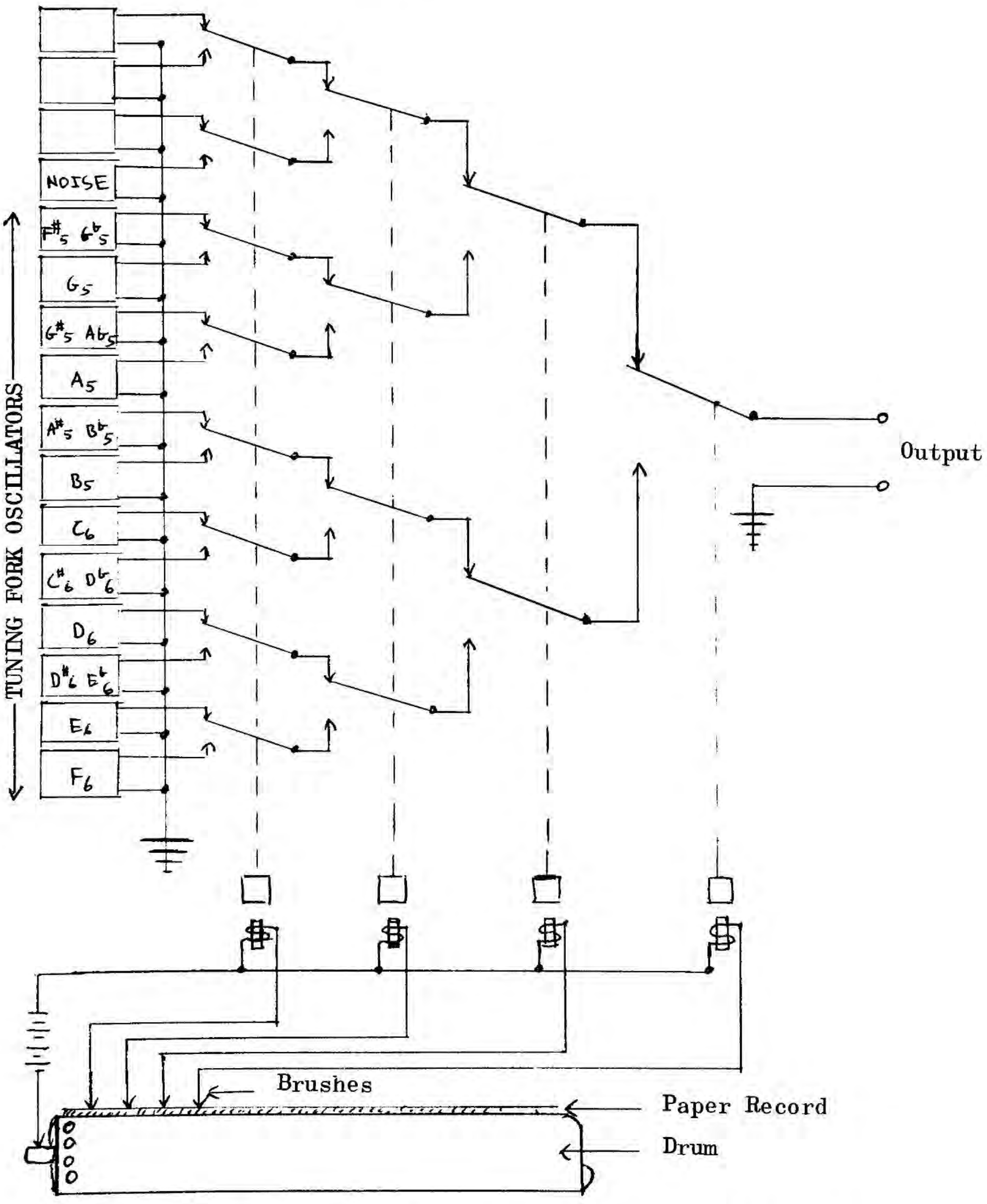


FIG. 2. Schematic diagram of the electronic music synthesizer.

FIG. 2. Schematic diagram of the electronic music synthesizer.

Two identical but functionally independent synthesis channels are provided, each sharing a common sound source bank. This consists of a white noise generator and twelve electrically driven tuning fork oscillators, designed to generate the twelve tempered pitches of the musical scale between F♯/G♭ = 739.989 Hertz: , and F = 1396.913 Hertz: . Accessing of these generators and control of the subsequent processing operations is achieved via a set of relay trees, activated by brush sensors which detect patterns punched in a steadily moving reel of paper tape. The latter facility permits a composer to programme his work as a series of punched instructions, and this aspect will be discussed in more detail later.

The first programmable function in each synthesis channel involves the selection of a sound source generator via a 16 pole relay tree which responds to digitally encoded instructions contained in four punched tape columns. The three spare relay tree connections are available for the addition of extra sound sources should these be required. The following diagram of the frequency selection network for one channel shows clearly how the control system is an adaption of the relay tree techniques employed by Olson and Belar in their earlier random probability machine.[23]



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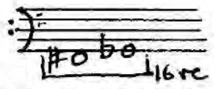
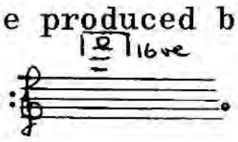
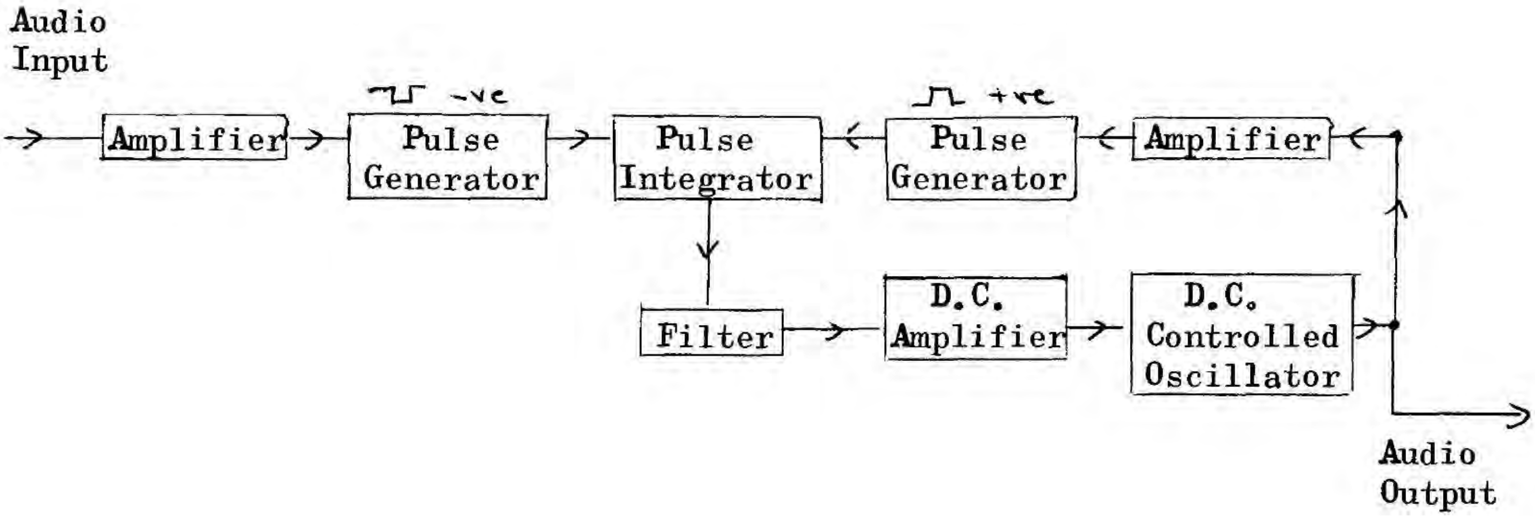
[23] Olson and Belar, 'Electronic Music Synthesiser', op.cit., p.598, fig.4

The next stage in the synthesis process involves the use of an octave selector, by-passed only when the white noise generator or the external inputs are selected. This device, operated by an 8-pole relay tree and an associated three-column punched tape record, accepts a frequency from the source bank, squares the sinusoidal wave, and passes it on to a set of octave multipliers and dividers. By combining the octave selection and source generator selection procedures, an eight-octave range of tempered pitches may be produced between F♯/G♭ = 23.124 Hertz: 

, and F = 5587.65 Hertz: 

. The wave is finally converted electronically from a square function - composed of odd harmonics only - to a sawtooth function - composed of both odd and even harmonics.

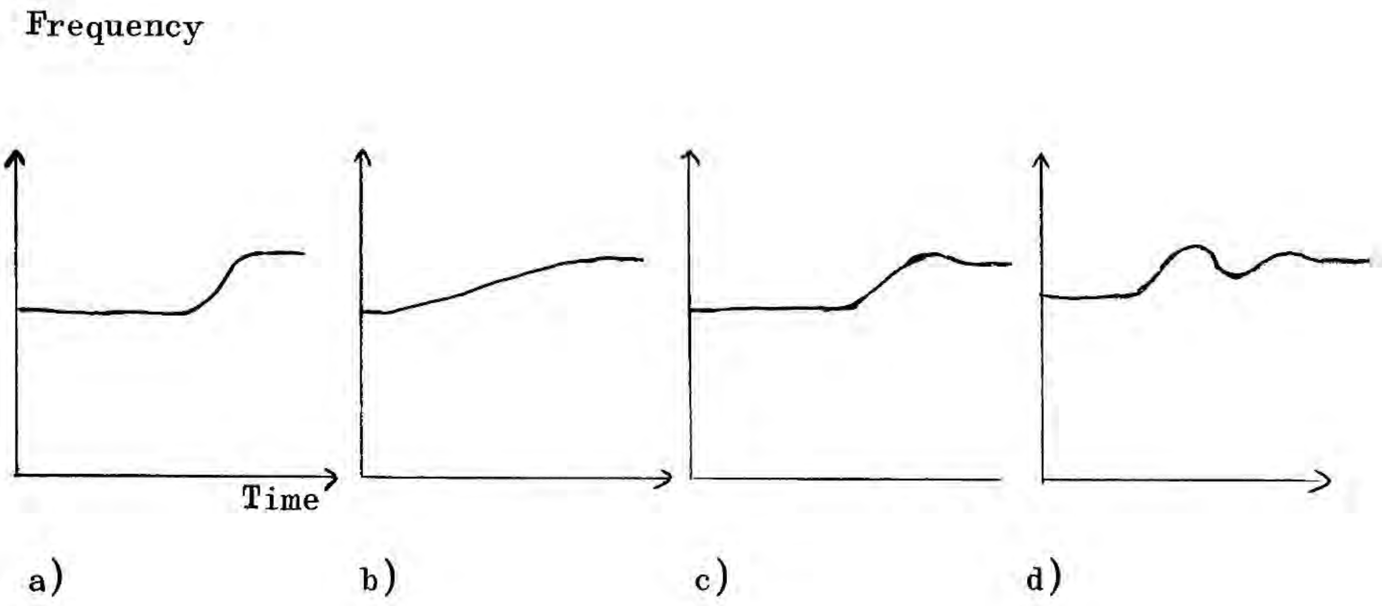
The output of the octave selector may, optionally, be connected to another device: a frequency glider, which permits a smooth transition to be made from one pitch selection to another. The electronic techniques employed here are of interest, for they involve the use of a control system to determine the frequency of a special oscillator. The incoming sawtooth wave is amplified and converted into a pattern of negative pulses. These are passed through a low-pass filter to produce a direct current signal, proportional in voltage to the applied frequency. This signal is fed to a D.C. controlled oscillator which generates a sawtooth function matching that of the incoming wave. The output frequency of the oscillator is locked to that of the source via a controllable feedback network, operating between the final output and the D.C. control system. This amplifies the oscillator signal and converts it into a series of positive pulses which are applied to one side of an integrator. Information for the other side of the integrator is obtained by tapping the line from the input pulse converter to the D.C. control input. The output of the integrator is then passed via a filter to the D.C. controlled oscillator, as a fine control.



Whilst the source remains at a fixed frequency setting the output oscillator is at the same frequency as that of the input. The output from the pulse integrator thus remains at zero, the positive pulses exactly cancelling the negative pulses. A change in the source frequency upsets this equilibrium, for an imbalance is created in the pulse rates applied to the integrator. The latter responds by generating a voltage, which is transmitted to the D.C. control system. In a series of approximations the D.C. oscillator adjusts its setting until, once again, the frequency of output matches the frequency of input. By providing electronic controls to vary both the transition time and also the way in which the new setting is approached, a series of frequency glides may be produced.

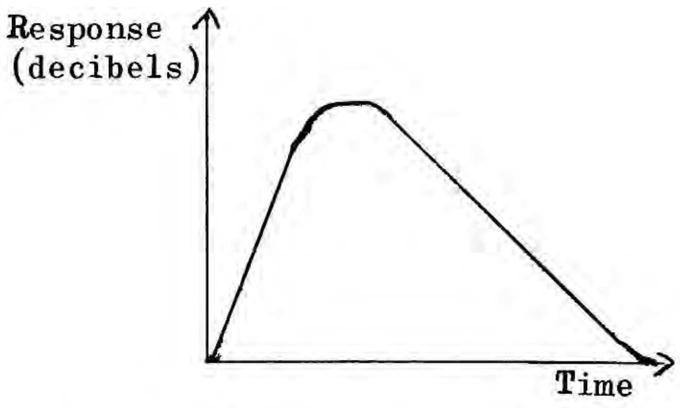
The types of transition available range from a progressive pitch change to a damped oscillatory pattern:[24]

[24] Olson and Belar, 'Electronic Music Synthesiser', <u>op.cit.</u>, p.603, fig.15



The frequency glider functions as the last device in the sound generachain. All the signals generated in this section of the synthesiser are passed to the first processing stage. This is concerned with the shaping of sound amplitude as a function of time between zero intensity and a pre-set maximum level to produce attack and decay characteristics. Eight different patterns may be selected using an 8-pole relay tree, activated by a three-column punched code in the master control tape. Each selection involves a different set of time constants, creating attack times which vary from 1 millisecond to 2 seconds, and decay times ranging from 4 milliseconds to 19 seconds.

In addition to simple attack and decay functions based on exponential characteristics, for example[25]



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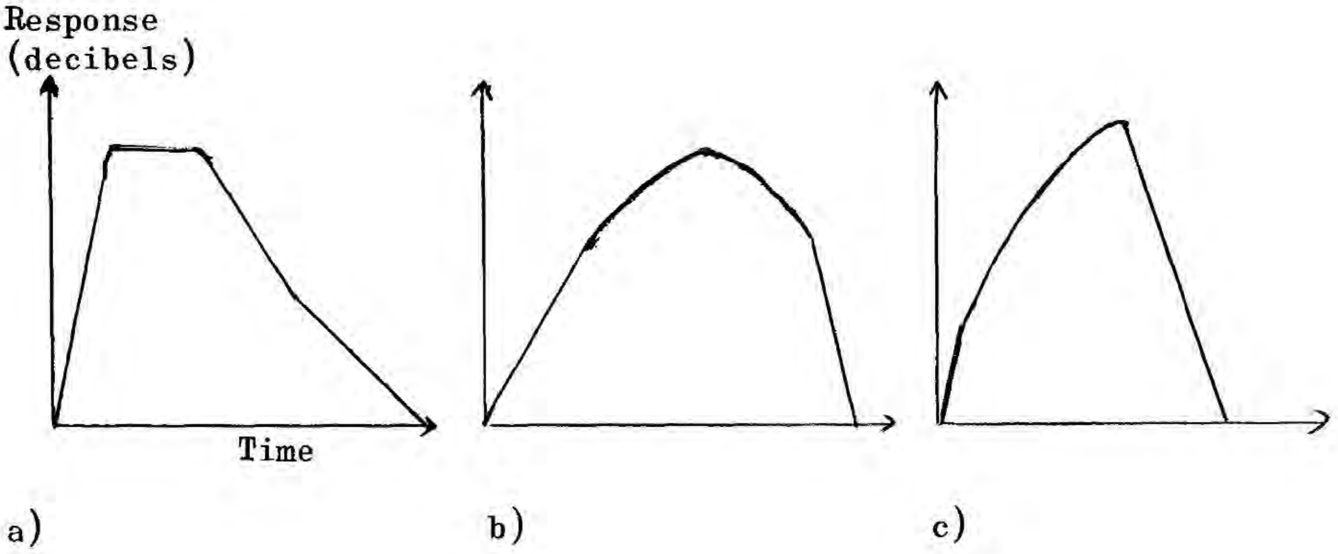
[25] Olson and Belar, 'Electronic Music Synthesiser', *op.cit.*, p.602, fig.12

functions of a more artificial nature may be specified. Furthermore, by changing the punched tape code during the course of the envelope, the functions may be combined to produce hybrid curves:[26]



The signal passes next to a pair of filters supplying high pass, low pass or band pass characteristics according to the position of a selector switch. Eight different centre frequency (or frequency cut-off) settings may be selected via four punch columns in the master tape control system, although the existence of this facility is not clearly indicated on the system diagram.[27] These filters suffer from one particular disadvantage: they cannot be made to track the incoming signal automatically. As a consequence it is not possible to produce a pitch scale which employs a consistent harmonic spectrum.

The output from these filters is connected to a programmable, general amplitude control system. This device, operated by a relay tree and an

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[26] Olson and Belar, 'Electronic Music Synthesiser', <u>op.cit.</u>, p.602, fig.12

[27] See page 215

associated four-column punched tape code, provides a total of sixteen different degrees of attenuation. Pre-sets are provided for regulating the attenuation range over which the device may be operated, and the circuits are arranged to ensure that these settings create equal steps of intensity between the two selected limits. The restrictions arising from a switched attenuation system offering only sixteen steps of intensity are of some importance, for unless the extreme operating limits are kept within a range of fifteen decibels or less, each step will be perceived as a distinct jump in volume level.

It may be deduced from the system diagram and general description that the designers intended this amplitude control system to be used as an ancillary to the attack and decay circuitry, acting in this capacity to provide sixteen different levels of maximum intensity for each enveloped event. If the synthesiser is used to imitate conventional instrumental textures by producing a sequence of articulated musical pitches, these operational characteristics may well prove adequate, if not entirely ideal. Composers, however, are usually more concerned to manipulate amplitude levels discretely as a major procedural feature in their work, and this requires the provision of far more comprehensive response characteristics. Such a facility could have been provided if the amplitude control system had offered a complete range of intensity settings. Taking into account the characteristics of the punched master control system, a six-channel tape code could have been programmed to control sixty-four different intensity settings, or a seven-channel tape code one hundred and twenty-eight; more than sufficient in the latter instance to supply a finely differentiated series of levels over the complete dynamic range of the synthesiser.

The next stage in the processing chain provides a facility for the production of low frequency modulation. The device employed here consists of an amplitude modulator operating at a frequency of about 6-7 Hertz with a modulation pattern which approximates to a sawtooth wave. The use of such an artificial vibrato technique to 'humanise' electronic sounds has been discussed earlier,[28] and the inclusion of such a facility in the R.C.A. synthesiser Mark 1 offers evidence to support the view that Olson and Belar were more concerned with the construction of a machine to imitate conventional instrumental sounds than with the provision of a comprehensive sound generation and processing system for musical composition.

The last device consists of a bank of eight electronic resonators, employed to augment the timbre shaping facilities offered by the earlier described pair of filters. Each resonator consists of an amplifier equipped with a regulated feedback loop, tuned via a resistive/capacitive network to respond to excitation at a particular frequency. The effect on any incoming signal is to boost in strength any frequency components at, or very close to the point of resonance, creating a band pass type of response. By reducing the overall gain of the amplifier so that the output emerges at a more normal system level, the circuit will act as an attenuator for frequencies outside the pass band. The sharpness of the band response may be controlled by varying the degree of feedback. By means of a toggle switch the resonator function may be inverted to produce a notched filter response, attenuating instead of boosting the frequency band. The frequency of resonance itself for each unit may be switched manually to a range of

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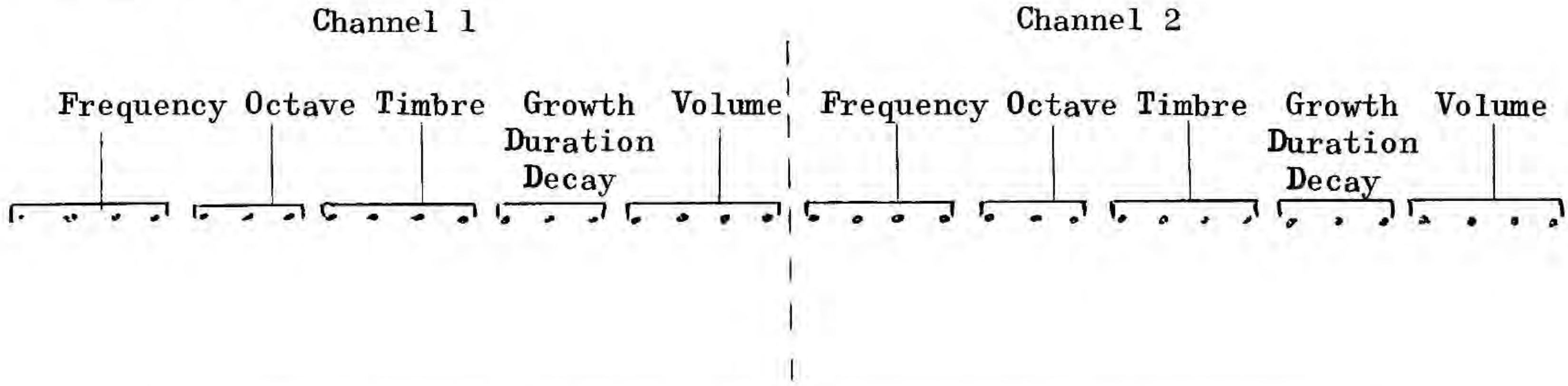
[28] See chapter 2, page 155.

settings in steps of approximately one-third of an octave. Selection of these units may be programmed via the master control system, by removing plug connections between the pair of filters and its associated relay tree and substituting ones to the resonators.

The master control system provides the primary means of communication between composer and synthesiser in the realisation of a work. It may be seen from the preceding description of individual devices that adjustments to manually operated controls are necessary at several strategic points in the processing chain. These, however, are essentially pre-operative functions, determining the range of sonological characteristics upon which the programmed sequence of instructions will draw.

The punched paper tape is fifteen inches wide, carrying device control information coded in thirty-six columns, eighteen for each channel, arranged as follows:



The tape is sprocketed on both edges to engage with matching teeth on a circular drum, the latter being linked to a motor with a variable speed drive which transports the tape at speeds ranging from two to eight inches per second.

A key-punching system permits the composer to enter his instructions as a sequence of punched hole patterns. Duration thus becomes a function of distance measured along the paper tape, according to the speed at which it is to be played. The punching mechanism ensures that holes are spaced equidistantly, the tape advancing a fixed distance after each complete row of punches has been entered. The tape reading system consists of a set of spring loaded brushes which press down onto the surface of the paper. When a hole passes under a brush, the latter makes electrical contact with the drum beneath, activating the relay line assigned to that particular punch column. Several springs are connected to each brush to ensure that electrical contact is maintained until the tape has moved to the position for the next punched row. If a new hole is detected the circuit remains complete. If no hole is encountered, however, electrical contact is broken and the relay line returns to its 'off' state.

The output from the two synthesiser channels may be monitored via a pair of loudspeakers and also recorded directly onto disc. The choice of such a cumbersome recording medium, where a groove once cut cannot be re-used, contrasts sharply with the programming flexibility of the punched tape input system. The Radio Corporation of America, however, as a company with an extensive interest in the gramophone industry, was naturally interested in the development of specially designed disc cutting equipment for the synthesiser. It must also be appreciated that the use of taping systems was under-valued in many branches of the recording industry at this time, and there was a tendency to cling to and seek improvements in traditional techniques rather than to explore new ones. Another factor may also have influenced this choice. The use of a gramophone turntable, with its comparatively low speed of rotation and high inertia, facilitated the use of a direct drive system, powered via a flexible cable from the motor employed to transport the punched control tape. This simple linkage between input and recording systems ensured accurate mechanical synchronisation between the sequential control instructions supplied by the former, and the acoustical product registered by the latter.

The disc cutting lathe was nevertheless extremely inflexible and wasteful in its operation, and it is surprising to discover that this recording system was retained in the Mark 2 version supplied in 1959 to the Columbia-Princeton Electronic Music Center.[29] By this time multi-track tape recorders were widely accepted as standard items of recording studio equipment, and the engineers at Columbia quickly replaced the direct drive disc cutter with an electronically synchronised four-channel Ampex tape recorder.[30]

The mechanical ingenuity of the R.C.A. engineers should not be overlooked, however, for they tried within the limitations of the medium to provide a practicable re-recording system. The cutting lathe employed a 16-inch lacquer rotated at $33\frac{1}{3}$ revolutions per minute. Instead of a single groove, six concentric tracks could be cut, each providing a maximum of about three minutes' recording time. By encoding both synthesiser channels at the same time, only three operations were required to produce six multi-tracked channels of sound. A second cutting lathe, driven by the same motor, provided a means of blending the six channels together as a single track on a new lacquer, using a set of six playback pickups and a mixing

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[29] See chapter 2, pages 193-194

[30] Milton Babbitt, 'An Introduction to the R.C.A. Synthesiser', Journal of Music Theory, VIII, 2 (School of Music, Yale University, New Haven, Conn., U.S.A.), pp.257-258

system. By repeating this process this secondary lacquer could be used to record thirty-six channels of information, supplied from six primary lacquers. Similarly, six secondary lacquers could be employed to provide 216 separate channels on a single tertiary lacquer.

The three minutes of recording time per channel available on each disc compared slightly unfavourably with a maximum control tape time of about four minutes when the latter was run at its slowest speed. The assembly of complete pieces involved the playing of a sequence of completed lacquers alternately on the two drives, considerable manual dexterity being required to ensure smooth joins between sections. The eventual abandonment of the disc cutting system in favour of reel-to-reel tape not only facilitated the use of splicing and editing routines but also permitted the overlapping of sections during the processes of montage by commencing individual track recordings at different positions. The restriction on the length of each run on the synthesiser nevertheless could not be removed, for it was not practicable to increase the capacity of the control tape system significantly.[31]

The use of a punched tape control system for the R.C.A. synthesiser created a landmark in the development of electronic music systems. Despite the inevitable shortcomings of an all-mechanical approach, the system offered for the first time a programmable approach to the constituent processes of electronic sound production. The ability to alter individual parameters in a composite sequence of events, by pasting over existing holes and punching new patterns, facilitated certain operational procedures

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[31] Four minutes' running time at a roll speed of eight inches per second would require a reel of paper 160 feet long.

which in a 'classical' electronic music studio would prove awkward if not at times impracticable. The construction of a succession of enveloped and timbrally shaped pitches at Cologne using discrete oscillators, for example, required an elaborate sequence of manual device settings, followed by a series of recording and splicing routines, whereas the same sequence of events might be conveniently punched onto a control tape for the R.C.A. synthesiser in a fraction of the time. The availability of such facilities, however, does not in itself result necessarily in a more musically useful synthesis system, and it is important to evaluate carefully both the benefits and the disadvantages which result from the introduction of such features.

The article by Olson and Belar on the R.C.A. synthesiser Mark 1, studied above, reveals at least in part the design philosophy upon which the machine was based, and several aspects of the latter merit closer examination. The abstract which prefaces their account, for example, claims that 'The electronic music synthesiser provides the musician, musical engineer and composer with a new musical tool, with no inherent physical limitations.'[32] This suggestion is further expanded in the main introduction which states that the synthesiser is capable of generating 'any tone produced by a voice or musical instrument by employing an electronic system.'[33]

These attributes concerning the capabilities of the synthesiser are very misleading, for although the control system provides for the specification of a substantial number of different sonological events, the devices

[32] Olson and Belar, 'Electronic Music Synthesiser', op.cit., p.595

[33] Ibid., p.595

themselves cannot be employed to synthesise adequately several important characteristics associated with naturally produced musical sounds. The use of filter networks acting upon a sawtooth wave source offers a means of generating a wide range of timbres according to the type of shaping employed. This technique cannot, however, be considered equivalent to the processes of Fourier synthesis, where the harmonic spectra of individual tones may be accurately generated by adding together individual sinusoidal components, specified both in amplitude and phase.

Olson and Belar also make no reference to one of the primary difficulties encountered in any electronic imitation of musical sounds: the synthesis of transients. The nature of these phenomena is extremely complex and not fully understood even today. Nevertheless, several basic characteristics were known to acousticians of the time, and the contemporary observations of both Schaeffer and Meyer-Eppler have already been discussed.[34] In particular, before making the claims outlined above, account should at least have been taken of the fact that a primary feature of the attack of many musical sounds is the independent growth of each harmonic component. By only using a single envelope shaping system, acting upon a harmonic spectrum which has been determined in the manner described above, the R.C.A. synthesiser, in common with many other electronic music systems, is not capable of reproducing this essential acoustical feature.

The designer's intention to produce a machine primarily suited to the imitation of instrumental sounds has already been noted, and the musical restrictions which result from such an outlook on electronic music system

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[34] See chapter 2, pages 72-73 and 145-149

design will become increasingly apparent in the later study of the characteristics of voltage-controlled and computer-aided systems. In explaining the procedures involved in assembling a complete recording using their disc system, Olson and Belar make these objectives quite clear:[35]

> The synthesiser is purposely limited to the production of two simultaneous tones. The reason will be evident in the description which follows. In general, due to the characteristics of most musical sounds, the system is actually limited to a series of single tones. That is, the system can simulate any single wind instrument, such as a clarinet, saxophone, oboe, trumpet, etc., or one string of a stringed instrument, such as a guitar, violin, etc., or one finger playing of a keyboard instrument such as a piano, accordion, organ, etc. Thus it will be seen that in order to simulate an orchestra, each individual instrument must be coded and recorded separately and then the group of instruments combined.

A further indication of the designers' views on the musical application of electronic synthesis may be deduced from the following comment:[36]

> Conventional instruments produce various noises such as the rustling of wind in wind instruments, bow scratch in the viol family, various clatters and rattles in plucked and struck string instruments ... These undesirable noises do not exist in the electronic music synthesiser.

The underlying suggestion that all the acousto-mechanism by-products which result from the playing of musical instruments are 'undesirable', points to another misconception on the part of Olson and Belar regarding

[35] Olson and Belar, 'Electronic Music Synthesiser', op.cit., p.606

[36] Ibid., p.595

the aesthetics of musical creativity, for it postulates a world of clinical perfection rather than natural ordering.

The R.C.A. synthesiser was thus designed around the concept of individual, tempered tones, providing the primary source of material for composition with electronic sound. This feature accounts for the interest first shown in the machine by composers with such different musical outlooks as Babbitt, Luening and Ussachevsky. The latter two protagonists saw electronic synthesis, at least in the first instance, as a powerful extension of 'classical' instrumental techniques, providing new textures and feats of simulated dexterity which might be specified in terms of conventional scoring and tonal organisation. The designers themselves were quite content to demonstrate its capabilities by merely realising existing works, including the popular tunes Obelin, Sweet and Low, and The Old Folks at Home, relics, no doubt, of their Stephen Foster composing machine. Olson and Belar also ventured into the classical repertoire, reproducing works such as Brahms's Hungarian Dance No.1, 'In the gypsy style' and Bach's Fugue No.2 from Book I of the Well Tempered Clavier 'In the style of ancient struck and plucked strings in several variations'.[37] These exercises might well be seen as the antecedents of the era of Switched On Bach, associated with the popular exploitation of the Moog synthesiser in the late 1960s and early 1970s.

Babbitt, on the other hand, was concerned with developing the principles of total serialism, where 12-tone set procedures are used to determine every aspect of each individual event in a composition. The precise control

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37 Olson and Belar, 'Electronic Music Synthesiser', op.cit., p.612

provided by the R.C.A. synthesiser over the specification of such elements as pitch, duration, dynamics and timbre again made the use of the machine a highly attractive proposition, despite its functional limitations.

The achievements of the synthesiser and its successor, the Mark 2, in providing useful means of generating and manipulating sound material for the large number of composers who have come, and continue to come to the Columbia-Princeton studio, must be clearly acknowledged. Allowances must also be made for the limitations imposed by the control system itself, for from a pedagogic point of view the very simplicity of the input system ensured that composers and potential designers gained an appreciation of the problems associated with the programming of musical procedures as a series of sequential instructions at a very basic level.

The limitations of the two processing channels, each capable of producing only single chains of individual sounds, are nevertheless significant when considering the synthesiser's potential as a self-contained sound generation system. Any desire to employ tone mixtures as source material, for example, might only be fulfilled by employing the recording system as an intermediate means for building up the constituent elements, and then playing the results back into the synthesiser as a special input. This not only adds an extra stage to the operating procedures but also limits the composer's scope for direct experimentation with sound complexes.

A description of the R.C.A. synthesiser Mark 2 is given by Olson and Belar in the March 1960 number of the Journal of the Acoustical Society of America.[38] Their account is more concise than that produced for the Mark 1

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[38] Harry F. Olson, Herbert Belar, J. Timmens, 'Electronic Music Synthesis', Journal of the Acoustical Society of America, XXXII, 3 (Lancaster, Pennsylvania, U.S.A., March 1960), pp.311-319

synthesiser five years previously, and there is a notable absence of the type of controversial claims discussed above.[39] Despite the similarity of design with the Mark 1 version, some features would appear to have been implemented in the light of experience gained from the use of the former for serious composition, rather than popular interest. One improvement is the provision of four processing channels, doubling the output capabilities of the machine. The sound source bank is also considerably enlarged. In addition to a set of twelve tuning fork oscillators providing an octave of tempered pitches and a noise generator, two sets of twelve variable frequency oscillators are provided, each of which may be tuned to any desired frequency setting between 8130.0 to 16180.0 Hertz. Used together, the complete set of oscillators may be employed to provide an octave scale with thirty-six divisions. The frequency range of the fixed oscillators is changed from F♯5/F6 (739.989/1396.913 Hertz) to C9/B9 (8372.016/15804.272 Hertz) providing the top octave of the tempered range.[40] An octave divider, similar to that employed in the Mark 1 model, is employed to provide the ten-octave range of pitches between C0 = 16.352 Hertz and B9 = 15804.272 Hertz. The twenty-four variable frequency oscillators may also be fed into the octave divider to produce octave transpositions of their settings.

Two relay trees are employed to control generator and octave selection

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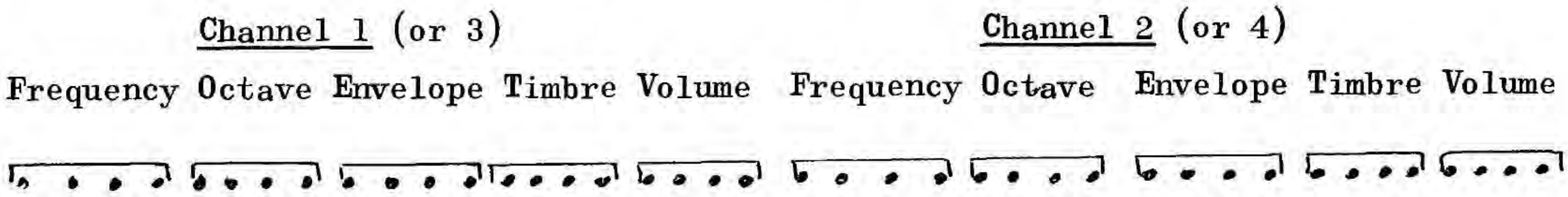
[39] With one exception: 'Practically any overtone structure whatsoever may be obtained by means of frequency discriminating systems employed in the timbre control system.' Olson, Belar and Timmens, 'Electronic Music Synthesis', <u>op.cit.</u>, p.315

[40] The tuning fork oscillators actually resonate in the range C7/B7 (2693.004/3951.068 Hertz), but they are each subjected to a fixed, two-octave multiplication process before being passed to the programmable octave divider.

for each channel, as before. The availability of a ten-octave dividing network in place of the former eight, however, necessitated provision of a four-bit coding system for this function. Despite the provision of twenty-four extra source oscillators no change is made to the size of the relay system for selecting generators, which still operates on a four-bit code commanding a maximum of sixteen different sources. Each synthesis channel may thus call upon only a selection of the oscillators, allocated manually via a special switchboard. Furthermore, the digital control system still only permits the processing of one source per channel, at any particular instant.

The frequency glider, attack, duration and decay, and volume control systems for each channel are retained in the Mark 2 design with no major change in their operational characteristics. A low frequency modulator is also included, offering, however, a slightly wider range of amplitude modulation speeds, variable from 5 to 10 Hertz. The timbre control system, as in the Mark 1 synthesiser, consists of high and low pass filters, which may be combined to give band pass functions, and also resonator chains. Improvements, however, are to be found in the range and operation of the various options. In particular two sets of resonators are provided for each synthesis channel, and the operational flexibility of these devices is enhanced by providing a comprehensive patch panel for signal inter-connections and control functions.

One general feature concerning the whole system merits particular attention. The layout and basic operating procedures for the synthesiser take full account of the need to provide a studio system which is essentially modular in construction, for no pre-conditions are imposed regarding the way or the order in which the individual devices are to be used. In any one of the synthesis channels the routing of signals from source bank to output is thus entirely a matter of choice for the composer concerned, buffer amplifiers being built into the input and output of each constituent unit to avoid problems of mis-matching. Considerable freedom is also afforded in the use of the paper tape control system, for the designation of the punched instructions may be altered at will. Two mechanically linked paper tape drives are employed, each providing control information for a pair of the synthesis channels. The standard designation of punched colums to relays, with the addition of an extra column for the octave divider, is identical with that used in the Mark 1 model.



These designations within each channel may be altered by interchanging the plug connections to the sensors. For example, the holes punched for frequency may be used to control volume, and <u>vice versa</u>. Furthermore, the destinations of the individual lines activated by the relay trees may also be interchanged or even paralleled, in the latter case weighting particular device functions and by-passing others. An alternative input system using continuous lines in place of punched holes, drawn with a felt-tip pen and read by an optical scanner, was introduced during the 1960s.

The installation of the R.C.A. synthesiser at the Columbia-Princeton Center served as a major catylist to those concerned with the advancement of electronic music systems. In particular, the inclusion of a programming system, albeit elementary, aroused general interest in the possibilities of techniques for controlling dynamically the functions of analogue devices. It was in such a climate of investigation that the commercial voltage-controlled synthesiser was born.

The voltage control revolution in the design of electronic music systems during the 1960s was brought about primarily through the pioneering efforts of two Americans: Robert Moog and Donald Buchla.[41] Both these individuals, working quite independently, established commercial manufacturing companies which adopted their names as trademarks, and began to exploit a large and receptive market for their products. The expansionist policies of universities and colleges during this period, coupled with a ready interest in this new form of sound production, led to a rapid escalation of activity as the phenomenon gathered momentum. Firms such as Tonus, manufacturing under the name of A.R.P., and Electronic Music Studios, London, soon joined the pioneers as leading manufacturers, and by the end of the decade numerous smaller companies were contributing to a sales campaign which had, by then, become of world-wide importance.

The marketing of these synthesisers has been accompanied by the publication of considerable quantities of supporting literature, extolling the virtues of voltage-controlled sound generation as a means of composing

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[41] The work of one other pioneer, Paul Ketoff, should not be overlooked at this point. Working in Rome, he produced a small but very individual voltage-controlled synthesiser in 1964, known as the Synket. This was subsequently manufactured on a very modest scale as a performance unit for use in concert in association with other instrumental combinations.

electronic music. In much of this information particularly where manufacturers or their associates have been involved in its preparation, there is a marked tendency to emphasise the functional characteristics of partcular voltage-controlled systems without paying sufficient attention to their musical value. In more substantial publications[42] it is also unusual to discover more than a passing reference to the problems encountered in the creative use of synthesisers, and even in such a technically comprehensive book as Electronic Music by Allen Strange[43] the vital subject of composer/machine communications is not discussed in any depth.

This inclination to allow technical considerations to influence musical application rather than the reverse is clearly disturbing. To return to the closing observations of the previous chapter, it may be noted that the mass-production of synthesisers has resulted in the propagation of a limited number of system designs, produced to provide their manufacturers with a compromise between flexibility and accuracy of specification, and profitability. In such a situation the advancement of the medium has been largely determined by the policies of these firms, who are naturally unwilling to consider making any alterations to their products which do not appear to be of commercial benefit. This has led to a distinct threat of stagnation and a lack of critical awareness in the development of electronic music systems, encouraged by the continuing sale of identical synthesisers to an ever increasing number of studios, only too ready to pay for the convenience of complete packages without proper regard for their musical value.

[42] See Bibliography

[43] Allen Strange, Electronic Music (Wm. C. Brown Co., U.S.A., 1972)

Some larger studios, notably Utrecht Institute of Sonology, Holland, have resisted these pressures and developed their own voltage-controlled systems built entirely to their own specifications. These are, unfortunately, exceptions to a situation where most studios merely provide functional units, playing no part themselves in what must be treated as vitally important programmes of research into better techniques of sound generation and manipulation to serve an ever expanding field of compositional activity. A study of individual synthesiser characteristics is to a large extent inappropriate in the present context, for it would merely serve to illustrate individual design interpretations of a common technology. It is the technology itself and its musical advantages and limitations which provides the basis for a critical study of voltage-control systems and their significance as a link between 'classical' and computer-controlled electronic music studios.

The 'classical' analogue studio consists of a series of sound generation and treatment devices which may be connected together to provide a complete synthesis chain from source to output. All devices are usually set individually by hand, and the content of the sound complexes produced may, in general, only be changed by making discrete alterations to selected controls. Such systems require composers to rationalise their ideas into closely defined technical procedures, giving detailed specifications for the settings of each source and treatment device, and their manual manipulation.[44]

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[44] It may be seen that the provision of a punched tape control system for the R.C.A. synthesiser did not remove this requirement, but provided instead a convenient means for programming precise device control instructions as an ordered sequence of events.

In electronic terms the principles of voltage control are essentially a logical development of traditional analogue techniques. An impetus for this development was created by the rapid expansion in the use of transistor circuits during the late 1950s and early 1960s which facilitated the use of miniaturised devices as the basis for assembling complete system packages.

In order to explain the basic characteristics of voltage-controlled systems it is necessary to examine the functional characteristics of the constituent devices in closer detail. Most sound generation or processing devices offer one or more variable functions. An oscillator, for example, will normally permit its frequency, amplitude and, in certain designs, waveshape to be manipulated. In a conventional analogue device these parameters will each be controlled by a specific knob or slider which, in most cases, will vary a resistance or capacitance value in a particular part of the circuit. In a voltage-controlled system the same manual facilities for device control are usually available to the user, but the internal electronic functions are significantly different. The control circuits for selected device characteristics operate by responding to changes in an applied voltage level, which may be varied within limits standardised for the whole system, typically plus and minus 3 or 5 volts. Each of the associated manual controls accordingly functions as a direct current voltage regulator. The circuits are usually designed so that there is a clear relationship between the perceived response characteristics of the various device parameters and linear changes in voltage levels. This is of particular relevance to the frequency input for a voltage-controlled oscillator, where it is usual to relate equal changes in voltage to equal changes in pitch. Moog oscillators, for example, function on the principle that a change of one volt results in a one octave shift in frequency. Similarly such a logarithmic response is desirable for amplitude control, although this is not always strictly adhered to.

The level of voltage applied need not be regulated solely via a uniquely assigned knob or dial. If breakpoints are inserted in the lines from these primary regulators to the device control inputs, the voltage levels supplied may be added to or subtracted from by an external voltage source. The provision of such a facility permits a single source to act as a control for several different devices. Three oscillators, for example, might be interlocked by connecting the breakpoints, or what might better be described as external control inputs, in parallel with a common regulator. Providing the response characteristics are as described above, all three oscillator frequencies may be transposed together up or down from their original settings while preserving their original pitch interval structure.

Discrete voltage steps may be supplied by connecting a voltage-producing keyboard to the control input of a device. Facilities vary from manufacturer to manufacturer, but the majority of units are monophonic,[45] producing one voltage level at a time.[46] In their simplest mode of operation

[45] A.R.P. (Tonus) manufactures a highly sophisticated polyphonic keyboard capable of generating simultaneously a whole series of different voltage outputs for connection to different devices.

[46] If two or more keys are depressed at the same time, usually the highest key voltage is selected for output.

they are usually capable of generating a tempered range of pitches either from their own internal oscillator circuit or from an external oscillator via a control line connection.

Many designs permit the keyboard voltage sensitivity to be varied, increasing or decreasing the overall size of each step proportionally. One or two models allow, additionally, the voltage level assigned to each key to be individually pre-set. The former facility is especially useful for generating musical scales based on equal pitch intervals other than the tempered semitone. If such a keyboard is connected instead to the amplitude control input for a device, a discrete series of dynamic levels may be produced. Provision of a pre-set facility for regulating the voltage output of each key independently frees the composer from the limitations of equal voltage steps, and permits the use of the keyboard as a more versatile, manually operated means of producing sequences of events. Buchla, however, is the only leading manufacturer to have marketed this facility as a standard feature, and its models usually employ touch-sensitive plates as the means of control in place of the more traditional black and white press-keys. The reasons for this divergence of marketing policies may, at least in part, be attributed to the different marketing potentials available to these firms. Elliott Schwartz, writing in his book Electronic Music, notes in this connection that

> The Moog and A.R.P. originated in the pitch concerned, 'control'-orientated, 'academically' inclined north-eastern part of the United States, while the Buchla is a product of the sound-sonority-textural inclinations of the American West.[47]

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[47] Elliott Schwartz, Electronic Music - A Listener's Guide (Secker and Warburg, London, 1973), p.76

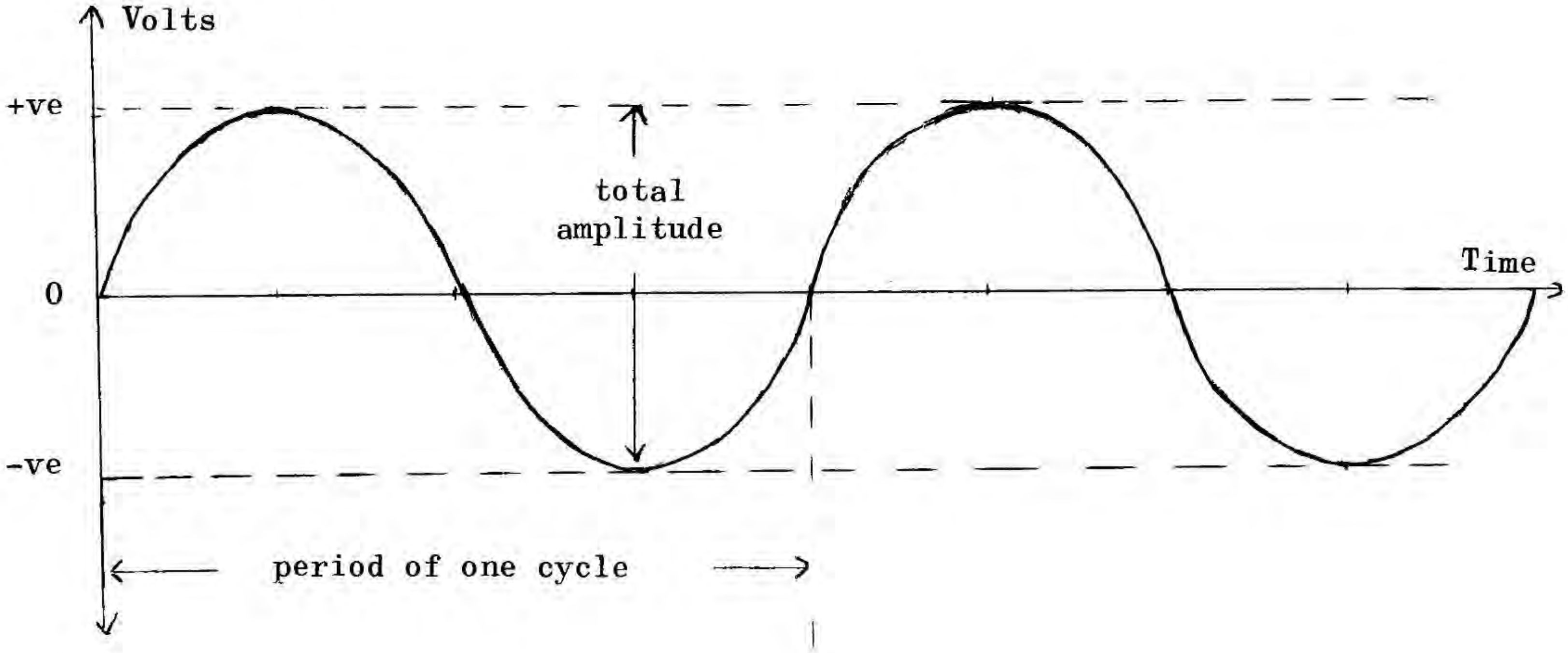
The use of conventional keyboards as input systems for synthesisers is sometimes criticised on the grounds that this traditional method of instrument control is too readily adopted in situations where other input methods might be more appropriate. Several important issues are raised here, not least the wide range of viewpoints which exist regarding the musical uses of electronic synthesis. If one is concerned with the imitation or augmentation of traditional instrumental sounds, a keyboard will prove in many circumstances an effective means of communication. The argument here lies not with the means of device control, but with the musical relevance of the whole operation.

There are plenty of individuals only too willing to indulge in the exploitation of public taste for commercial gain rather than artistic merit. To the general public, for example, the Moog synthesiser is known primarily as an electronic imitator, used to provide such successful realisations as Switched On Bach (Walter Carlos). It would be misplaced reasoning, however, to suppose that composers in general are unduly influenced by this particular aspect of electronic sound production. Any truly creative artist working in this sphere will be concerned with the realisation of his own sonological ideas, utilising whatever prove to be the most useful channels of composer/machine communications. It must therefore be acknowledged that keyboard control systems, used intelligently, may well prove to be of value in a variety of compositional situations.

During the course of this chapter an important line of development will be traced from keyboard methods of device control to the use of digital sequencers, which in turn has led to the introduction of computer-controlled synthesis techniques. The importance of the first-mentioned device in initiating this progression of events must thus not be overlooked. The generation of discrete pitches from oscillators is but one aspect of this method of voltage control, and reference has already been made to the possibility of controlling amplitude levels in discrete steps. As this account of voltage-controlled systems progresses it will become clear that an ability to generate exact and repeatable voltage levels when manipulating a wide range of studio functions is an operational feature of major usefulness.

The development of increasingly complex methods of sound production is nevertheless only of limited value if no positive attempts are made to improve further the methods of communication between composer and machine. In the case of voltage-controlled systems this consideration has become submerged by the ease with which complicated sequences of sound events may be produced from a small number of shorting pin or patch-chord connections, and device settings. One result has been a proliferation of mediocre compositions from less gifted composers who have readily accepted what have now become clearly recognisable system clichees as the basis for their realisations. Apart from a few devices such as joysticks, which may be employed to provide two-dimensional control facilities, and touch-sensitive plates, little manual input sophistication has been introduced as a by-product of commercial voltage-control technology. Knobs, dials, faders and switches still provide the main interfaces for individual devices, and these dictate the manner in which specifications must be made.

The main interest in voltage-control techniques lies, nevertheless, not just in the ability to execute more flexible manual operations, but in the application of the output of certain electronic devices as dynamic input controls for others. One of the commonest examples of the use of this technique is the application of the output of one oscillator as a frequency or amplitude control function for another. The majority of audio electronic equipment is designed around the principle of the constant current amplifier, where signal informaion is produced in the form of a varying voltage pattern. The output of a sine wave oscillator thus consists of a repeating voltage function which fluctuates about a selected median voltage level.



The amplitude of this wave may thus be measured in terms of the maximum displacement of the voltage function from zero potential:[48] the greater the amplitude the greater the displacement. The period of the

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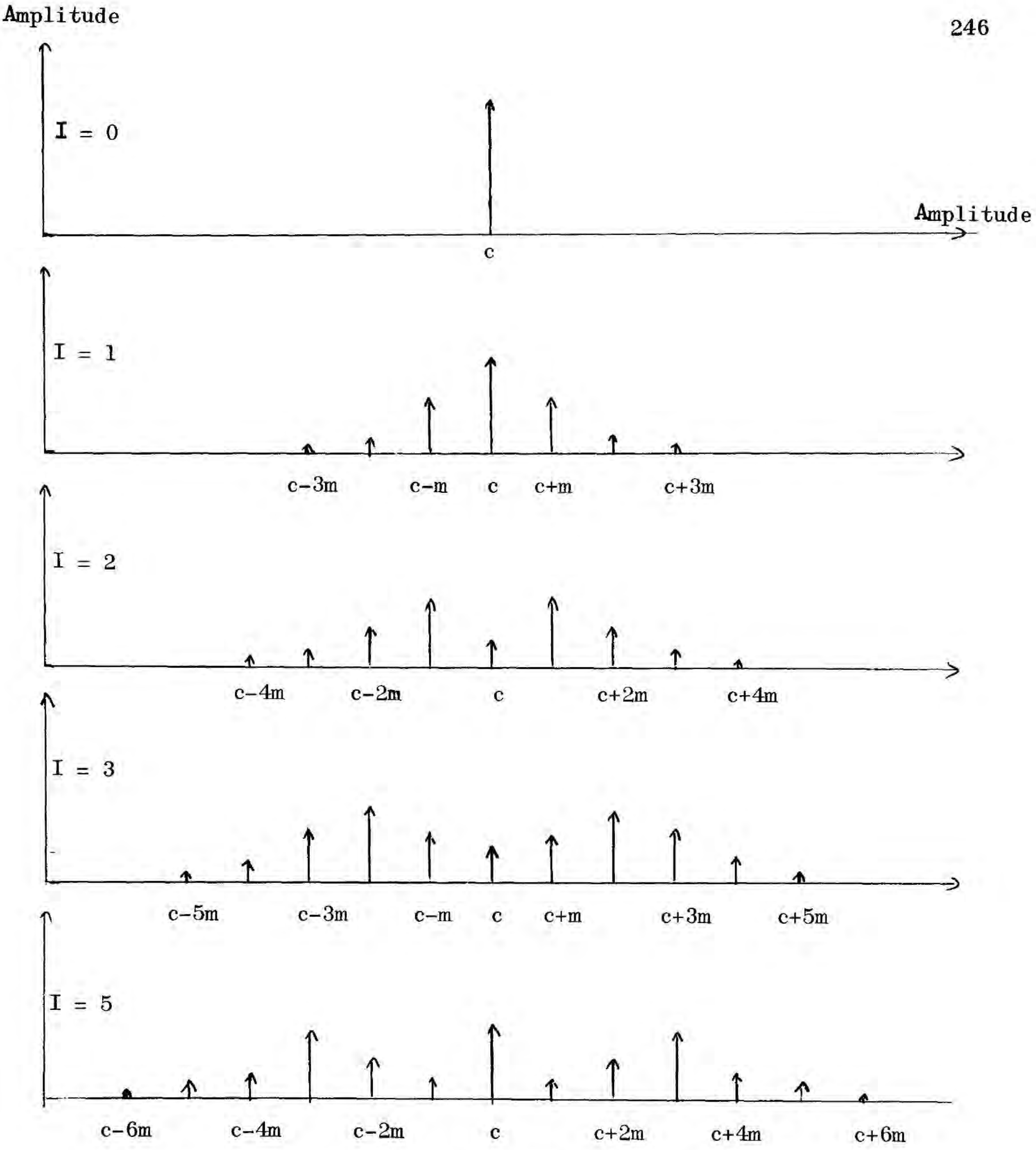
[48] Alternating current theory states that the effective overall amplitude of a sinusoidal voltage function is given by the root-mean-square of the wave over one cycle, approximately 0.707 times the maximum displacement.

wave is measured as that time taken for the function to complete one cycle, and the frequency of the wave is given by the reciprocal of the period, providing the latter is expressed in seconds or divisions thereof. A wave which repeats every 100 milliseconds (0.1 of a second), for example, will produce a frequency of $1/0.1 = 10$ Hertz. If the amplitude of this wave is kept within the control voltage range of the studio system, the oscillator function may be used to manipulate the characteristics of another device. If the feature selected is the frequency control input for another oscillator, for example, the output of the latter will be subjected to frequency modulation. The size of the pitch deviation, or depth of modulation, will be determined by the amplitude of the control wave, the speed of modulation will be determined by the frequency of the control wave, and the modulation function will be determined by the wave shape itself.

If this frequency of modulation is fairly small, perhaps 8 Hertz or less, with a relatively shallow modulation depth employing a smooth function such as a sine or triangle wave, the output of the oscillator which is being controlled will display pitch vibrato. If the control oscillator frequency is reduced further the speed of the vibrato will slow correspondingly. It then becomes possible for the ear to follow clearly frequency fluctuations associated with much greater depths of modulation. In order to operate at these low speeds control oscillators must be designed to work at sub-audio frequencies, producing waves which may usefully take as long as 20 to 30 seconds to execute each cycle at their slowest setting. Under these conditions it becomes possible to perceive the functional characteristics of the control wave very clearly, the shape of the latter becoming of critical importance in determining the progression of events. A compound wave containing overtones, for example, will produce a significantly different oscillatory pattern to that produced by a pure sine wave. In these operating conditions electronic wave shapes such as ramp/sawtooth, or square have particular applications, the former providing a method for repeatedly sweeping device functions upwards or downwards, and the latter providing a method for switching continuously between two steady states.

Above about 10 or 12 Hertz the output of a frequency modulated oscillator becomes increasingly blurred, the ear being unable to follow the fluctuating pattern, and a complicated series of side bands is produced. In outline the basic technical principles of fequency modulation are as follows. If a steady sine wave of frequency c Hertz is modulated by another of frequency m Hertz, two series of side bands are generated either side of the steady frequency in the progressions c-m, c-2m, c-3m, ... and c+m, c+2m, c+3m, ...

Only a limited number of bands are produced, regulated by a factor known as the modulation index; I. This may be calculated by dividing the peak frequency deviation; d, produced by the modulating wave, by the modulating frequency itself. As an approximate guide, side bands are produced in a range slightly exceeding the frequency of deviation either side of the average frequency. The distribution of amplitudes, however, is rather complex, determined by what are known as Bessel functions. Overall, as I increases from zero, energy is 'stolen' from the average frequency and distributed amongst an increasing number of side bands. The following diagrams give some indication of the effects of different modulation indices:



At a much later stage in this study; at the end of chapter four, the characteristics of frequency modulation will be examined more closely in relation to some interesting developments which have emerged from the work of John Chowning, at Stanford University, America, during the 1970s. It is important here, nevertheless, to draw attention to the problems of musical specification in this context, for there is no easily definable relationship between the perceived nature of frequency modulated sounds and their scientific basis. To describe frequency modulation in terms of side bands is more of a mathematical convenience than of acoustical significance.

Ancillary equipment items such as a cathode ray oscilloscope and a spectrum analyser will aid a composer's study of the phenomena, but no straightforward guidelines may be laid down which will allow specific compositional ideas to be equated with equivalent device settings. The system, in practice, dictates a range of operating characteristics to the user, who has to evaluate the relevance of these sonological products to his conceptual ideas.

An account of the principles of ring modulation, which technologists often treat as a special form of amplitude modulation, has already been given in the previous chapter,[49] and certain aspects of this process will be raised again shortly. The use of voltage control techniques to produce amplitude modulation, however, involves a rather different set of operating procedures, more closely related to those already described in connection with the production of frequency modulation. As in the latter case these modulation characteristics are determined by the amplitude, frequency and wave shape of one oscillator controlling the performance of another. The difference lies in the choice of control input, the effect being a fluctuation in volume instead of frequency. This allows signals of any origin to be modulated, including natural sounds captured via a microphone.

At low modulation speeds, from about 10 to 12 Hertz downwards, the effect of the control wave is perceived as a distinct pulsing in the volume level. If the amplitude of this wave is increased sufficiently, the

[49] See chapter 2, pages 102-105

output of the signal oscillator may be modulated between zero intensity and a peak level, similar in effect to the use of pulses as one of the inputs for a ring modulator, described previously in connection with the techniques employed by Stockhausen in his realisation of Kontakte.[50] Under these conditions, if the control function is a sine or a triangle wave the amplitude will vary continuously and regularly between the two states of 'fully on' and 'off'. If, however, the control function is a square wave, abrupt switching, or what is commonly referred to as 'gating', may be produced.

Increasing the control oscillator frequency above about 12 Hertz leads to a blurring of the changes in amplitude and the generation of a single pair of side bands in addition to the primary signal. The positionings of the side bands are determined by frequency of the control oscillator: the upper side band consisting of the sum of the signal and control oscillator frequencies, and the lower band of the difference. These bands are not as strong as the primary signal and are thus perceived as non-harmonic overtones and subtones. A 200 Hertz control wave modulating a 1000 Hertz signal, for example, will produce side bands at 1200 Hertz and 800 Hertz. If either the control or signal oscillator generates waves which contain overtones, one set of side bands will be developed for each component, leading to the production of more complex timbres.

Parallels may now be drawn between the acoustical content of ring modulated and amplitude modulated information. Both involve the interaction of two different frequency sources. In the former process only

[50] See chapter 2, pages 168-169

sum and difference products are generated, whilst in the latter the frequency of the signal being modulated is also present in the output. It may now be seen more clearly why Enkel in his description of the Cologne quadripole modulator[51] referred to the mixing of one of the source signals applied to a ring modulator with its products as a method of producing amplitude modulation, for providing the source chosen is that which lies between the side bands, and its amplitude level is also suitably adjusted, the appropriate timbre spectrum may indeed be generated. The argument levelled against this description was musical rather than technical, for it must be borne in mind that electronic music systems are intended for the creative production and organisation of sound. Due consideration must therefore be given to the needs of composers who wish to develop a musically based understanding of the practical way in which devices perform.

The fact that these two different electronic techniques may, in certain circumstances, produce identical characteristics is thus of only limited interest. It is far more important for a composer to know that ring and amplitude modulation are in practical terms primarily associated with two quite different studio operations each exhibiting a particular range of functional characteristics. A ring modulator will accept as inputs any two analogue signals with no restrictions as to their origin. An amplitude modulator of the type described above, on the other hand, is based around the functional characteristics of a specific studio device, undergoing direct modulation by a suitable oscillator.

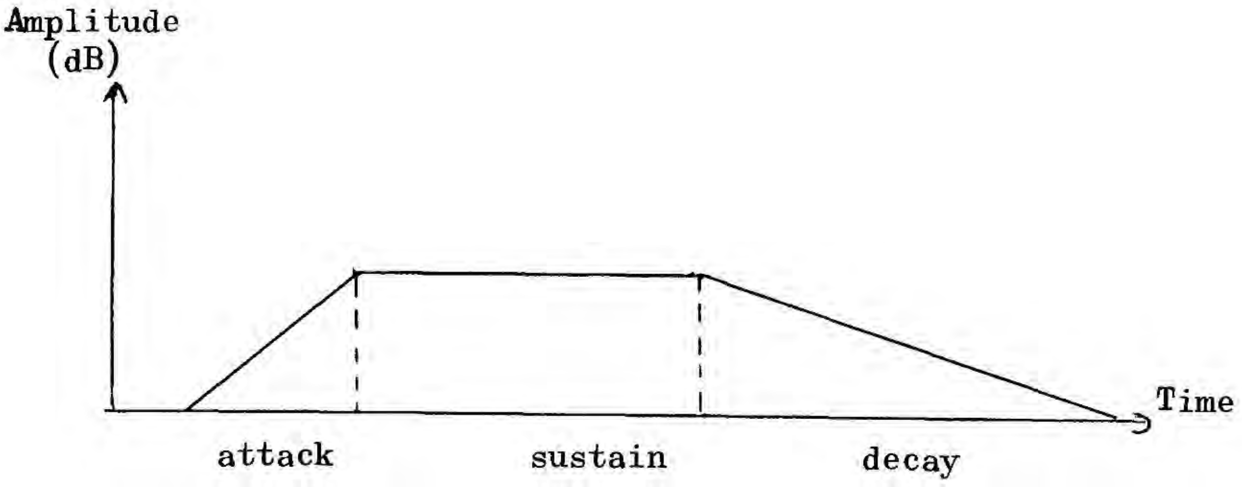
Reference should be made here to a special form of modulator developed during the 1950s for use in 'classical' studios, and occasionally

[51] See chapter 2, pages 105-108

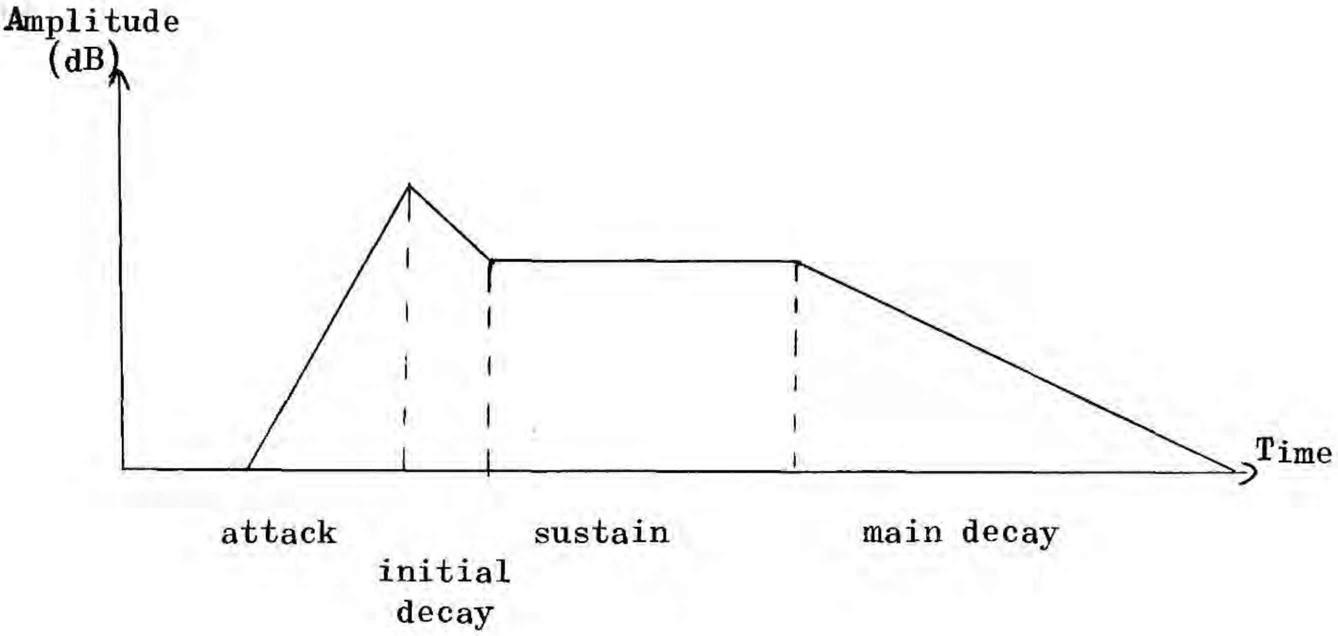
included in present-day voltage-controlled systems as an extra feature; the single side band generator or frequency shifter. This device, as the first description suggests, isolates either the summation tones or the difference tones in a ring-modulated signal. Some models offer just one of the side bands; others supply both. The isolation of summation tones is of particular musical value, for the timbres produced may be made rich in spectral content without being unduly complex. If one of the inputs consists of a steady sine wave the device may be treated as a means of transposing frequency components in a second signal by absolute amounts. Sum modulation of a signal of 500 Hertz, with overtones at 1000 Hertz, 1500 Hertz and 2000 Hertz, against a sine wave of 20 Hertz will result, for example, in a transposition of the fundamental to 520 Hertz and the harmonics to 1020, 1520 and 2020 Hertz respectively. The acoustical effect will be not only a change in pitch but also a marked change in timbre, for the partials will no longer belong to the harmonic series of the fundamental. Increasing the sine wave frequency further will lead to a distinct frequency compression of the partials, for the perceived change in pitch of the spectral elements will reduce steadily from fundamental to the highest harmonic.

It might be noted at this point that the ring modulator and the single side band generator are electronic devices employing internal functions which are essentially fixed in their specification. The manual controls normally made available are thus associated with additional attenuator circuits which may be attached, if so desired, at both the input and also the output. Aids such as these, which may be usefully employed at several strategic points in a studio system, are primarily concerned with regulating the overall levels of signals as a mixing function. Designers may choose in certain instances to make these amplitude functions voltage controllable, in which case these regulators may be considered as independent amplitude modulators for any signals which are passed through them.

The envelope shaper is an extremely important processing device in any synthesis sytem, for it provides a flexible set of amplitude control characteristics which may usefully be varied over wide operating limits. In voltage-controlled synthesisers the simplest designs normally supply manual control over the durations of the growth or attack function, the 'on' or sustain time at which the signal is maintained at a steady level, and the decay function. It is not usual, unfortunately, for any provision to be made for varying the nature of the attack and decay functions themselves, which are usually fixed to approximate a logarithmic response.



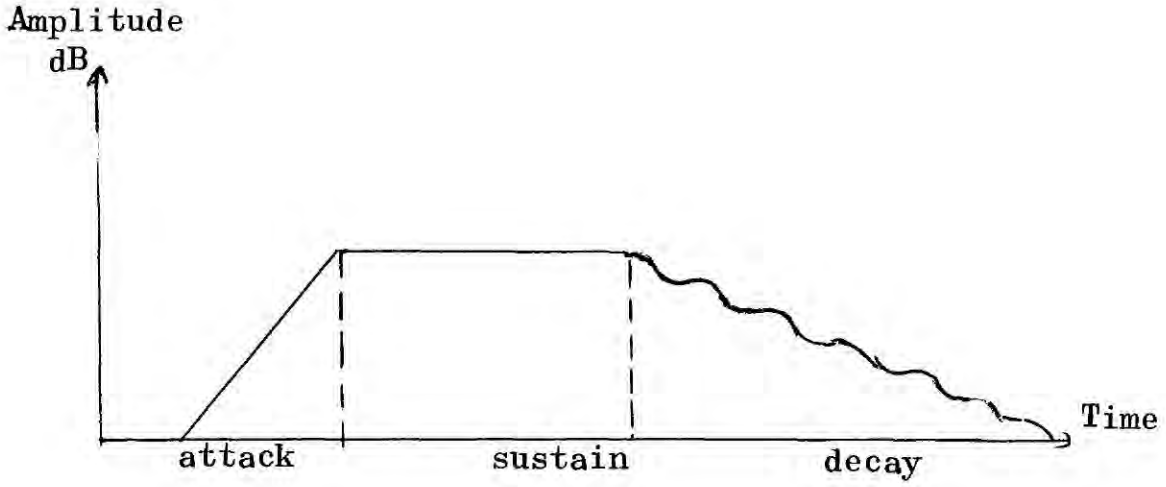
More sophisticated designs insert an initial decay function after the main attack to highlight the acoustical impact of the latter:



The only additional controls required to operate such envelope shapers are a sustain amplitude level regulator for the former design, attack peak and sustain level regulators for the latter, and some means of triggering the complete sequence of events. The trigger function may be supplied in several different forms, of which the simplest and most basic is a manually operated press button. More conveniently, if the synthesiser includes a keyboard system, the envelope shaper may be triggered externally via a special control line whenever any key is depressed. If the output of a device or a chain of devices being controlled by the main keyboard voltage system is then passed through the envelope shaper, an articulated sequence of individual sound events may be produced. A recycling facility is often included, allowing the envelope shaper to trigger itself automatically. In this situation the 'off' time which is allowed to elapse between the end of one envelope and the beginning of the next provides a further variable characteristic.

Designers usually provide a comprehensive set of manual controls for all the above-mentioned variables, but they often limit the provisions for external voltage control in some instances to the decay function only. Envelope shapers are primarily intended for use as processing devices at speeds of operation where their amplitude functions may be clearly perceived. If the time intervals for the attack, sustain and decay functions are all made too short the result will be the production of a series of 'blips' of uncertain acoustical content.[52] If the device is then made to recycle rapidly, a rapid gating of the applied signal will occur, generating side bands such as may be produced by a conventional amplitude modulator.

There is little practical value in applying a rapid modulation to any of the durational characteristics themselves, for the ear will not be able to detect any significant change in the enveloped output. Lower modulation speeds, of less than about 40 Hertz, however, may sometimes prove of value. The decay function, for example, may be modulated to simulate a reverberant decay:



or, if the control wave is especially slow and the envelope shaper is set

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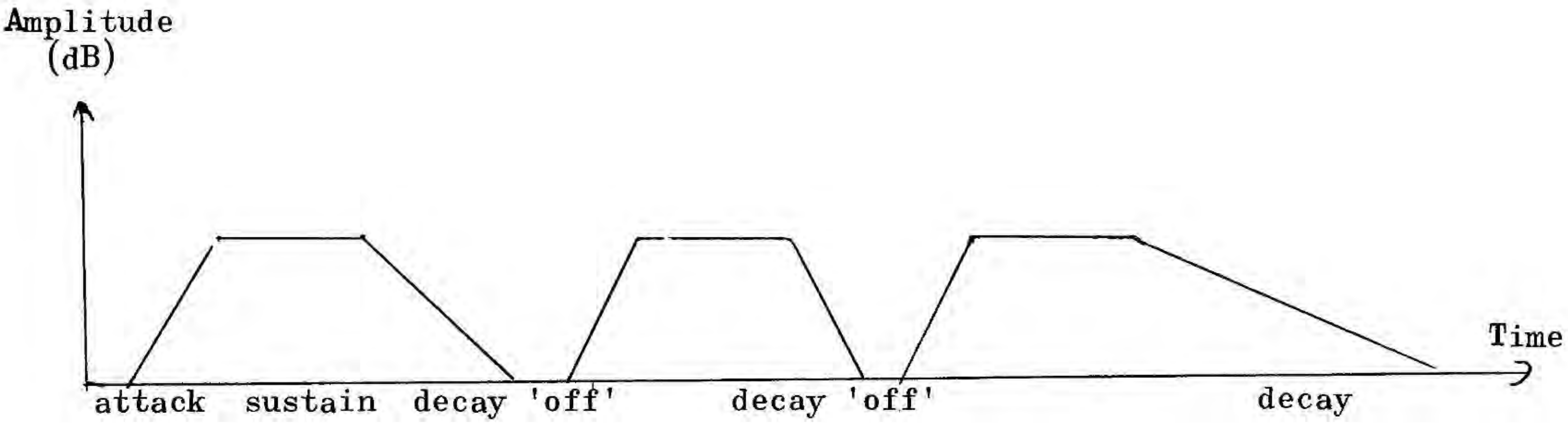
[52] See chapter 2, pages 48-49, 126 and 142-144

to recycle, decay times may be created which vary significantly from one triggering to another.

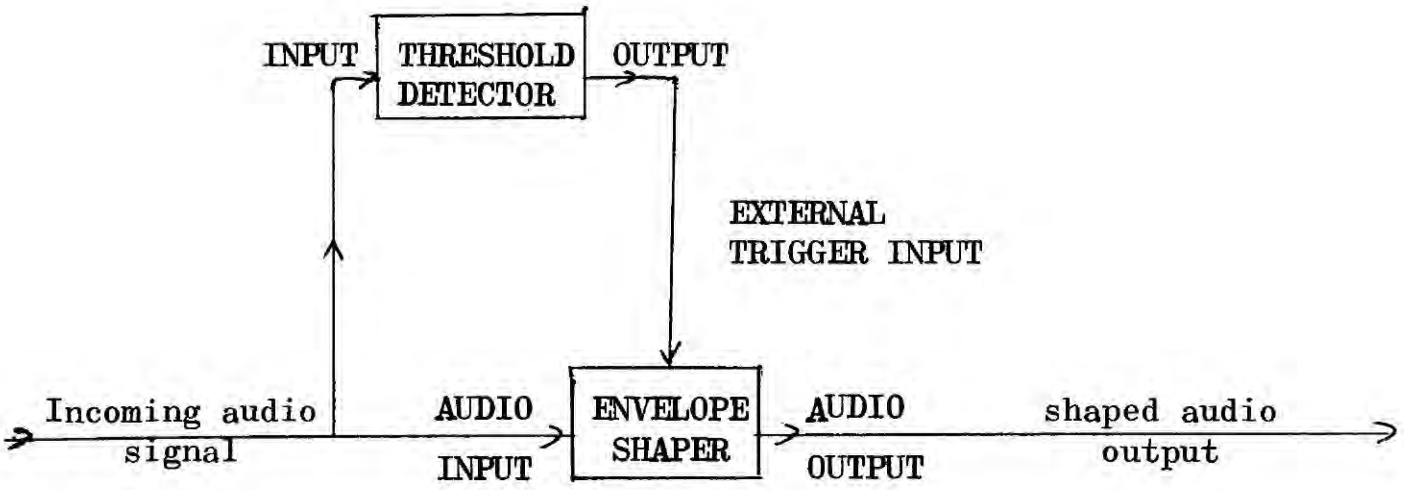


A more precise control over decay times may be exercised by applying stepped control voltages from a keyboard to the decay control input.

Modulation of more than one duration function is electronically feasible but such extra provisions are of restricted value, for the device then becomes an unwieldy multi-processor, generating characteristics which could better be achieved in many circumstances by other techniques. The latter include gating with a pulse generator, or using two different wave forms, perhaps at different frequencies, superimposed to provide a composite control function for a single voltage-controlled amplitude modulator. The use of electronic patterns as control functions for devices in a manner such as the above must also be treated with some caution, for it provides an opportunity to create repetitive and hence musically limited patterns of events. Discrete control of the durational characteristics, which will inevitably require a detailed specification of the individual sonological events on the part of the composer, proves in many cases to be a far more useful method of approach.

More sophisticated designs sometimes offer a level detection facility which will trigger the envelope function as soon as the applied audio signal reaches a predetermined level.

The electronic circuit employed for this purpose is known as a threshold detector or Schmitt trigger. This consists essentially of an amplitude sensitive switch which closes to create an 'on' state only when a signal applied to its input reaches or exceeds a pre-set level. This state is then maintained until the signal strength falls back again below the threshold point, whereupon the switch re-opens again. If such a detector is inserted in the control line for the attack function of an envelope shaper and a feed taken from the audio input of the latter to the input of the former, the shaper may then be made to trigger itself in response to amplitude peaks in the applied signal.



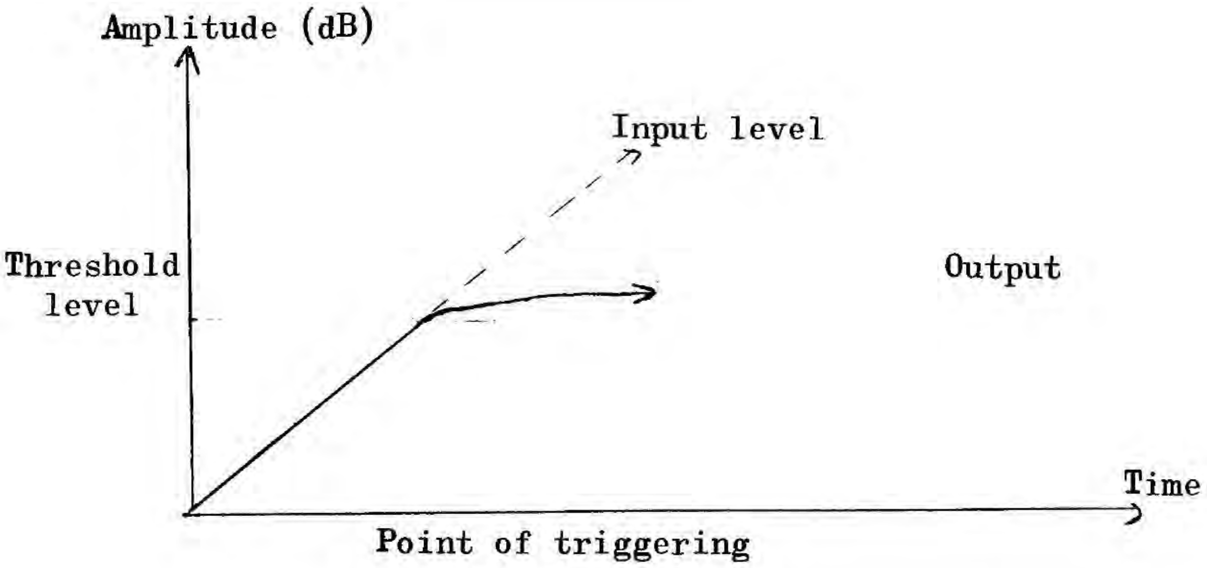
For such an application only the 'off' to 'on' switching function would be utilised to trigger the device, the re-opening of the switch being ignored.

The Schmitt trigger has provided the basis for a number of special amplitude processing devices, several of which more recently have become generally available in larger studios. In view of the common electronic principles involved it is possible to combine usefully the more important functions in a single unit to offer a selection of operating modes. The writer and Dr. John Emmett, working in Durham University Electronic Music Studio, have designed and assembled a bank of such units, each of which offers a choice of five functions.

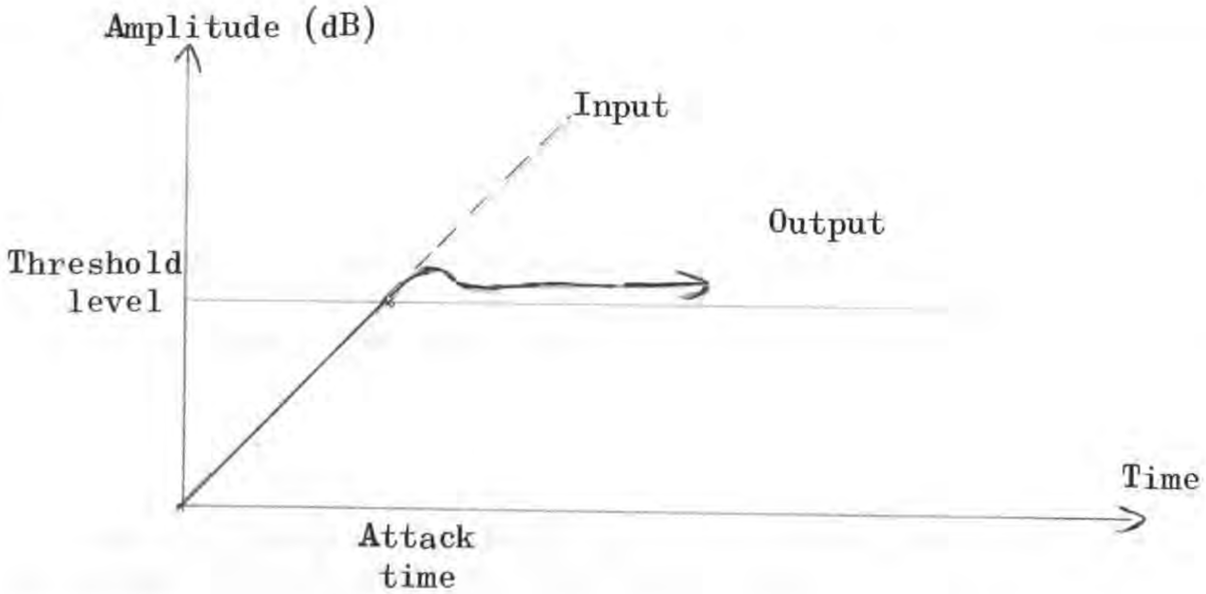
The simplest of these is a 'gate' function, which acts to block all incoming audio signals which are below a selected amplitude level, and to pass unaltered all those which are above. This threshold point may be continuously varied from 0 dB (studio line level) to -40 dB, this range of settings being common to all five modes. Gating has been referred to earlier in connection with the application of low frequency square waves to the control input of an amplitude modulator. This elementary technique, however, provides no correlation between the switching process and the amplitude contour of the incoming audio signal. The more sophisticated facility currently under discussion permits several useful processing operations to be carried out. For example, the inevitable 'leaking' of steady oscillator tones through a ring modulator may be suppressed whenever the complementary input signal is not present, or just the loudest events may be extracted from a passage of previously generated material.

If the gating circuit responds instantaneously to incoming signals which rise above, and then subsequently fall back again below the threshold level, audible clicks will be generated as the sound material cuts in and out. These clicks will become particularly noticeable if the amplitude of input signal fluctuates rapidly around the detector setting. This effect may be reduced or removed by introducing attack and decay circuits to increase and decrease the amplitude of the processed material over a specific time interval whenever the detector switches the gate on or off. An enveloping function is then created, controlled by the signal itself. Unlike the conventional envelope shaper, however, these time intervals are normally kept fairly small. The Durham studio unit, for example, allows the attack time to be varied between 0.5 milliseconds and 100 milliseconds, and the decay time between 10 milliseconds and 1 second. Again, these functions are common to all the other operating modes.

Two of the other functions are complementary: 'compress' and 'expand'. In compression mode the threshold detector is used to introduce circuits which restrict the amplitude peaks of incoming signals. All amplitude characteristics below the selected threshold level setting are passed through the processor unaltered. Any signals rising above this level are subjected to attenuation with rapidly increasing severity. This effect may best be explained in the form of a diagram, comparing a steadily increasing input signal against the processed output:



It may be seen that the output curve rapidly flattens to give a maximum level a few decibels above the threshold point. The sharpness of this characteristic varies from design to design. Some circuits provide a more gentle curve, flattening towards a point several decibels above the threshold level. Others clip all signals so firmly that the maximum level is only just above the threshold level itself, providing, to all intents and purposes, a straight line transfer characteristic. The Durham unit compromises with a curve which flattens at about 6 decibels above the threshold. Problems arise again, however, with abrupt changes in signal levels, and the inclusion of attack and decay characteristics are again desirable. In this mode the effect of the former is to allow rapidly increasing input amplitudes to rise above the threshold point until the effect of the attenuator circuits comes fully into force.



This has an effect of artificially highlighting transients, for the shorter the rise time of the input signal, or the longer the attack function time, the more pronounced the initial overshoot. The decay function provides a matching recovery time when the signal falls back again below the threshold point. A wide range of processing characteristics may be obtained by varying both these response delaying controls, according to the nature of the source information.

The expander provides a mirror response to the compressor, artificially boosting signals which rise above the threshold point to an increasing degree. Theoretically, providing the transfer characteristics are curved functions and exactly matched, a signal should be recoverable from the process of compression by subsequent expansion to its original form. In practice it is extremely hard to achieve these conditions using these relatively simple processors. Far more sophisticated approaches are adopted for such compression/expansion applications as tape noise reduction systems commonly in use in professional recording studios today.[53] Used independently as studio processing techniques, both these functions provide powerful means of altering the dynamic contours of fluctuating signals.

The fourth mode is a 'limit' function, employing the threshold detector to supply the information necessary for supplying a constant amplitude level at the output. The associated circuitry thus attempts to boost all

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[53] One such technique, the Dolby A noise reduction system, compresses incoming signals, divided into four frequency bands, in an inverse fashion to that described above: the quietest signals are artificially boosted about 10 to 15 decibels, this degree of alteration decreasing steadily at higher levels to no boost at all at normal studio line levels. The information in this electronically altered form is then recorded onto tape. On playback an accurate expansion process is applied, restoring the signals to their normal amplitude states. The 'hiss' noise inherent in the tape itself is also subjected to this expansion, and as an already quiet signal element is reduced to a still lower level.

incoming signals below the selected threshold point, and attenuate all those above. Practical considerations demand some restriction on the operating range of this function, otherwise the gain of the device would rise to 75 decibels or more on an open input, boosting the inevitable residual noise of the system to an intolerable level. The Durham unit, for example, restricts the maximum gain/attenuation of the limit amplifier to about 40 decibels either side of the threshold setting. The use of attack and decay functions again permits the response of the limiter circuits to be retarded.

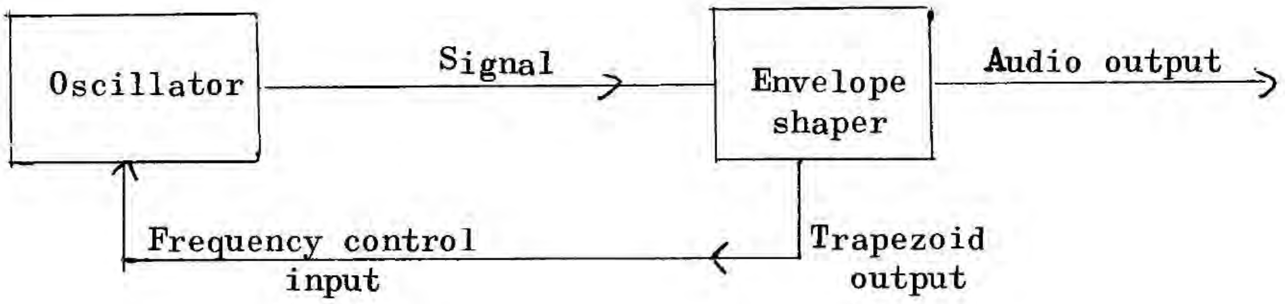
The most important application of the limiter in an electronic music studio is for the suppression of envelopes in signals. If a recording of a single piano note which initially exceeds the selected threshold setting is input to the limiter, for example, the circuits will try to remove the natural decay characteristic, first by attenuation and then by boosting. Depending on the length of the attack setting, an element of the initial transient may reach the output before the limiter attenuates the gain to that of the threshold level. As the natural decay falls below this point the limiter will then boost the signal correspondingly until the maximum allowed gain factor has been reached. It must be appreciated, however, that, as noted above, this increasing degree of amplification applies not only to the signal itself, but also to the residual system noise, which, once it has reached the threshold of audibility, will grow perceptibly louder. Limiters must thus be used with care if such undesirable effects are to be kept to a minimum. They are nevertheless useful as amplitude regulators, especially when applied to smooth, essentially continuous information, such as might be supplied, for instance, by the output of a band-pass filter of variable centre frequency, employed to isolate elements in a tone mixture.

The fifth function, 'invert', is more experimental, for it attempts to invert the dynamic levels of applied signals around the selected threshold setting. Again, similar restrictions over the attenuation/amplification ranges are applied.

An extra option is available in all five modes, involving a device known as a side-chain filter. This may be switched into the line which feeds the threshold detector from the incoming signal, acting to select particular frequency areas in the latter. The Durham unit offers three options, isolating high, mid-range and low frequency information via suitably tuned band-pass filters. The effect of these circuits is to modify the response of the threshold detector, and hence the associated processing function itself. If the range switch is in its 'low' setting, for example, the trigger will only respond to bursts of energy in this frequency region. An incoming signal will thus be processed according to the characteristics of its low frequency elements and not those of higher regions.

One additional output is available from these units: an envelope follower function consisting of a control voltage which fluctuates according to the amplitude of the applied signal. This characteristic may be used as an external control to another of the processor units, or applied elsewhere in the main voltage control system. It thus becomes possible, for instance, to use the characteristics of one signal to operate a gate which is acting upon another.

The circuits providing the amplitude characteristics of an ordinary envelope shaper may also be harnessed to supply an equivalent control voltage output, sometimes described as a trapezoid. If this function is applied to the control input of a device, the audio output of which is also being processed through the envelope shaper itself, the dynamic shaping and the acoustical content of the emergent signal will become interlinked functions, controlled by the settings specified for individual envelope characteristics. If the device chosen is a frequency controlled audio oscillator, for example, the output from the envelope shaper may consist of a tone which rises in pitch by a regulated amount as its amplitude grows, and falls again as it decays.[54]



One possibility opened up by this technique is the use of the trapezoid output to control a filter unit, providing a means for shaping both the amplitude and the timbre of an applied signal.

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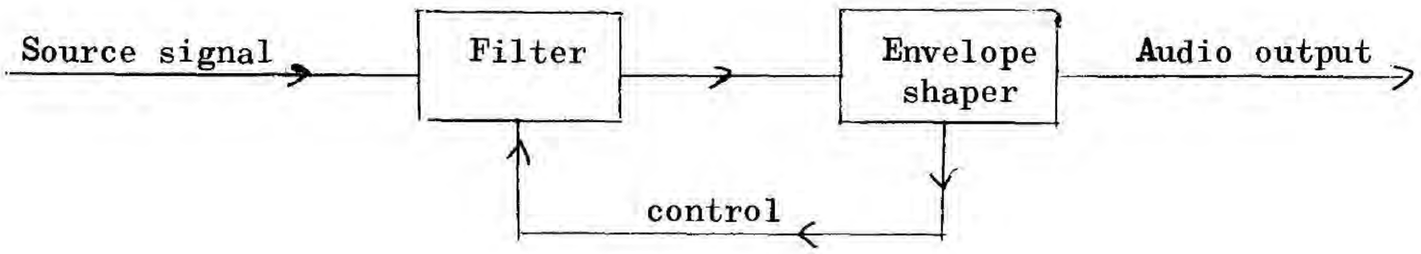
[54] It is possible to invert this, or any other voltage characteristic by means of an inverting amplifier; the tone could thus fall in pitch on attack, rising again on decay.



The difficulties associated with the synthesis of transients have already been mentioned earlier, when discussing the characteristics of the R.C.A. synthesisers, and the significance of this voltage control facility merits further scrutiny in this context. Firstly, however, the characteristics of voltage-controlled filters must be outlined, for not all the filter types encountered in 'classical' studios are suitable for modification to this method of operation. A standard filter bank, for example, consisting of a set of parallel band-pass filters of fixed response and centre frequency, is too complicated a device to be usefully controlled by simple voltage functions.[55] The manual operating procedures involve the manipulation of as many as twenty-eight different amplitude controls in the case of a third-octave bank, any number of which may prove critical in shaping the spectrum of an applied signal. The potential of such a processing technique is considerable, and it is a matter of some concern that many synthesiser manufacturers have considered this 'classical' device expendable.

The filters most commonly adapted for voltage-controlled operation

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[55] It will be seen in the fifth chapter, however, how the possibility of controlling a studio system by a computer radically alters this situation.

are single units of the band-pass, high-pass or low-pass type, where the primary controllable variable is the centre frequency, or frequency of cut-off. This provision facilitates several operating techniques. If a low-pass filter, for example, is employed to attenuate the upper harmonics of a square wave, and the frequency control inputs to both the filter and the oscillator concerned are connected to the same voltage output of a keyboard, the two devices will automatically track together.[56] This constant relationship between the two settings will result in the same harmonic spectrum being generated whatever the pitch selection. If the keyboard offers two output control voltages of independently variable sensitivity, or a second set of keys is provided, the filter and generator control settings may be made to vary relative to one another. In the former case timbre may become a function of frequency, and in the latter, each pitch may be associated with a set of discretely specifiable timbres. In common with other voltage-controllable devices these filters may be modulated by control wave generators, either sweeping the frequency spectrum of a selected audio signal, or tracking the modulatory characteristics of other studio devices connected to the same control source. In the first case both frequency and amplitude modulation characteristics are produced. The former effect is caused by the modulation of the centre frequency (or frequency of cut-off) of the filter, and the latter occurs because the proportion of the total sound energy which is passed will vary as a function of the frequency setting (unless the source is pink noise

[56] Provided the designer has taken care to see that the frequency/voltage characteristics for both devices are the same, or may be adjusted to be so.

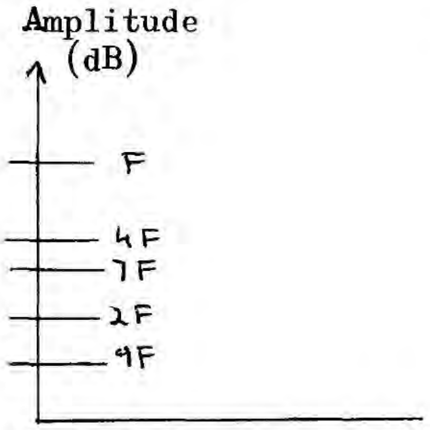
and the filter is in band-pass mode).

The acoustical effect of this type of modulation is particularly distinctive, a feature which is not immediately obvious from the above observations. If a complex sound source, rich in overtones, is processed by a low-pass filter, for example, set near the bottom of its range, only the lowest frequency components will emerge, all higher elements being heavily suppressed. As the filter is opened up, progressively higher frequency components will be allowed through, leading to an increase in the overall volume level and an associated brightening of the sound quality. If the filter is used to sweep continuously the frequency range of the applied signal, the effect is that of a changing timbre which may best be described onomatopaeically as a 'wah-wah', a term invented by electric guitarists who were amongst the first to exploit this technique.

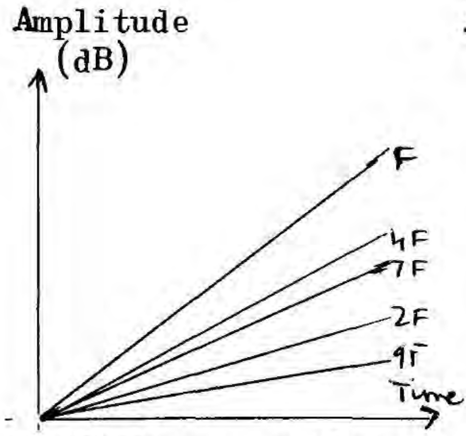
From a composer's point of view the primary interest in this technique is its potential to produce sounds which may be made to vary dynamically in acoustical content. However, it is still only possible to produce a very general shaping of his material.

A picture of some significance is now beginning to emerge regarding the characteristics of voltage-controlled systems. The most important general feature to have resulted from the introduction of this technology is an ability to operate devices electronically via a control network, which may be considered in several respects to be independent of the generation and processing devices themselves. The production of musical material thus became heavily dependent on the type and range of voltage functions made available by the system. Repeating wave forms generated from oscillators designed to work at lower frequency ranges have provided the most convenient method of producing voltage-controlled characteristics which vary dynamically without continuous manual intervention. The inflexibility of such repetitive functions, however, has proved to be a major procedural stumbling block and, as will be seen shortly, several designers have come to realise that the key to the advancement of these electronic music systems lies in the development of more sophisticated methods of device specification and control.

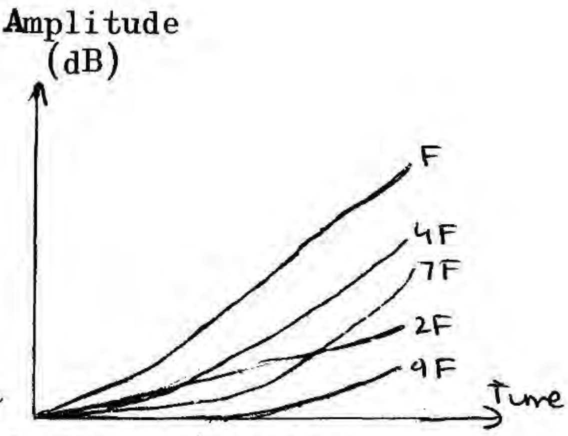
The linking of a filter and an envelope shaper together in the manner suggested earlier to produce a superimposition of envelope and timbre functions would appear to provide a more flexible system for shaping sound events. The effect of a sharp attack on a harmonically rich signal, for example, would be both a rapid growth in amplitude and also a change in timbre, which may be varied in degree by regulating the sensitivity of the trapezoid control output of the envelope shaper. The resulting transient would be more complex than that associated with a simple envelope function, for the relative strengths of each harmonic component will change during the attack. Consider the following illustration of the possible effect of a low-pass filter acting on selected harmonics of a signal source:



1 Relative strength of selected harmonics in a source signal

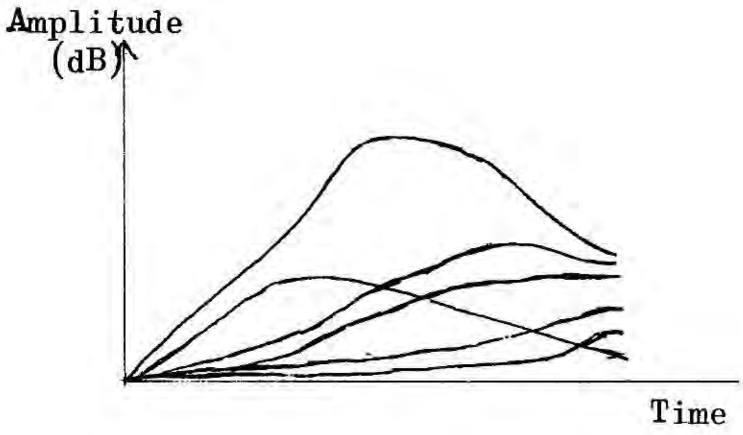


2 Ordinary envelope attack (expanded time scale)



3 Filter/envelope attack using a low-pass characteristic, and a filter sweep which increases beyond the frequencies of the selected harmonics

If a band-pass filter is employed in this manner, a far more varied characteristic will be produced, dependent on both the frequency sweep and the band-width. If the former is fairly large and the latter fairly small, the filter may affect the overall amplitude function of the envelope considerably. Using the previous illustration as a basis, the following attack curve might be produced:



The harmonics now are executing quite different attack characteristics, and the growth of the sound will be far more varied. Furthermore, the timbre of the sound, when the end of the attack is reached, will also have been affected considerably. The composer, however, still does not have precise control over the function of each individual harmonic component and the range of characteristics open to him are still very limited. A filter/envelope is all too easily adjusted to produce only 'wah-wah'-like effects, described earlier, and it should be noted that shapes such as these have proved to be one of the most over-used sounds in the construction of popular electronic jingles.

The underlying problem encountered in transient generation, and indeed timbre/dynamic shaping in general, still remains one of specification. The only method of achieving some measure of independent control over the characteristics of individual harmonic components in a pitched sound event is to synthesise each element separately, using a set of accurately tuned sine wave generators and an associated bank of envelope shapers connected to a common trigger circuit. The true potential of such an approach, however, can only be realised if complete control may be exercised over the characteristics of the attack and decay curves. Unfortunately, as indicated earlier, most envelope shapers supplied in synthesisers only provide single function curves, the electronic intricacies associated with multi-function circuits proving too costly to be considered commercially viable.

Two further restrictions are likely to be encountered in attempting the latter approach. Firstly, a major drawback in all but the most expensively designed voltage-controlled devices is their fairly poor stability and accuracy. In the case of oscillators, it is extremely difficult to set up a Fourier series of overtones from a selected fundamental with sufficient precision, except where the operation is restricted to the very lowest order components. Secondly, most electronic music studios include only one or two envelope shapers as part of their systems. This factor raises one very important general consideration in evaluating synthesiser design, which has not yet been discussed: modularity.

The functional characteristics of any 'classical' or voltage-controlled system are determined by two primary factors: 1) The type and number of the different devices which are available; 2) The actual facilities offered by each device. Where a studio is unable or unwilling to design and develop its own equipment it must be accepted that the second factor will be entirely determined by the philosophy of the chosen commercial manufacturer. There is no fundamental reason, however, why the first should not be a matter which the composers and engineers associated with a particular studio should be able to determine. If each device is available in a modular form, and care has been taken to provide mutually compatible input and output characteristics, it will then prove possible for a system to be purchased initially in a very basic form, to which extra devices and ancillary equipment may be added as funds become available.

Certain manufacturers such as Moog have been sufficiently far-sighted to make their devices available as individual units. These may be purchased in standard configurations, but individual modules may be added to nearly all of their systems with a minimum of difficulty. Other firms such as E.M.S.(London) have concentrated almost entirely on the production of synthesisers, both large and small, which are entirely fixed as regards the selection and layout of facilities.

At the bottom end of the equipment scale special circumstances arise which favour the use of compact, fixed design units. These are associated with the use of electronics in live situations, where a synthesiser is treated as a performance box, either as a sound generation system in its own right, or as a sound processor for instrumental sources. Under these conditions a performer will often regard the box itself in instrumental terms, and will thus wish to explore only a specific range of functional characteristics. Ease and simplicity of manual operation in such circumstances becomes of greater relevance than freedom of device substitution, providing that the synthesiser offers a sufficiently useful set of basic characteristics. A total lack of freedom in the choice of the latter is nevertheless still a drawback, and it is very disconcerting to discover models from the same manufacturer marketed concurrently, such as the E.M.S.(London) V.C.S.3 and A.K.S., which would appear, superficially, to supply different facilities but which prove in reality to offer only limited variations on a fixed theme.

At the other end of the scale, involving systems costing upwards of £10,000, the attractions of being able to purchase a complete synthesiser off a manufacturer's shelf should not be allowed to outweigh several important musical disadvantages. Firstly, a studio, in accepting a mass-produced system design, is imposing a single philosophy of electronic synthesis on all who choose to use it. Secondly, the inability to alter the choice of facilities leads to a tacit acceptance of the system as a fixed entity. Thirdly, this inevitably restricts composers in the development of their own particular methods of working with sound material, unless it should prove possible to increase the potential usefulness of the existing devices by providing externally an improved method of procedural control.

Solutions to the latter situation, as already noted, have been created by the introduction of computer-based facilities in certain studios, providing a fully programmable interface between the composer and the machine.[57] It must be realised, however, that the computer in effect, replaces many of the existing functions offered by voltage-controlled procedures, and this involves an entirely new set of operational characteristics. In the absence of such facilities the all too real limitations of fixed designs will prove a major source of frustration for those

[57] Chapter 4 of this thesis is concerned with a study of the problems of musical communication encountered in developing computer-based sound synthesis systems.

composers who wish to explore the possibilities of analogue sound generation and processing to the fullest extent. Modularity in itself does not offer solutions to many of the major problems encountered in sound synthesis, but it at least preserves an important degree of flexibility in allowing the development of a studio according to the requirements of its principal users.

The research and development which led eventually to the construction of these computer-controlled studios grew out of a realisation during the 1960s that the control facilities of the early commercial synthesisers were far from adequate. As investigations into computer-based techniques advanced, certain hybrid devices were incorporated into the designs of both large and small voltage-controlled synthesisers in an attempt to increase their versatility.

The link between digital technology and voltage-controlled synthesis was developed from the analogue side in the first instance by a study of the characteristics of electronic pulse generators, in particular their ability to provide gating functions. When a square wave is employed to modulate the amplitude of a voltage-controllable device between the two states of 'on' and 'off', as discussed earlier, it creates a switching function as it changes between two steady D.C. voltage levels at equally spaced time intervals.[58]

More complicated switching patterns may be obtained by employing a multi-function pulse generator in place of the square wave oscillator.[59] This device is sometimes based on the techniques of threshold detection, discussed earlier in the chapter in connection with signal-controlled

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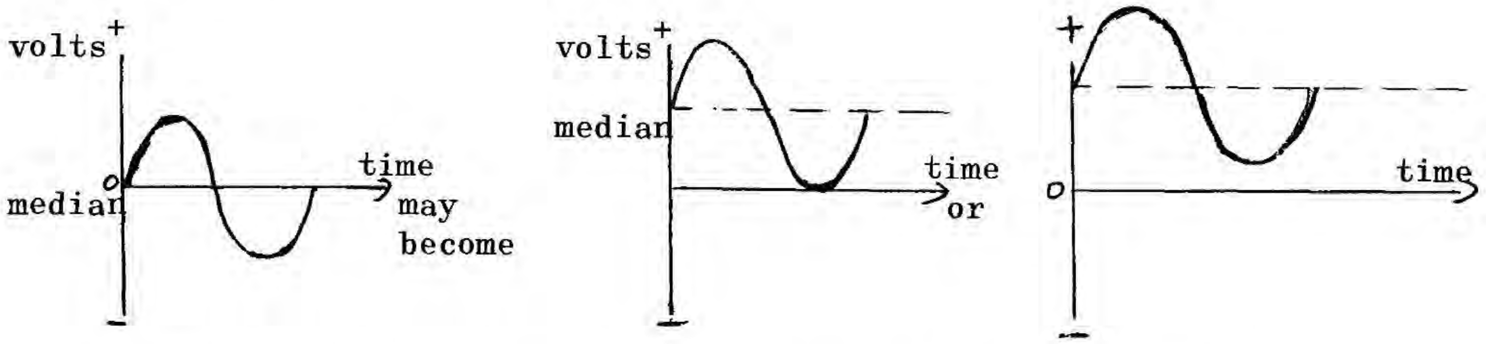
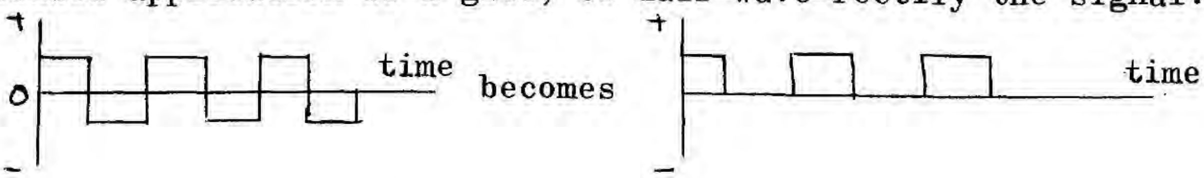
[58] A distinction should be made between unipolar and bipolar control functions, for the difference between the two has proved a source of confusion for composers who are not fully conversant with electronic theory. In the description of a sine wave oscillator given earlier in the chapter its function was described in bipolar terms, fluctuating positively and negatively about a median potential of zero volts. This is electronically convenient for it provides an unchanging central reference point for sinusoidal patterns of different amplitudes. Such a function may be made unipolar if a fixed positive D.C. voltage bias is added to the output, raising the potential of the lowest part of the wave to a value of zero volts or above. The corresponding effect on the median level is to raise its potential to some positive value. Whenever a control wave oscillator is applied as an external input for a device function which is also manually controllable the latter may supply just such a bias:



In the case of pulse activated devices, such as are shortly to be discussed, it is more convenient to design their control inputs to be unipolar, such that an 'off' state is represented by zero volts and an 'on' state by a positive voltage value. If a suitable bias is added to a bipolar pulse source such as a square wave generator it is possible to raise its potential by the required amount, but it is usually more convenient in view of the nature of the source and its intended application as a gate, to half-wave-rectify the signal:



[59] The use of pulses as a source of audio signals has already been discussed in chapter 2, pages 163-169.

amplitude processors. These are applied to produce a voltage-controllable source of trigger pulses using a modified Schmitt trigger, where the pattern of pulses produced may be made dependent on the characteristics of an external voltage source. The basic circuit consists of the detector, a control input, a threshold level regulator and an output. If a voltage function is applied to the input, the threshold detector senses when the potential reaches its pre-set value. The switch then activates and the trigger fires, producing a D.C. voltage pulse at the output. Two options are available at this point depending on the design of the associated circuitry. The trigger, for example, may remain on, or may switch on and off repeatedly until the voltage level falls back below the threshold level again. By coupling the device to an oscillating control source such as a sine wave generator and then varying the threshold detection voltage level, assymmetric pulse patterns may be produced.

A refinement of this triggering system is employed in a device known as a timing pulse generator. Three operating modes are normally provided: 1. 'single shot', manual firing via a push button, one pulse only being produced each time the button is depressed; 2. repetitive mode firing, where the generator fires repeatedly at a rate proportional to an applied voltage level; 3. pulse burst firing, where the generator starts and stops firing at a selected rate in response to external trigger pulses. If these pulse patterns are produced at audio speeds, highly complex timbres may be generated.

The use of two fixed voltages to determine the information content of a signal is directly related to the techniques employed in digital processing, where all data and instructions are specified in terms of binary number patterns, the digit 0 being used to indicate an 'off' state and the digit 1 being used to indicate an 'on' state. The digital equivalents of gating circuits are a series of four logic functions known mnemonically as AND, NAND, OR and NOR. Each of these functions examines the states of two digital inputs and generates an output pulse if particular logic conditions are detected. An AND gate will only produce an 'on' pulse if both inputs are activated simultaneously. A NAND gate conversely will produce a pulse if neither or just one of the inputs is activated but not both. An OR gate will produce a pulse if either one, or both inputs are activated, and a NOR gate will pulse only if neither of the inputs is activated. These characteristics may best be illustrated diagrammatically as a truth table:

Inputs		Logic gate function			
A	B	AND	NAND	OR	NOR
			X		X
	#		X	X	
#			X	X	
#	#	X		X	

= input in active 'on' state

x = output pulse generated

It may be seen from this simple comparison of analogue and digital equivalents that the use of logic functions for the latter considerably enhances the potential of such circuits for switching applications.

Pulses by their very nature cannot directly provide analogue voltage characteristics. They may, however, be usefully employed for controlling the operation of special voltage generation systems known as sequencers. This descriptive title is a little misleading, for it may generally be applied to a number of studio devices. A keyboard, for example, is a manually operated sequencer since its output consists of a series of discrete voltages which may be manipulated to produce a continuous progression of events. In larger studios, however, the term is normally reserved for devices which produce a serial output of voltage levels under electronic control, and these are sometimes labelled more explicitly as variable function generators.

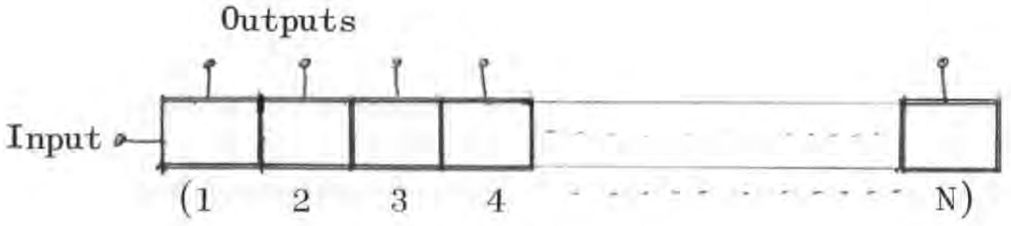
The basic sequencer consists of a set of D.C. voltage supplies, each of which may be individually regulated, and a switching system which connects each supply in turn to an output line. These devices are usually unipolar in output providing an operating range from 0 volts to about +5 or +10 volts. A negative D.C. shift may, however be applied to make the device bipolar, should this be required.[60] Some early prototypes used an electromechanical switching system consisting of a relay activated rotary switch. Most modern designs use electronic switching techniques which are not only less cumbersome in terms of physical design but also capable of far faster speeds of operation.

The total number of individual voltage steps which may be produced varies from one design to another. Some smaller models provide less than twenty elements and are thus somewhat limited in their application

[60] See note 58

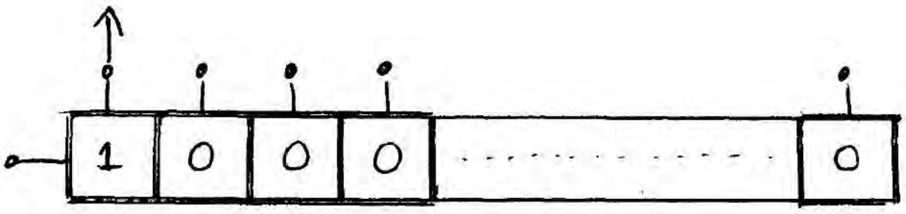
as programmers of events. More comprehensive designs are capable of sequencing fifty, or even upwards of a hundred successive events. Three modes of operation are normally available: 1. single step, where the switch system may be manually advanced, position by position; 2. single run, where in response to suitable clock pulses the sequencer will work through the bank of voltages only once; 3. repetitive mode, where the switching system, on encountering the end of the bank, will automatically loop back to the beginning and continue to cycle until the source of control pulses is terminated. Composers will often wish to specify a definite number of voltage steps for a particular sequence, and it is usual for a thumbwheel switch or numbered dial to be fitted to the input which may be used to truncate the working length of the bank to the required size.

Modern switching systems are usually digital, based on the characteristics of a shift register. This device is a digital counter, consisting of a bank of binary storage locations wired in series.

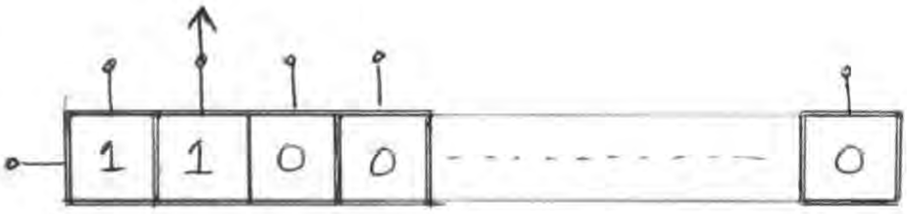


Each location may either be de-energised (indicated below by the figure 0) or energised (indicated by the figure 1). The input to the register consists of a single terminal at one end of the bank, and the output consists of a string of individual terminals wired one to each location.

Initially the register is cleared so that all locations are de-energised. At the start of a sequence the register accepts an 'on' pulse from the input, which energises the first location. This results in a pulse being sent down output line no.1, which in turn is used to effect a connection from the first voltage setting to the output of the sequencer.



When the next 'on' pulse is received at the input, the charge in location no.1 is pushed into location no.2 by the incoming charge which itself enters the chain. The change of state in location no.2 results in a pulse being sent down its own output line, the 'on' state of location no.1 being now ignored. This in turn causes the second voltage setting to be connected to the output of the sequencer in place of the first.



Further input pulses result in a continuation of this progression, the leading 'on' state moving progressively down the register, activating output lines one at a time, until the last location is reached. Depending on the state of the mode switch, the register will either stop at at this point or be cleared to restart the sequence.

An alternative method of register operation involves the inclusion of a start pulse generator and a shift pulse generator in the switching system. At the beginning of each cycle the register is cleared as in the former method, and the start pulse generator connected between the pulse input and the register. The first pulse received activates the start pulse generator which inserts a '1' in shift register location no.1, resulting in turn in the connection of the first voltage step to the sequencer output. The start pulse generator is then automatically disconnected and replaced by the shift pulse generator. The next incoming pulse activates the latter, which by means of a logic gate produces a '0' output. This enters the register in position one, displacing the '1' to location no.2. Subsequent incoming pulses result in further zeros entering location no.1, pushing the single '1' pulse steadily down the register, activating outputs until the end position is detected, when the whole process is either stopped or recycled. This method of operation avoids the need to inhibit pulses subsequent to the first.

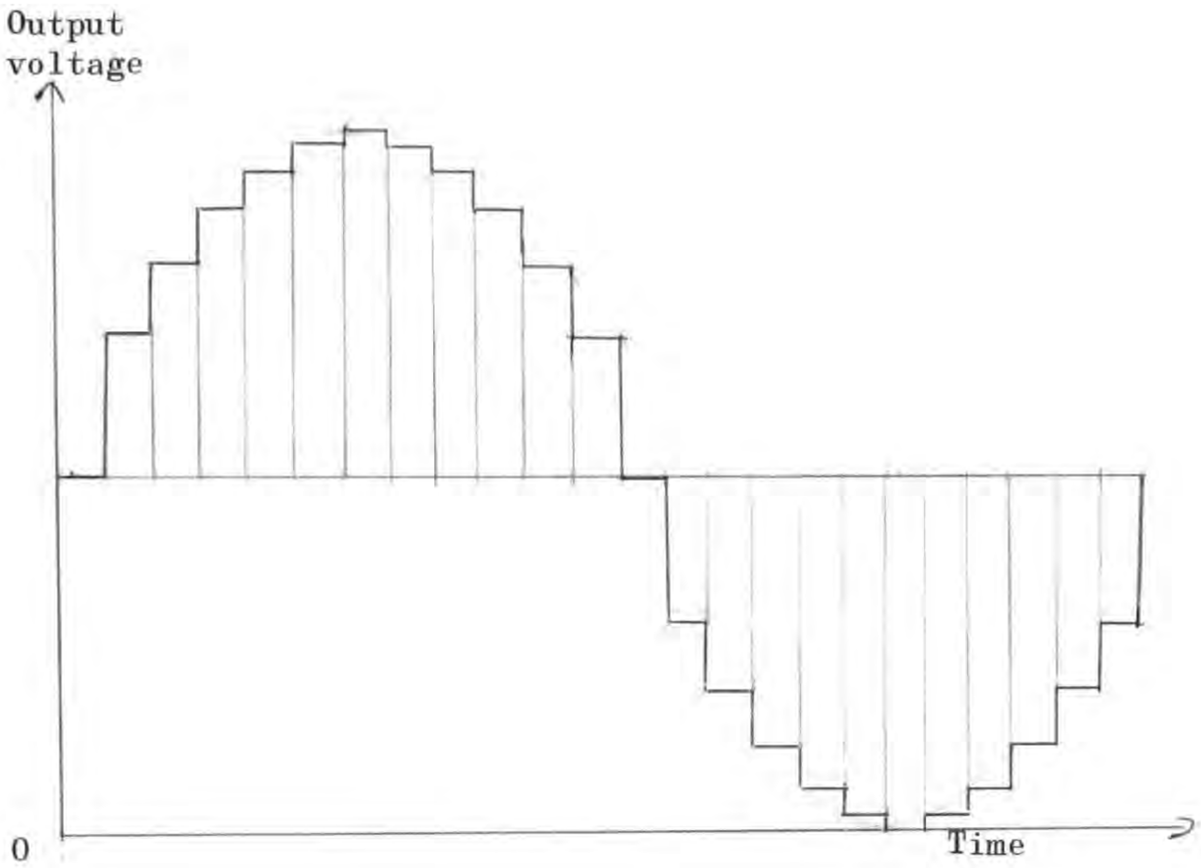
Sequencers need not be restricted to the production of single voltage functions. Some models provide two or even three parallel banks with independent outputs. Normally, however, a common switching system is employed, ensuring that the step by step changes in each bank occur simultaneously. This facilitates the synchronous control of two device characteristics; for example, frequency from one bank and amplitude from another. Given an external pulse rate control facility, more complicated rhythmic patterns may be produced by applying the voltage output from a second (or third) bank to this input. The time taken to transfer from one step position to the next then becomes a function of the voltage levels in this bank, creating a voltage-to-time conversion system.

From a musical standpoint the primary characteristic of a sequencer is its ability to realise a programmed series of instructions. The usefulness of such a device centres on the range of voltage functions which may be produced, and the sophistication of the means by which they may be specified. Sequencers which only offer staircase functions, that is, progressions of D.C. voltage steps, suffer from one important limitation: they cannot provide a continuously varying output. If, for example, an attempt is made to simulate a continuous glissando, using a sequencer to control the frequency input of a generation over an octave range, almost the entire resources of a 100-step bank will be required, each level set with painstaking accuracy by hand. Such an operation would be extremely time consuming for an effect which could, with care, be achieved by a single sweep of a manual joystick control. This illustration is an extreme case, but it highlights the problem of using discrete steps to create a gentle progression of parameter changes, particularly where these concern pitch information. A solution to this difficulty might be to enlarge the size of the sequencer bank to 250 or even 500 elements, increasing the number of variable voltage steps available for specifying changes of events. Such an improved accuracy, however, could only be achieved if the composer concerned is prepared to expend considerable time and effort converting his or her musical ideas into a far greater number of individual settings.

It will be seen later that the use of computers, either as control systems for analogue studios or as synthesis systems in their own right, involves the specification of procedures entirely in terms of discrete steps. The availability of arithmetic and logic programming facilities, however, permits the composer to specify creative ideas in terms of general functional descriptions delegating to the computer the task of calculating the individual steps internally. A simple sequencer cannot, unfortunately, provide such a sophisticated facility.

Step functions are of particular value when a sequencer is employed to control the production of musical events which involve simple and precise specifications of pitch, duration and timbre. The 'note' concept of electronic synthesis has already been outlined in discussing the R.C.A. synthesiser, and it may be seen that the elementary sequencer is particularly suited to such applications as the electronic imitation of instrumental textures and the acoustical realisation of serially based compositional techniques.

One other application of this type of sequencer meriting study concerns its use as an audio wave generator in its own right. If the bank is switched steadily in repetitive mode at a sufficiently fast speed the output may be used directly as an audio signal source. Varying the voltage levels of the bank elements thus affects the timbre of the wave (and also its amplitude). Twenty-four elements might be set to create the following approximation to a sine wave:



The general characteristic of this wave, as may be seen from the above diagram, is approximately sinusoidal, and will be perceived aurally as a note with a clearly defined fundamental frequency. The stepped irregularities, however, introduce a considerable amount of harmonic distortion. If the number of individual elements is increased, the distortion factor will correspondingly decrease, for the outline of the wave shape becomes progressively smoother. The ear, unfortunately, is extremely sensitive to such defects and for lower range audio frequencies up to 500 or more, accurately set voltage steps may be required in certain situations to achieve an acceptable degree of purity.[61] Once again, such a proposition is far from practicable as a manual function

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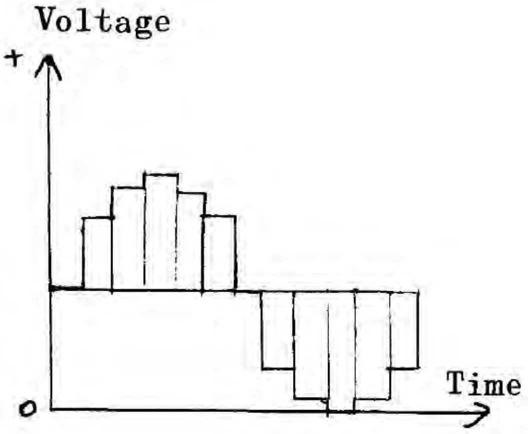
[61] This principle of wave quantizing in terms of discrete levels provides the basis for generating acoustical information directly from a computer via a digital to analogue converter, and the characteristics of this technology will be discussed in greater depth in the fourth chapter.

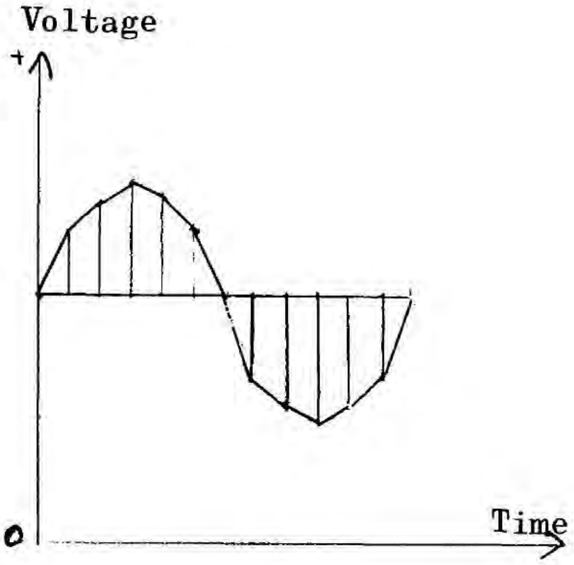
in view of the inordinate number of device settings involved.

One or two designers have developed sequencers which are capable of producing voltage gradients in addition to voltage steps. This extra facility enables a composer to achieve more accurate approximations to continuous curves, with far fewer elements. Contrast a sine wave constructed out of twelve D.C. voltage steps with one constructed out of twelve gradients:

i)

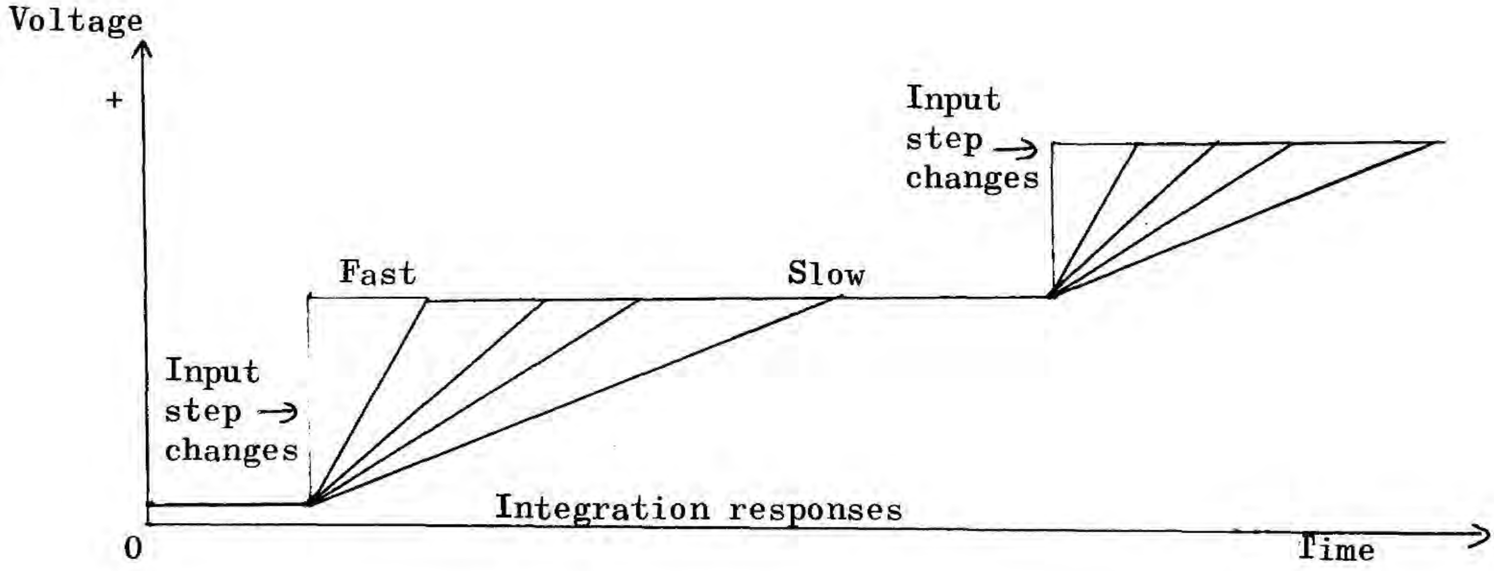


ii)



The gradient technique clearly produces a better approximation. As few as twenty-five elements are required, using this method, to achieve a sine function sufficiently pure for many studio applications. Such a facility also greatly enhances the use of the sequencer as a voltage source for the control of other devices, for it becomes possible to generate continuously changing, as opposed to step, characteristics.

The production of a voltage gradient between one D.C. level and another requires an electronic function capable of interpolating between two potentials. This is provided by an analogue integrator; essentially a charging circuit which will change potential positively or negatively at a controllable rate until inhibited, usually by a level detection circuit such as a Schmitt trigger. Some of the large voltage-controlled systems offer an integration facility as a device in its own right, known generally as a slew limiter. This is a unity gain amplifier which will track changes in applied input voltage levels with a response time which may be varied either manually or by voltage control. A slew limiter is of particular value for keyboard-controlled operations, and this device is sometimes included as an integral part of the keyboard system itself. The response characteristic is approximately linear:



The use of such a technique to smooth the staircase characteristic of a sequencer raises special problems, for it is vital that the response time of the integrator adjusts for each step to match both its size and its duration. This necessitates a buffer system which will measure these quantities before the integrator is applied and an output generated.

A sequencer capable of producing such gradients (described as a double variable function generator) has been specially constructed by the Institute of Sonology, Utrecht for use in their main voltage-controlled studio.[62] This device consists of two 100-step voltage banks which are linked synchronously, corresponding step positions being controlled by twin potentiometers mounted on common spindles. All the standard sequencer facilities are supplied: variable step bank size, single step/single run/repetitive mode operation, internal/external pulse rate control, and voltage/time conversion, using the second bank's settings as a regulator for the clock of the first. Useful monitoring facilities are also provided, including an oscilloscope to display the voltage characteristics, a digital voltmeter to allow each voltage step to be set accurately, and a number tube display to indicate the position of the step switch along the bank.

In addition, the following facility is available to convert the staircase outputs of the two banks into a single function of gradients: The potentiometers associated with the first bank operate to give voltage settings over the normal output range of 0 to +5 volts. The steps produced by the second bank are given a negative D.C. bias of -2.5 volts to make their characteristics bipolar, and then amplified by a factor of two to give an operating range of -5 to +5 volts. The steps of this bank are then sent to an integrator which starts from zero potential and integrates positively or negatively to the specified voltage level for each step over the time interval specified. At the end of each step the integrator is discharged to zero potential, ready to start the next integration.

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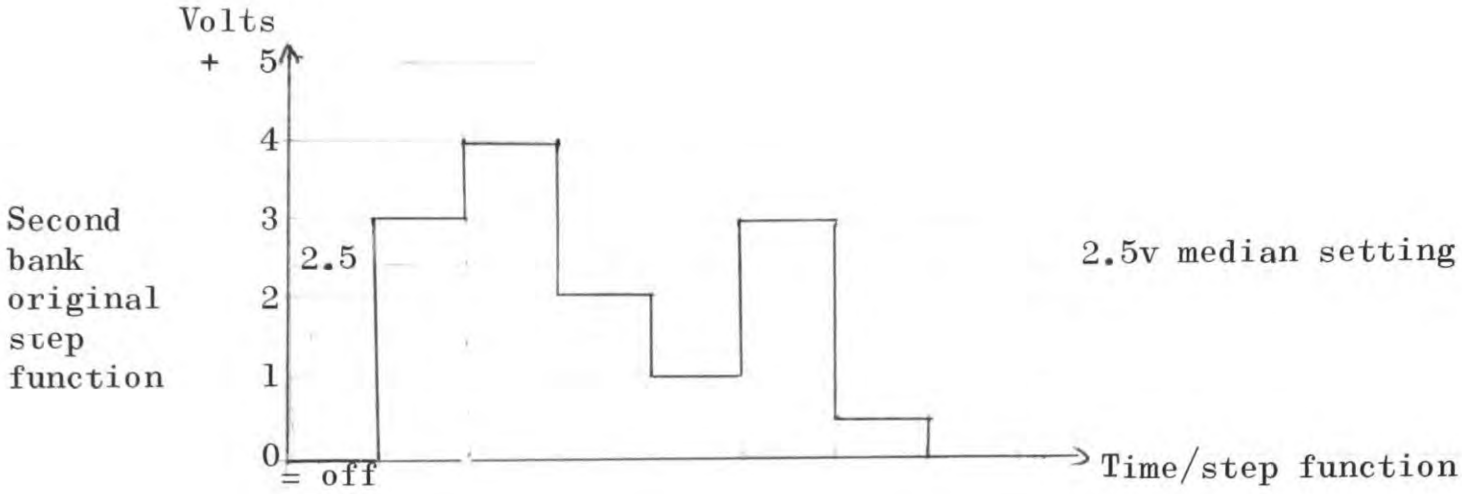
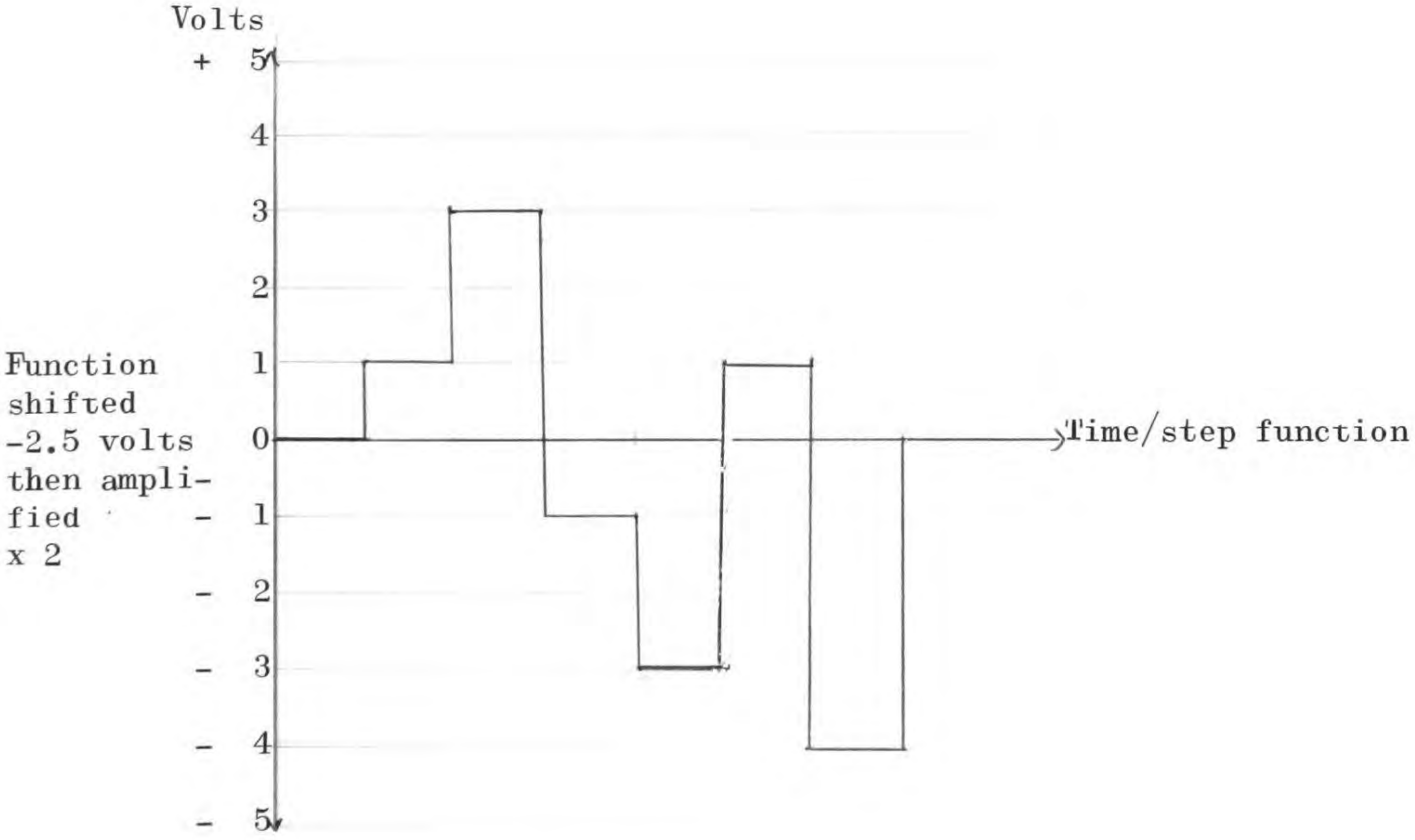
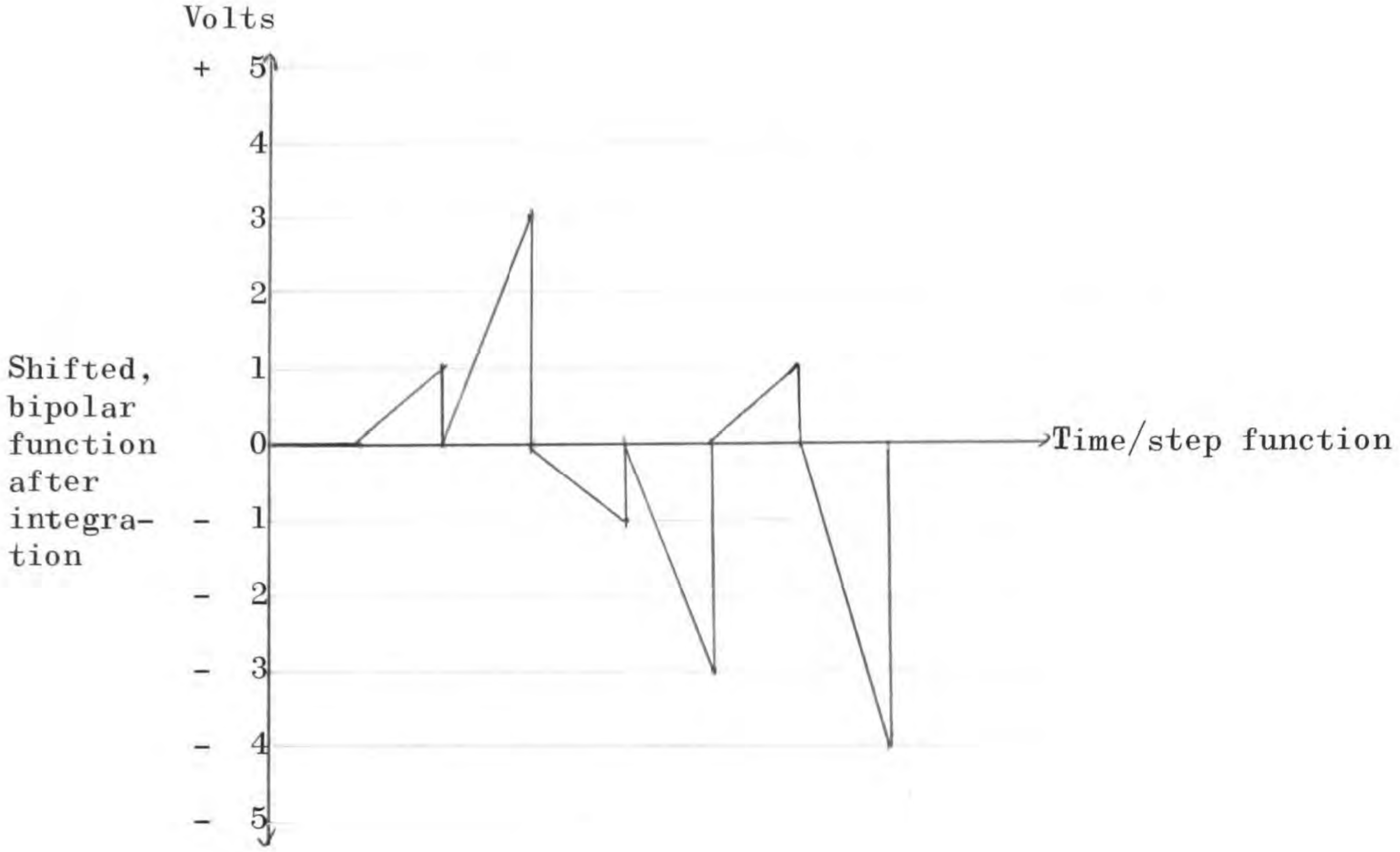
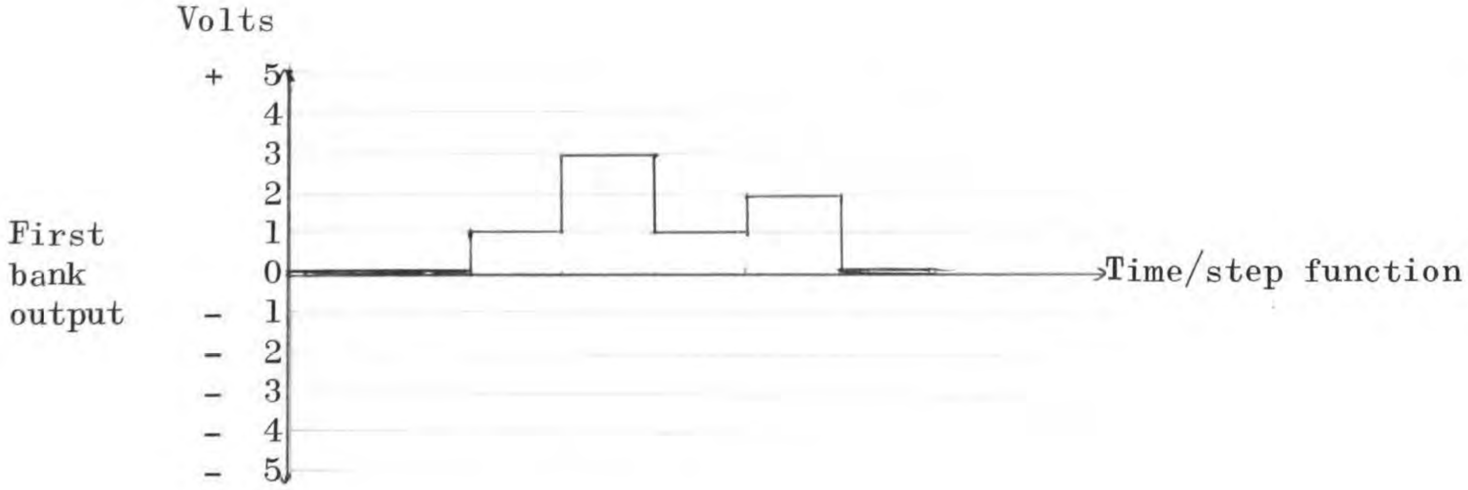
[62] See Stan Tempelaars, 'A Double Variable Function Generator', Electronic Music Reports, 2 (Institute of Sonology, Utrecht State University, Netherlands, July, 1970), pp.13-31

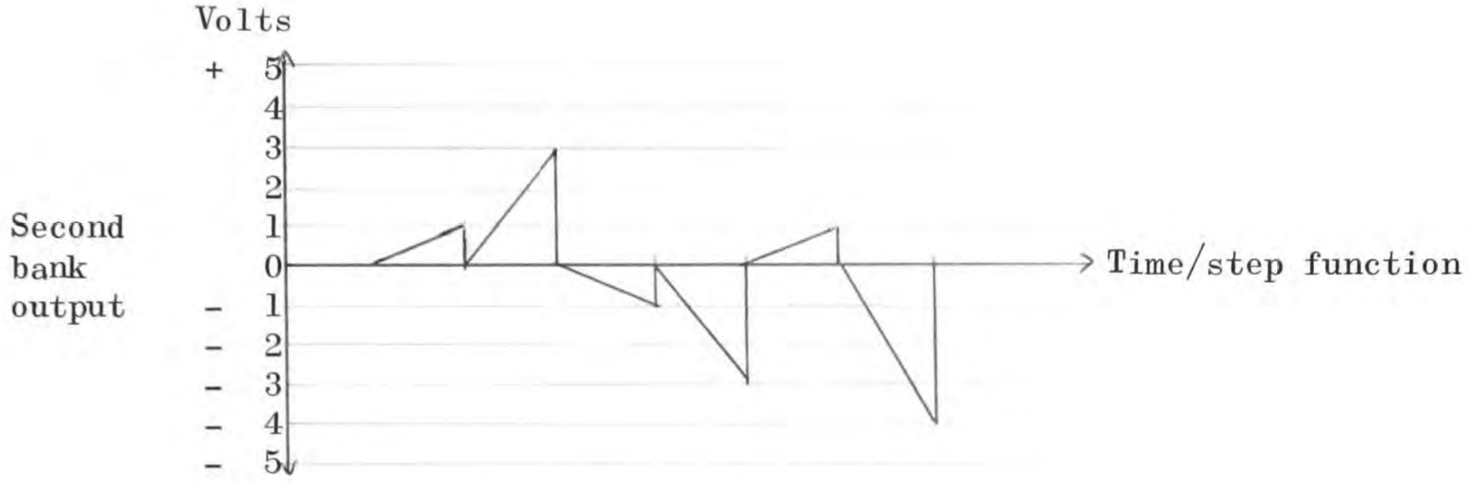

Volts
+ 5
4
3
2.5
2
1
0
= off
Second bank original step function
2.5v median setting
Time/step function

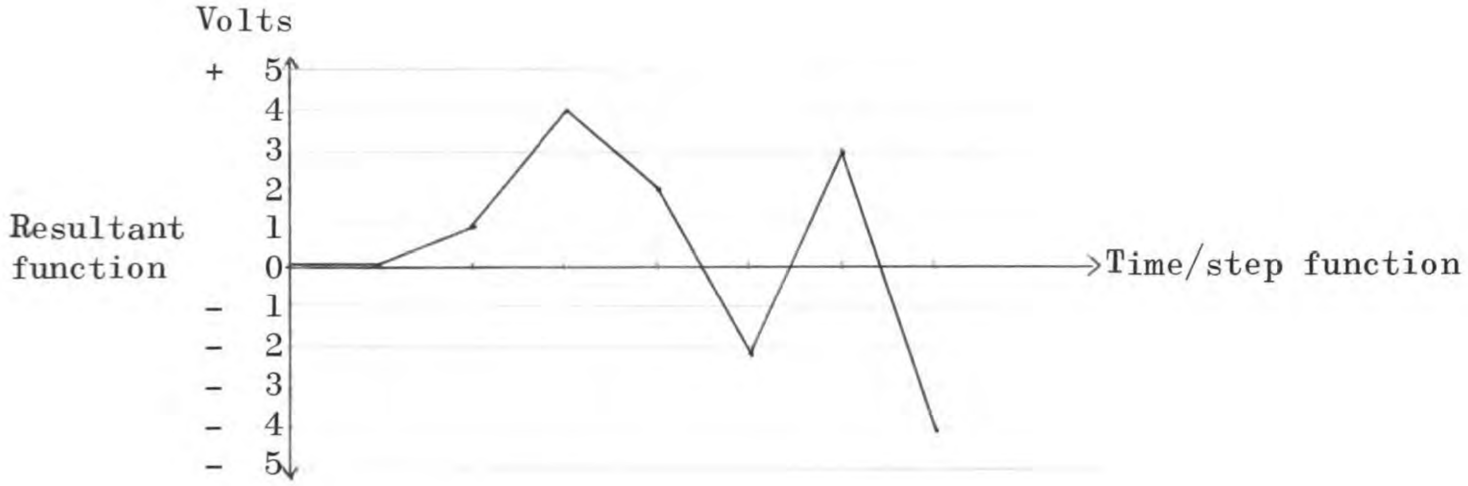

Volts
+ 5
4
3
2
1
0
- 1
- 2
- 3
- 4
- 5
Function shifted -2.5 volts then amplified x 2
Time/step function



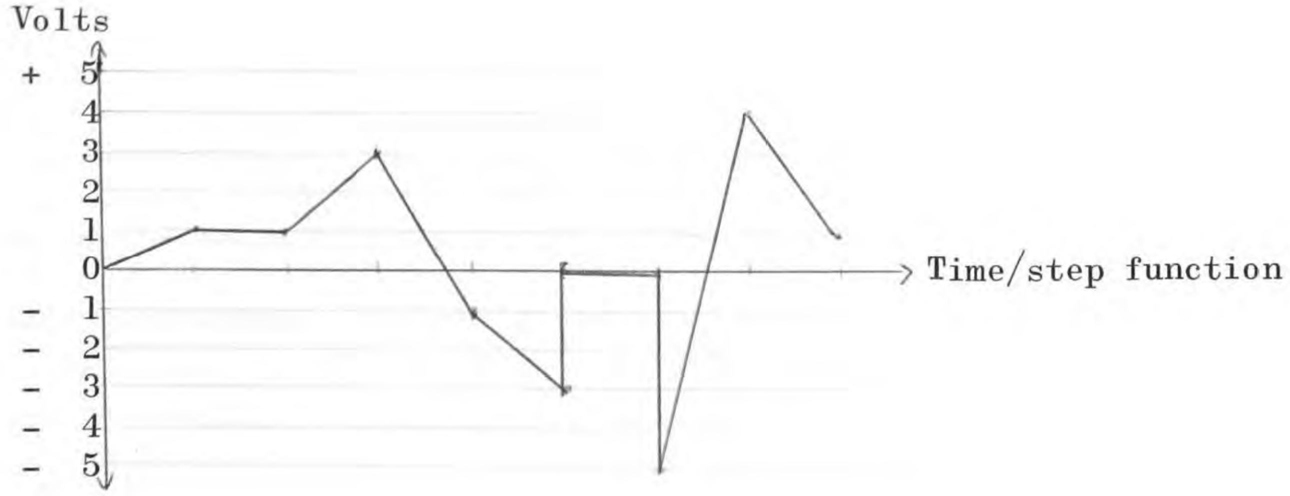
The output of the integrator is then added to the stepped output of the first bank. Provided suitable combinations of settings for the two banks are chosen, a continuous function may be generated.







Functions need not be continuous; for example, a mixture of steps and gradients may be produced. This is of particular value when employing the sequencer as a control source for other devices.



The engineers have given careful consideration to the composer in designing the system, for it is possible to assemble functions visually, using the oscilloscope as a monitoring device, simply by manipulating the two potentiometers available for each sequencer step. The 'start' voltage level is set by manipulating the control for the first bank only. A gradient, varying from flat to positive or negative, may then be created by adjusting the control for the second bank above or below its median setting. This graphical means of specification adds an important new dimension to the characteristics of input systems so far studied, and the ability to construct wave shapes directly in terms of line approximations merits closer study.

Earlier discussions have drawn attention to the problems encountered in attempting Fourier syntheses in an electronic music studio, in particular the number of different sine wave generators required and the high degree of tuning accuracy and frequency stability demanded of each unit. The direct specification of wave shapes, by contrast, provides a powerful degree of freedom in the construction of stable harmonic spectra, the only major constraints being associated with hand-setting errors, and also the associated limitations over accuracy encountered in the use of line approximations for higher order harmonics. From the composer's point of view the process suffers, nevertheless, from an important practical restriction. If individual generators are used for Fourier synthesis, it is relatively easy to make dynamic alterations to the frequency spectrum by continuously varying the strengths of individual generators. Using a sequencer, the slightest change in harmonic content will involve careful manipulation of several of the potentiometer settings making it impossible to effect continuous changes in timbre, and making the process of altering wave shape specifications dynamically a time consuming if not impossible task. On the positive side, however, it should be noted that once a characteristic has been specified, the sequencer may easily be manipulated as a special source generator by varying the speed of the switching system to alter its frequency, and also by fitting an attenuator onto the output to control amplitude. The former operation would be difficult to implement if individual generators were employed, for it will prove impracticable to control together the frequency inputs for all the units with sufficient accuracy.

One feature which is not offered by the types of sequencer so far described is a facility for registering and storing patterns of control voltages produced elsewhere in the system. The basis for a design capable of functioning in this manner may be found in a device known as a sample/hold processor. In its simplest form this consists of a voltage detection circuit which will record the instantaneous value of a fluctuating voltage when activated by a trigger pulse. This information is then output as a steady D.C. voltage until over-ridden by the new level associated with the next pulse. If a regular series of pulses is employed, a step pattern approximation to the applied voltage characteristic will be produced. This process is thus essentially the reverse of that associated with the standard sequencer described above.

Analogue circuitry may be employed for the sample/hold function if the only requirement is the production of 'real time' steps from a studio control-voltage source. If these voltage steps are to be stored, however, some form of electronic memory bank must be provided. Although analogue voltage storage circuits are perfectly feasible, designers have generally preferred to develop digitally based systems, for these are not only more compact but also far more versatile.

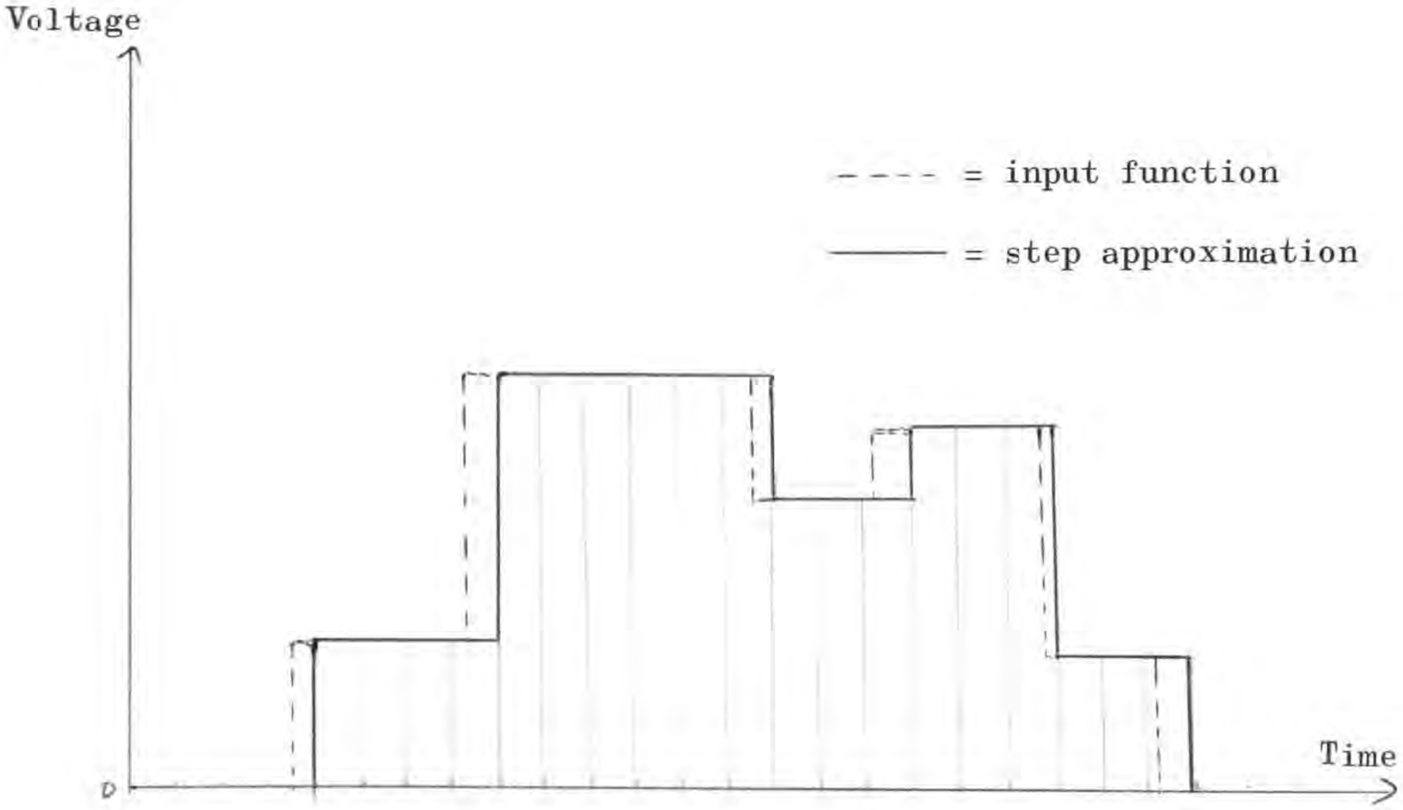
In their basic form these devices consist of a bank of digital storage locations, known as 'bits' organised into groups or 'words' of a fixed size. To such a bank is attached a switching system for altering the position of an input/output pointer from word to word, a clock for controlling the speed of switching, and a pair of converters, one for translating analogue voltages into digital equivalents, and the other for performing the reverse operation. At the beginning of a sampling sequence the word memory is cleared and the word pointer set to its start position. The clock is set to give a suitable rate of switching pulses, the control voltage function is connected to the input and the system switched on. At the first clock pulse the instantaneous value of the input voltage is read and converted into a digital numerical equivalent as a pattern of 'on' and 'off' states using the analogue to digital converter. This information is then read into the first word in the memory. The word pointer is advanced to the second location and the process of sampling, conversion and storage repeated on receiving the next clock pulse. This process may then be repeated until the end of the word bank is reached, at which point the sampling process automatically terminates, unless the bank is re-cycled and previous function values overwritten. The sequence of voltage levels thus recorded may be recovered by reversing the process. The contents of each word are examined in turn, a pattern representing their contents sent to the digital to analogue converter in response to a clock pulse and an equivalent D.C. output voltage generated.

One interesting feature which should be considered is the effect of varying the input and output clock rates independently. Such an operation makes it possible to alter the rate of change of events on reproduction without affecting the frequency of the acoustical content.[63] The use of a low speed input clock rate permits, for example, a long, slowly changing progression of keyboard voltages to be encoded economically in terms of a relatively few number of samples. On recovery the whole sequence may then be reproduced exactly as recorded, or speeded up simply by increasing the clock rate. Since the actual voltage levels do not alter, the characteristics produced by the devices to which they are applied change only in duration. Conversely, a short, rapidly changing voltage function may be sampled at a high clock rate for subsequent reproduction at a slower speed.

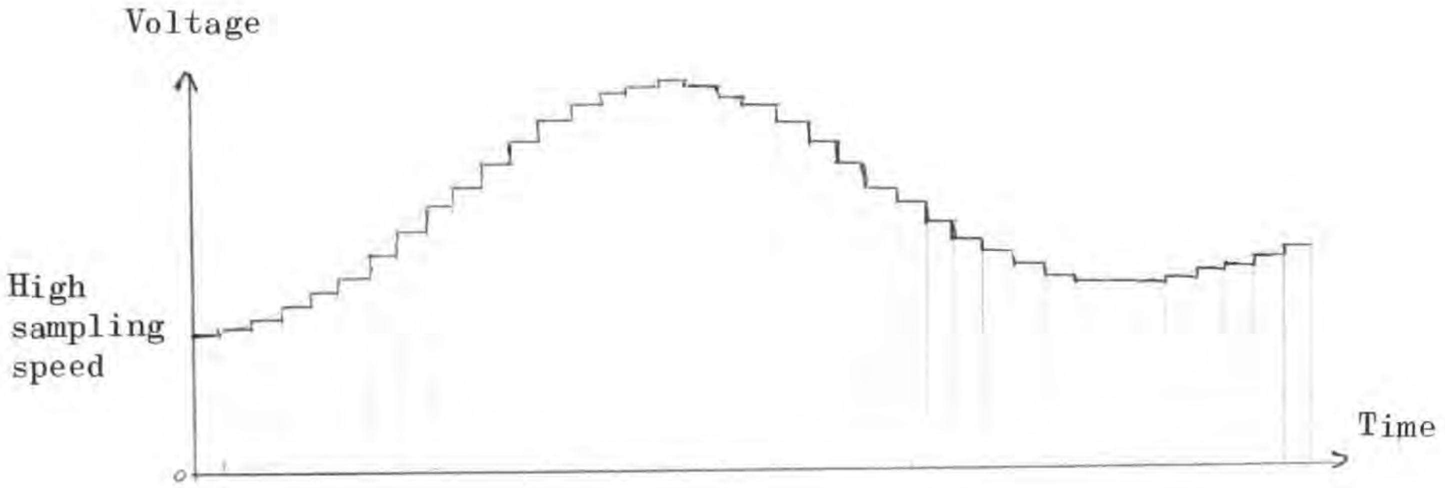
Some designs of these digital sequencers not only allow the clock rate to be slowed down to a dead stop but also permit the clock to be run backwards, reversing the direction of scan through the memory. If a link switch is provided for connecting the last location to the first, the contents may be read out cyclically. The functional limitations of this storage and retrieval system are determined both by the number of words available in the bank and also the accuracy of the digital

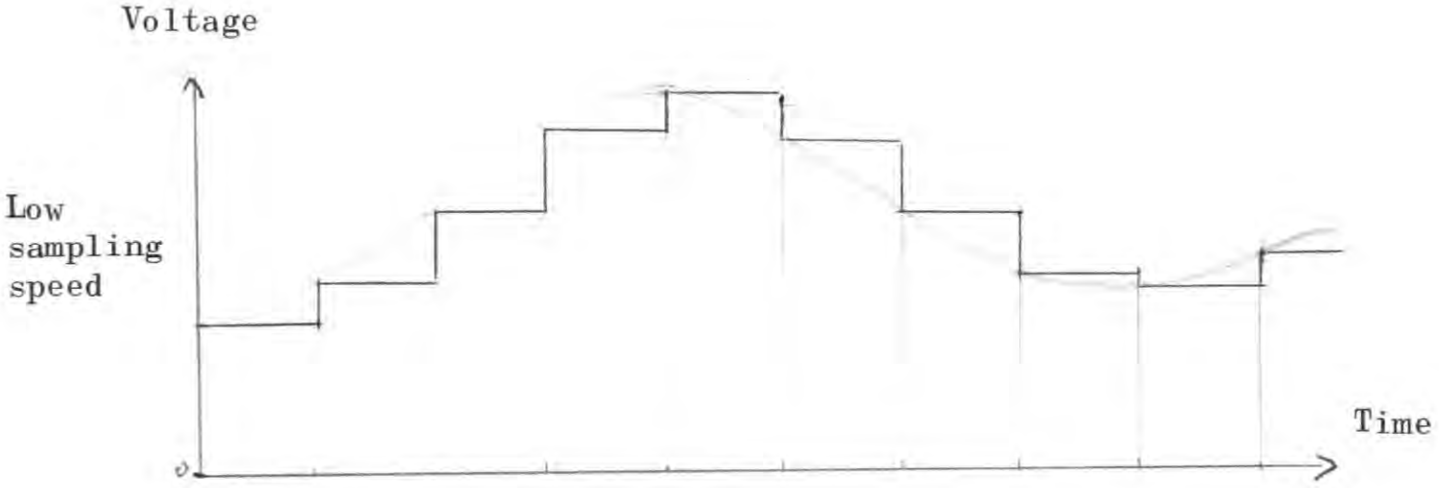
[63] In the discussion regarding variable speed tape techniques in the previous chapter, it was noted that this particular method of altering the rate of change of recorded events also affects the frequency of the resultant information. Halving the tape speed, for example, not only halves the speed of event changes but also drops the pitch of the contents by one octave.

representations of each sample. The significance of the former factor in determining the accuracy of a stepped approximation of a continuous voltage pattern has already been discussed earlier in connection with the conventional sequencers. It is equally important here that the sampling speed selected should take adequate account of the voltage changes occurring at the input. If the source consists of a series of slowly changing D.C. voltage steps, such as might be generated from a keyboard control output, the speed of sampling need only be rapid enough for the ear not to detect the resultant re-timing of any events which occur between pulses:



If, however, a continuously varying voltage is to be registered, the sampling speed must be considerably higher if major inaccuracies are not to occur.





Higher sampling speeds involve unfortunately a more rapid use of the bank's storage locations and the size of the latter then becomes an extremely important factor, limiting the length of a control function which may be adequately sampled.

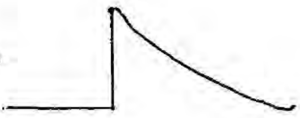
The accuracy of the numerical equivalents themselves, however, is a feature of equal importance. If a voltage level is sampled and converted into a digitally expressed numerical equivalent, a discrete approximation must be made to a continuously variable quantity. The accuracy of this representation will depend on the range of numbers available to quantify the voltage function between its extreme limits, and this in turn will depend on the digital word size employed. A six-bit word, for example, will provide $2^6 = 64$ different values, clearly suitable only for fairly crude approximations. An eight-bit word will provide 256 different values, a ten-bit word 1024 values, and a twelve-bit word 4096. Some small synthesisers provide simple digital sequencers of only eight-bit resolution over a single memory length of 128 words or less. Such a lack of precision severely inhibits the usefulness of these memory facilities, particularly for registering frequency control functions. The limiting factor governing word size is not only the cost of providing extra bits but more significantly the accuracy of the analogue to digital and digital to analogue conversions themselves. Converters capable of handling eight-bit functions are relatively easy to design and hence cheap to build or purchase. Each extra bit of resolution, however, involves an escalating increase in component accuracy, and hence cost. As a result it is extremely hard to produce reliable converters capable of handling more than twelve bits or thirteen bits of information.

Comprehensively equipped digital sequencers will usually provide three, four or even more parallel storage/retrieval banks of at least 256 words and ten-bit resolution equipped with individual sets of analogue to digital and digital to analogue converters. Controls to vary the analogue input and output sensitivities are often included, permitting the composer to modify the range of characteristics produced by recorded voltage patterns.

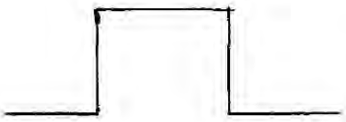
The procedures employed for composition with electronic sounds naturally vary from composer to composer. These creative processes are nevertheless organic by their very nature, entailing at least some degree of interplay between composer and machine. The precise nature of sounds and their methods of organisation are rarely completely evident before work commences in a studio. Patterns of sound events will frequently be built up in stages, components being evaluated both in isolation for their individual characteristics and also in combination with other sounds. Facilities for manipulating many individual strands of material and studying such complex acoustical problems as the spatial distribution of sounds must be readily available in any comprehensively equipped studio. Larger synthesisers usually provide basic output mixing facilities which may be operated under voltage-control. It is rare, however, to encounter an effective substitute for a professional multi-channel studio mixer, since the latter item may well cost as much as the synthesiser itself. Lack of adequate taping facilities imposes an equally frustrating limitation on creative working. Not only should provisions be made for multi-track tape facilities but also a bank of conventional two-channel machines should be available if restrictions on the methods available for the organisation and juxtaposition of material are to be kept to a minimum.

The use of voltage control for panning facilities creates the possibility of location modulation, where the spatial location of sounds becomes a variable quantity which may be manipulated in a manner similar to that employed for amplitude or frequency modulation. The provision of voltage control facilities for specifying degrees of reverberation mix, however, have proved less useful, for these have offered no new solutions to the problems encountered in creating a sensation of depth of field for sound images. Synthesisers usually provide reverberation springs for this purpose or supply external control lines for plate units. The mechanical imperfections of such devices, however, result in the production of readily identifiable artificial characteristics rather than an acceptably natural sounding degree of sound reinforcement.

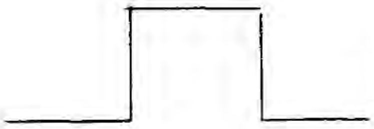
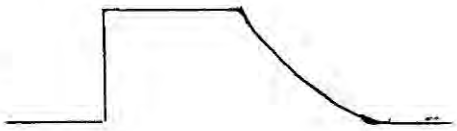
Mixing facilities for voltage-controlled synthesisers should not necessarily be restricted to audio signals. The control voltage network functions as the primary regulation system for the shaping of device characteristics, and provision should be made not only for flexible interconnections but also for the superimposition and even modification of control sources. Such facilities are particularly important for overcoming the limitations of simple repetitive control wave shapes, which all too easily result in monotonous cliches of limited musical value. The most useful types of control voltage transformations are associated with the use of special filters designed to process low frequency signals. A low-pass filter, for example, may be used to smooth the steps in a staircase curve, making it appear more continuous. Resistance/capacitance networks may be used to produce split function low-pass filters, one function smoothing the rise time of an increase in voltage, the other instantly discharging such a change in potential to produce a fall time characteristic.

 may thus become either  or 

Combining the two functions will provide a more normal low-pass smoothing characteristic:

 then becomes altered to 

High-pass filters may be used to provide modifications similar to the second type of split function indicated above, smoothing negative changes in voltages rather than discharging voltage peaks.

 becomes altered to 

It is possible to employ further types of transformation such as subtraction, multiplication and division. There are considerable doubts which must be raised about the value of these last mentioned techniques as practical features, however, for they introduce a level of complexity which demands a high degree of logical organisation. Voltage-controlled systems, more especially those which have been custom designed by individual studios, have nevertheless expanded their ranges of control facilities in other directions in an attempt to create more flexible operating facilities for the composer. The characteristics of the envelope follower have already been described earlier, and this device is of great value in converting the overall amplitude characteristics of any audio signal into equivalent control voltage functions.

A frequency-to-voltage converter provides a useful complement to this facility, for such a device may be used to provide control voltages which are proportional to an applied frequency. A common design for such a converter consists of a voltage-controlled low-pass or band-pass filter which is rapidly swept upwards from a frequency setting below the source signal range by a sweep oscillator. As in the case of the envelope follower, the amplitude of the filtered signal is monitored by a threshold detector. When the gain rises above a carefully pre-set threshold level, indicating that the fundamental frequency of the input has been reached, the frequency setting of the filter at this point is converted into a voltage equivalent. The filter returns to its minimum frequency setting for a new sweep and the process re-cycles, the registered voltage level being held constant until replaced by a new setting from the threshold detector. If the sweep speed is sufficiently fast the output appears to be a continuous voltage function. Such a technique will only work satisfactorily on source signals which offer clear fundamental components, ideally single sine waves of sufficient amplitude. Frequency complexes which change in content are liable to produce spurious, although occasionally interesting, voltage functions. A far more satisfactory design involves a technique of pulse counting. The incoming signal is converted into a series of square pulses of the same frequency using a Schmitt trigger. These pulses are passed to a resistance/capacitance network which converts the square pulses into spiked pulses and then on to a blocking circuit which removes all negative components. The result is a series of positive spikes produced whenever the incoming signal passes its zero cross-over point in a positive direction. These are used to trigger single shot, square pulses which are of a constant width with respect to time. The resultant pulse pattern is then used to charge a capacitor which develops a voltage proportional to the frequency of these pulses which in turn matches the frequency of the input signal.

These converters facilitate the application of analogue tape recorders for transforming, recording, editing and retrieving control voltage information. The development of such techniques has remained largely in the hands of specialist studios, and the potential of such systems remains largely unknown to composers who have to rely exclusively on commercial synthesisers. In certain respects analogue voltage registration on tape in the form of frequency patterns offers distinct advantages over digital storage and retrieval facilities. For example, there are no sampling or quantizing problems involved, and the duration of control voltage sequences is restricted only by the length of tape employed and the speed at which it is run. Even if the tape has to be run at 30 inches per second, a 2400' reel will still give fifteen minutes' registration time, far in excess of that usefully obtainable from any normal digital sequencer. Problems arise, however, over the choice of operating characteristics for the conversion systems, for these need to be carefully regulated. The simplest design frequency-to-voltage converter offers a linear transfer characteristic, the voltage output being directly proportional to the frequency input, and this must be matched by a similar response from the voltage-to-frequency encoding system. A special voltage-controlled sine wave oscillator is normally employed for the latter function, offering a linear frequency control input. The 'carrier' frequency of the latter is set as high as practicable, usually at a point about 20 to 30 per cent below the top frequency limit of the recording system. This factor varies according to the speed at which the tape is run. Typical settings are 10 Kilohertz for a tape running at 7½ inches per second, 20 Kilohertz for a tape at 15 inches per second and 40 Kilohertz for a tape at 30 inches per second.[64] If bipolar control voltages are then applied to the oscillator input they will cause the carrier to be frequency modulated. The maximum degree of frequency deviation permitted by extreme control voltage variations is accordingly restricted to about ± 20% to keep the highest peaks within the system's band width. The maximum frequency of the control waves themselves must be less than one fifth of the frequency of the carrier, otherwise phase distortion is introduced. The highest control wave frequency which may be used with a 10 Kilohertz carrier, for example, is 2000 Hertz. At the other end of the scale, however, no such restrictions exist and it is thus possible to register D.C. control voltage levels.

Care must be taken to ensure that the recording system itself does not produce distortions. In particular wow and flutter may be introduced from fluctuations in the drive system. The response sensitivity of the frequency-to-voltage converter must also be accurately matched to the response of the input modulator if the output control voltage characteristics are to be identical with those applied at the input to

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64 The tape recording/replay bias characteristics for these higher speeds usually have to be modified to achieve these optimum settings.

the system. Any deviations will result in non-linearity, particularly at extreme voltage levels. The speed of playback may be varied to alter the rate of change of voltage levels, providing the gain of the frequency-to-voltage converter is suitably re-set and a D.C. bias applied to the output to compensate for the change in carrier frequency.

In assessing the musical value of the developments which have taken place in electronic music studio designs, one factor of overriding importance becomes central to the discussion. Increasing sophistication in device control processes contributes little to the advancement of the medium if such technological refinements are not matched by improvements in the facilities for man/machine communication. This has been touched upon several times during this account, and it is relevant in the light of the foregoing descriptions to study the matter further. The following commentary on voltage-controlled synthesis by Allen Strange merits attention.[65]

> In many instances of recorded electronic music, the composer fills a dual role of creator and performer. In the older studios of Europe, it was a common practice for the composer to realise his work through the aid of an engineer; he would seldom come in direct contact with the equipment. But due to simplified methods and design, along with the contemporary composer's increasing knowledge of electronic methods and an instinctive curiosity about the internal workings of his art, the electronic composer today has a tighter reign over the compositional processes. One of the major appeals of electronic music is that it offers the composer an opportunity to come into direct physical contact with the various parameters in which he is interested. This is appreciated by some composers because of the unrestricted control it affords; it is appreciated by others because of the actual kinesthetic sensations involved...

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[65] Allen Strange, <u>Electronic Music</u>, <u>op.cit.</u>, p.21

> The electronic composer may control the parameters of the sound he produces through the use of control voltages. An entire composition may be the function of voltage application, consequently the composer is often more concerned with methods of voltage production than the manner of application.

This provides an illustration of the differences which may arise from a technical, rather than a musical appreciation of the desired characteristics and purposes of sound synthesis systems. It is true that the equipment to be found in the studios of the early 1950s was both unwieldy and also limited in its range of facilities. These disadvantages, however, were heavily outweighed by the very environment created by teams of composers and engineers working in close cooperation. In such situations the studio engineer or technical assistant provided an intelligent, interactive, human interface between the system technology and the artistic aspirations of the composer. The latter, far from being removed from the system processes, was able to develop clear rationalisations of the technical problems created by his or her particular artistic intentions through the channels of team discussion and experimental evaluation. Within its absolute functional limitations, the equipment characteristics could thus be exploited to the fullest extent leading, where appropriate, to the development of additional technical facilities to satisfy particular musical objectives.

Voltage control technology has indeed opened up a whole new range of sound generation and processing techniques, but it must be realised these have not in themselves provided any major solutions to the fundamental problems of specification encountered by the studio composer trying to translate creative ideas into practical equivalents. To state, as quoted above, that the electronic composer has a 'tighter reign over the compositional processes' and that he is 'often more concerned with methods of voltage production than the manner of application' is in the former instance misleading, and in the latter most disturbing, for there is an underlying suggestion that the system and not the composer should be allowed to dictate the manner in which the medium is employed. It is certainly unavoidable that the former will limit the range and type of operations which may be executed in a particular studio, but this should never be treated as an unreserved justification for the technology itself.

Such an attitude leads to a form of inbreeding in designs, which in the long term is counter-productive to the creative advancement of the medium as a whole. The very commercialism of the voltage-controlled synthesiser, with the mass-production of units offering a compromised range of facilities has cultivated just such an introspective attitude towards the use of electronics in musical composition. This environment has encouraged the tacit acceptance of many operating procedures on the grounds of electronic convenience rather than musical relevance.

The development of visual input systems offers, potentially, one of the most valuable methods of achieving improvements in the processes of composer/machine communications. The possibilities of 'drawn sound' techniques have been investigated for many years by Daphne Oram, who established her private studio at Fairseat, Wrotham, Kent in 1959. With the aid of a grant from the Calouste Gulbenkian Foundation she has established a unique sound synthesis system, Oramics, which merits study.[66]

One of the most powerful advantages of a graphical input system is the ability to dispense with intermediate specification languages. In Oramics, the composer is able to specify directly, in terms of drawn shapes and patterns, the three primary parameters of pitch, amplitude and time, and also, to a certain degree, timbre, associated with a monophonic progression of sound events. Polyphonic textures may be obtained at present only by mixing separate recordings. Miss Oram, however, hopes to build a smaller, more compact, machine in the near future, offering up to four separate sound synthesis channels.

The control system consists of ten sprocketed, 35 mm. clear plastic film strips mounted in parallel and transported synchronously by a common drive mechanism from right to left over a bank of illuminated photocells. Opaque shadow masks or markers are entered onto these strips by the composer to produce modulations in the light intensity reaching the photocells. The latter translate these fluctuations into voltages which are then employed to control system functions.

The time parameter thus becomes a linear function of distance measured along the strips in terms of their speed of transport, normally set at 10 centimetres per second. These strips are divided into two banks of five, the upper bank being employed for digital control functions, and the lower for analogue ones. This distinction is reflected in the construction of the photocell system. In the case of the lower bank, one photocell is provided per strip, measuring the fluctuations in light

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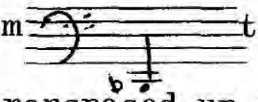
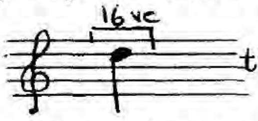
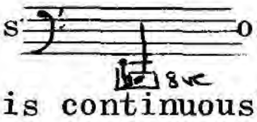
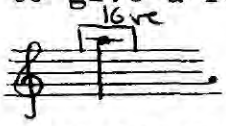
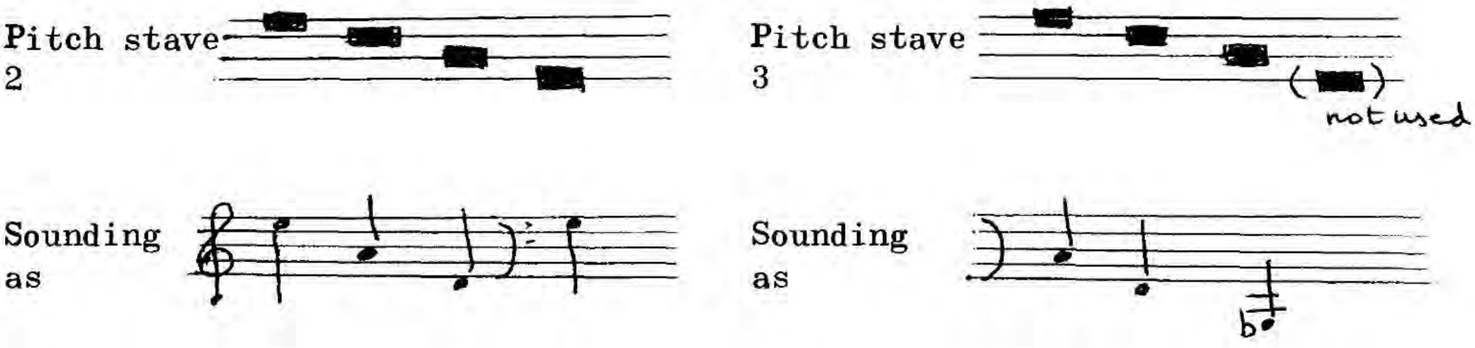
[66] I am indebted to Miss Daphne Oram both for granting access to her system and also for generously providing information concerning its design philosophy and constructional detail.

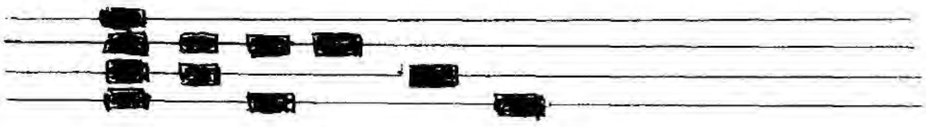
intensity caused by the blanking contour of the shadow masks. For the upper bank, four parallel photocells are provided for each strip, their positions being indicated by a faint four-line stave engraved in the film. These detectors function as light sensitive switches, activated by patterns of black rectangular 'neumes', entered by the composer on the stave lines. The main purpose of this bank is to provide pitch control information, using a special coding system entered across three of the film strips. A fourth strip may be used to switch ancillary studio equipment, and the fifth, at present, is left unused. None of the strip designations within each bank is rigidly assigned to a particular function, and these may be varied to suit individual requirements.

The pitch code in its standard setting may be used to generate a tempered chromatic scale from  to 

the whole scale, however, may be externally transposed up or down via a master regulator to give a range starting as low as 

or finishing as high as 

. Since this function is continuously variable it provides a useful fine tuning facility. Each line on the three staves is associated with a particular frequency setting. The lower two staves are normally tuned to give fifths as follows:

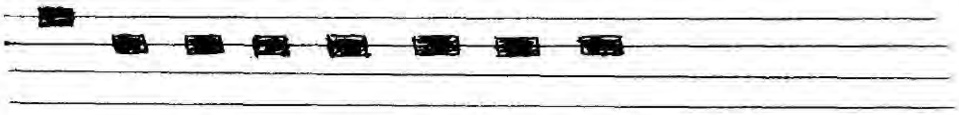


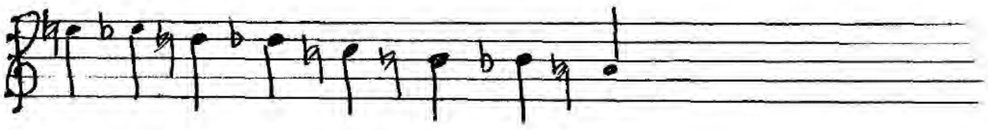
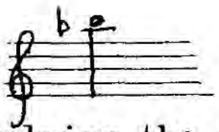
The lines on the uppermost pitch stave are used in combination with these primary tunings to provide chromatic intervals by means of a frequency addition process. For example:

Pitch Stave 1



Pitch Stave 2



Sounding as

Pitches above  are obtained by more complicated neume combinations involving the extension of frequency adding principles to pitch staves 2 and 3 also.

The process of pitch generation involves the synthesis of waveforms, drawn as shadow masks on glass plates and scanned optically via photo-multipliers. The speed of the latter is controlled by a common time base circuit responding to the commands produced by the pitch code system. Up to four different wave forms may be generated simultaneously, and independently regulated in amplitude via four of the five available analogue optical tracks. The mixed output will then consist of a monophonic pitch sequence varying both in amplitude and also timbre. This, however, leaves only one analogue optical track for any further signal processing and in some situations it will prove desirable to use perhaps only two timbre generators at any one time, the fourth digital track being employed to provide a means of switching between the four available sources whenever required. In the system diagram which follows two optical tracks are shown controlling the amplitude of each generator instead of one. This permits separate control to be exercised over the envelope functions and the overall dynamic level. As illustrated, the former parameter is varied separately for each of the two generators via a pair of optical tracks and the latter parameter is regulated as a common function via a third. In the case of the envelope shadow masks the circuitry provides a positive amplitude response to the contour outline, the highest output being associated with a complete blanking of the photocell. The overall dynamic control function, on the other hand, responds in reverse. No masking allows the envelope characteristic to peak at maximum output. Increasing degrees of masking lead to a corresponding reduction in this response range.

Pitch vibrato may be introduced using an analogue optical track switched to modulate the time base generator, the range of frequency deviation being determined by a sensitivity knob. The effects produced may be refreshingly different to those associated with regular, mechanical vibratos, such as may be generated on the R.C.A. synthesisers, or in a voltage-controlled system. Carefully drawn mask contours provide for delicate nuances by means of varying modulation speeds and depths. A reverberation chamber is the only other treatment device currently available. This consists of a large steel drum fitted with a loudspeaker at one end, and a microphone at the other. Although it is not possible to adjust the reverberation time of this unit, the degree of reverberation mix may be controlled via an analogue optical track.

The system diagram given below shows Oramics in a particular operational configuration. It must be remembered, however, that the operational characteristics may be varied by altering the designations of the optical control tracks:[67]

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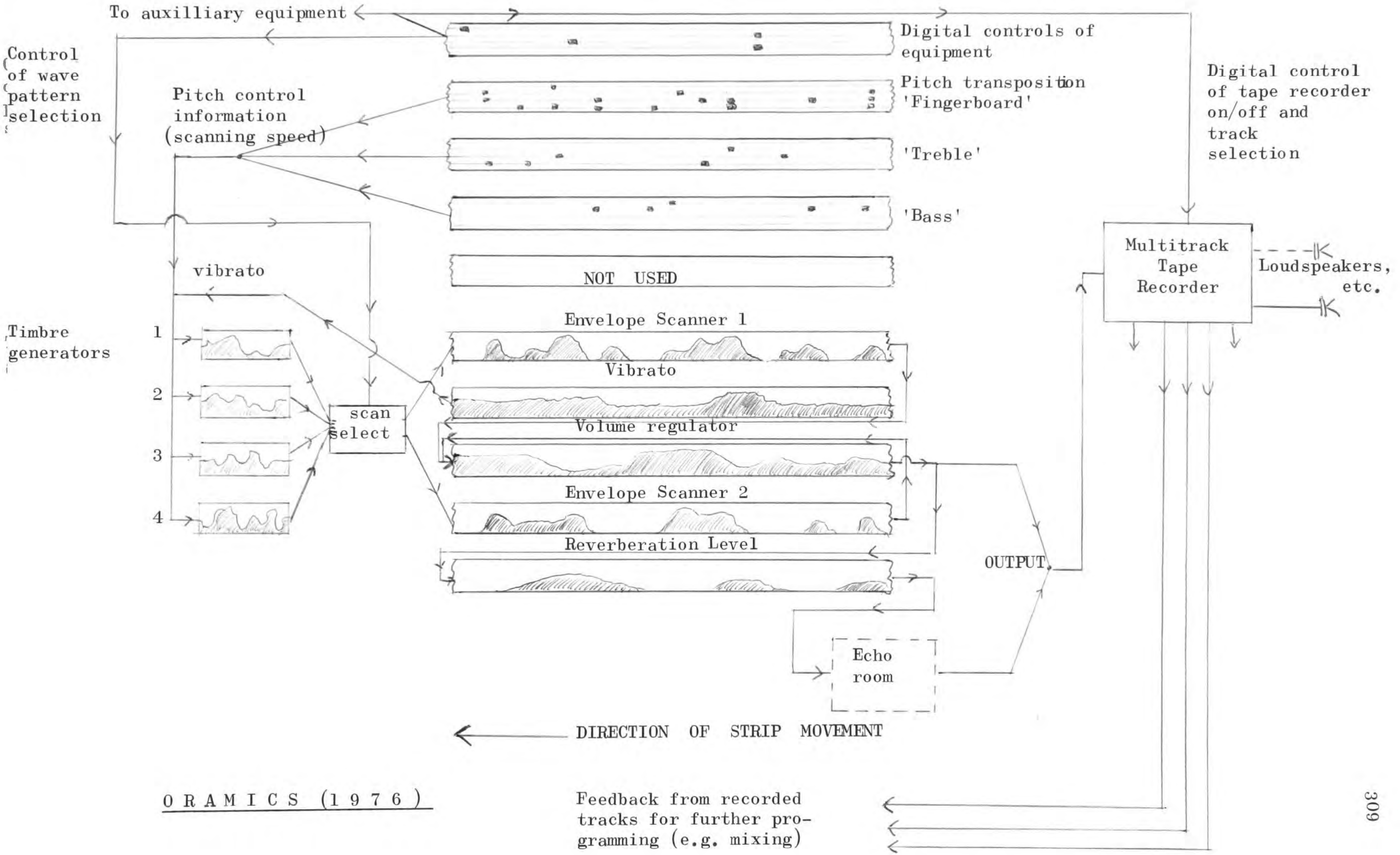
[67] Earlier versions of this diagram are to be found in the following two sources:
Alan Douglas, Electronic Music Production (Pitman Publishing, London, 1973), p.95
Daphne Oram, An Individual Note on Musical Sound and Electronics (Galliard, 1972), rear flyleaf



ORAMICS (1976)

Oramics provides a technique of sound generation which is both simple to operate and also readily intelligible to the composer. The subtleties offered by the specification system are of considerable musical importance, for they offer facilities for the creative expression of ideas which cannot be matched by any of the systems so far described. Perhaps the most significant disadvantage of the system, apart from its present restriction to a monophonic output, is the difficulty encountered in generating timbres with a rich harmonic content. The only filters in the system, unseen by the composer, act to smooth irregularities in the hand-drawn wave shapes. These inevitably restrict the range of harmonics which may be generated from a fundamental frequency, a drawback particularly noticeable in the lower pitch regions. The immediacy of contact between the composer and the sound generation facilities is, nevertheless, remarkable and demonstrates effectively the powerful potential of graphic input systems.

This study of the development of electronic music systems has highlighted one particular characteristic: an increasing preoccupation over the years with the development of ever more advanced specification facilities. The stage has now been reached where the application of computer technology to the medium must be considered in detail, both as a means of providing programmable control facilities for analogue studio devices, and also as a means of generating sound information directly.

A critical study of these major innovations, in particular their significance with regard to the cultivation of more meaningful man/macnine communication methods, thus forms the second part of this thesis.