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METHODS OF SOUND SIMULATION AND APPLICATIONS IN FLIGHT SIMULATORS

K.-P. Gaertner and K. Hillmann

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16. Abstract

An overview of methods for electronically synthesizing sounds is presented. A given amount of hardware and computer capacity places an upper limit on the degree and fidelity of realism of sound simulation which is attainable. Good sound realism for aircraft simulators can be especially expensive because of the complexity of flight sounds and their changing patterns through time. Nevertheless, the flight simulator developed at the Research Institute for Human Engineering (Forschungsinstitut fuer Anthropotechnik) shows that it is possible to design an inexpensive sound simulator with the required acoustic properties using analog computer elements. The characteristics of the sub-sound elements produced by this sound simulator for take-off, cruise and approach are discussed.

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Preface

This report is part of a series of publications, describing the work on the project Force Steering. The project is carried out by the Department of Control Elements of the Research Institute for Human Engineering in Meckenheim.

The aim of the project is to construct a research simulator for the two-jet business plane HFB 320 Hansa jet and to study the force steering concept. The project includes the study of the force steering concept during landing approach and in a later phase concerning other control tasks. The force steering device represents the separation between automatic and manual control of an aircraft so that an expression of the division of work between man and machine when landing an aircraft may be expected.

The present report gives a survey on the methods of sound simulation, with an introduction into the physical and physio-psychological aspects of sound followed by the discussion of actual aircraft noise simulation containing a sound analysis, sound synthesis and sound presentation. The sound simulator of the Research Institute for Human Engineering has a full electronic design according to the traditional procedure, employing exclusively analog components of computer technology. The synthesis of sound portions, jet engine, wind, rolling, landing gear and actual landing requires a subjective and an objective sound analysis gained by means of tape recordings from original sounds of HFB 320 for starting, flight and landing approach. The simulated cockpit sounds were described by HFB pilots as very much in agreement with the original sounds of the HFB 320.

Subsequently a brief survey on avionics sound simulation is made, limited in the case of the sound simulator of the Research Institute for Human Engineering to the synthesis of radio beam and approach markers.

The following reports on the project force steering have already been published or are in preparation:

1. "Problems and Methods in Evaluating Force Steering and Other Flight Control Modes: A Study and an Experimental Design", Pitrella, F.D.

The force steering concept, a relatively novel method of flight control, is described. The development of an experimental plane for experimental study of this method using the HFB 320 Hansa jet simulator of the Research Institute for Human Engineering is then discussed.

2. "Flight Mechanics and Programming of a Flight Simulator Using the Example of the HFB 320", Holzhausen, K.-P. and P. Kuehne.

The motion equation of an aircraft are derived and simplified. This simplification makes it possible to carry our programming of HFB 320 simulation on a small digital computer.

3. "Force Steering Simulation of Flight Simulators", Gaertner, K.-P. and W. Kruse (being prepared).

A survey on technical methods for generation of simulated manual forces of the control elements is prevented. The solution applied to the HFB 320 Hansa Jet Simulator of the Research Institute for Human Engineering is described with the application of torque motors.

Here we wish to thank our associates who have contributed to the present work and its publication as critical partners and subjective audience. The manuscript was read critically by Dr. G. Raw and K.-D. Schulz-Helbach. Mrs. H. Lanzerath drew the figures and Mrs. G. Wolf wrote the final text. We extend our thanks to all who helped.

Meckenheim, April 1975

K.-P. Gaertner
K. Hillmann

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List of Symbols

f	Frequency
h	Amplitude of a Partial Wave
H, ALT	Altitude of Aircraft
I	Sound Intensity
I_o	Sound Intensity of Threshold of Audibility
IAS	Aircraft Velocity
L	Sound Pressure Level
L_s	Volume Level of Standard Sound
p	Sound Pressure
p_o	Sound Pressure of Threshold of Audibility
p(x)	Probability Density
RPM	Torque
S	Loudness
S_o	Loudness of Standard Narrow Band Noise
T	Wave, Measuring Time
v	Partical Velocity
VS	Vertical Velocity of Aircraft
x_F	x Coordinate of Aircraft
y_F	y Coordinate of Aircraft
z	Subjective Tone Quality
δ, Δ	Difference
ω	Angular Frequency
σ	Phase angle, Roll Angle of Aircraft
ϕ	Standard Deviation

METHODS OF SOUND SIMULATION AND APPLICATION IN FLIGHT SIMULATORS

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Summary /1*

The report gives a survey on methods of sound simulation. "Sounds" are understood as acoustic sound vibrations, stimulating the human ear and causing a hearing response. The sound vibrations may reach the ear in the form of physical quantities such as tones, harmonious sounds, sounds or bangs. Noise may be described objectively by loud sound vibrations. It is, however, generally the expression for an acoustic event intolerable to man and is therefore evaluated subjectively.

A number of relationships may be established between physical stimulus and psycho-physiological response. Important quantities for the description of physical and psycho-physiological aspects of sound are sound pressure, sound pressure level, sound intensity, volume level, loudness, subjective tonal quality and masking.

The process of sound simulation includes analysis, synthesis and sound presentation. A mathematical model of sound sources is constructed according to subjective and objective analysis to be carried out. These determine sound spectrum for technical synthesis, presented via loudspeakers.

Mathematical model and technical sound synthesis is presently carried out by means of three different procedures. In order of increasing complexity these are the traditional, the poly-voice and the digital procedure. The mathematical modeling of sound sources shifts here from pure hardware into software. Estimation of work and costs in realizing this project depends on the aim of total simulation of aircraft and is explained in this report using the example of the sound simulator for the HFB 320 simulator constructed in the Research Institute for Human Engineering.

In addition to the sounds generated by the moving aircraft, sounds may have an affect on the pilot, for example, via morse and warren signals or radio. The possibilities of simulation of these surrounding sounds will be briefly discussed.

1. Introduction /4*

A simulator for an aircraft system is described by means of the following effect and information circulation: The pilot receives information on aircraft behavior in the form of sense stimuli, he compares these with the task indicators and derives from this comparison information on alteration in aircraft behavior which he carries out via control procedures. In the place of the actual aircraft with its information presentation and control devices in the case of a simulator, technical reproductions of these devices are used. The central problem in this case is the reproduction of devices for information representation [1]. The stimuli received by the pilot include the visual, kinesthetic, audio and tactical stimuli. This report is concerned with reproduction of the total sound of all contributing sound sources for auditive responsives, audible in an aircraft cabin and giving information on the operational condition of the aircraft.

Designs of sound simulators usually are specific to the aircraft to be simulated in each case. When simulating flight sounds, for example, reproduction of engine sounds, wind sounds caused by cockpit and landing gear, landing sounds, etc. is carried out. These sound portions have been controlled up to now in the traditional procedure individually by computer, i.e. by the mechanical flight model [2,3]. There are, however, already modern approaches aiming at synthesizing the desired sounds in the frequency level in the computer and transferring these in the time range via a fast inverse Fourier transformation (FFT, Fast Fourier Transformation) [4,5]. In this procedure, however, it is often overlooked that synthesis is only one aspect of sound simulation. For complete description in addition to synthesis the analysis and sound presentation are also necessary. According to subjective and objective analysis made on the mathematical model the technical synthesis of a sound spectrum is carried out out, which are then presented via amplifiers and loud speakers. As shown in Figure 1 a sound simulator consists of a technical device for synthesis and presentation of sound. The analysis carried out during the design phase determines the mathematical as well as technical extent of the actual sound simulator. The rapid technological development in the area of /5 computer technology and electronics is presently of great assistance to synthesis, while the statements presently valid on analysis and presentation will probably remain valid for future procedures of sound simulation.

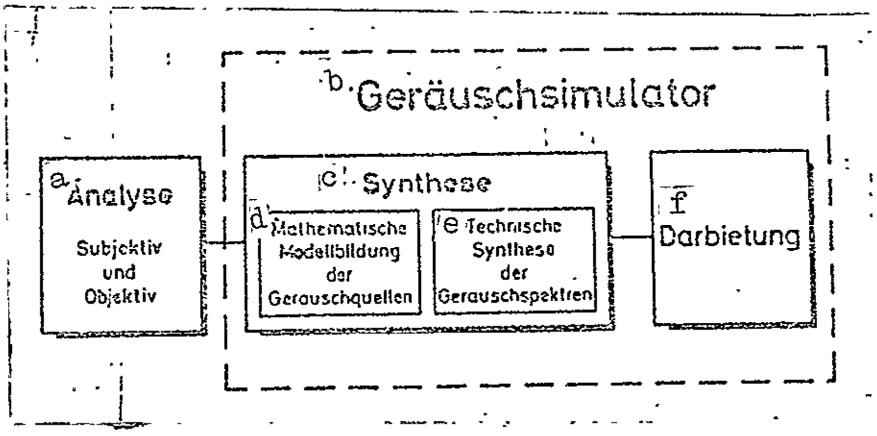

Figure 1: Process of Sound Simulation

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Key a. Analysis - Subjective and Objective
b. Sound Simulator
c. Synthesis
d. Mathematical Model of Sound Sources
e. Technical Synthesis of Sound Spectrum
f. Presentation

2. Basis of Sound Simulation

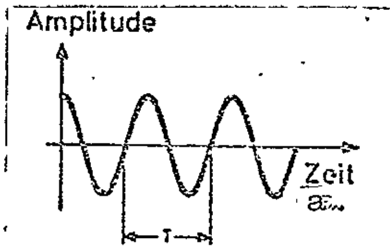
Sound vibrations may be described as tones, harmonious sounds, sounds, bangs or noise. The word sound is used in sound simulation as a collective concept and includes periodic and non-periodic vibrations. The vibrations of bangs and noise are, as already indicated in this chapter, not subject of a sound simulation. A series of relationships and regulations apply between physical stimulation and physio-psychological response, explained in the following.

2.1 Sound Vibrations

Sound is understood as an elastic vibration of gaseous, liquid or solid material. Molecules or grid structures are caused to leave a resting condition by external impulse, whereby vibrations are generated. These include adjacent particles, so that a sound wave is created, running through the elastic medium. Sound waves in the frequency range of 16 Hz to 20 kHz [6,7] reach the ear in the form of acoustic stimulus releasing the response tone, harmonious sound or sound in the "communication receiver ear".

2.1.1 Tones /6*

A tone is created physically by a sine wave. Its psycho-physiological descriptive quantities are tone amplitude and tone intensity. Air density or variations in air density find application as a physical measure of tone intensity.



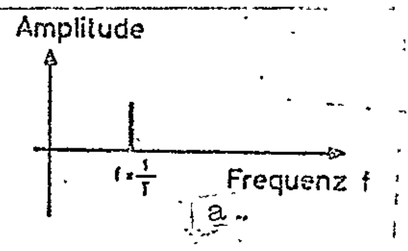


Figure 2a: Time Course of a Sine Wave with Period T

Figure 2b: Amplitude Spectrum of a Sine Wave with Period T

Key a. Time

Key a. Frequency

Figures 2a and 2b show the two usual representations of a sine wave.

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2.1.2 Harmonious Sounds

Harmonious sound is described as a hearing response, generated by a composite of periodic oscillations in the range of audible frequencies. Periodic oscillations are understood as rectangular, triangular, saw-tooth waves, etc. radiated by loud speakers, but also, for example, bell tones and tones of string instruments. Generally a harmonious sound stimulus is understood as a combination of base tone and over tones. The harmonious sound stimulus may therefore be taken apart into sine waves, corresponding to partial tones of the harmonious sound, whereby that partial tone of lowest frequency is termed base tone (harmonious sound analysis).

On the other hand, harmonious tones may also be put together artifically from sine waves (harmonious sound synthesis). Figure 3 shows a rectangular wave, composed of base tone and over tones [6]. The entire course of the wave from three partial tones already indicates /7* the form of rectangular wave desired.

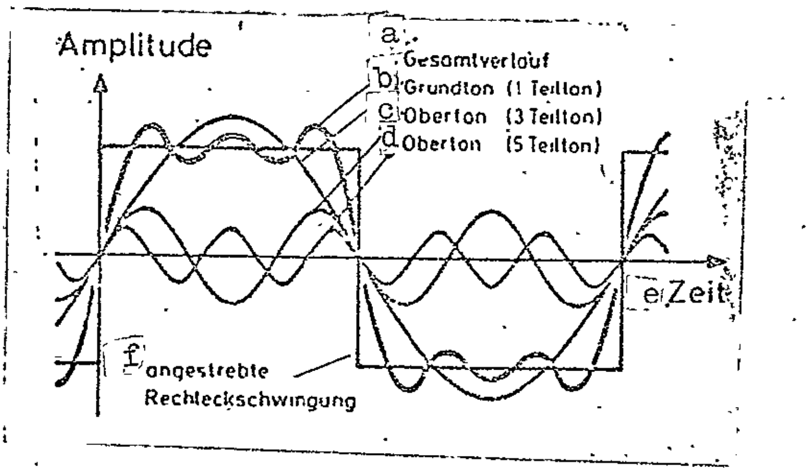

Figure 3: Synthesis of Rectangular Wave from Bases and Over Tones

Key a. Total course
- b. Base tone (first partial tone)
- c. Over tone (third partial tone)
- d. Over tone (fifth partial tone)
- e. Time
- f. Rectangular Wave Desired

In addition to the descriptive quantities tone amplitude and tone intensity already known, the tone color may be named as a third quantity of harmonious sound. Tone color is determined physically by the number and amplitude of the over tone over the base tone, not however by phase difference between the partial tones [6].

Harmonious sounds of varying tone color therefore have varying waves, but varying waves may not necessarily have harmonious sounds of different tone coloring. An infinite number of wave curves correspond to a certain tone color, differing from one another only by means of phase relationships between the partial tones. This fact is known to facilitate, for example, the synthesis of harmonious sounds for the electronic organ, but also every sound simulation, since the phase relationships between partial tones may be completely disregarded in the concept of sound sources. As an example, Figure 4a shows the partial tone and the resulting total course of a harmonious sound with the base tone of period T_1 and the over tone of the period T_3 without a relative phase difference.

/8*

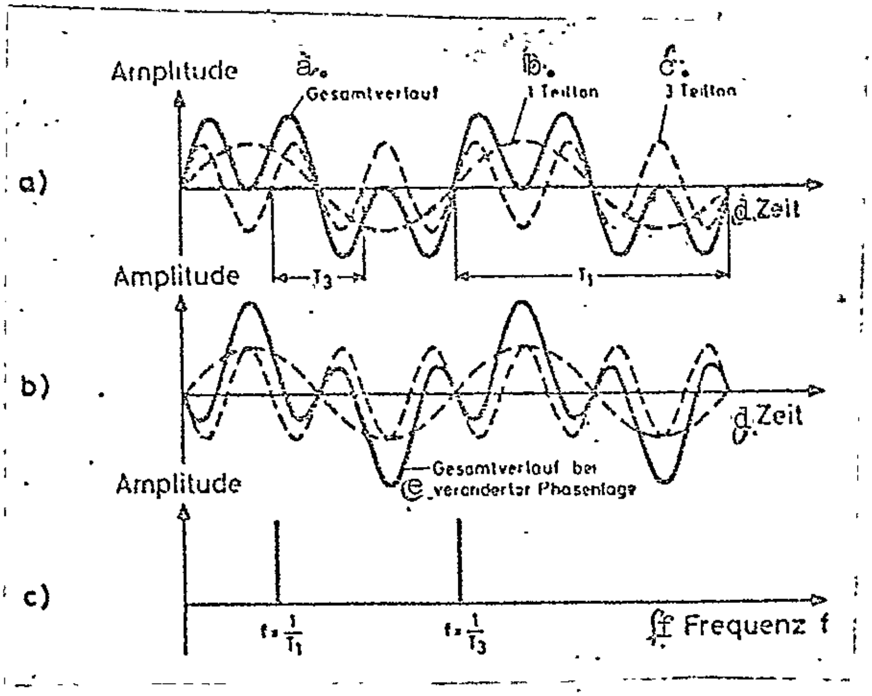

Figure 4: For the Dependence of Tone Coloring on Phase Length [6]

a. Time course of two waves of Periods T_1 and T_3 (= $\frac{1}{3}$ T_1) and the resulting total course
b. Time course of the same waves by altered phase length and the altered total course
c. Representation of points a and b, showing the same amplitude spectrum

Key a. Entire course
b. First partial tone
c. Third partial tone
d. Time
e. Entire course with altered phase length
f. Frequency

Figure 4b shows an altered total course where phase length is altered. Although the two total courses demonstrate varying wave forms, the same sound impression reaches the ear in both cases. The amplitude spectrum (figure 4c) is identical for both wave forms, since in the case of this type of representation the phase relationships are not taken into consideration.

Harmonious sounds and tones become audible when the vibration generator, for example, strings of stringed instruments, tones of brass instruments, membranes of loud speakers or even vibrating motor and machine parts transmit vibration to resounding constructions, also a vibrating and bringing a large amount of air particles to vibrate. The resounding construction must not always receive an impulse from the base frequency, but may often prefer over tones. These resounding locations are essential for creation of sounds and are called formance. The formance of musical instruments is known, those of machines (E.G. aircraft engines) must be determined by means of analyzers. These /9* sounds may be easily reproduced synthetically with the aid of a formant filter .

In addition to the formance, the fade-out and build-up processes determine tone coloring [7,34]. If, for example, a solo instrument recorded on a tape recorder is scanned by the tape recorder in such a way that the beginning tone is missing, i.e. the build-up process is cut off, the instrument may be recognized only with great difficulty or not at all. In the reproduction of harmonious sounds the build-up and fade-out processes must therefore be taken into consideration which are, however, easily controlled in the frame work of electronic production of harmonious sounds.

The usual representation of harmonious sounds in amplitude spectrum, as shown in figure 4c, requires of the sound that it is of infinite duration and periodicity. For harmonious sounds one characteristic is the build up and fade out process and on the other hand a periodicity which is not strictly maintained (extension of spectral lines in figure 4c). In sound simulation tones, harmonious sounds and sounds are in the final analysis evaluated by the ear and after several fractions of seconds responded to by the ear as stationary. The necessary technical approximation for generation of tones, harmonious sounds and sounds may then only be employed as permissible approximation if the duration of the presentation for all periodic waves continues for approx. a second [9].

2.1.3 Sounds

Tones and harmonious sounds may be represented in the amplitude spectrum as individual spectral lines. They may be conceived of as a finite number of sine waves. Sound spectrum which may not be represented by a finite number of waves are called sounds and are represented by a continuing spectrum. Amplitude, frequency and phase of the infinite number of partial tones change continuously and there is no regular connection between these quantities.

In the case of sound simulation it is expedient to begin with a spectrum in which no frequency section differs from another and which exhibits the same characteristics in each time section. Such a spectrum /10* is called "white noise". In the case of such noise the sound intensity density is independent of frequency, i.e., in any equal-sized frequency sections of the spectrum equal sound intensities occur.

Through filtering a white noise may be transformed into a noise with various levels of sound intensity density as a function of frequency. Where the spectrum covers a wide frequency band, this is termed a wide band noise. If a narrow frequency band is filtered out of the white noise spectrum by means of a band pass filter, narrow band noise is the result.

Typical wide band sounds are the sound of the ocean waves and of rain, the hissing of jet engines or wind noises of rapid vehicles. Narrow band sounds occur when acoustic resonators filter frequency ranged from wide band sounds.

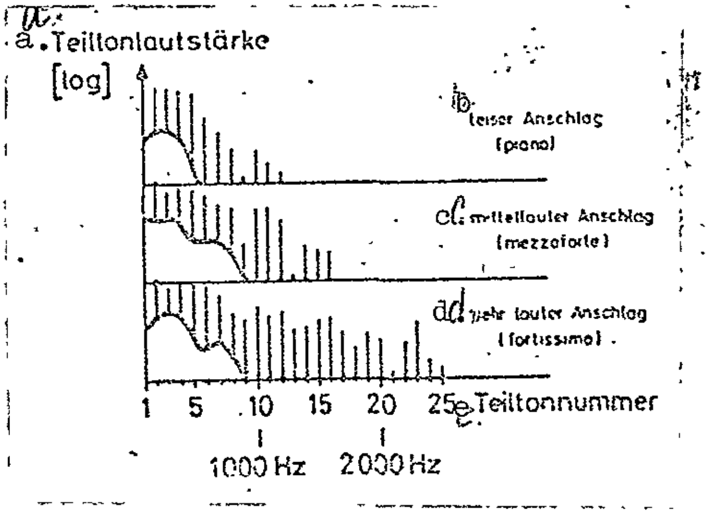

Figure 5: Sound Spectrum of a Piano Tone [7]

Key a. Partial tone volume
- b. Low tone
- c. Medium loud tone
- d. Very loud tone
- e. Partial tone number

In addition to the periodic sound the narrow band sound is typical to musical instruments and gives them the characteristic qualities. Figure 5 shows the sound spectrum of a piano tone. Over a continuous narrow band noise there is positioned a line spectrum. The string of the instrument engaged does not only generate the periodic sound but also excites the resonance body which prefers one frequency range (narrow band noise) in the range of the tone played. Such sounds also occur in speaking and singing, also in the case of a motor or a cockpit, noises in vehicles.

2.1.4 Bang /11*

A bang [7] is a hearing response of short duration. The duration amounts to only several fractions of a second. In this time the amplitude is reduced from extremely high values to zero. The spectrum is continuous as in the case of sounds. The response to a bang depends on the adaptation condition of the ear to the desired surrounding noise. All relatively loud and short sound events are registered as bangs and are relatively simple to simulate. The simulation of the bang noises having a very high amplitude, i.e. with sound pressure levels near the pain limit, is difficult to conduct. It is better to employ a "genuine noise simulator".

2.1.5 Noise

Noise is understood as every type of disturbing tone, harmonious sound and sound [35, 36]. Evaluation is subjective and may include all frequency ranges and volumes. In this report noise is not a primary subject of sound simulation. Sounds generated by a sound simulator, however, may be received by the evaluating persons as disturbing noise.

2.2 Physical and Psycho-physiological Aspects of Sound

Sound is understood as a wave-shaped motion in an elastic medium. In this case the motion is limited to air. Independent of size and condition of walls of the room surrounding the sound source, as well as the distance of hearer from sound source three main types of sound fields may be defined: the even sound field, the spherical sound and the diffuse sound field [8]. In a homogeneous medium this sound spreads from a point-shaped source in the form of a sphere. Adjacent to the sound source this sound field is spherical, relatively further away the field may be considered plane. The sound pressure is not on a spherical surface here but rather constant on a plane vertical to the direction of motion. When sound is studied in closed rooms with reflection and absorption surfaces, a diffuse sound field occurs. Because of multiple reflection and absorption the sound no longer reaches the observer from a single direction, but also coincidentally from all directions with differing intensities. /12*

When simulating sounds in vehicle simulation the sound field is usually diffused, in which sound events may be faded in for shorter or longer periods of time from a direction in relation to the observer, as, for example, the ignition of an engine, coming from the left right or from behind, may be audible.

2.2.1 Sound Pressure and Sound Pressure Level

Sound is created by rapid variations in air pressure at a localized point. Therefore sound pressure has the same unit of measure as air pressure. At a frequency of 1 kHz sound with an effective pressure of 2×10^{-4} µbar is just audible. The dynamic range of the ear reaches over 6 decades up to 200 µbar. The pain threshold lies between 200 and 2000 µbar as shown in table 1.

	a. Schalldruck p [µbar]	[N/m²]	b. Intensität I [W/m²]	c. Teilchengeschwindigkeit v [m/s]	d. Schalldruckpegel L [dB]
e. Schmerzschwelle	2000	200	100	5×10^{-1}	140
	200	20	1	5×10^{-2}	120
	20	2	10^{-2}	5×10^{-3}	100
f. Hörbereich	2	2×10^{-1}	10^{-4}	5×10^{-4}	80
	2×10^{-1}	2×10^{-2}	10^{-6}	5×10^{-5}	60
	2×10^{-2}	2×10^{-3}	10^{-8}	5×10^{-6}	40
g.	2×10^{-3}	2×10^{-4}	10^{-10}	5×10^{-7}	20
Bezugspunkt (Hörschwelle)	2×10^{-4}	2×10^{-5}	10^{-12}	5×10^{-8}	0

Table 1: Sound Quantities of Air at Normal Conditions [8]

Key a. Sound pressure p
b. Intensity
c. Particle velocity v [m/sec]
d. Sound pressure level L
e. Pain threshold
f. Audible range
g. Reference point (threshold of audibility)

For representation of the large dynamic range a logarithmic scale is employed for sound pressure. The threshold of audibility is used as reference point for this scale with $p_o = 2 \times 10^{-4}$ bar. Since logarithm only result from numbers and not from quantities, every sound pressure P is converted via division by reference sound pressure p_o into the pure number $\frac{p}{p_o}$. The decadent logarithm of this number is termed sound level L [6]. /13*

$$\boxed{L = 2 \log_{10} \frac{p}{p_o} \text{ [Bel]}}$$

The unit Bel is added to the number value $\frac{p}{p_o}$ in order to express that the decadent logarithm of a ratio of sound pressures has been formed. Since it is advantageous to use whole numbers where possible in acoustics, the tenth part of this unit, the decible (dB), is employed, resulting in a value for the sound pressure level

$$\boxed{L = 20 \log_{10} \frac{p}{p_o} \; [\mathrm{dB}] \qquad (1)}$$

When sound pressure increases from p to n x p, the sound level increases to

$$\boxed{L = 20 \log_{10} n \cdot \frac{p}{p_o} = 20 \log_{10} \frac{p}{p_o} + 20 \log_{10} n \; [\mathrm{dB}]^{1)} \qquad (2)}$$

2.2.2 Sound Intensity

The sound energy radiated from a sound source decreases with increasing distance from source in such a manner that the sound energy is distributed on an ever increasing cylinder surface, but maintained the same value. The intensity I, i.e. the energy contained in a surface unit vertical to direction of motion is

$$I = p \times v \qquad (3)$$

where v indicates the particle velocity (see also Table 1).

This is an expression of the relationship between intensity and /14* sound pressure, i.e. for a given particle velocity there is a reference intensity corresponding to reference sound pressure. Therefore, equation (1) for sound pressure level may also be expressed in intensity [9].

$$L = 10 \log \frac{I}{Io} \; [\mathrm{dB}] \qquad (4)$$

2.2.3 Volume Level

The quantities sound pressure, sound intensity and sound pressure level are physical quantities. The sound which is described objectively by these quantities is responded to subjectively by these quantities is responded to subjectively by the ear at a certain volume. In acoustics the standard value determined for volume response is a 1 kHz tone or a narrow band noise with an average frequency of 1 kHz [9]. This standard sound is coordinated with the volume level L_S in phons,

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1) If sound pressure is doubled, then n = 2, i.e. $20 \log_{10} \times 2 = 20 \times 0.3$. The sound level then increases with a relative doubling of sound pressure by 6 dBI. It can be seen from Table 1 that when sound pressure increases tenfold the sound level increases by 20 dB.

equal to the sound level in dB [9]. The volume level of other tones, harmonious sounds and sounds is measured by having a representative number of hearers compare these with the sound pressure level of the standard sound. As is shown in figure 6 a sound evaluated subjectively as having the same volume as an 80 dB standard sound has a volume level of 80 phons.

Figure 6 shows the range of audible sine tones for a plane sound field, characterized by the threshold of audibility for lower volume levels and by the pain threshold for higher levels. The surface resulting from these limits in the range of 20 Hz to 20 KHz is termed the audible area. Within this area the range of music, designated by vertical hatching, and the range of speech, indicated by horizontal hatching, are found. The curves show, for example, that a sound pressure level of 50 dB is coordinated with a 100 Hz tone of 40 phons, while a 1kHz tone of the same volume corresponds to a sound pressure level of 40 dB.

Curves of the same volume levels may not be drawn for a diffuse sound field, because no diffuse sound field may be constructed from a sinus-shaped standard tone due to the varying reflection and absorption of tones by the objects in the room and due to creation of stationery waves. Loudness comparisons in a diffuse sound field, however, may be /15*
carried out by applying a narrow band noise or by frequency modulated tones.

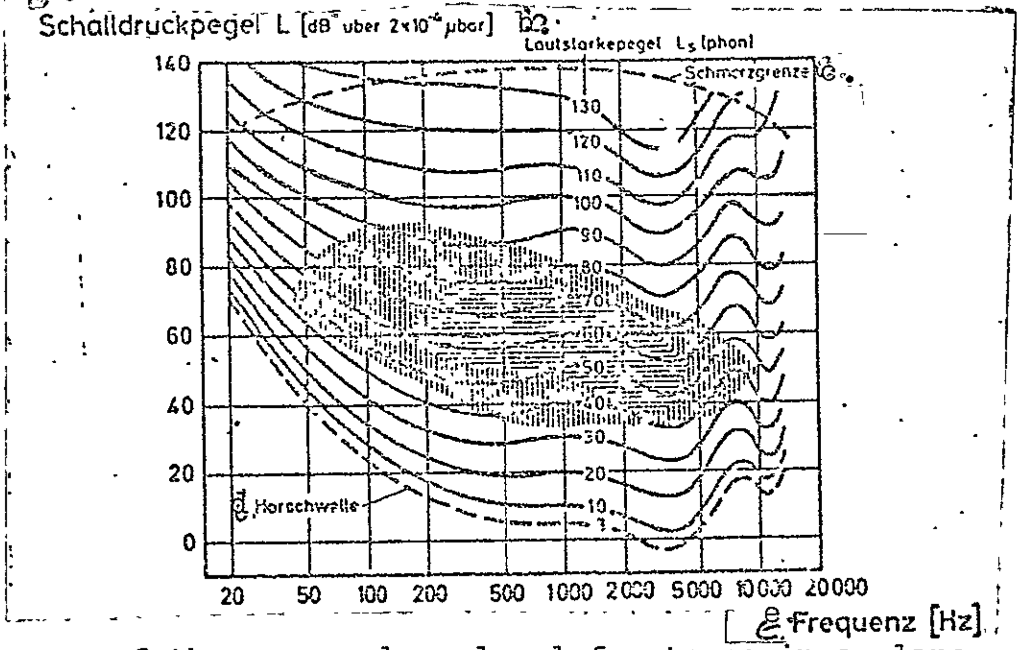

Figure 6: Curves of the same volume level for tones in a plane sound field
Vertical hatching: audible areas in the range of music
Horizontially hatched: Audible areas in the range of speech
(compiled from [8] and [9])

Key a. Sound pressure level L [dB over 2 x 10^{-4} bar]
b. Volume level L_s (phon)
c. Pain threshold
d. Threshold of audibility
e. Frequency [Hz]

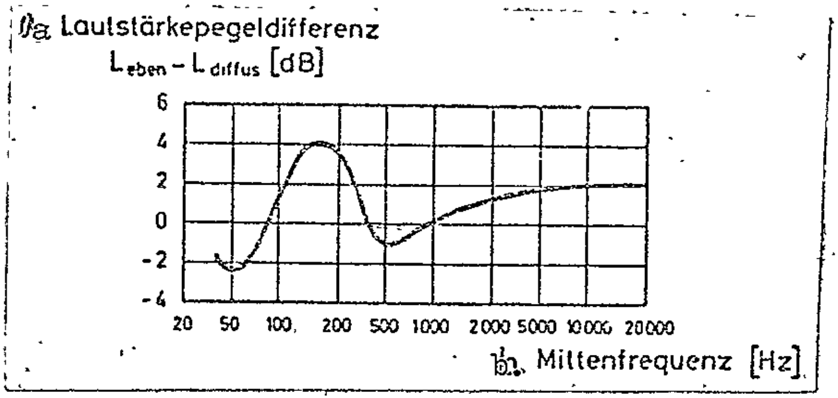

Figure 7: Deviation of curve between plane and diffuse sound field at the same volume [9]

Key a. Volume level difference $L_{\text{plane}} - L_{\text{diffuse}}$ [dB]

b. Mean frequency [Hz]

Measurements have shown [9] that only the sound pressure present at the ear drum is responsible for volume. Therefore curves of equal volume may be calculated for the diffuse sound field from that of the plane sound field according to $L_{\text{plane}} - L_{\text{diffuse}}$. Figure 7 shows the difference between the levels in plane and in the diffuse sound field of narrow band noise of equal volume. /16*

2.2.4 Loudness

The volume level is a subjective as well as objective measure, since it is gained by means of test audience via a comparison with the sound pressure level of these standard sounds. The subjective concept of loudness [10] developed from the question on how much louder or softer a sound to be measured in comparison to a standard sound. The objective physical quantity sound level is coordinated with the subjective response quantity loudness. As reference point for the sound level series loud, twice as loud, three times as loud, etc., and loud, half as loud, one-third as loud, etc. an international agreement was made on assigning a sine tone with the frequency of 1 kHz and a sound level of 40 dB in a plane sound field the loudness 1 sone. The /17*
relationship between loudness S and the volume level L_S is given in the

$$S\,[\text{sone}] = 2 \times \exp\left(\frac{L_s\,[\text{phon}] - 40}{10}\right) \qquad (5)$$

At values of less than 40 phons loudness follows the equation [9].

$$S\,[\text{sone}] = \frac{1}{16}\left(\frac{I_{1\,\text{kHz}}}{I_0}\right)^{0,3} = \frac{1}{16}\left(\frac{p_{1\,\text{kHz}}}{p_0}\right)^{0,6} \qquad (6)$$

ure level L [dB]

Table 2 shows several typical comparison values of volume level and loudness. It may be seen that doubling the volume level from 60 dB for a quiet conversation to 120 dB for the starting noise of a jet plane is not subjectively responded to as double loudness[1]. The statement that a jet plane is 50 to 60 times louder than a normal conversation is much more in agreement with experience.

a. Lautstärkepegel (Phon)		b. Lautheit (Sone)
140	c. Schmerzschwelle	1024
120	d. Düsenflugzeug	256
100	e. Lastkraftwagen	64
80	f. Sprechen	16
60	g. Leise Unterhaltung	4
40	h. Geräuscharmer Raum	1
20	i. Blätterrascheln	
3	j. Hörschwelle	

Table 2: Relationship of several values of volume level and loudness [8]

Key
- a. Volume level
- b. Loudness
- c. Pain threshold
- d. Jet aircraft
- e. Truck
- f. Speaking
- g. Quiet conversation
- h. Sound proof room
- i. Rustling of leaves
- j. Threshold of audibility

With the aid of the two equations the loudness of tones at other frequencies than 1 kHz may be easily determined using the curves of equal volume according to figure 6. Volume level values must merely be converted into loudness values; this conversion is carried out for figure 6 and is given in figure 8.

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[1] Also not with the expression 1000 times louder (60 dB ≙ 1000 ÷ 1)

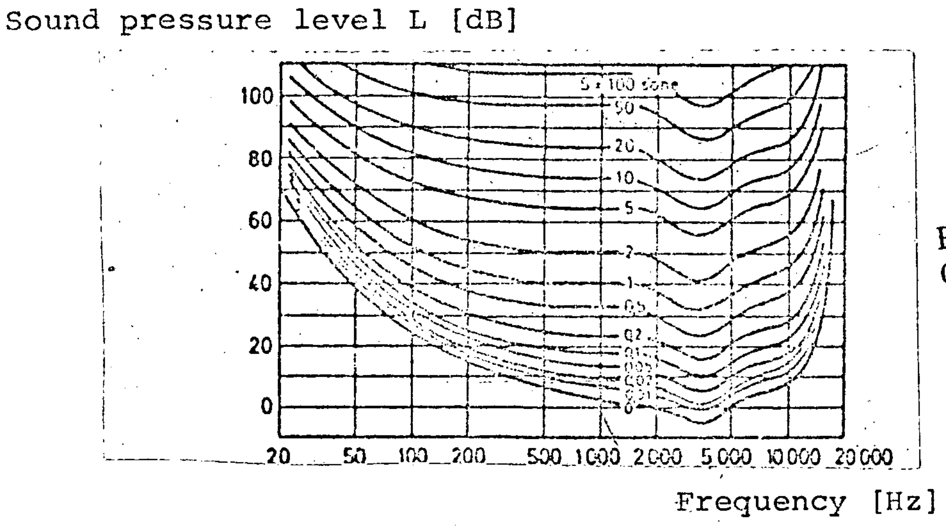

Figure 8: Curves of Equal Loudness for Tones in a Plane Sound Field [9]

Wide band noise is perceived by the human ear as much louder than narrow band noise of equal sound pressure level. In figure 9a a representation is given of the spectrum of three noise at a mean frequency of 1 kHz at equal sound intensity densities, but with differing band widths. The three sounds of the equal sound pressure level are perceived by the ear as differing in loudness. It is demonstrated in this case that for band widths smaller than the critical band width of 160 Hz, sounds of equal sound intensity den- /19* sities are perceived as being equally loud and that for band widths greater than 160 Hz, the subjective loudness as shown in figure 9b, increases in a linear manner with increasing band width. At constant sound pressure level of 60 dB the loudness for a band width of 2 kHz increases by the factor 2.5. /18*

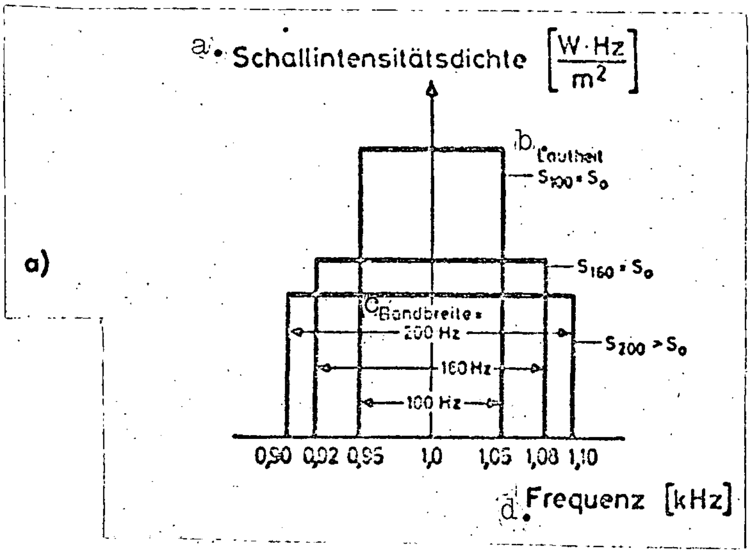

Figure 9: Influence of Band Width on Loudness at Constant Sound Pressure Level of 80 dB for a Band Noise of a Mean Frequency of 1 kHz [8]

Key follows on page 14 following figure 9b.

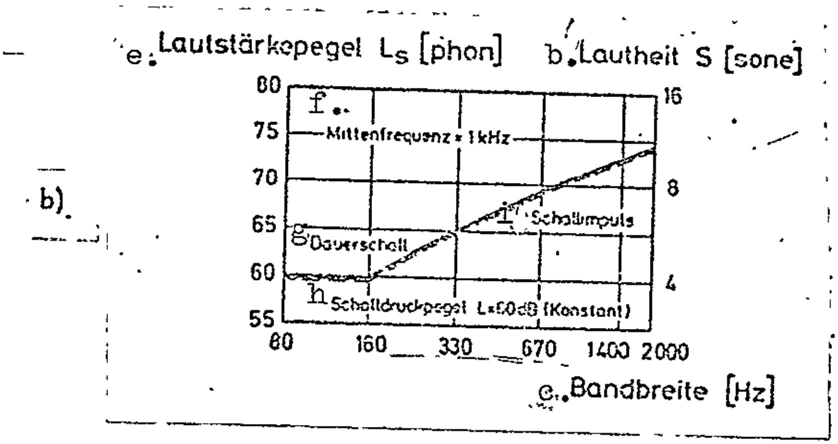

Figure 9: Influence of Band Width on Loudness at Constant Sound Pressure Level of 80 dB for a Band Noise of a Mean Frequency of 1 kHz [8]

Key a. Sound intensity density
b. Loudness
c. Band width
d. Frequency
e. Volume level
f. Mean frequency
g. Sound of continuing duration
h. Sound impulse
i. Sound pressure level L = 60 dB (constant)

As shown in figure 9 the required resolution accuracy in the spectral analysis needs no filter with narrower band width than the critical band width, since spectral distribution in band width smaller than the critical one do not influence loudness.

The sense of hearing divides the audible frequency band into 24 closely arranged frequency groups [8], where the upper frequency of one group is simultaneously the lower frequency level of the next group. The audible frequency range of 20 Hz to 15.5 kHz is therefore covered by 24 filters for the loudness analysis.

2.2.5 Subjective Tone Level

The critical bands of the human ear also seem to be related to a further characteristic of hearing, with that of the subjective tone level [11]. The subjective tone level indicates how the ear compares frequencies of differing tones.

If an audience is given a reference tone or a reference narrow band noise of 8 kHz, for example, and if these persons are given the opportunity of selecting that tone on an adjustable second tone source, which is half as high as the reference sound in their opinion, the results is that a tone of 1.3 to 1.4 kHz is perceived as half as high.

Figure 10 demonstrates the relationship between transmitted test tone with the frequency f_1 and the subjective adjusted tone f_2 with the frequency $\frac{1}{2}$ x f_1. Instead of the frequency as stimulus quantity the quantity tone level is introduced as response quantity for the concept pitch. The unit of tone level is the mel.. The frequency f = 131 Hz of the tone c_o was chosen as reference point, as shown in figure 11, and was set equal to 131 mel. In the case of values under 50 Hz the numerical value of the frequency equals the numerical value of tone level. The representation in figure 11 results from figure 10. The tone f_1 and f_2 have tone levels with numerical values differing /20* by the factor 2. While the frequencies of tones increase to 16 kHz, the subjective response reaches tone levels of up to 2400 mel.

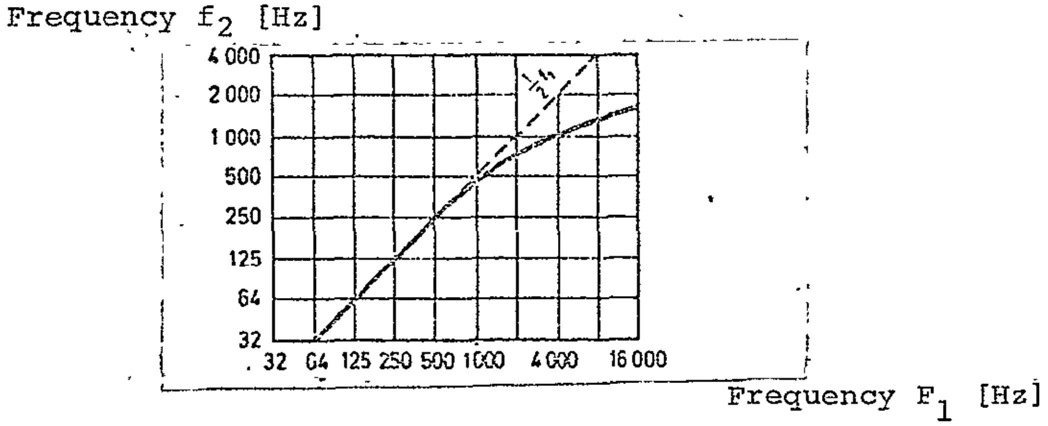

Figure 10: Frequency f_2 of a tone with a subjective pitch response half as high as the pitch of a tone with the frequency f_1 [9]

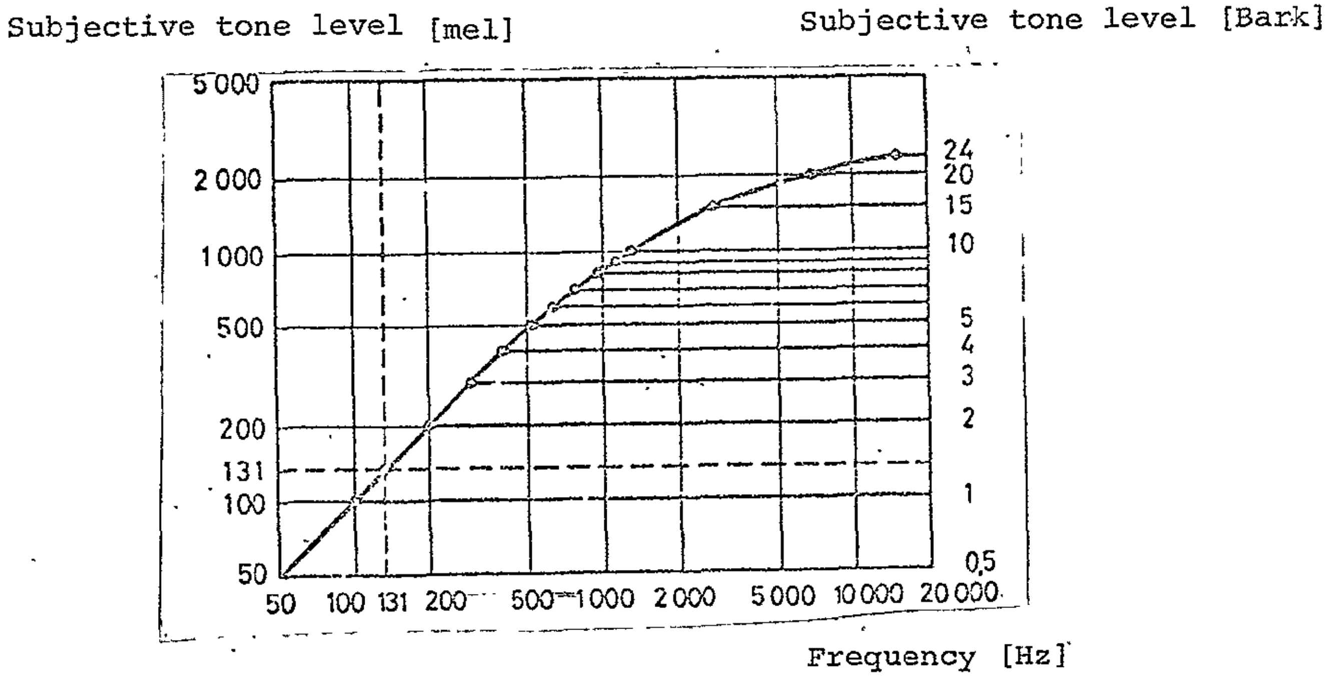

Figure 11: Pitch z as a Function of Frequency

The ordinates indicated on the right in figure 11 show the relationship between the experimentally determined 24 frequency groups (compare figure 9) and the tone level scale. The scale results from the equation definition 1 Bark = 100 mel. /21*

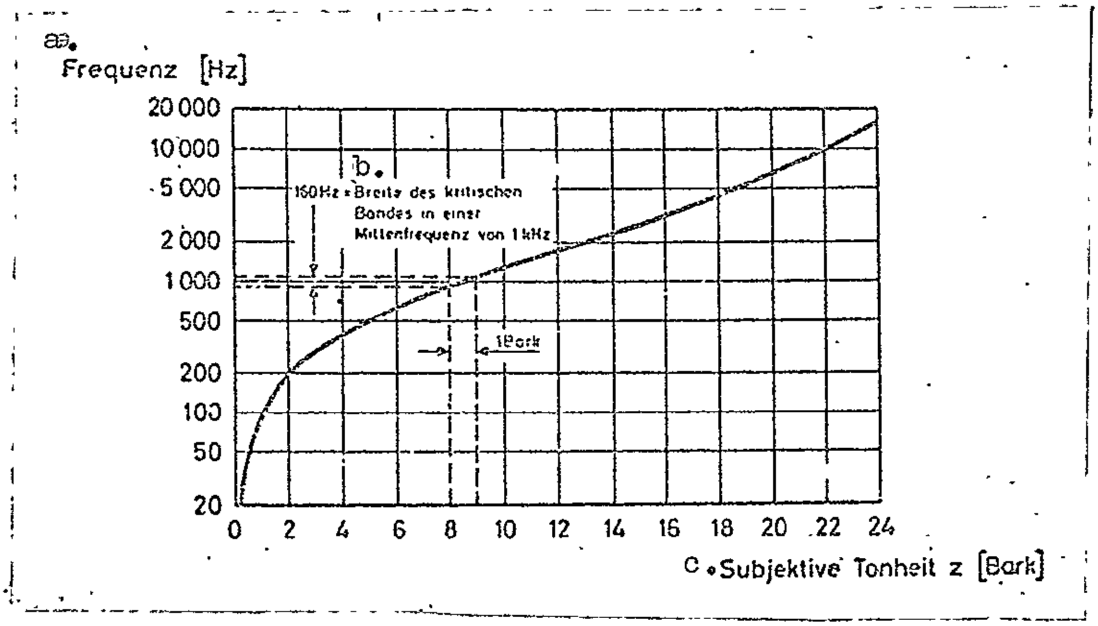

Figure 12: Subjective Tone Level as a Function of Frequency

Key a: Frequency
b. Width of critical band in a mean frequency of 1 kHz
c. Subjective tone level

The values 1, 2, 3 ... 24 Bark correspond to the uper frequencies of the 24 frequency groups.

Figure 12 demonstrates the relationship between tone level and frequency resulting in figure 10. Generally this frequency scale is employed in loudness measurements, where the linear scale is in subjective tone level. Up to 500 Hz the relationship is linear while above 500 Hz and 5 Bark the frequency and the tone level are in a logarithmic relationship.

2.2.6 Masking

When two tones are heard simultaneously, differing in frequency but equal in volume level, it would be expected that the loudness would be perceived as being greater than with only one of the two tones. As experiments show the ear perceived this addition only in the case of tones with a sufficient difference in frequency. The closer the frequencies lie to one another, the more influence they have on one another so /22* that the total loudness lies under the sum of the partial loudnesses. This effect is termed partial masking [8, 9, 11].

The masking effect is explained by the receptor structure and in the mechanics of the human inner ear. An acoustic stimulus reaching the ear does not only excite the receptors of the basilary membrane corresponding to this stimulus, but also adjacent areas of the receptor field [9, 37]. Masking studies have provided results that even spectral lines excite large areas of the receptor field and contribute in this way to a physiological spread of the lines [9]. This stimulation of sensory cells to adjacent frequencies may be qualitatively expressed by the quantity a loudness density.

$$\text{Loudness density} = \boxed{\int_{z=0}^{z=24\ \text{Bark}} \frac{ds}{dz}\, dz} \qquad (7)$$

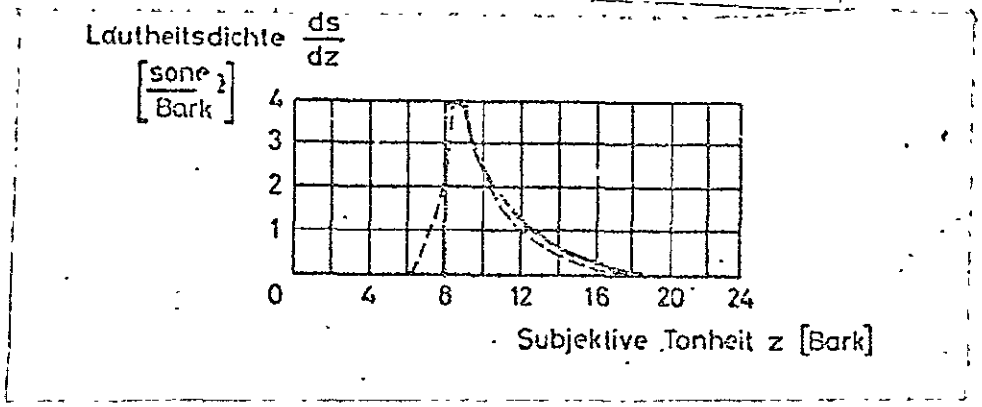

Figure 13: Masking Curve for a Pure 1 kHz Tone with a Loudness of 13 sone$_g$ 1). The solid line curve of equal range represents an approximation which has proven itself good in practice [8].

Figure 13 shows the loudness density of a 1 kHz tone with a sound level of 77 dB as a function of subjective tone level. The range of experimentally determined broken line curve and the solid curve approximation are equal and amount to 13 sone$_g$ 1). The horizontal section of the curve amounts to 1 Bark and represents the band loud density. The masking curve was determined experimentally for the 24 bands and for differing sound pressure levels.

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1) The index g indicates the area of excited adjacent frequencies, i.e. one of the 24 frequency groups (see also figure 12).

/23*

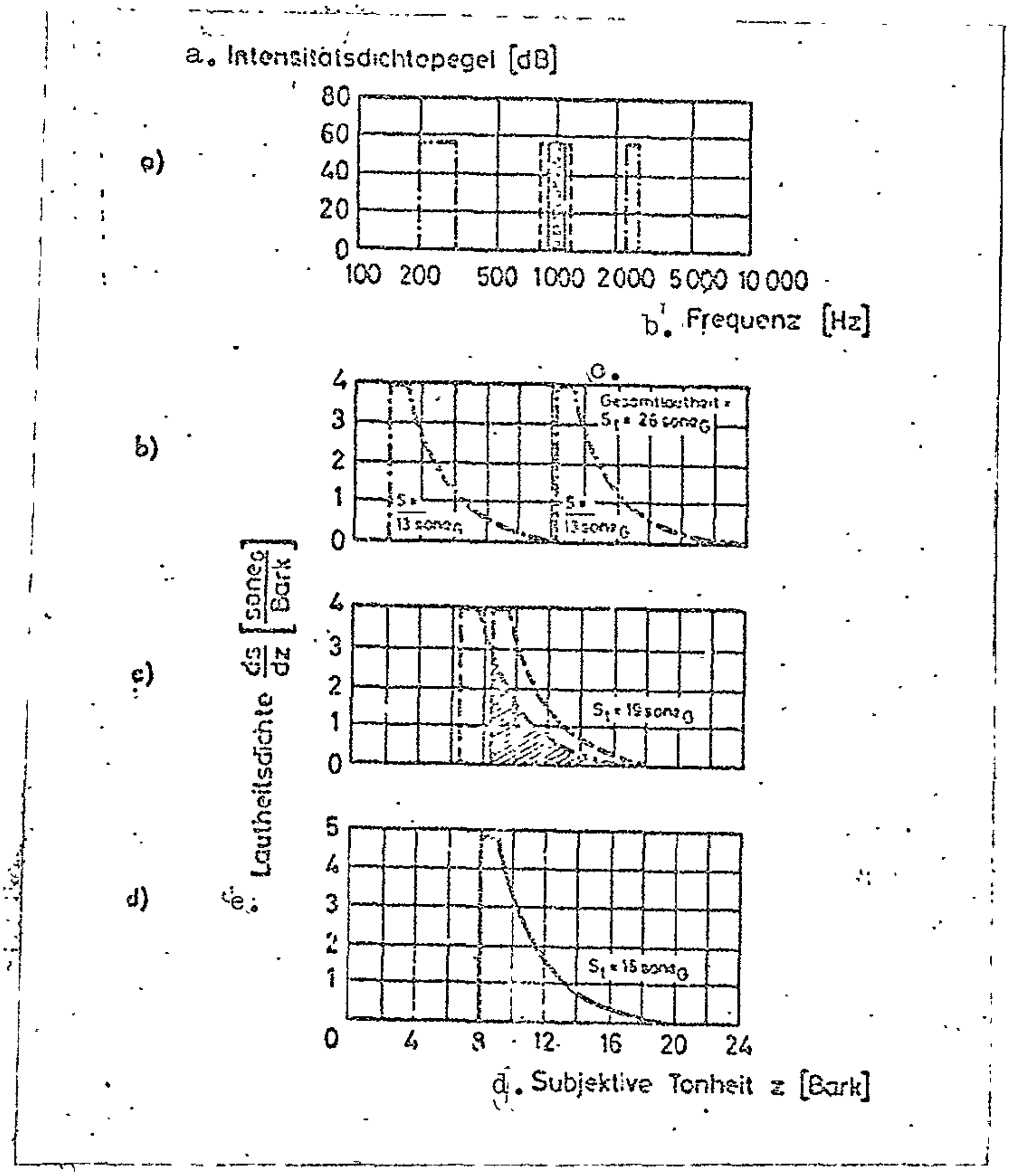

Figure 14: Masking Effect (Explanation see text) [8]
(Note the expanded Ordinate in Figure portion d)

Key a. Intensity density level
b. Frequency
c. Total loudness
d. Subjective tone level
e. Loudness density

The band pass noise indicated in Figure 14 has a sound pressure level of 77 dB. Figure 14a shows the spectra of noise pairs, broken lines for a small frequency difference and dash and dot lines for a large frequency difference of the two noise pairs.

The sounds are filtered out of a white noise and have in each case the width of the frequency group. In the case that the two sounds are far apart from each other, as in Figure 14b, no masking occurs. /24* Two completely separate partial loudnesses are formed and the loudness of the double sound is then double as large as the loudness of each single sound. At a sound pressure level of 77 dB the loudness value of 13 sone$_g$ results according to equation (5), i.e. both sounds with a total of 26 sone$_g$ are perceived as doubly loud as one sound, signifying an increase in loudness level from 77 dB to 87 dB. If the mean frequencies of the two narrow band sounds are arranged close to one another, as shown in figure 14c, the excited areas of the inner ear over lap to a great extent and a common excited level is created with a single maximum. The course of the maximum presents the ear from recognizing that it was caused by two narrow band sounds. This area, indicated in the figure by hatching, shows how much smaller the loudness of the double sound becomes. In this example the total loudness is reduced from 26 sone$_g$ to 19 sone$_g$. In figure 14d the mean frequencies of individual sounds succeed one another. The sound intensity is doubled, resulting in a volume level increase of 3 phon to 80 phon according to equation (4). This level corresponds to a loudness of 16 sone$_g$. The increase in volume level of 7 phon calculated, which results from the separation of individual sounds according to frequency, is in agreement with experience.

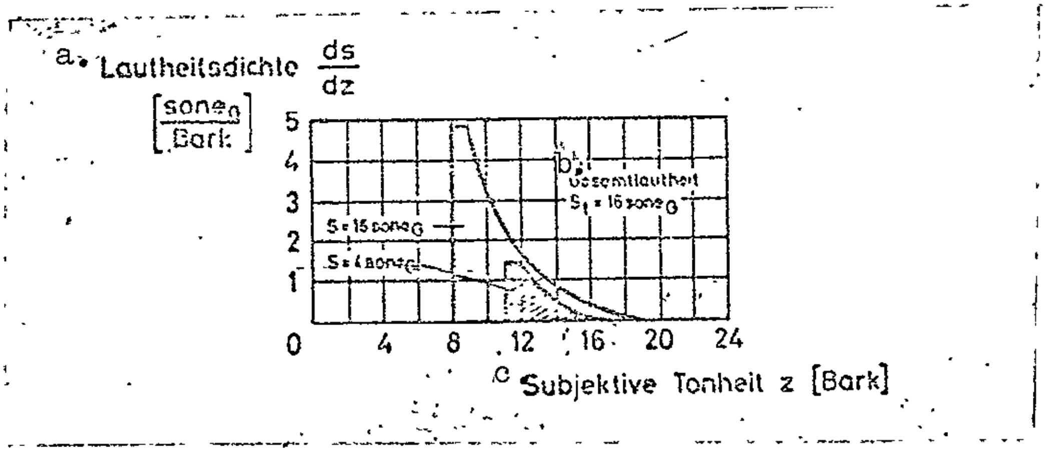

Figure 15: Complete Masking of a Tone

Key a. Loudness density
b. Total loudness
c. Subjective tone level

Generally tones, sounds and narrow band sounds have /25* differing volume levels. It may therefore occur that a strong tone partially or completely masks a weak tone, i.e. makes it inaudible. If this extreme case occurs, as shown in figure 15, the physically present but not audible tone does not contribute to the total loudness.

For loudness analysis representing the basis for synthesis of sounds this masking effect is of special interest, since frequency portions present in the spectrum, not perceived by the ear because of the masking effect, also do not have to be taken into consideration in sound synthesis.

3. Aircraft Sound Simulation

The sound simulation presented in this report includes mainly motion and engine noises of the vehicle, for example, aerodynamics and landing gear sounds. Only aircraft sound simulation will be presented here, but the procedure is also transferable to land and ocean-going craft. The reproduction of any sound via electronic means presents no problem today, but usually demands enormous technical equipment. This may be reduced when the characteristics of the human ear are employed in the reproduction of sound. As can be seen from chapter 2, the ear judges sound of identical spectra to be identical, independent of phase relationships between the elementary sound sources. In addition frequency portions lying close to one another in the spectrum are not received by the ear because of the masking effect. In this chapter the subjective and objective analysis of original sound to be carried out and procedures of sound synthesis are reported.

3.1 Sound Analysis

Via the sound analysis the parameter are determined which later alter the characteristic of the sound of aircraft to be simulated in the simulation model. In addition to the changing volume and direction of various sound sources the frequency spectrum has to be analyzed. The spectrum is analyzed in this case objectively and subjectively.

3.1.1 Objective Analysis /26*

An objective analysis is understood as the recording of the amplitude-frequency-spectrum changing in time. In the case of non-stationery sounds periods of time are used which are more or less close to one another and assumed to be quasi-stationery for analysis. As described in Chapter 2.2 the determination of physical quantities such as frequency and sound levels is not necessary for the evaluation of sounds, but rather the response quantities tone level and loudness as well as the related masking.

Sound level measurement devices are available for simple sound level measurements. With these the sound pressure is conducted through an evaluating network, reproducing the frequency half of the ear and indicating the sound pressure in the logarithmic form. Sound level measurement devices are simple in design. However they have the drawback that they can only record one tone. A wide band network may not simultaneously evaluate a low frequency tone of high level and a high frequency tone of low level. Since this device also does not take the masking effect into consideration, it is unsuitable for sound analysis.

The frequency range between 45 Hz and 14 kHz is divided into bands with almost critical widths in the case of the Zwicker procedure (DIN 45 631) [8, 9, 10]. The sound is taken apart with the aid of filters and the sound pressure level of each band is measured. The partial loudnesses resulting are recorded in a diagram, automatically taking masking into consideration. Since the required filters are not generally available for the critical bands, the procedure was modified [8, 9] in order to apply one third octave filters. The error occuring due to amplication of one third octave filters may be neglected in the range of 280 Hz to 14 kHz [8, 9].

In the case of frequencies lower than 280 Hz the one third octave bands are substantially more narrow than the frequency groups. Here the frequency group level results from the addition of the sound intensities in the one third octave. According to DIN 45 631 in this range a grouping of the sound into two octave bands and a two thirds band is required. In this manner the hearing range is covered by /27* twenty filters instead of by 24 critical bands. The differing loudness densities of individual bands resulting from this procedure are taken into consideration by means of scale corrections in the Zwicker diagram.

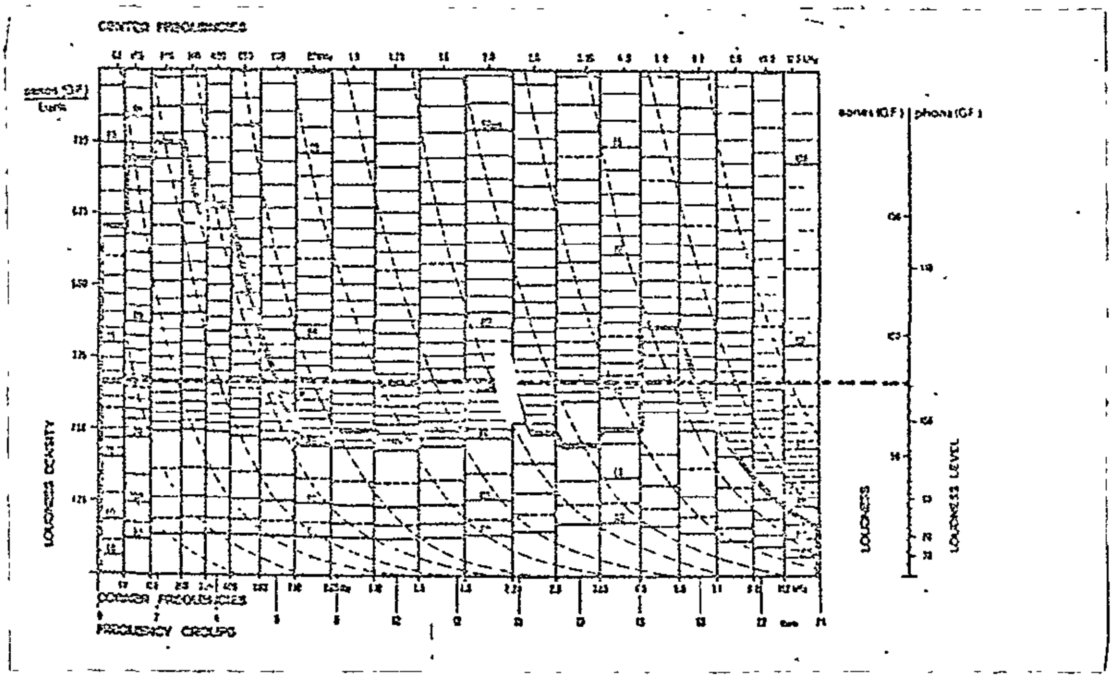

Figure 16: Zwicker Diagram. The boldface curve represents the analysis of a sound.

There are ten different Zwicker diagrams with varying sound levels, five for the plane and five for the diffuse sound field. Figure 16 shows a diagram for the plane sound field in the case of a loudness range to 120 sone$_g$. The loudness density in sone$_g$/Bark is plotted as ordinate and the 20 bands as abcissa and for comparision the frequency groups (tone level z in bark). The lines plotted vertically to the horizontal lines bordering the bands correspond to the core loudness of the band. The course of core loudness may be discovered by following the band level, for example 80 dB. In the resulting curves the frequency path of the ear may be recognized. The broken line curve correspond to the approximation for the upper edge loudness, as explained in figure 12. The increase to core loudness is plotted as vertical lines. The coincide with the vertical lines limiting the bands.

The sound spectrum drawn in figure 16 was derived in the following manner: The core loudnesses determine per band are indicated by a horizontal line. From the right side of the core loudness the upper edge loudnesses are plotted in such a manner that they run parallel to the broken lined curve. The lower edge loudnesses are plotted almost vertically. The resulting area represents the total loudness of the sound. The area determined planimetrically may be converted into a rectangle of equal area with the base 24 Bark. The horizontal unbroken line now indicates in an equivalent manner the total loudness of sound recorded. The value is given on the right scale as loudness in sone_g or as volume level in phon_g.

The Hewlett-Packard loudness analysator 8,051A was employed for frequency analysis of the aircraft sounds studied in this report. This analysator generates a new loudness spectrum every 25 msec. and a display of the total loudness. In the following the essential points in favor of a loudness analysis as opposed to a frequency analysis are summarized once again:

1. The sound is recorded as perceived by the ear and is taken apart corresponding to the ear curve.

2. As a measure for the frequency the response quantity tone level is used.

3. As measure for amplitude (sound level) the response quantity loudness is used.

4. Partial masking of tones is evaluated as it would be heard, i.e. the total loudness of two tones is not equal to the sum of the individual loudness of each tone.

5. In the case of total masking the masked tone does not contribute to loudness; although it is present physically it is not audible and does not need to be taken into consideration in synthesis.

The figures 17-21 give examples of a comparison of amplitude-frequency spectrum with the loudness-tone level diagram (Zwicker Diagram) of several periodic waves. /29

Figure 17 gives a sine wave which appears as a line in the amplitude-frequency spectrum. The frequency of the tone is plotted on the abcissa and the intensity density is plotted on the ordinant, i.e. the amplitude in relation to frequency interval, here to the frequency value of the tone. The base wave of the spectrum is designated by f_1, also for figures 18-21. According to the Fourier synthesis or analysis every periodic spectrum may be put together or taken apart from its or into its base end over waves. Figure 18 and 20 show the rectangular and the triangular oscillation. It is noteworthy here that both oscillation forms only contain an odd number of over waves, i.e. form only an "incomplete spectrum". The two spectrums differ due to various rates in amplitude reduction of individual over waves. Because of the quadratic reduction in amplitude of the triangular wave this has already achieved the same value at the ninth over wave as the rectangular wave achieves at the 127th over wave. Figures 19 and 20 show a rectangular impulse and a saw-tooth wave. Both wave forms contain a "complete spectrum", i.e. the even numbered as well as odd numbered over wave are present. In this case the impulse wave is reduced more rapidly than the saw-tooth wave. It achieves the same amplitude value at the sixth over wave as the forty eighth over wave of the saw-tooth wave.

The figures 17 to 21 show the corresponding loudness-tone level diagrams (Zwicker Diagram) for comparison with the amplitude-frequency spectrum. In Figure 17 the loudness area is indicated with the continually decreasing upper edge loudness for the sine wave, which does not lie in the selected example in the frequency group with core frequency of 400 Hz. In the figures 18 and 20 it may be observed that the even numbered over wave are lacking in the loudness diagram for the rectangular impulse and for the saw-tooth wave. Figure 22 shows a comparison of amplitude-frequency spectrum with the loudness-tone level diagram for white noise. It may be seen from the Zwicker diagram that the loudness clearly increases in case of higher frequencies.

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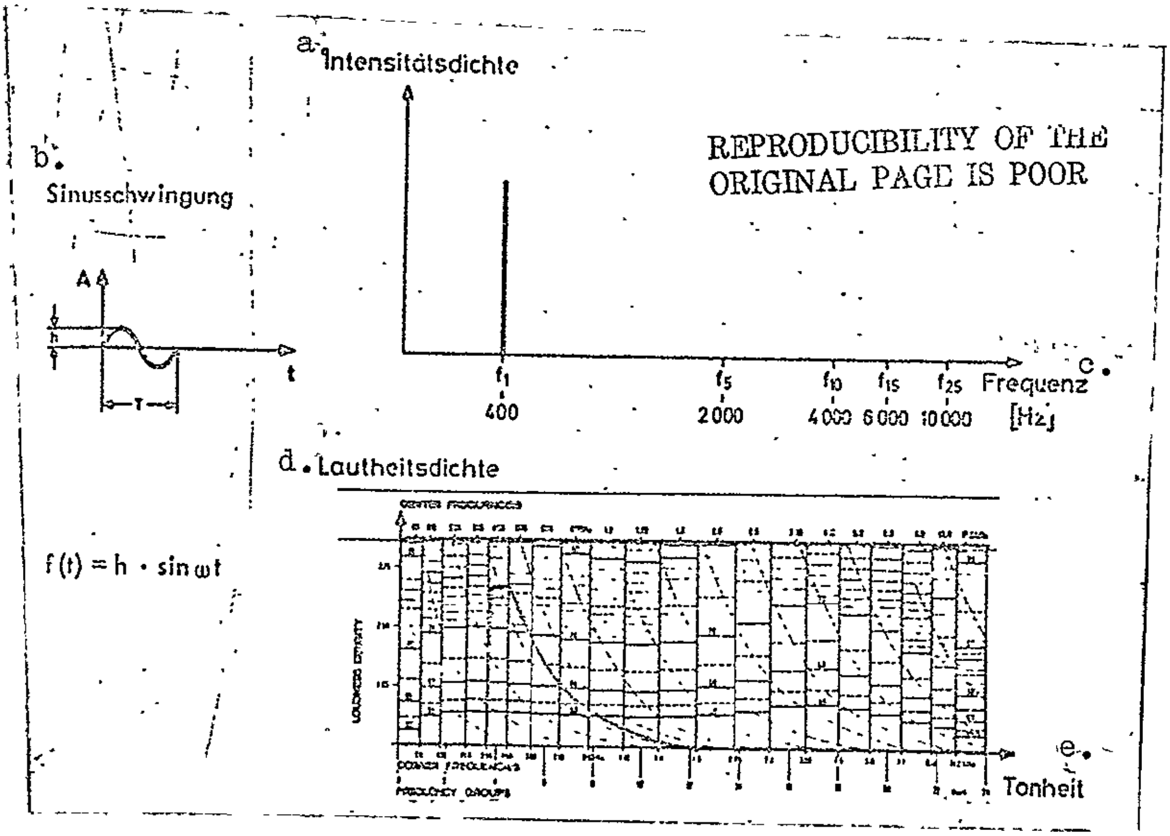

Figure 17: Comparison of Amplitude-Frequency Spectrum with the Loudness-Tone Level Diagram of a Sine Wave

Key a. Intensity density b. Sine wave c. Frequency
d. Loudness density e. Tone Level

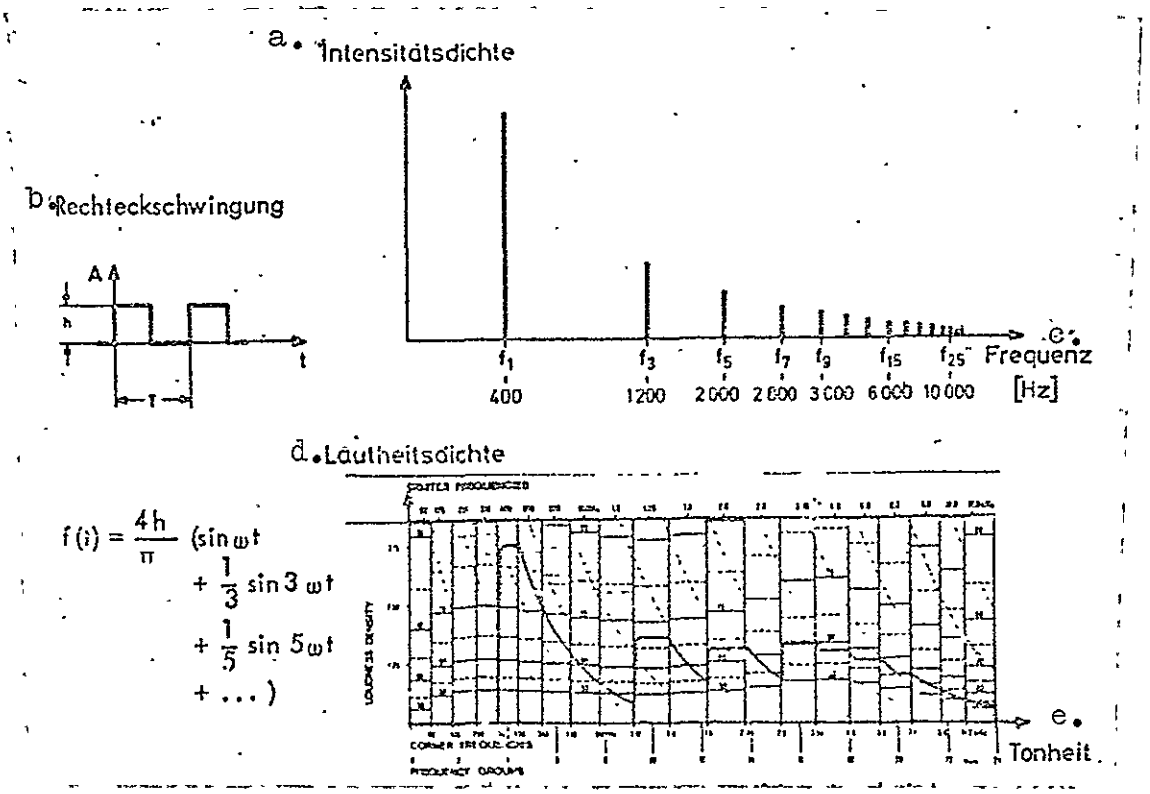

Figure 18: Comparison of Amplitude-Frequency Spectrum with the Loudness-Tone Level Diagram of a Rectangular Wave

Key a. Intensity density
b. Rectangular wave
c. Frequency
d. Loudness density
e. Tone level

/31

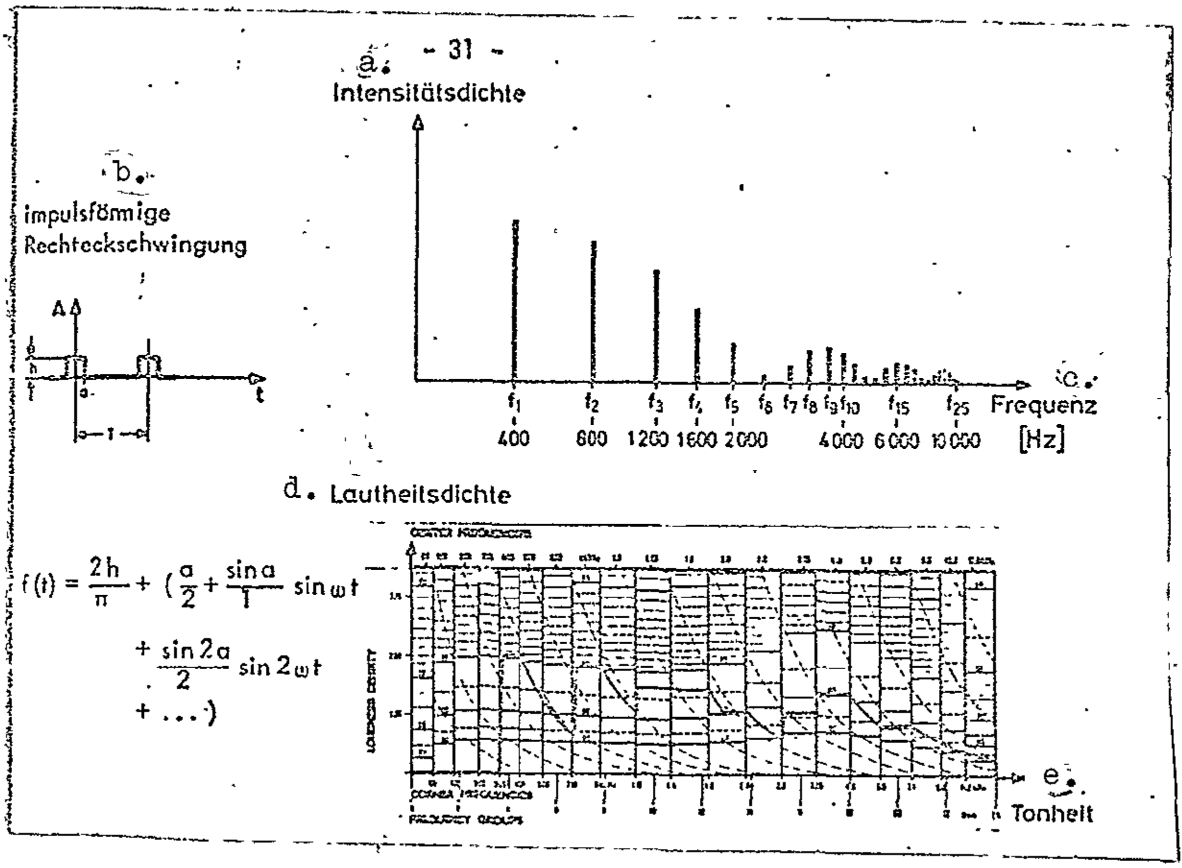

Figure 19: Comparison of Amplitude-Frequency Spectrum with the Loudness-Tone Level Diagram of a Rectangular Wave in the Shape of an Impulse

Key a. Intensity density
b. Rectangular wave in the shape of an impulse
c. Frequency
d. Loudness density
e. Tone level

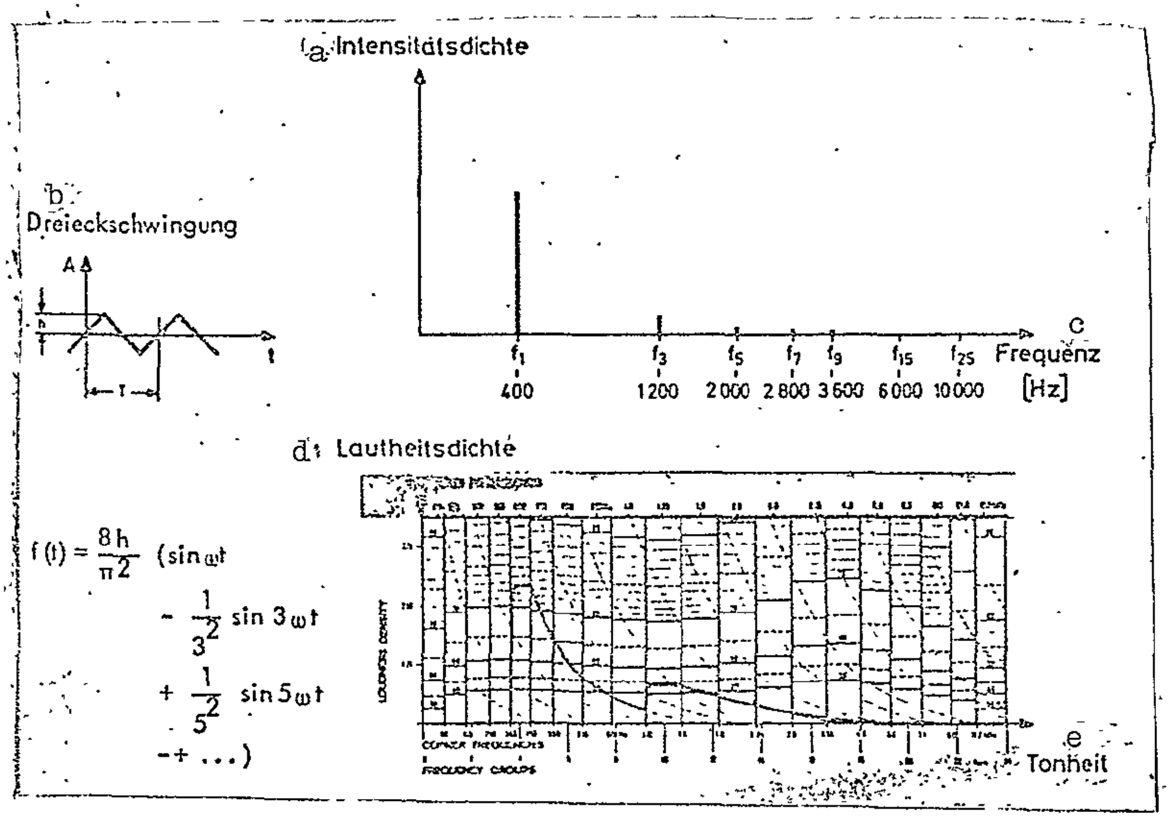

Figure 20.: Comparison of Amplitude-Frequency Spectrum with the Loudness-Tone Level Diagram of a Triangular Wave

Key a. Intensity density
b. Triangular wave
c. Frequency
d. Loudness density
e. Tone level

/32

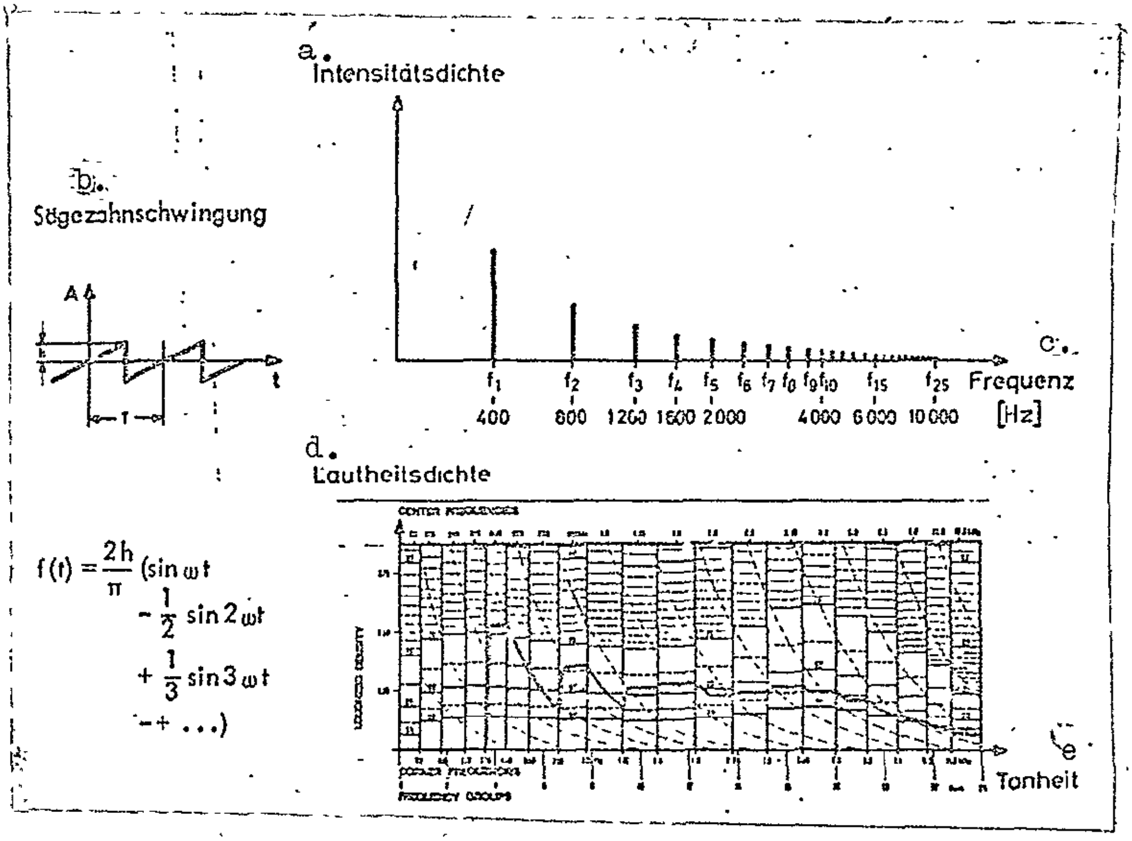

Figure 21: Comparison of Amplitude-Frequency Spectrum with the Loudness-Tone Level Diagram of a Saw-Tooth Wave

Key a. Intensity density
b. Saw-tooth wave
c. Frequency
d. Loudness density
e. Tone Level

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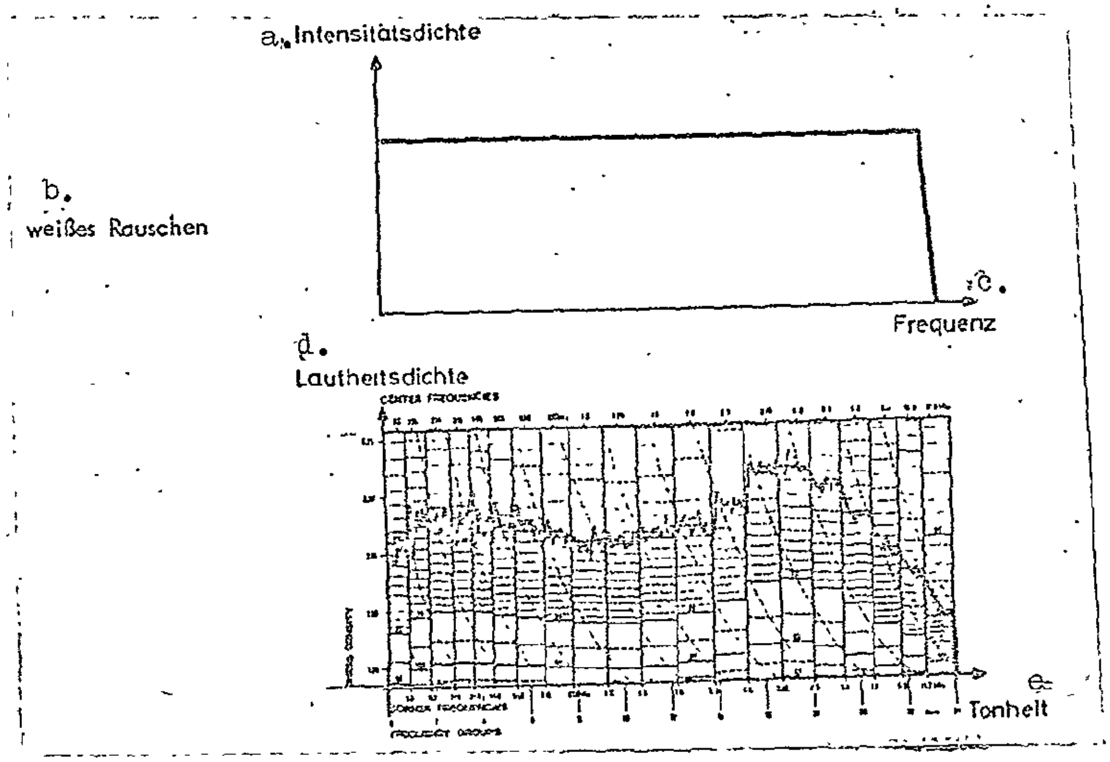

Figure 22: Comparison of Amplitude-Frequency Spectrum with the Loudness-Tone Level Diagram of White Noise (Upper Limit Frequency 20 kHz)

Key a. Intensity density
b. White noise
c. Frequency
d. Loudness density
e. Tone level

While in the representation of intensity density-frequency diagram the amplitude of over waves for rectangular, triangular and saw-tooth waves decrease at a constant rate, follow the course sin x/x for the rectangular voltage in the form of an impulse (see also text accompanying Figure 24b) and remain constant for white noise, the representations in the loudness diagram show that the subjective loudness of the waves perceived by persons does not decrease in a linear manner with frequency, but is perceived as louder for higher frequencies. The masking of tones, in this case over waves, may also be seen in the diagrams, since two or even three over waves fall into a frequency group through the frequency band distribution of the Zwicker Diagram for higher frequencies. /33

3.1.2 Subjective Analysis

Subjective analysis is understood in connection with sound simulation as an evaluation supporting the objective analysis, in relation to simulation quality and sound dynamic. First trial persons are acquainted with the original sound in all load cases, for example, engine sounds arising in normal driving and in the mountains as well as the wind sounds dependent on velocity. This stored knowledge on sound and the functional relationships typical for vehicles is released in a subsequent evaluation of synthesized sound. It may be stated that the first set of the subjective analysis consists in understanding the relationships of the noise accompanying the total dynamics of the vehicle.

Normally in the case of vehicle simulators the sounds are reproduced as they occur in the cockpit. For evaluation of the sound in the synthesis a tape recording is made at the location of the sound area of interest. In order to insure that the sound recording was made with a high degree of reproduction quality, in a second step an additional subjective evaluation should be made via a comparison of sounds in the original vehicle with those of the tape recording by several test persons. In addition to the evaluation of sound spectrum the direction of various sound sources, for example, from above or from below, or from the right or from the left, etc., are also to be analyzed by the test persons. The recording of volume is not critical, since it has been shown that the users of vehicle simulators adjust the volume of the sound simulator subjectively. /34

The objective analysis of the recorded sound spectrum is often extremely difficult, since sound portions with similar frequencies from different sources may not be discerned with the aid of sound analysators. For example, the aircraft turbine sound and the wind sounds in the cockpit together result in the function "climbing". It is almost impossible to separate the two sound portions from one another by means of analysator. The reason for this lies in the fact that these two sub-spectrums strongly overlap. An aid in this case is one characteristic of hearing, known as the Cocktail Party effect.[12]. Where many are conversing at the same time, most people are capable of hearing and understanding a single partner in the voice tangle. This capability of tuning out of background information is of course dependent upon the relative intensity of desired and unwelcome conversation. This effect is employed in the third step of the subjective analysis when listening to the recorded sound spectrum as well as in the evaluation of original vehicle sounds and is noticeably intensified after repeated listening, i.e. it leads to separation of similar frequency spectrums, not technically distinct.

In a few cases criteria on a sound may also be defined without available data gained through analysis. The desired sound based solely subjective knowledge is "adjusted" on a universal sound simulator. If this path is taken, an extremely flexible system for synthesizing sound is required. This idea leads finally to the development of modular sound simulation systems, capable of simulating any indicated sound without hardware modification.

3.2 Sound Synthesis

The process of sound synthesis includes the establishment of mathematical models of sound sources as well as the technical synthesis of sound spectrums (see also figure 1). Realization of these two portions of synthesis may be achieved today in a purely analog manner and also in a purely digital manner. In addition the portion of the hardware may be reduced more and more in favor of the more flexible software. The mathematical model of a jet engine, for example, or the wind compacting on the cockpit each represents a sound source generated in conjunction with vehicle sounds, which in the technical sound synthesis must be constructed as sound spectrum from individual elements. This section deals with the necessary theoretical and technical basis for generation of periodical and non-periodical sound waves as well as the applied technical procedures for synthesis of such sound spectrums. /35

3.2.1 Mathematical Synthesis of Sound Spectrums

Sound spectrums contain periodic and non-periodic portions, as described in section 2.1. In analysis the spectrums are examined as to periodicity. Periodic waves may result from simple individual waves, such as saw-tooth, rectangular wave, etc. by addition, subtraction, multiplication, etc. and assume the desired form by means of subsequently arranged filters. For this purpose it is most practical to choose those individual waves containing all over waves and of these again those, in which attitudes of over waves decrease slowly. The saw-tooth and impulse-shaped rectangular wave are especially suitable for this purpose. Wind and hissing sounds are non-periodic sounds gained from the white noise spectrum.

3.2.1.1 Periodic Spectrums

A periodic wave is characterized by its base wave. Frequency of the base wave is the lowest component contained in the wave. Along with the base wave the over waves with higher frequency and usually with lower amplitude of values form the periodic signal. Ordinarily these components are harmonics of the base wave, i.e. the frequencies are whole number multiples of base wave frequencies.

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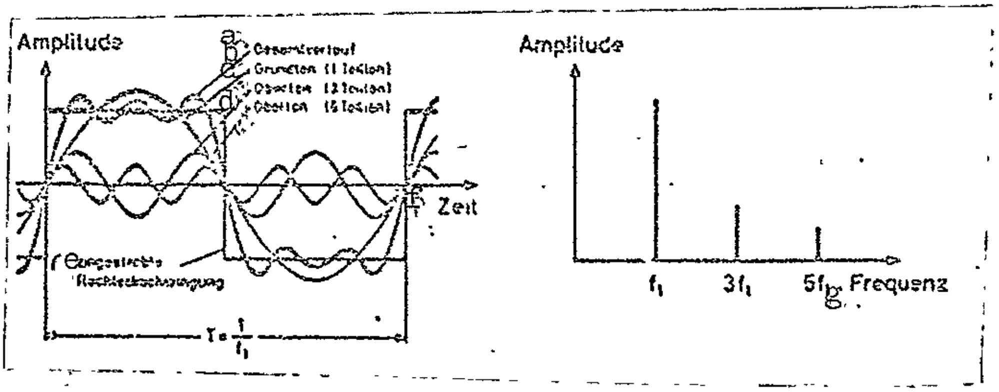

Figure 23: Amplitude/Time and Amplitude/Frequency Diagram of a Synthesized Rectangular Wave from Base Wave and Two Over Waves

Key a. illegible b. illegible c. illegible
d. illegible e. illegible f. Time
g. Frequency

Figure 23 shows the synthesis of a rectangular tone wave from the base wave and the third and fifth over wave. Analysis findings are that the saw-tooth wave and the impulse-shaped rectangular wave are most suitable as starting individual waves for synthesis. These tone waves with full range of over tones are easy to convert with filters. Considerations leading to the selection of a starting wave shape for synthesis are explained in the following employing the impulse-shaped rectangular wave. Figure 24 shows the wave form in the amplitude-time diagram, as well as the corresponding amplitude and performance density spectrum. The most important quantities described in this figure are:

1. The base frequency is 1/T, i.e. it is the lowest in the course of the wave and simultaneously,the periodicity (figure 24a).

2. The over wave are the harmonics of 1/T with the frequencies of 2/T, 3/T, 4/T etc. (Figure 24b).

3. The envelope of the over waves in figure 24b has zero
points at frequencies of whole numbered multiples of
the inverse of impulse width 1/ΔT. The envelope behaves
according to the general expression (sin x/x). The curve /37
demonstrates negative amplitude values1), indicating
phase lengths "shifted 180°" to the positive amplitude
values (figure 24b).

4. The number of over waves between the frequency 0 and /38
the first zero point 1/ΔT and between the zero points
1/ΔT and 2/ΔT etc. are proportional to the ratio ΔT/T
(figure 24b).

5. The performance spectrum indicates the energy or performance with which each harmonic over wave contributes to total performace of the spectrum. The performance is proportional to (amplitude)^2 and is plotted as a logarithm, corresponding to an agreement (figure 24c). The performance spectrum contains no phase information and may not be categorized as having any special time function (compare also figure 4).

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1) Figure 24b gives the theoretical "technical amplitude spectrum", figure 19 gives the same spectrum recorded by a spectrum analysator.

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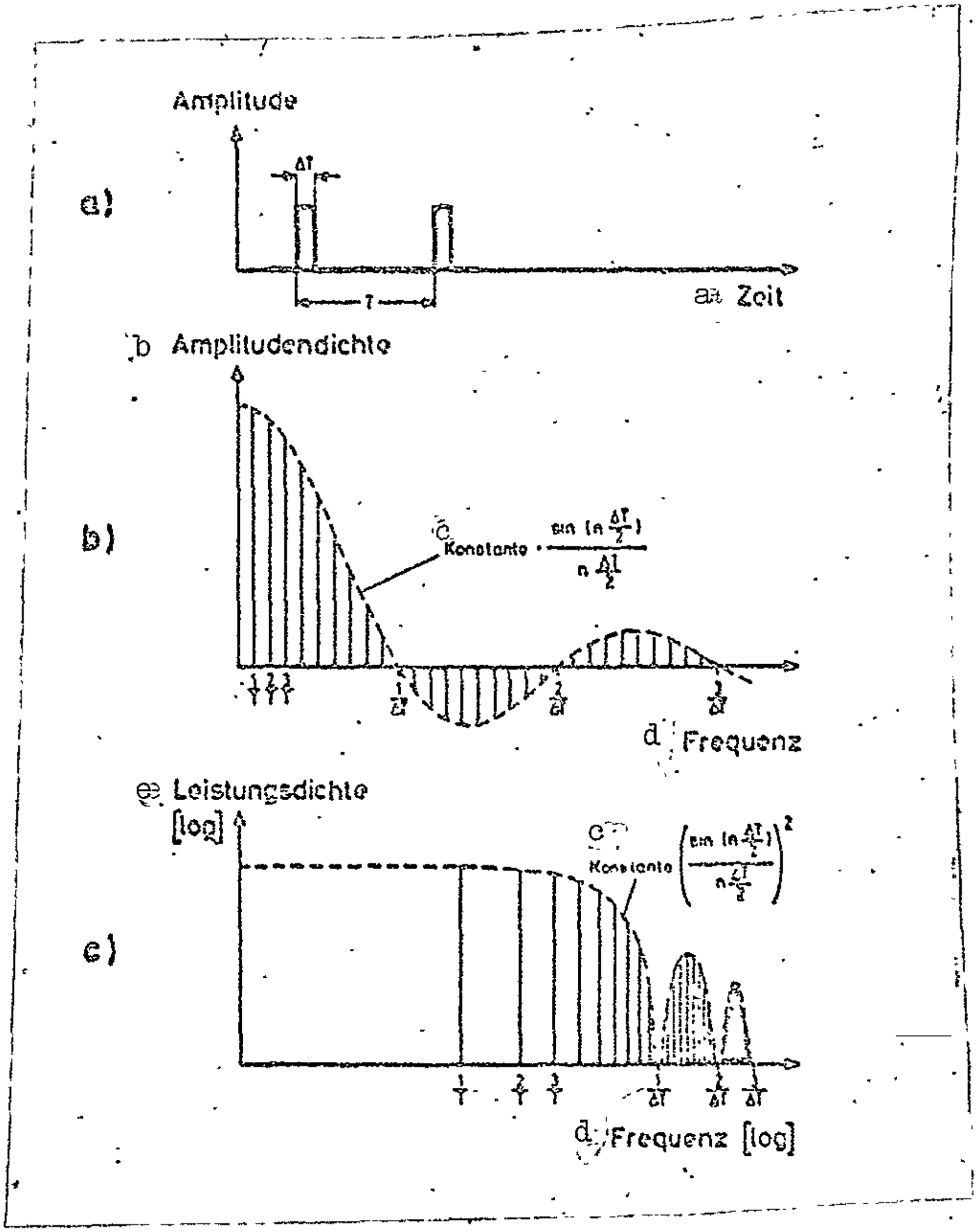

Figure 24: Amplitude Spectrum [1] and Performance Density Spectrum of an Impulse-Shaped Rectangular Wave

Key a. Time
b. Amplitude density
c. Constant
d. Frequency
e. Performance density

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[1] Figure 24b gives the theoretical "technical amplitude spectrum", figure 19 gives the same spectrum recorded by a spectrum analysator.

With the aid of the performance spectrum of starting signal the effect of a filter on the corresponding input signal may be determined. Figure 25 shows the performance spectrum of an impulse-shaped rectangular wave with differing characteristic quantities ΔT and T for a predetermined band filter with the limit frequencies A and B. The Fourier development of functions are indicated in the figure. When the .T (figure 25a) is doubled, i.e. the base frequency is halved to 2T (figure 25b) with the same impulse width (ΔT), the number of over waves in a filter range is doubled. The figure shows that the performance density was doubled in frequency range from zero to 1/ T. When the impulse width is doubled to $2\Delta T$ in relation to starting function (figure 25a), the total performance density is maintained, the value $1/2\Delta T$ is shifted to lower frequencies and "presses" the lines together so that the performance density per filter band is also doubled (figure 25c).

These mathematical relationships must be adhered to when designing sound sources for transmitting periodic signals. The initial turning of a turbine wheel, for example, is reproduced by means as such variable controlled generators.

3.2.1.2 Non-Periodic Spectrums /40

Sound spectrums do not only contain periodically spectrums such as tones or harmonious sounds consisting of individual spectral lines, but also of spectrums containing non-periodic portions. The non-periodic spectrum may not be taken apart by means of a Fourier analysis into whole numbered multiples of the base wave. They do not form a distinct line spectrum but a continuous spectrum. Of the many possible continuous spectrum that one is interesting for sound simulation, in which no frequency section differs from another and in which no frequency section may be distinguished from another. If these two characteristic requirements are met with a precision satisfying the perception capability of hearing, such a sound is termed "white noise"[1].

In addition to the non-periodic spectrum of white noise the quasi non-periodic spectrum, gained by means of pseudo-stochastic event generators[2], play an important role in sound simulation. Pseudo-stochastic event generators [13] generate events apparently withour rules, which, however, were gained in a starkly deterministic manner and which therefore demonstrate a systematic character when closely esamined. A sound generated in this manner is termed "pseudo-stochastic noise" and forms a discrete amplitude spectrum in contrast to white noise. If the period is chosen with a sufficient length and if the entire frequency band of the ear is covered, this may not be differentiated subjectively from white noise.

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[1] In acoustics the spectrum of white noise contains the frequency range from 16 Hz to 16 kHz. This range corresponds to human hearing [9].

[2] Pseudo-stochastic event generators generate a condition succession with a pseudo-stochastic course, i.e. seemingly coincidental.

/39

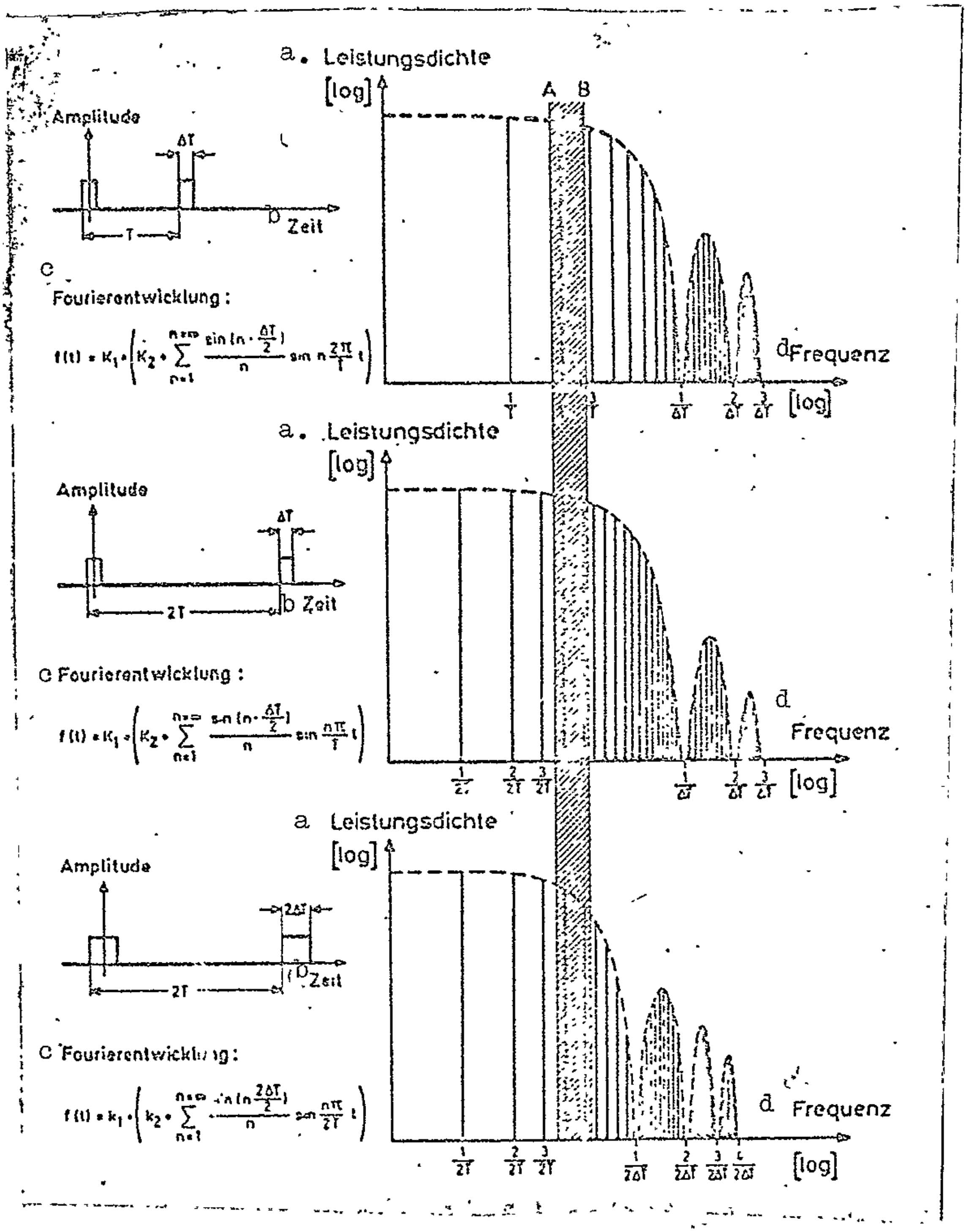

Figure 25: Performance Density Spectrum of an Impulse-Shaped Rectangular Wave with Differing Characteristic Quantities ΔT and T

Key a. Performance density b. Time c. Fourier development
d. Frequency

According to the considerations of the Fourier synthesis white noise consists of very many, closely adjacent sine waves[1]. These are represented by a sum in the form: /41

$$f(t) = \sum_n h \times \sin(\omega_n t + \varphi n)$$

The amplitudes h of all partial waves should be of equal size. The frequencies should fill the entire frequency band with an equal distribution and the zero phase angle φn should be distributed independent of one another and statistically equally over the angle range 0 to 2π. In the case of a sinus wave the ratio of the peak value to the effective value of amplitude is defined by the value $\sqrt{}$ of 2. In the case of white noise peak values occur far exceeding the $\sqrt{}$ of 2 times its effective value. Only a statistic statement on the postioning of a momentary value may be made. The distribution is determined according to a Gaussian probability function.

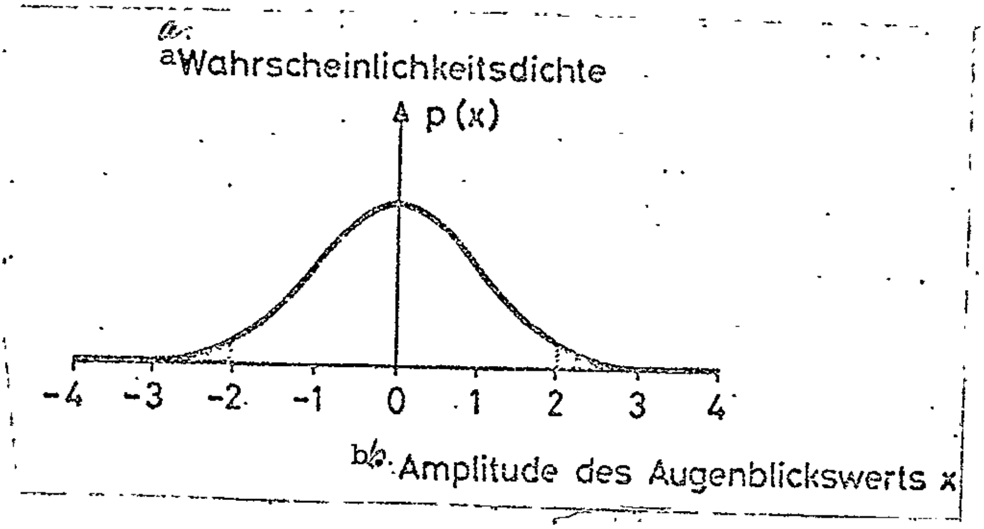

Figure 26: Probability Density p(x) of Momentary Values of White Noise [9]

Key a. Probability Density
b. Amplitude of momentary value x

Figure 26 shows the probability density p(x). It may be read from the figure, that the probability of finding x between two abcissa values equals the area under the bell curve limited by these two values. The total content of the area is 1. The hatched area

---

[1] The sine waves may not be described as whole numbered multiples of a base wave.

indicates the probability for a momentary amplitude exceeding the double effective value or not achieving it1).

By means of filters with certain attenuation distortions a white noise may be converted into a noise with another sound intensity density level. This is no longer constant, but a function of time. A narrow frequency band may be filtered out of the white noise spectrum with the aid of a band pass filter. Figure 27 shows several oscillograms of band pass noise [9]. The figure demonstrates that the noise assumes the character of an amplitude-modified wave as the frequency band becomes narrower. /42

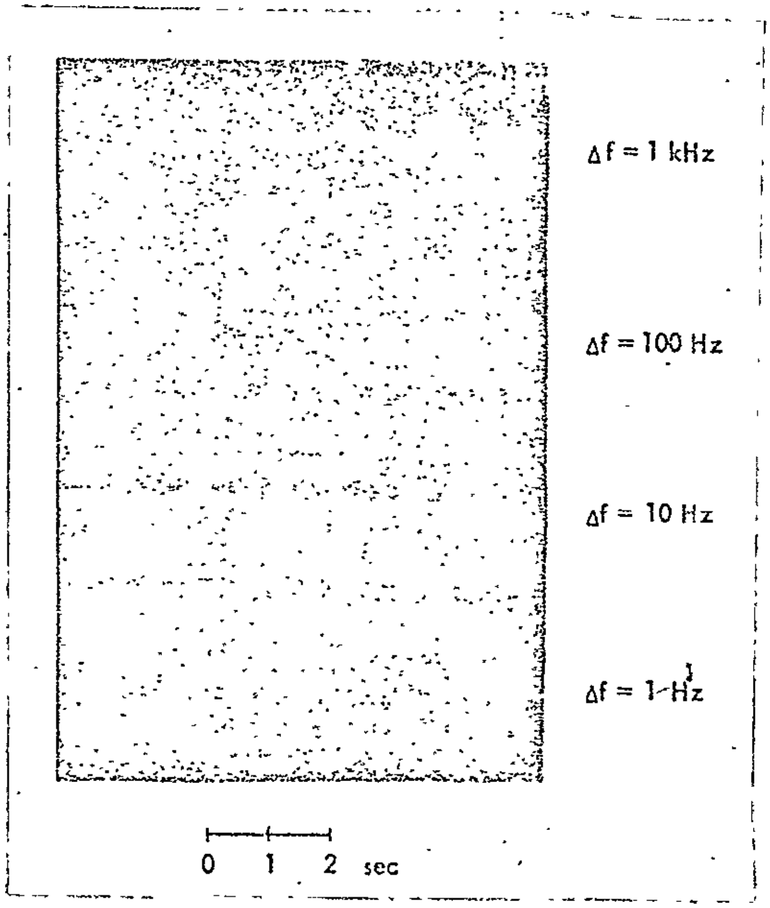

Figure 27: Oscillogram of Band Pass Noise of Varying Frequency Band Widths (Mean Frequency 1 kHz) [9]

The wave changes in an irregular manner and the "frequency" of the envelope curve becomes smaller. The envelope is again determined only statistically. The band pass noise shown in Figure 27 may be interpreted as a wave, simultaneously shaped in amplitude and frequency /43

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1) Exceeding and not achieving signifies that the value of the amplitude of the sine wave according to a convention may be positive or negative. This same consideration applies to the sound pressures p.

as a low pass noise. In this case attention must be paid to the design of "rhythmic" noise spectrum when simulating sound, since the ear may perceive an amplitude modulation as a volume change or a frequency modulation pitch variation.

Electric sound generators are employed as white noise source. The unordered heat motion of electrons in a resistance wire is utilized here, creating a white noise voltage spontaneously at the ends of the wire. In contrast to this, pseudo-stochastic distribution are gained by means of binary pseudo-stochastic event generators. Figure 23 shows a binary succession distributed in a pseudo-stochastic manner, in which the change from the value 0 to the value 1 and back occurs at the rate of the frequency rate $1/\Delta T$. The corresponding performance spectrum is shown in figure 28b. The zero positions appear here as multiples of frequency rate $1/\Delta T$ and the over waves as multiples of the lowest occuring frequency $1/T = 1/N\Delta T$. N signifies the number of times required for a pseudo-stochastic binary succession. Figure 28c shows the course of performance spectrum where frequency rate is divided by two, leading to a doubling of the smallest possible impulse width ΔT and a doubling of performance density within a band width A - B. Where the frequency rate is changed the performance contained in a wave shape is not altered, but the performance within the spectrum is merely redistributed.

The pseudo-stochastic noise just explained is binary, i.e. a noise with only two amplitude values. With respect to frequency it contains practically all frequencies occuring in white noise when N values are sufficiently high, it does not have, however, the usual continuous form. Although white and binary noise demonstrate identical envelope of performance density spectrums, the image in time is totally different. The performance density is not sufficient for complete description of a noise. An expression of the amplitude characteristic in time must also be made. The amplitude characteristic is gained by means of low pass filters (compare 3.2.1.2). In order to estimate this /44
amplitude characteristic the proportional times between ascending and descending signal are added, occuring in an amplitude window δx during a measuring period, as shown in figure 29. The resulting curve is the probability density of momentary value of white noise amplitudes (figure 30), already presented in figure 26. The most usual probability density function for natural noise is the Gaussian or normal distribution

$$\boxed{p(x) = \frac{1}{\sigma\sqrt{2\pi}} \exp \frac{-x^2}{2\sigma^2}}$$

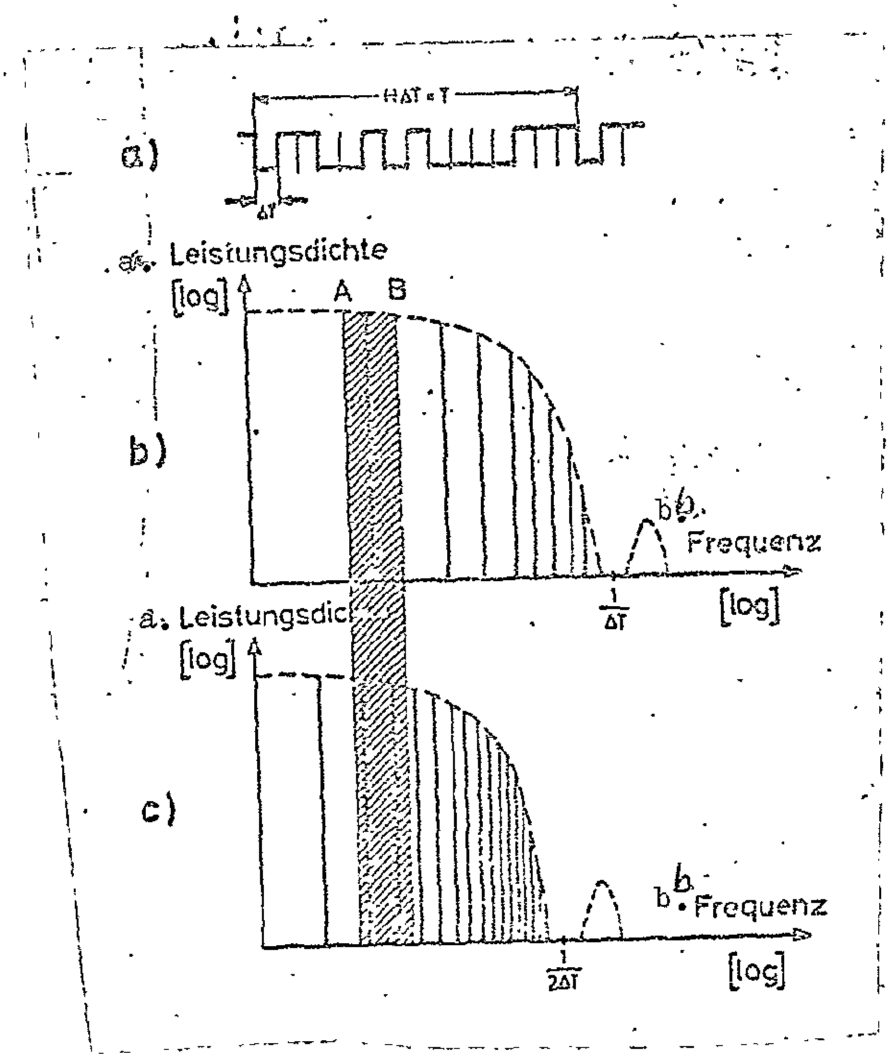

Figure 28: a) Pseudo-stochastic Binary Succession
b) Performance Spectrum of a Pseudo-stochastic Binary Succession
c) Performance Spectrum b) at Frequency Rate Divided by Two [33]

Key a. Performance density
b. Frequency

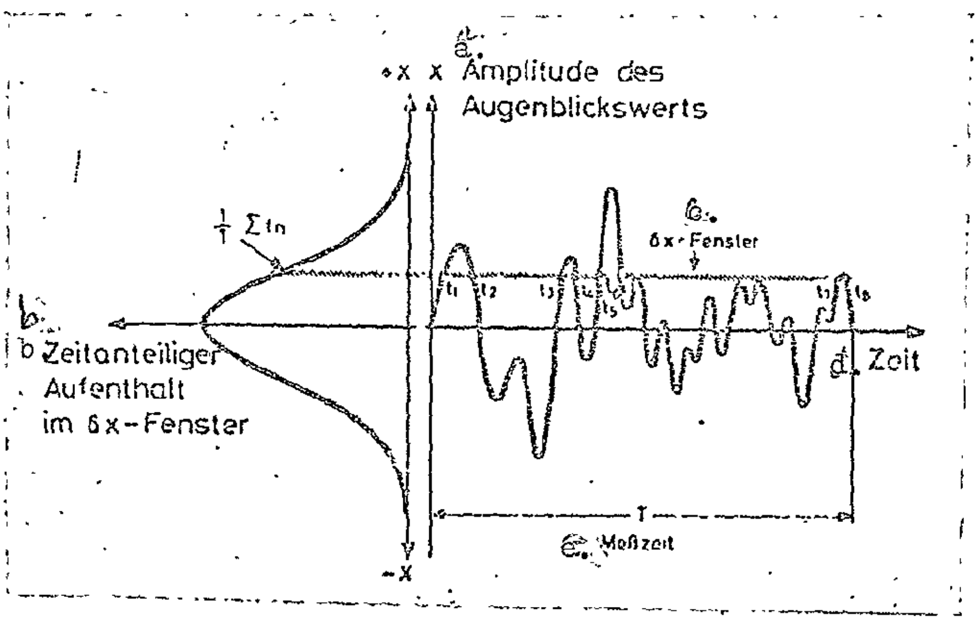

Figure 29: Probability Density Function of a Continuous Noise Signal [33]

Key a. Amplitude of momentary value
- b. Proportional time of stay in window
- c. Window
- d. Time
- e. Measuring time

The abcissa in figure 30 is not designated with 1, 2, 3, ... , as in figure 26, but rather with 1σ, 2σ, 3σ, etc. The quantity sigma of the amplitude axis indicates the standard deviation, a measure for the extension of amplitude values beyond this mean value. Sigma equals the effective value of the signal for a noise signal with mean value of zero.

The probability density distribution in figure 20 indicates that a signal remains between the amplitude values ± most of the time, and that values above the ± 3 hardly occur. It may be further seen that even values with infinitely high peaks may occur. Since the pseudo-stochastic noise with an amplitude continuum is gained via electronic filters, and even in the case of white noise electronic components are employed in generation, the "technical probability density function" deviates from normal distribution. This noise with a limited amplitude is plotted in figure 13 by hatching and shows that amplitudes may assume no value greater than ± 2.5σ.

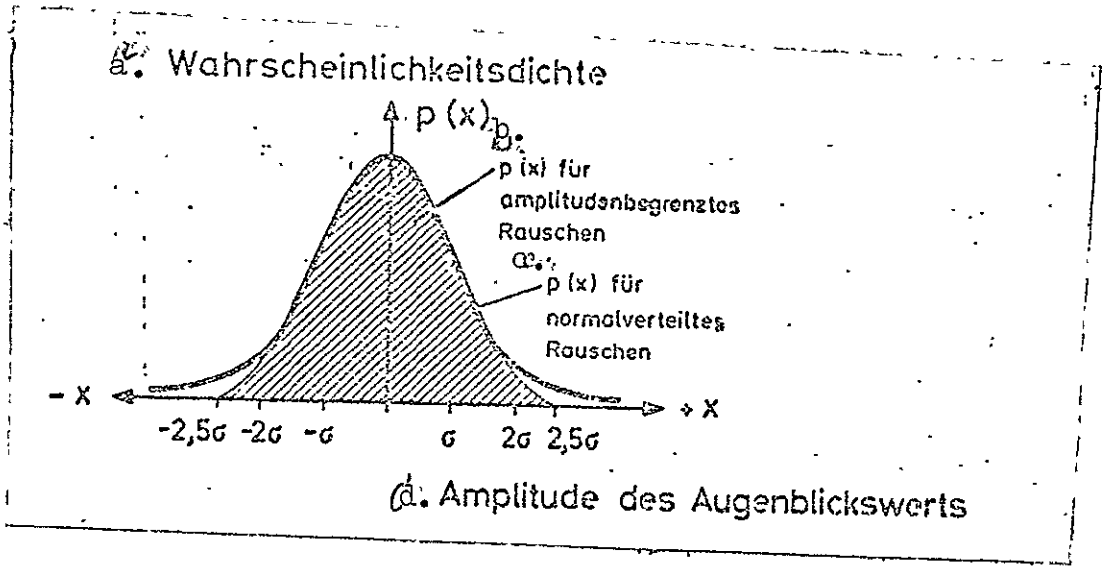

Figure 30: Probability Density Function of a Theoretical Noise with Normal Distribution and a Noise Created Technically with Amplitude Limits [33]

Key a. Probability density
b. p (x) for noise with amplitude limits
c. p (x) for noise of normal distribution
d. Amplitude of momentary value

3.2.2 Technical Synthesis of Sound Spectrums /46

Analytical results demonstrate that the saw-tooth voltage and the impulse-shaped rectangular voltage are best suited for generation of periodic spectrums due to the complete over wave content (see also 3.1.1). The square-wave voltage (duty cycle 1 : 1), which represents a special case of impulse-shaped square-wave voltage, is often utilized as periodic starting spectrum in spite of its incomplete over wave content. The reason given is that the generation of this voltage form involves simple technology and is inexpensive. Advantages of the two square-wave voltage forms are also found in further processing which may be carried out analogously and also in a digital form. Because of the relatively high amount of technology needed in generating triangular voltage, this is a voltage form which is only suitable for analogous sound synthesis technology. Similar considerations may be made for the non-periodic voltage form noise. White noise is applied exclusively in analogous synthesizing procedures, while the pseudo-stochastic noise may be further processed in a digital as well as an analogous manner.

3.2.2.1 Generation of Periodic Spectrum /47

The generation of sound spectrum is carried out with electronic components generally available on the market. For the synthesis of periodic waves voltage-controlled generators are normally applied today, so-called VCG (Voltage Controlled Generator). In Figure 31 a square-wave voltage of constant amplitude is indicated. Direct voltage at the imput controls the frequency in a range from approx. 1 : 10,000.

In a subsequent voltage controlled amplifier (VCA) the amplitudes of the square-wave voltage is amplified or reduced, to be subsequently mixed with other portions. At the exit of the VCA two diagrams are supplied as a further explanation. In the upper one the intensity density of rectangular wave is plotted against frequency. The amplitudes of the odd numbered over waves decrease at a constant rate. The lower loudness diagram shows that the subjective loudness of rectangular wave perceived by a human ear does not decrease with frequency, but is perceived as louder for higher frequencies than for middle and low frequencies (see also section 2.2). An example for periodic sounds constructed in this manner is the synthesis for the howling of runners in an engine.

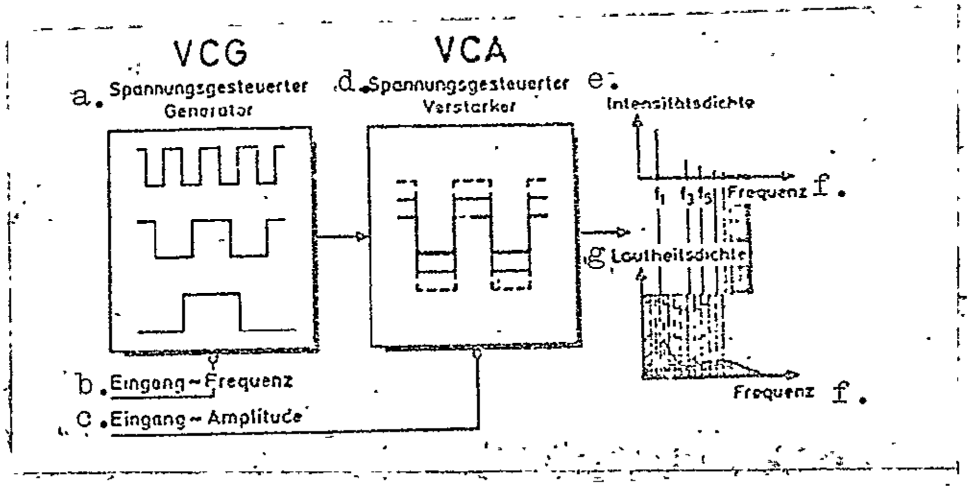

Figure 31: Components for Synthesis of a Periodic Sound Spectrum

Key a. Voltage Controlled Generator
b. Input frequency
c. Input amplitude
d. Voltage controlled amplifier
e. Intensity density
f. Frequency
g. Loudness density

In the following several technical designs of voltage controlled /48 generators (VCG) will be briefly explained. Because of rapid development of these components, representing a profitable market due to construction of electronic organs and pianos [14, 15, 16, 17], several components will be used as examples.

The Universal Analog VCO [18]

The switching circuits XR-2207 and XR-2307 produce square-wave, triangular, saw-tooth or impulse-shaped square-wave voltage according to circuit. The frequency range covers 0.01 Hz to 1 MHz. The switching circuit is supplied in a fourteen pin dual-inline-package. The oscillation range selected may be altered via direct voltage in the ratio 1000 : 1 (VCO operation).

Digital Saw-Tooth or Square-Wave VCO [16]

The switching circuit SAJ 205 is designed in MOS technology and supplies a saw-tooth and a square-wave voltage. The component is to be regulated via a voltage frequency converter or a serial digital signal. A divider chain divides the input frequency f binary up to f/256. The saw-tooth voltage, which is actually a step voltage, results from the summation of square-wave voltages achieved via division in a special digital analog converter. The component is supplied in a dual-inline-package. The oscillation range may be regulated in the range of the ten audible octaves.

Digital Impulse-Shaped Rectangular VCO [19]

The switching circuit XR-2340 is available as dual-inline component. The inputs and outputs are compatible with PTL. In this component the impulse width T for a given frequency 1/T (see figure 24a) may be regulated in the range of $1 \Delta T$ smaller than or equal impulse width smaller than or equal $255 \Delta T$ where $T = 256 \Delta T$. For this control an 8 bit serial-parallel counter is necessary.

3.2.2.2 Generation of Non-Periodic Spectrum /49

Generally noise generator, so-called WNG (white noise generator) are applied to the generation of non periodic spectrum, as demonstrated in figure 32.

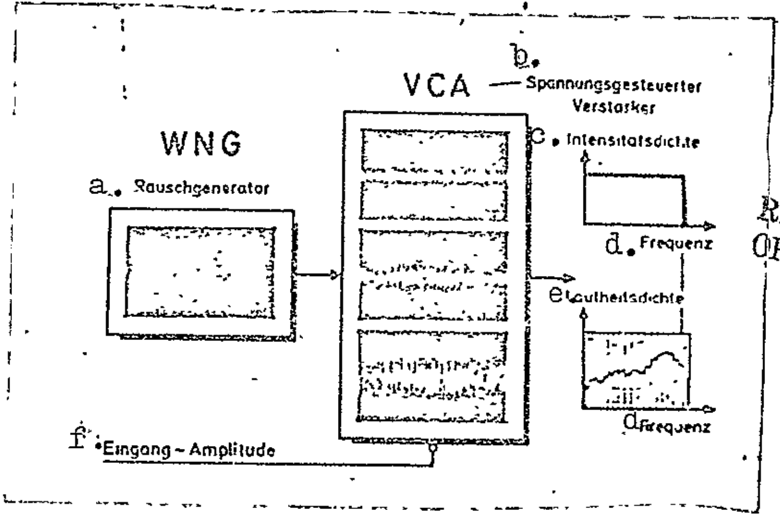

Figure 32: Components for Synthesis of a Non-Periodic Sound Spectrum

Key a. Noise generator b. Voltage controlled amplifier
c. Intensity density d. Frequency e. Loudness density
f. Input amplitude

In addition to these noise generators which generate events completely without rules, the pseudo-stochastic noise generators are of equal importance, only apparently generating events without rules (section 3.2.1.2). These events are voltage courses in figure 32. In the subsequent VCA the amplitude may be increased or decreased via the control input. Intensity density remains constant with frequency up to a defined limit frequency. The loudness diagram also shows a loudness of spectrum differing from that perceived by the ear, with the increase of loudness for higher frequencies being specially clear. In the following the technical design for generation of both noise spectrums is explained.

White Noise

As a source for white noise the unordered heat motion of electrons in a resistance wire may be utilized, already resulting at room temperature in a noise voltage of several micro-V with a resistance of 100 kΩ at a band width of 20 kHz. The self-generated noise of a diode, tubes, /50
transistors, etc. may be further used.

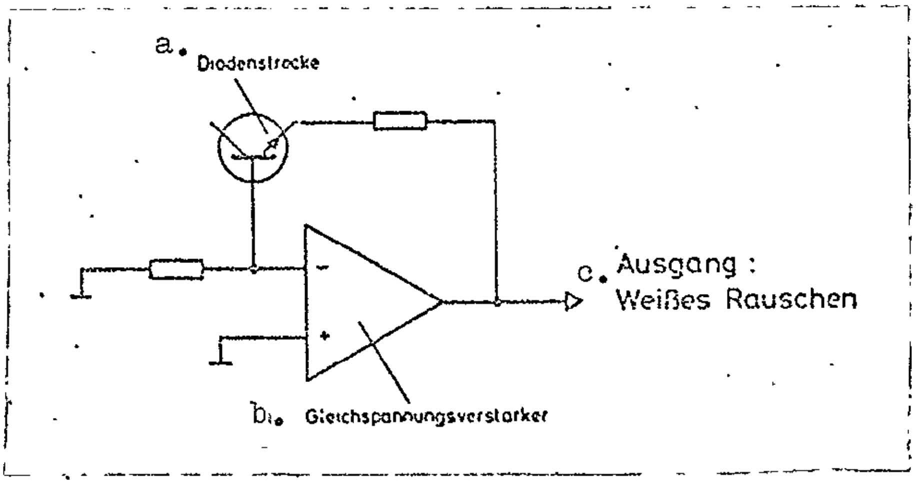

Figure 33: Circuit Arrangement for Generation of White Noise

Key a. Diode path
b. Direct current amplifier
c. Output: White Noise

Figure 33 shows a circuit arrangement for generation of white noise [20]. The diode path of a transistor is connected to the feed back half of a direct current amplifier. Introduction of a lower limit frequency serves for the protection of the moving-coil of the loud speaker radiating the spectrum. Upper limit frequencies are determined for differing sound sources. The limit frequency for rolling, for example, is substantially lower than that for the wind sound of an aircraft.

Pseudo-stochastic Noise

With the aid of digital information technology [13] a pseudo-stochastic noise may be generated. When in this case shifting

registers are fed-back in a suitable manner, a binary succession of signals with periods of great lengths results, demonstrating almost the identical coincidental event characteristic as binary stochastic processes [21, 22]. Figure 34 shows a circuit for generation of a logical coincidental series with a multiple-step shifting register. After N register steps the selection of tapping i again reaches the starting condition after 2^n cycles, for example, in the case of ten steps after approx. 1000 cycles. The long period exceeds memory capability of man, creating the impression of a random event sequence /51
not governed by any rules. When this sequence is converted[1] by means of a time-amplitude transformation (low pass filter) into a time function with Gaussian normal distribution, at the exit an analog signal appears, also only apparently not govern by any rules, but which sufficiently demonstrate statistic characteristics of desired white wide band noise when closely examined (see also figure 30).

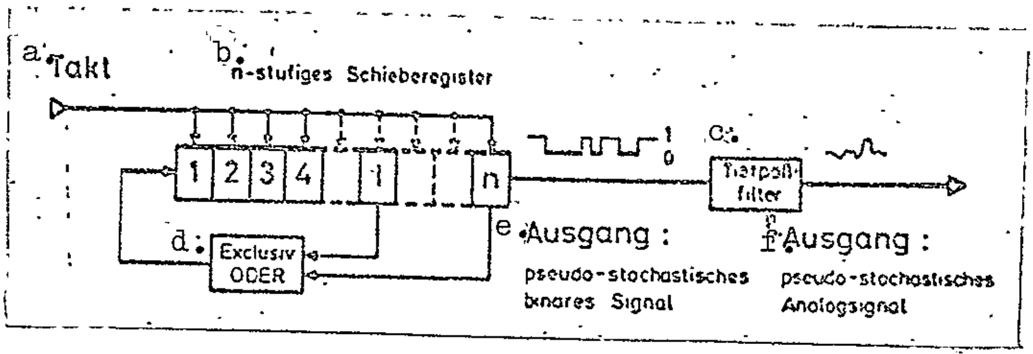

Figure 34: Generation of a Pseudo-stochastic Random Sequence [22]

Key a. Cycle b. Shifting register width n steps c. Low pass filter
d. Exclusive OR e. Exit:pseudo-stochastic binary signal
f. Exit: pseudo-stochastic analog signal

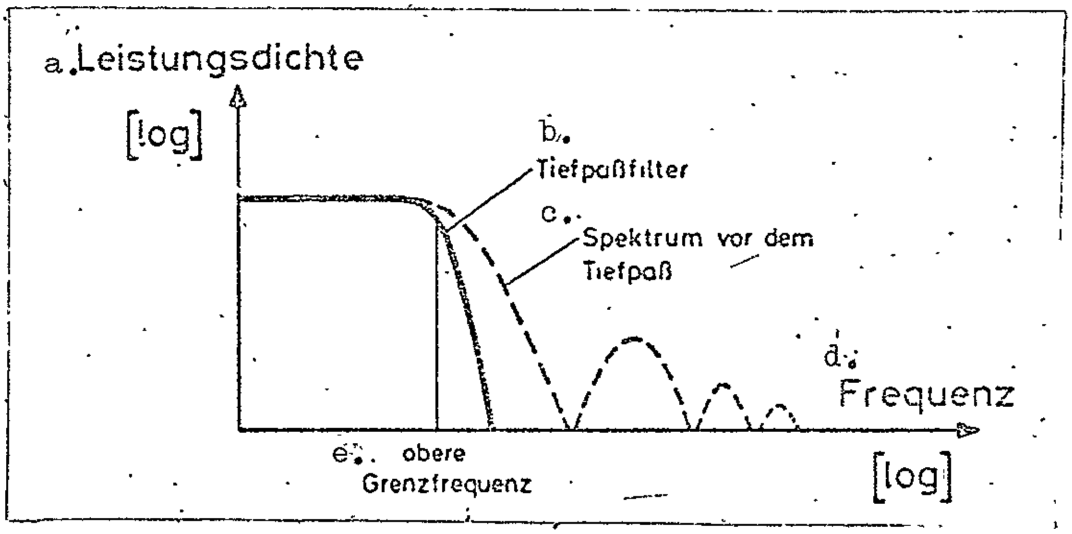

Figure 35: Performance Spectrum of Pseudo-stochastic Noise

Key a. Performance density b. Low pass filter c. Spectrum previous to low pass d. Frequency e. Upper limit frequency

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1) According to Giloi [13] only an approximation of this transformation is possible.

Figure 35 shows the performance spectrum of pseudo-stochastic noise, gained by means of low pass filtering of binary signal and demonstrating with sufficient exactness the behavior of white noise. /52 The corresponding loudness diagram (loudness density/frequency diagram) is given in figure 32.

3.2.3 Procedure of Sound Synthesis

The technical procedure for synthesis of noise applied today, i.e. supplying the simulator pilot with the normal three dimensional sound field, may be carried out through three different processes. These are in the order of increasing complexity

The traditional procedure
The poly-voice-procedure
The digital procedure,

represented in figure 36. In this case the mathematical design of sound synthesis shifts from pure hardware to software. Estimation of cost involved depends on the aim of total simulation of the vehicle and is explained in the following. Only synthesis of aircraft noise is reported, since the complexity of these vehicles justify the high costs of sound simulation in relation to the total cost of the simulation device. The methods to be presented are also applicable to land and ocean-going vehicles using the corresponding mathematical formulation [23, 24].

3.2.3.1 Traditional Procedure

In the case of the traditional procedure [2, 3, 25] quantities of aviation mechanics such as a velocity, altitude, engine speed etc. directly control sound simulation, completely designed in hardware (figure 37). In this case the noise portions "air", "engine" and "ground", arising in an actual aircraft with the surrounding air, through changes in engine speed and via ground contact, are reproduced in hardware as analagous physical images. In the subsequent synthesis mainly voltage-controlled generators (VCG) are applied for periodic waves such as engine howling while white noise generators (WNG) are applied for non- /54 periodic waves such as wind noises. The amplitudes of these noise portions are evaluated via voltage-controlled amplifiers (VCA).

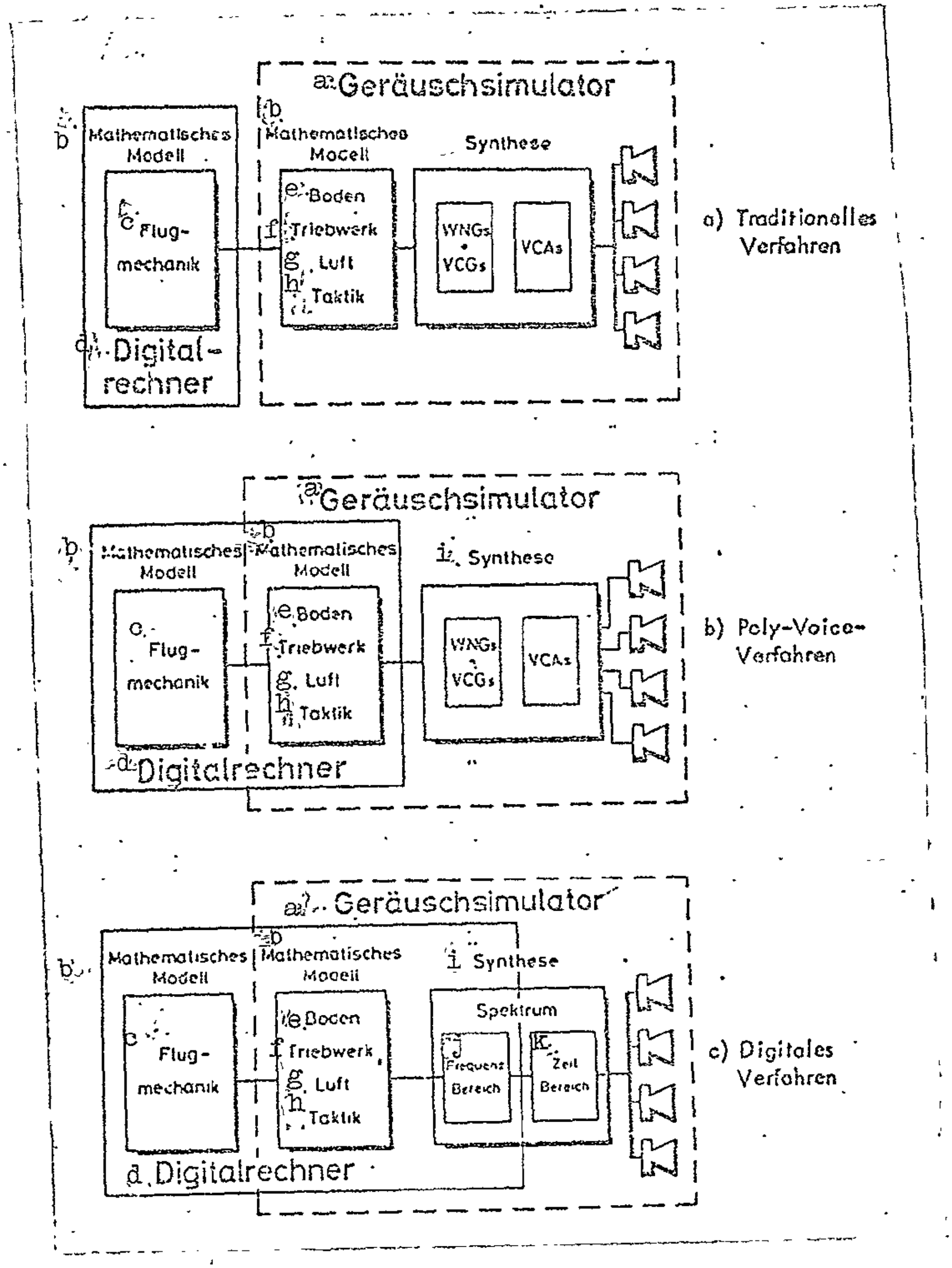


Figure 36: Survey on the Procedures of Sound Synthesis

Key a. Sound simulator b. Mathematical model c. Aviation Mechanics
d. Digital computer e. Ground f. Engine g. Air
h. Tactics i. Synthesis j. Frequency range k. Time range

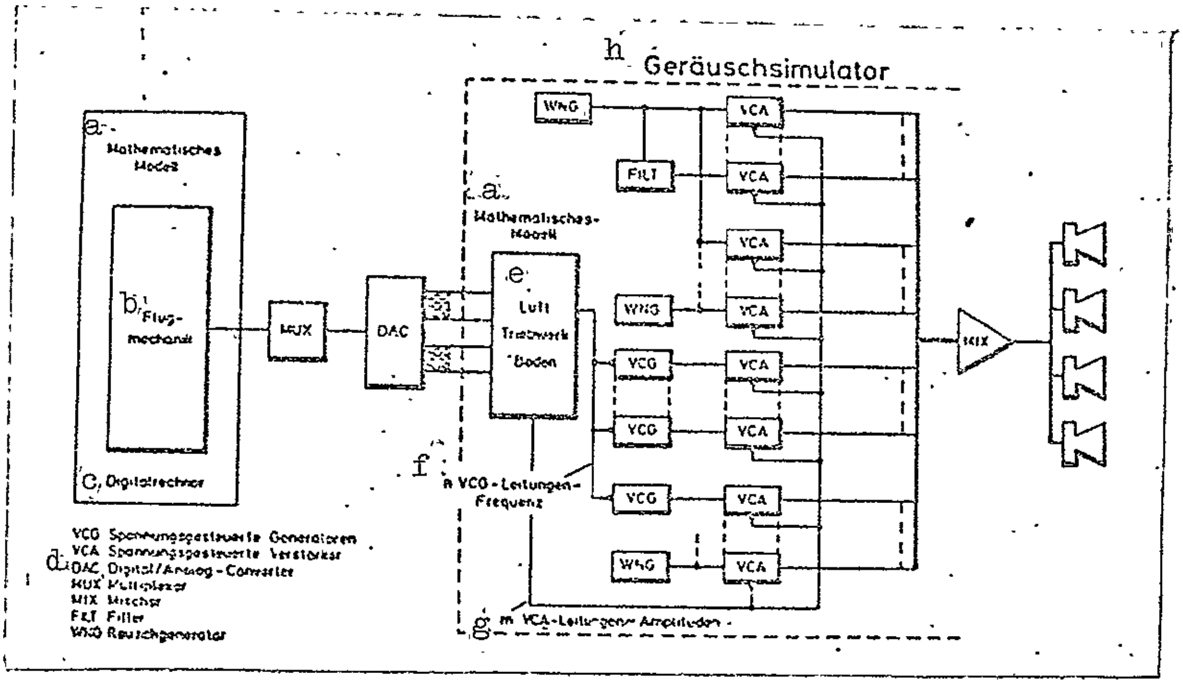


Figure 37: Block Circuit Diagram of a Traditional Sound Simulator using the Example of the Aircraft

Key a. Mathematical model
b. Aviation mechanics
c. Digital computer
d. VCG voltage controlled generators
VCA voltage controlled amplifier
DAC digital/analog converter
MUX multiplexor
MIX mixer
FILT filter
WNG white noise generator
e. Air
Engine
Ground
f. VCG lines-frequency
g. VCA lines-amplitudes
h. Sound simulator

The hardware simulation of the sub-noises air, engine and ground is shown in figure 37. The generation of air sounds as a result of air hitting the cockpit and of landing gear sounds audible in the cockpit is carried out by means of noise generators and filters, which may be altered by means VCA according to velocity, altitude, etc. of the simulated aircraft. In the case of engine simulation the VCG, controlled by the quantity rotations per minute (RPM), generate the engine howling and the noise generator generates the air intake and exhaust noise. When rolling on the gound is simulated the VCA evaluated noise generator supplies the bumping roll noise and the noise generator is rated by VCG suppling the tire noises. In figure 37 the possibility for sound synthesis variety is shown.

Since the design of the "traditional sound simulation" in computer-technological component groups of analog operation is physically analogous to the actual sound generation, design, testing and error search cause no difficulty. This approach provides a high degree of operational reliability and it is therefore not surprising that this procedure has been employed in almost all flight simulators. A further advantage lies in the relatively low costs. /55

This technique, however, carries considerable disadvantages for commercial manufacturing. On the one hand the hardware necessary increases in a linear manner with the number of different sub-noises, on the other hand a new aircraft to be simulated with all its detail characteristics requires in each case a new design of the corresponding circuits. The modular standard system of the SINGER Company [26] provides a way out of this difficulty, consisting of approx. ten standardized punch cards as VCG, VCA, WNG, mixer etc. Due to the standardization a close meshing of circuits is no longer possible, i.e. the required hardware further increases. This fact is a reason leading to the development of an improved synthesis principle, the poly-voice system [27]. As can be seen from figure 37 the one-channel sound is given over four (26) loud speakers, generating a sufficient spacial sound field in the cockpit for development and research simulators. This concept, however, is not sufficient for training simulators, since engine failures must also be simulated acoustically in the case of multiple-engine machines. The necessary hardware increases further, since the loud speakers must be regulated over several channels. This requirement may also be met with less cost through the poly-voice system. SINGER provides the hardware requirement for the sound synthesis conducted commercially with the aid of the modular standard system with a 19 inch plug-in unit as well as approx. 30 digital/analog converters and 20 one-bit control lines [28]. The technological requirement for a less expensive simulator of traditional design due to simplified demands will be discussed using the example of an HFB-320.

3.2.3.2 Poly-Voice Procedure /56

In contrast to the traditional procedure where flight mechanics factors control hardware sound simulation, in the poly-voice system [27, 29] simulation is divided into a software and a hardware portion, as shown in figure 28. Parameters of aviation mechanics effect mathematical models as "ground", "engine", "air" etc., which in turn regulate the hardware. Actual model design is carried out in the digital computer and the quality of simulated dependency of sound sources to be reproduced is only limited by the computer. The turbine noise, for example, is constructed as a model in software and regulated by aviation mechanics by the factor rotations per minute (RPM). Software includes, for example, simulation of the first and the second compressor, the turbine, the intake air and the exhaust. Each factor is calculated here as partial wave and regulates the VCG according to frequency and the VCA according to amplitude. At least five times two or ten factors are processed in a digital/analog converter, in contrast to the traditional procedure where one factor (RPM) is converted.

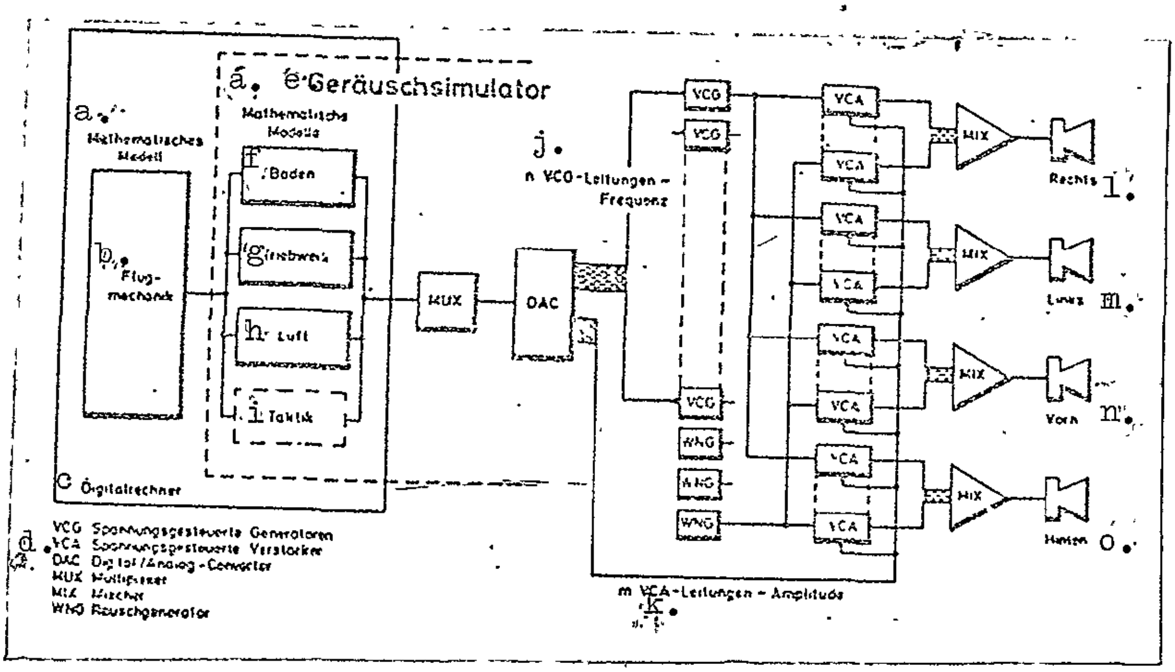


Figure 38: Block Diagram of the Poly-Voice Procedure using the Example of an Aircraft

Key a. Mathematical model
b. Aviation mechanics
c. Digital computer
d. VCG voltage controlled generators
VCA voltage controlled amplifier
DAC digital/analog converter
MUX multiplexer
MIX mixer
WNG white noise generator
e. Sound simulator
f. Ground
g. Engine
h. Air
i. Tactics
j. VCG lines-frequency
k. VCA lines-amplitude
l. Right side
m. Left side
n. Front
o. Back

The necessary hardware seems to be as extensive as that of the traditional procedure in spite of the shift of the model design into software. This conclusion may apply when a single unit is produced but for commercial manufacturing this procedure provides considerable advantages. Programs of the mathematical model carried out once in the expensive software may be correspondingly converted with little cost for varying aircraft types. The hardware is of modular design from standardized components. Component number is kept at a minimum due to design possibilities contained in the poly-voice procedure. It is possible, for example, to utilize the hardware components several times for the synthesis of such sounds which do not occur simultaneously, as landing gear operations, tire squealing during landing, taxiing sounds on the ground, etc. /57

Although the poly-voice system may be designed quadrophonically, due to the possibility of multiple hardware utilization no genuine four channel concept may be applied. The possibilities in this system for directional hearing in the cockpit as sound stimuli from the right, left, front and back is only simulated for the sound sources for which this effect is logical, or where this effect is desired by the customer.

The simulator design has disadvantages for test and service personnel, who have difficulty in understanding the operational manner of the cost-minimized sound synthesis. SINGER [28] indicates that the required hardware for poly-voice synthesis is one-half of a 19" push-in unit as well as more than fourty digital/analog converters.

3.2.3.3 Digital Procedure

The digital procedure is based on the synthesis of wave shapes dependent on time in the time plane [5]. This concept demonstrates a complete departure from traditional procedures as shown in figure 39. The entire sound simulation is carried out by a digital computer in the frequency range and controlled by corresponding factors of flight mechanics. In comparison to the poly-voice procedure (figure 38) it may be seen that the starting quantities of the mathematical model do not regulate the hardware, but instead are grouped in four bands corresponding to acoustic frequencies. These frequency bands are /58
processed through a rapid inverse fourier transformation (SFT) [4], in order to obtain the corresponding time functions. The actual synthesis is carried out in a digital hardware synthesizer [5]. The digital sound spectrum constructed in this manner is fed through a digital/analog converter and after filtering of the digital sounds conducted to the loud speaker groups. A quadrophonic effect may be produced in a simple manner in the cockpit by means of a multiplexer between the synthesizer and the digital/analog converter.

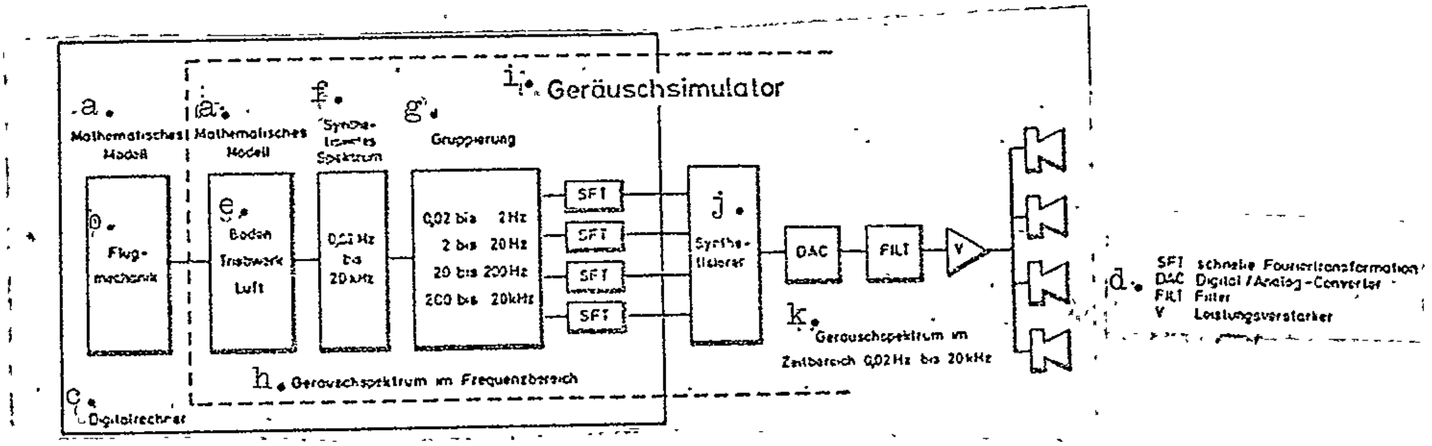


Figure 39: Block Diagram of a Future Digital Sound Simulator using the Example of an Aircraft [5]

Key a. Mathematical model
b. Aviation mechanics
c. Digital computer
d. SFT rapid fourier transformation
DAC digital/analog converter
FILT filter
V performance amplifier
e. Ground, engine, air
f. Synthesized spectrum
g. Grouping
h. Sound spectrum frequency range
i. Sound simulator
j. Synthesizer
k. Sound spectrum in the time range 0.02 Hz to 20 kHz

The advantage of this procedure lies in the reduced hardware requirement. The necessary software, however, is very complicated, with financial demands on programming which only a simulator manufacturer may meet. Sound synthesis requires the application of high velocity small calculators already available today and of special computers for the processing of the rapid fourier transformation.

3.3 Sound Presentation

The sound is presented via transmission systems for converting electrical into acoustical signals. This conversion involves special /59 difficulties. Any hi-fi fan can confirm this fact, who has ever chosen loud speakers for his stereo. The designer of a sound simulator finds himself in the same situation. He must be very selective about the transmission systems required and their arrangement in the cockpit of the simulator.

The selection of transmission system limits the quality of the radiated sound. Generally loud speaker systems commercially available are employed. Trials with sound transducer have recently been conducted, which have an advantage due to their small size that they may be positioned anywhere in the cockpit, as in the control elements, seats, panelling, indicator panels etc. They utilize the mechanical background on which they are mounted as resonance bodies. Trials of our own and some of which have been published [30] show, however, that these transducers are difficult to position, since in some portions of the cockpit undesirable resonance waves are produced by the transducer at the required frequency range of 9 to 10 octaves, which then considerably reduce quality of simulation.

The aim of generating a three dimensional sound field in the cockpits may only be achieved by means of a larger number of loud speakers of high quality. For training simulators sterophonic systems are required because of the possibility of simulation of engine failures. The poly-voice system is operated with a quadrophonic system of hi-fi quality. For information originating from the right-left-front-back four sound sources are perceived as average, eight as excellent [27]. A sufficient compromise is found in arrangement of six loud speakers.

3.4 Sounds to be Reproduced

The following sounds are reproduced in commercial flight simulators:

Sounds during starting and landing:
- taxiing sounds of landing gear
- tire squealing during landing
- air brakes
- engaging and disengaging landing gear including locking /60
- braking parachute

Engine noises: Ignition
Turbine howling
Air intake at the condenser
Exhaust
Thrust conversion
After burner
Engine failure during flight
Slow disruption in the compresser

Air Sounds: Wind as a function of altitude, velocity, position of the air brakes, landing flaps, leading wing edge and landing gear
Rain and hail
Transition from sub-sonic to super-sonic flight
Super-sonic flight

Sounds of Aggregates on Board; Starting and Operation of:
Pressurized ventilation system
Fresh air supplies
AC and DC voltage generators
Auxiliary aggregates
Hydraulic positioning members

(Dog Fight Sounds)

The simulation of dog fight sounds, for example machine gun fire or the firing of guided missiles is also practiced.

4. The HFB-320 Sound Simulator /61

In the Research Institute for Human Engineering a flexible flight simulator was designed for research projects employing many subjects of examination. The executive HFB-320 Hansa jet (figure 40) was constructed as fixed-seat simulator without arrangements for outside viewing.

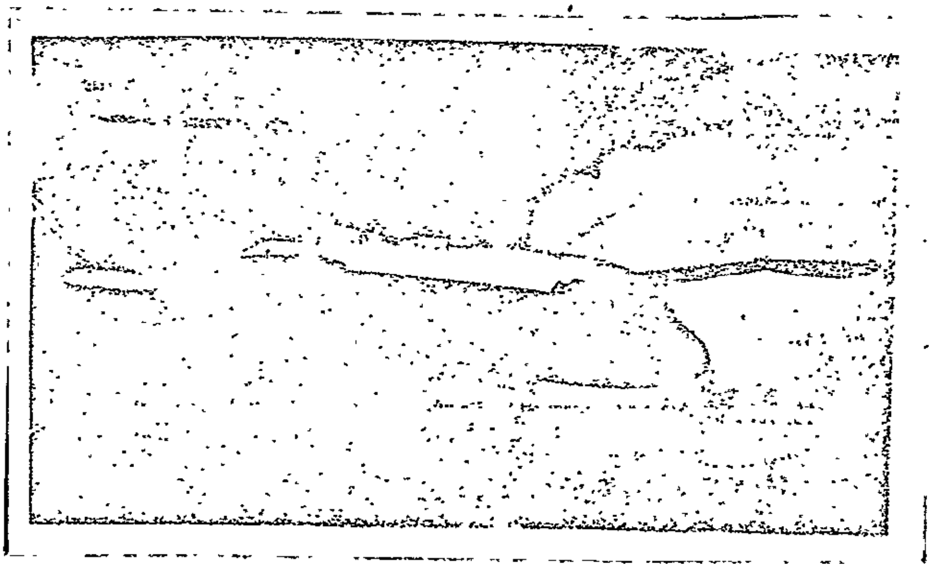


Figure 40: Aircraft HFB-320 Hansa Jet

4.1 Analysis of Individual Sound Portions

Selection of individual sub-spectrums, the summation of which result in cockpit sound, is carried out with the aid of a loudness analysator as well as through subjective evaluation. The tape recording made during actual flights of the interesting flight phases of starting, flight and landing were objectively analyzed by evaluation of corresponding loudness diagram. A comparison of actual flight and recorded sounds carried out subjectively has been embodied into the total analysis (section 3.1).

4.1.1 Subjective Analysis /62

Evaluation of "subjective judgement" supplies the subjective "audibility of individual sound portions" as a function time in relation to the evaluated flight phases of starting, flight and landing.

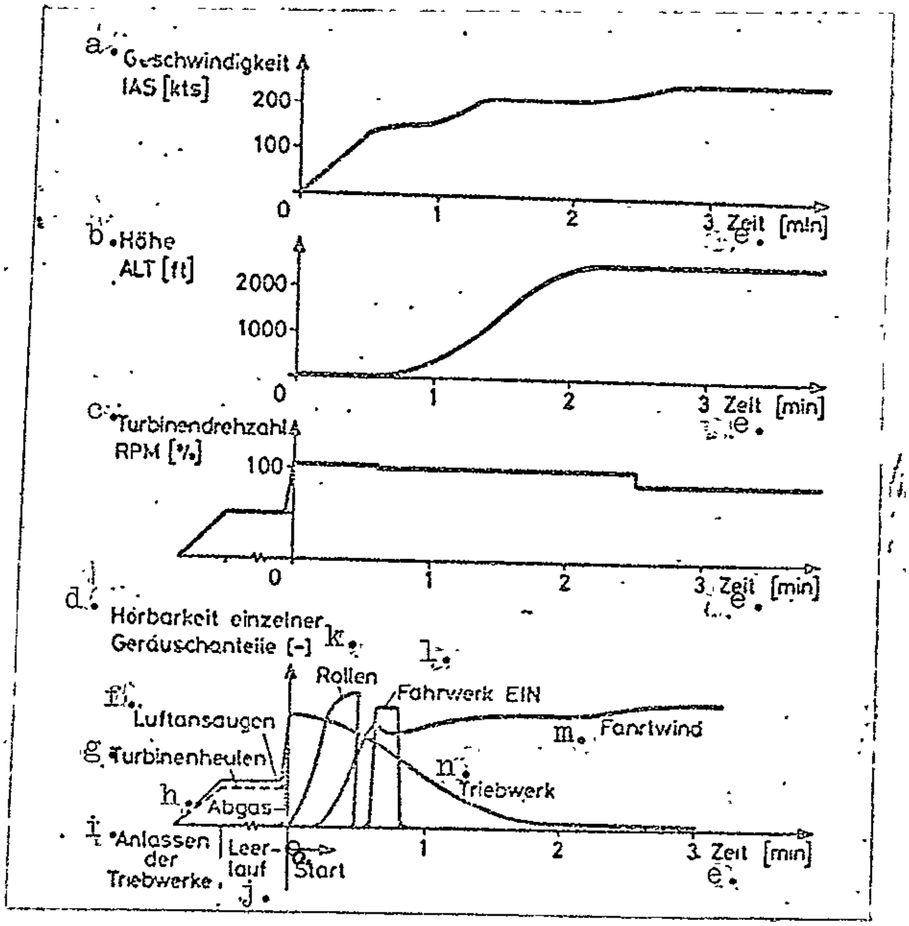


Figure 41: Audibility of Individual Sound Portions during Starting

Key a. Velocity
b. Altitude
c. Engine RPM
d. Audibility of individual sound portions
e. Time
f. Air intake
g. Engine howling
h. Exhaust
i. Engine ignition
j. Neutral
k. Taxiing
l. Landing gear
m. Wind
n. Engine
o. Take-off

The figure 41 is an example for take-off analysis. For simulation of cockpit sound of the HFB 320 only those sounds were considered necessary originating from

1. Engine
2. Wind
3. Taxiing
4. Landing gear /63
5. Landing

The subjective analysis of take-off is to be clarified using several figures as examples.

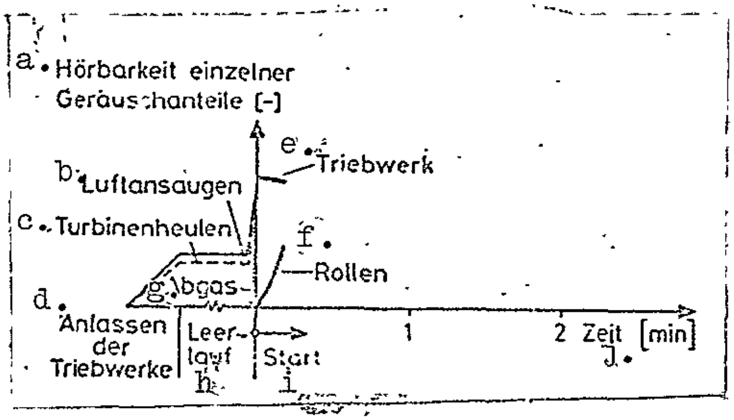


Figure 42: Audibility of Individual Sound Portions in an Aircraft on the Ground where RPM Range of the Engine is 0 - 100%

Key a. Audibility of individual sound portions
b. Air intake
c. Engine howling
d. Engine ignition
e. Engine
f. Taxiing
g. Exhaust
h. Neutral
i. Take-off
j. Time

When the engines are ignited, see figure 42, the volume of turbine howling and of air intake noise increases proportionally to number of revolutions. This is valid for the entire RPM range of 0 - 100%. The exhaust noise is not audible until RPM exceeds 50%, but then it is very loud and is more audible in the upper RPM range than the other sound portions of the engine. As time t = 0 at an RPM of 100% the engine volume reaches its maximum. For the further course of the diagram the three partial sounds are not considered separated, but are plotted as sound portion "engine". With increasing velocity the audibility of this sound portion is continually reduced and can hardly be perceived at volumes exceeding 200 knots. Taxiing noise may be heard immediately after taxiing begins (transition from figure 42 to 43) and at a velocity of 100 knots already exceeds (figure 43) the engine volume. It becomes clear in figure 43 that already after approx. 20 seconds the rolling sound portion dominates in audibility due to the continuous slow reduction in sound portion "engine" and the rapid increase in rolling sound. Until lift-off speed (approx. 130 knots) is reached, however, only a small increase in rolling noise is perceived. /64

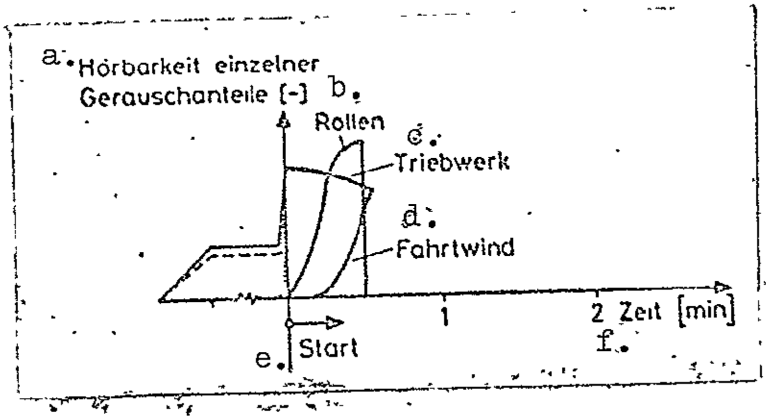


Figure 43: Audibility of Individual Sound Portions During the Rolling Phase of Aircraft Take-off

Key a. Audibility of Individual Sound Portions
- b. Rolling
- c. Engine
- d. Wind
- e. Take-off
- f. Time

Commencing with a velocity of approx. 60 knots wind sounds may be heard (figure 43), achieving the volume of engine sound after take-off, i.e. after the rolling noise is abruptly discontinued (figure 44).

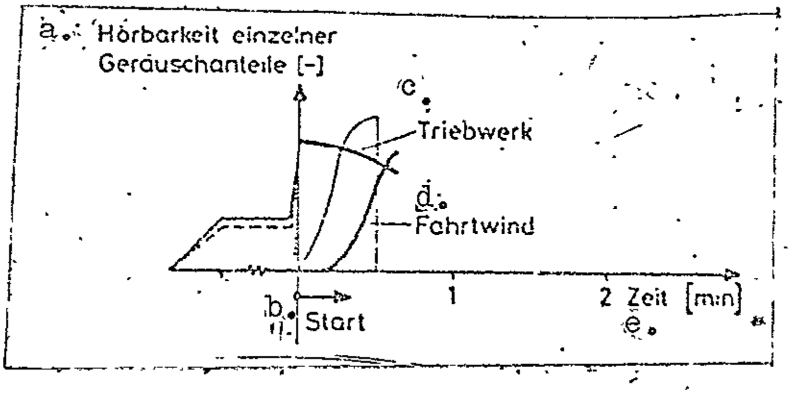


Figure 44: Audibility of Individual Sound Portions During the Take-off of an Aircraft Shortly after Lift-off

Key a. Audibility of individual sound portions b. Take-off
c. Engine d. Landing gear e. Time

It may be recognized here that the sound portion "wind" further increases with further velocity increases of the aircraft, while the sound portion "engine" decreases, resulting in the impression of a very strong wind noise even before the landing gear is disengaged. The sound of the moving landing gear (figure 45) is of constant volume while it is being retracted (approx. 12 seconds) and is /65 more audible than the wind, with an audibility which has been reduced after landing retraction due to the reduced air resistance. After landing gear has been locked and a further velocity increase the audibility of engine is very low compared to wind noise (figure 46).

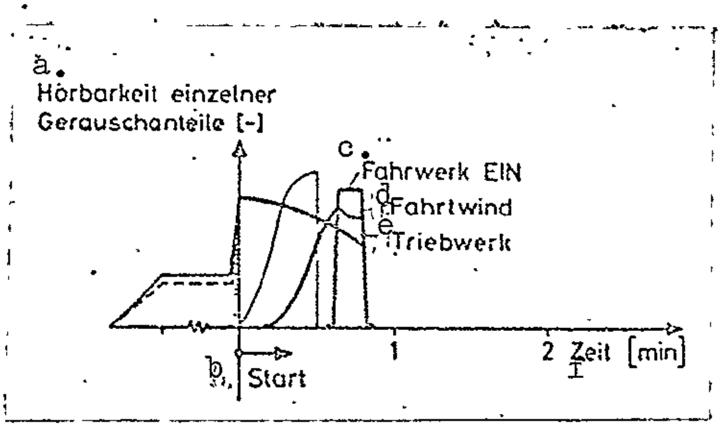


Figure 45: Audibility of Individual Sound Portions During Take-off of an Aircraft when Landing Gear is Retracted

Key a. Audibility of individual sound portions b. Take-off
c. Landing gear retracted d. Wind e. Engine f. Time

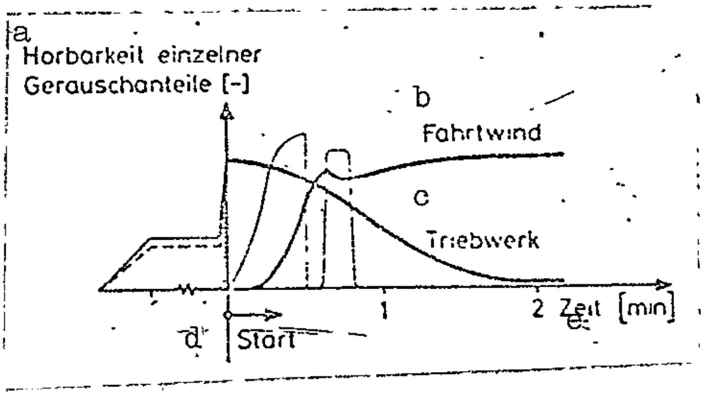


Figure 46: Audibility of Individual Sound Portions of an Aircraft During Flight

Key a. Audibility of individual sound portions b. Wind c. Engine
d. Take-off e. Time

During flight only wind sounds and/or engine sounds may be heard according to velocity. Audibility of the wind is influenced only to a small extent by velocity during flight.

/66

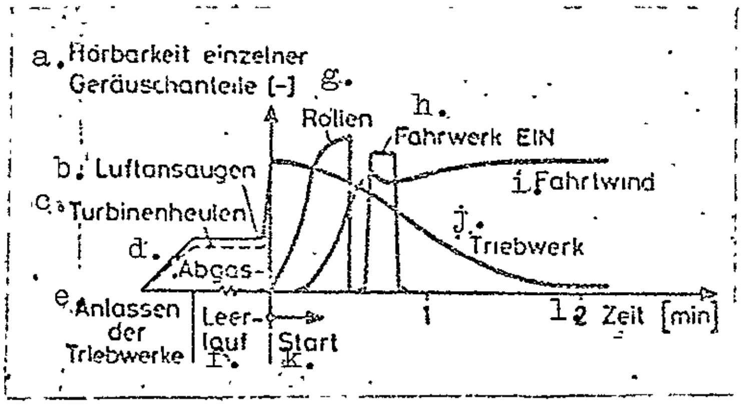


Figure 47: Audibility of Individual Sound Portions in the Cockpit of an Aircraft During Take-off (here HFB-320)

Key a. Audibility of individual sound portions b. Air intake
c. Engine howling d. Exhaust e. Engine ignition
f. Idleling g. Rolling h. Landing gear retracted
i. Wind j. Engine k. Take-off l. Time

Figure 47 represents the audibility of individual sound portions in its entirety. Corresponding to the time sequence of a take-off ignition of the engine, air intake engine sound portion, turbine howling, and exhaust are combined to a total engine sound and plotted, and the sequence of rolling and landing gear noise as well as the reduction in engine noise and the decrease in wind is apparent.

4.1.2 Objective Analysis

As already described in section 3.1.1 the objective analysis is understood as the recording of the amplitude-frequency spectrum changing with time. For analysis of the aircraft sounds examined here the Hewlett Packard loudness analysator 8051 A was employed. This analysator generates a new loudness diagram every 25 msec. and is therefore suitable for recording stationary and non-stationary sounds. The recording of a stationary spectrum requires approx. 75 sec. measuring time (figure 48a), i.e. only spectrums may be recorded with an amplitude distribution which does not alter greatly during this time, as in the flight already described. In this figure amplitude changes in time within the 20 frequency groups are easily recognized. If the recording of a non-stationary sound is required, such as a take-off, a much shorter time span, here approx. 25 msec., is employed by the analysator and is assumed to be /67 quasi-stationary. An example of the resulting recording is shown in figure 48b. This method of recording is of special importance for the take-off to be analyzed here because the "stationary measuring time" of 75 seconds assumes the value of the entire take-off time. During this time a large number differing amplitude spectrums are covered, which would be useless if represented in a loudness diagram.

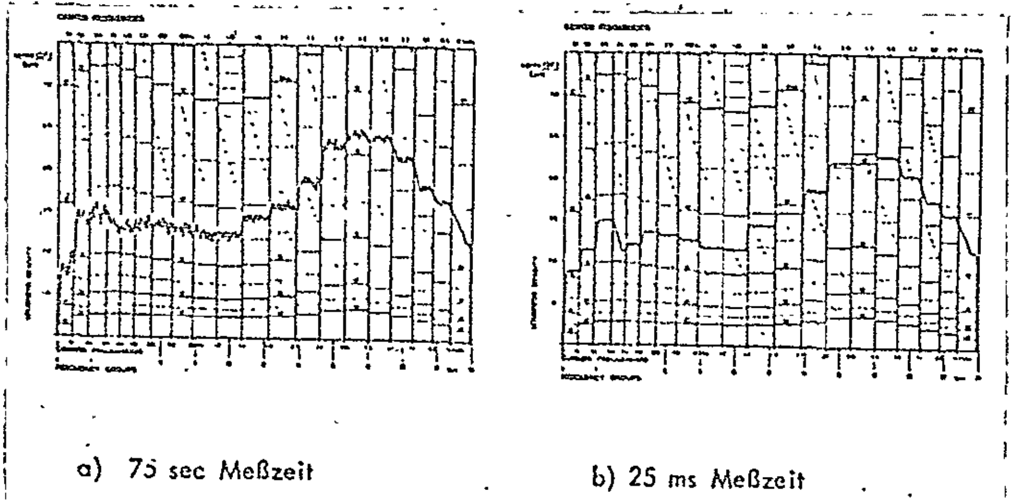


Figure 48: Loudness Diagram for a Flight
Here - HFB 320, 85% Thrust, 20,000 foot Altitude

Key a. 75 seconds measuring time
b. 25 msec. measuring time

The tape recordings were made during a regular flight of HFB 320 machine and contain the radio messages necessary for carrying out the flight. For this reason the measuring mode 25 msec. was selected for analysis, since it is possible to avoid disturbances of the flight sound spectrum changing constantly with this short measuring time and to obtain a sequence of loudness diagrams without gaps.

The following five pages show five loudness diagrams recorded in the HFB 320 compared to the diagrams synthesized in this simulator. In section 4.2 the generation of these spectrums will be further explained. A good agreement quality may be seen in these figures, also confirmed by HFB pilots by means of auditive subjective comparison. The figure sequence demonstrates the sound phases idleling (figure 49), stationary 100% thrust (figure 50), rolling at 50 knots (figure 51), rolling at 120 knots (figure 52) and flight with 85% thrust, 20,000 feet (figure 53).

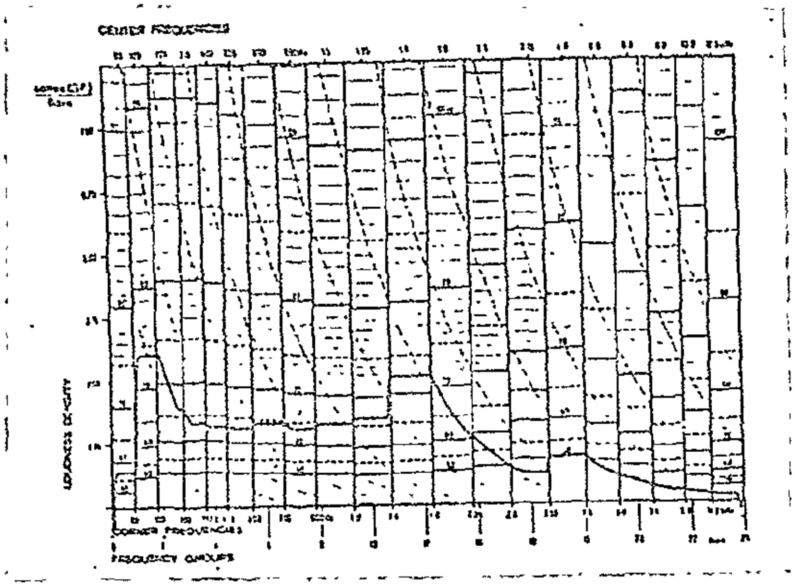


Figure 49a: Loudness Diagram of the Sound Spectrum "Idling" Recorded in the Actual Aircraft HFB 320

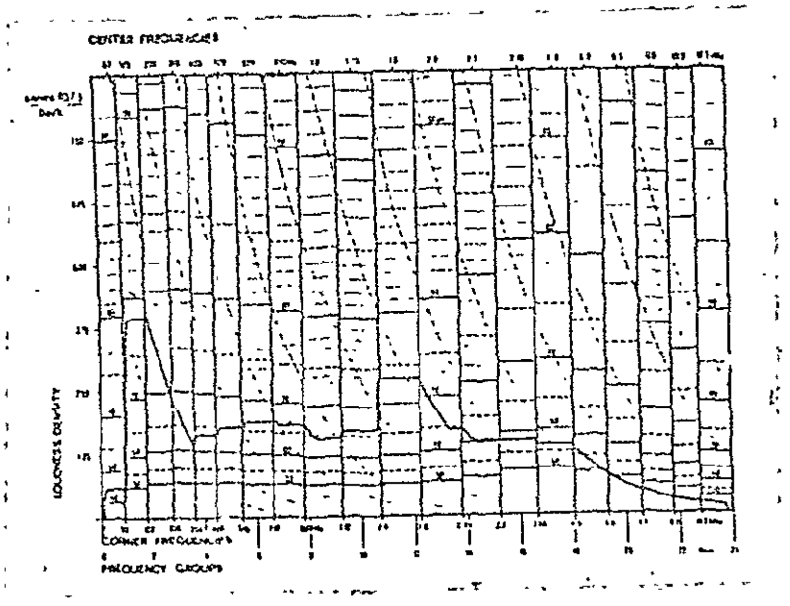


Figure 49b: Loudness Diagram of the Sound Spectrum "Idling" Recorded in the Flight Simulator HFB 320

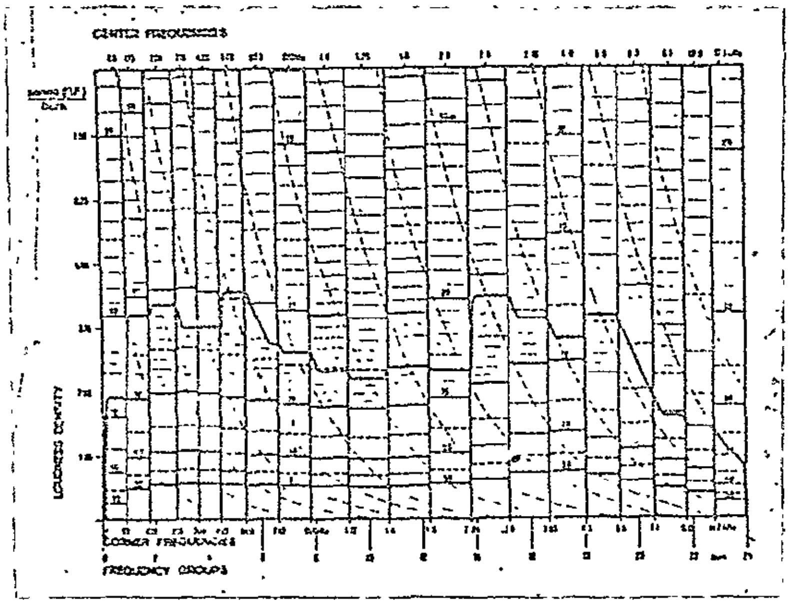


Figure 50a: Loudness Diagram of the Sound Spectrum "Stationary 100% Thrust" Recorded in the Actual Aircraft HFB 320

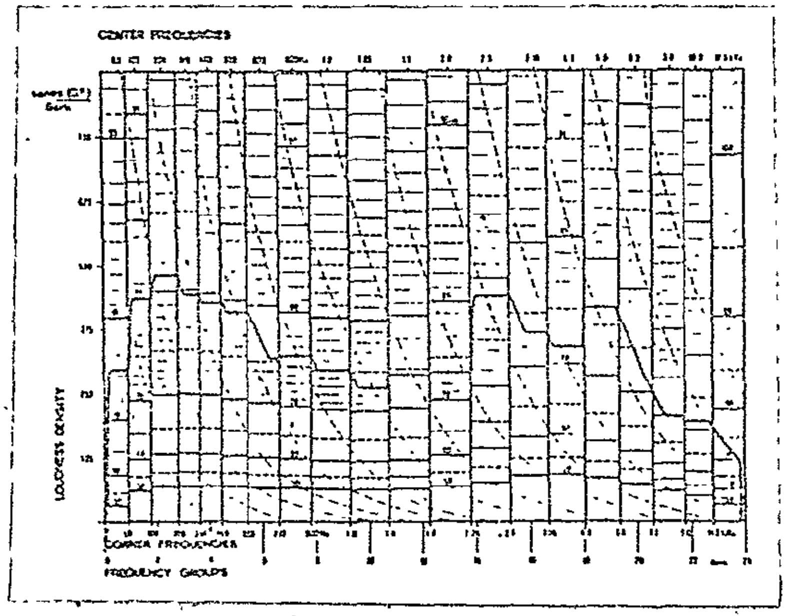
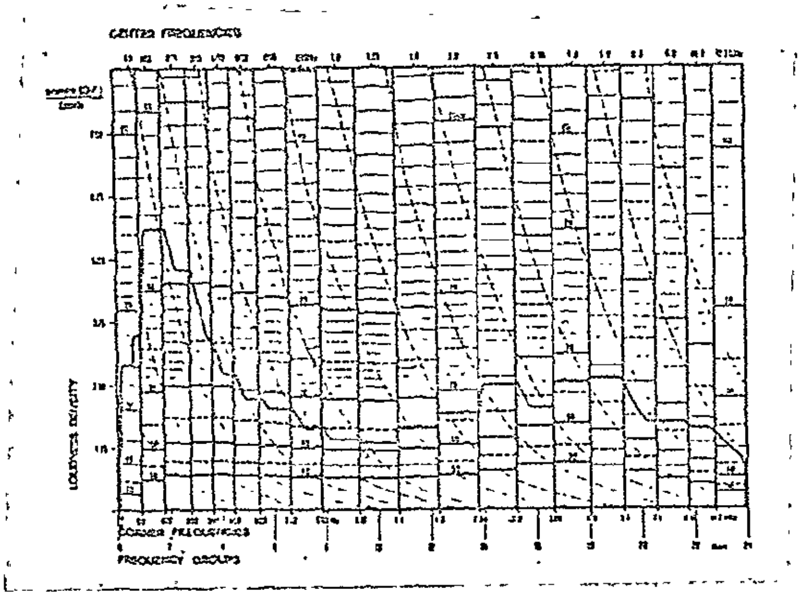


Figure 50b: Loudness Diagram of the Sound Spectrum "Stationary 100% Thrust" Recorded in the Flight Simulator HFB 320



-Figure 51a: Loudness Diagram of the Sound Spectrum "Rolling at 50 Knots" Recorded in the Actual Aircraft HFB 320

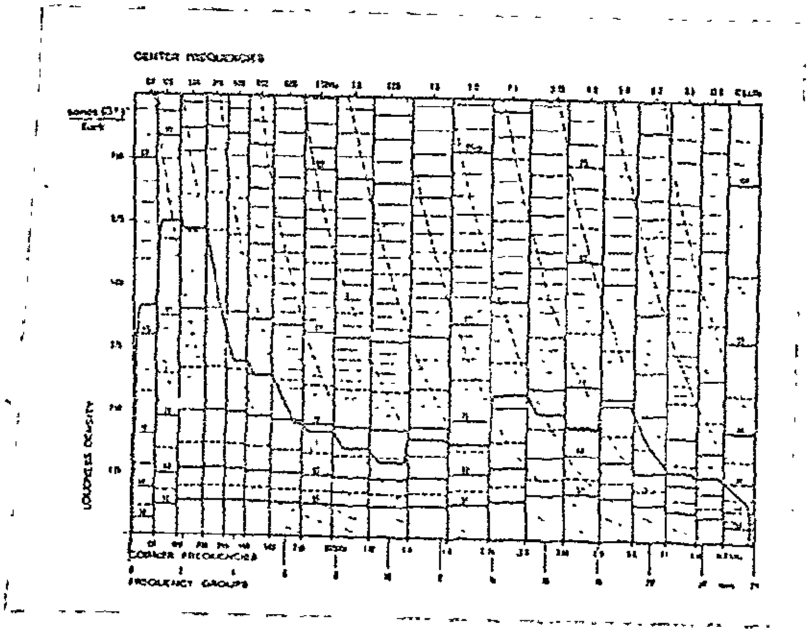


Figure 51b: Loudness Diagram of the Sound Spectrum "Rolling at 50 Knots" Recorded in the Flight Simulator HFB 320

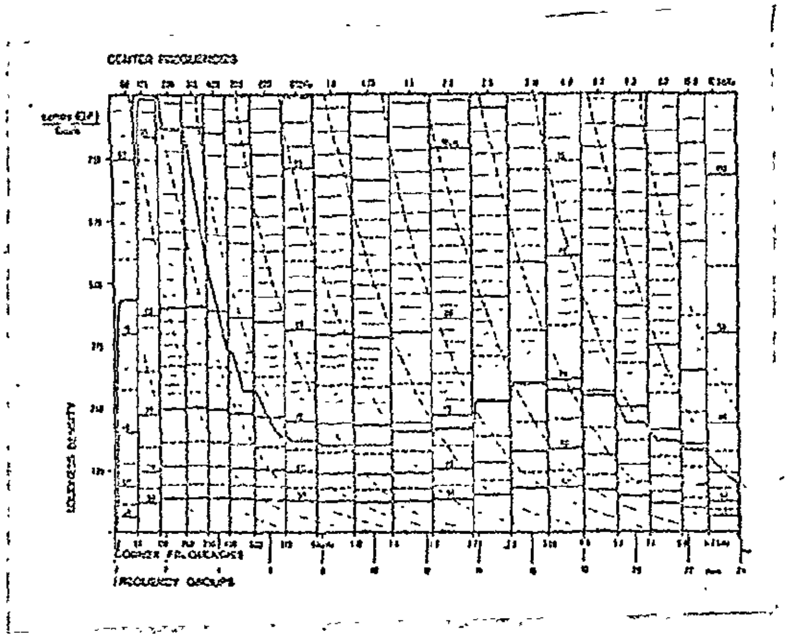


Figure 52a: Loudness Diagram of the Sound Spectrum "Rolling at 120 Knots" Recorded in the Actual Aircraft HFB 320

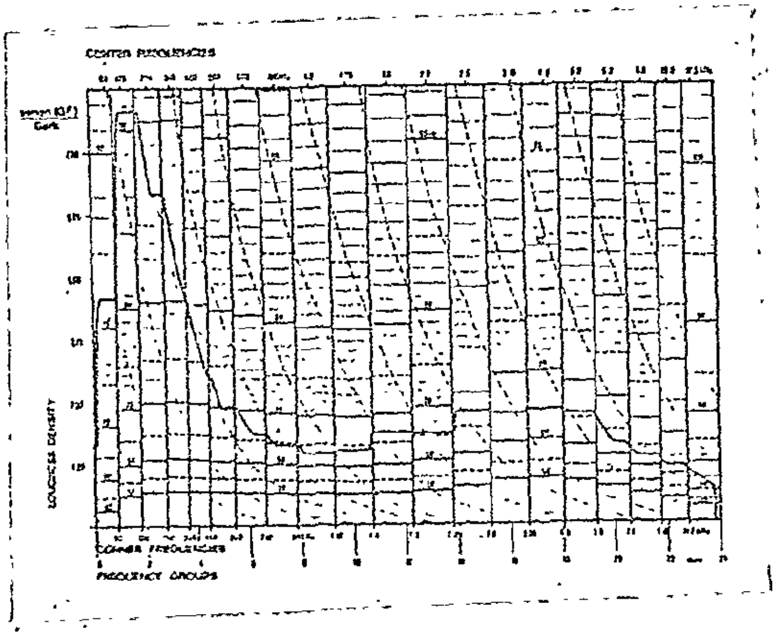


Figure 52b: Loudness Diagram of the Sound Spectrum "Rolling at 120 Knots" Recorded in the Flight Simulator HFB 320

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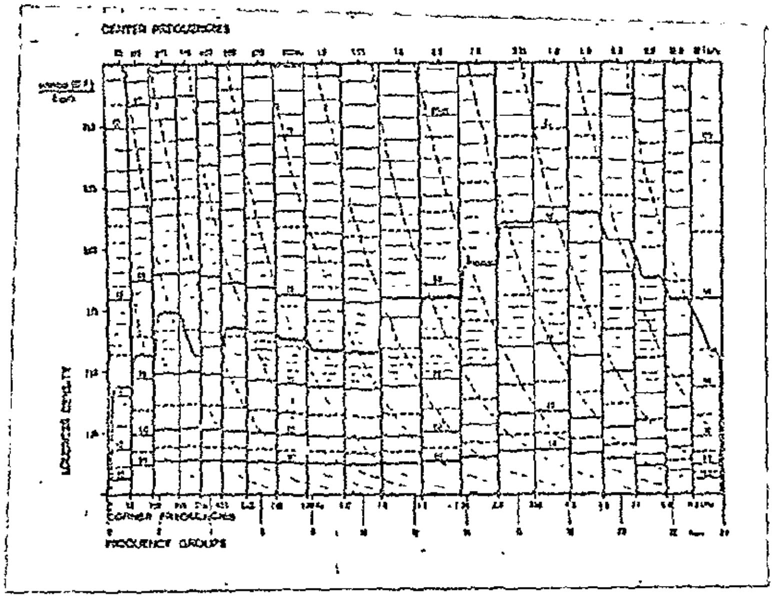

Figure 53a: Loudness Diagram of the Sound Spectrum "Flight with 85% Thrust, 20,000 feet altitude" Recorded in the Actual Aircraft HFB 320

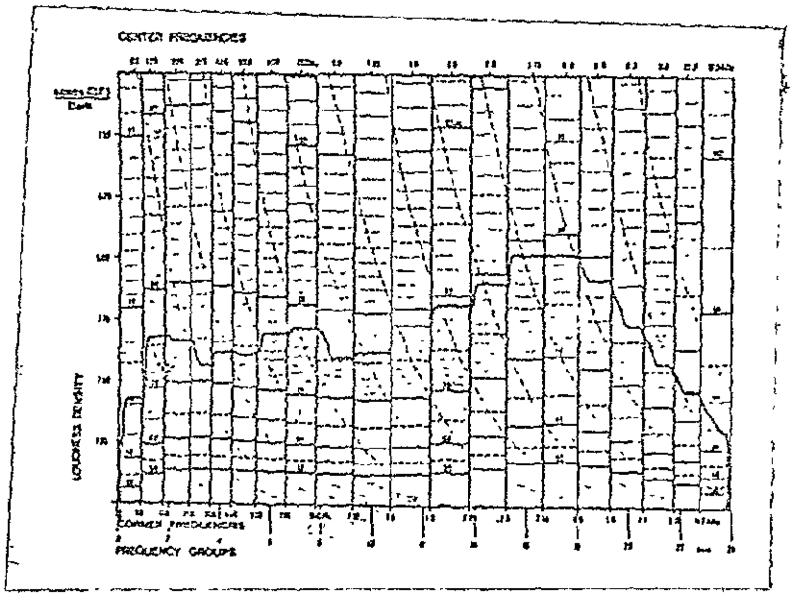

Figure 53b: Loudness Diagram of the Sound Spectrum "Flight with 85% Thrust, 20,000 feet Altitude" Recorded in the Flight Simulator HFB 320

4.2 Synthesis of Individual Sound Portions /73

For reproduction of cockpit sounds in the HFB 320 Hansa Jet the sound portions
Engine
Wind
Rolling, Landing gear and landing
were considered necessary for simulation due to the subjective and objective analysis conducted. The technology of the sound simulator was carried out employing the traditional procedure because of the simplicity of electronic circuitry and the relatively short development time.

4.2.1 Engine Sound Simulation

The engine sound of the HFB 320 was constructed from three partial sounds. These sound portions are produced by engine howling, arising from the rotation of impellers in the compresser and the turbine, by the sound of air in the compresser intake and by the enhaust sound in the thrust nozzle. Figure 54 shows a section of the HFB 320 engine.

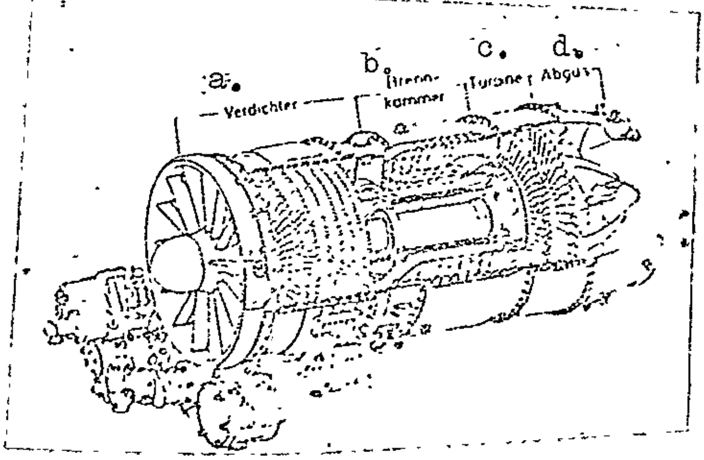

Figure 54: Sectional View of the HFB 320 Engine [34]

Key a. Compresser
b. Combustion chamber
c. Turbine
d. Exhaust

8 runners may be seen in the compresser with a impeller size decreasing toward the rear. The turbine consists of two runners. Figure 55 /74 shows the block diagram for sound simulation of an engine with the portions engine howling, air intake and exhaust. The combustion sound present in the combustion chamber was not simulated, since it is not audible in the cockpit as an individual sound.

4.2.1.1 Engine Howling

Frequency amd amplitude of howling are a function of engine RPM and of velocity (IAS). A quantity analog to RPM controls six square-wave voltage generators (VCO) in a parallel circuit, operating asynchronically with different frequencies (1) to (6)[1]. Corresponding to the differing number of impeller blades in the compresser and turbines the oscillators have differing frequency ranges, as may be seen from figure 55 only six of a total of ten impellers have been simulated. The subjective comparison of the simulated engine noise with the actual noise produced the result that the simulation of the remaining four impellers provides no contribution to the total sound and may therefore be disregarded.

The frequency range of oscillators (VCO) is determined as follows:

Compresser:	0 to 13.0 kHz 0 to 13.3 KHz 0 to 10.0 kHz 0 to 11.0 kHz	corresponding to four impellers
Turbine:	0 to 6.0 kHz 0 to 6.6 kHz	corresponding to two impellers

The frequency mixture of oscillators feeds the signal imput of voltage controlled amplifier (10). Amplitude regulation of this VCA is carried out by means of the controlled voltage from (13), resulting from the subtraction of the input factor (RPM and the velocity (IAS), since the audibility of engine howling increases with increasing RPM - decreases, however, with increasing aircraft velocity (compare figure 47). The amplifier exit (10) therefore supplies engine howling regulated /75 in frequency and amplitude, fed into the mixer amplifier (16).

4.2.1.2 The Air Intake Sound

Audibility of the air intake sound is a function of the RPM and velocity (IAS), as is the audibility of the engine howling. For simulation of this sound the same control voltage may therefore be applied from (13) which also served for amplitude control of the engine howling. The voltage controlled amplifier (11) controls the amplitude of the noise generator (7) for generation of an air sound and therefore supplies the air intake sound dependent on number of rotations of the engine and on the aircraft velocity. This signal is led to an input of the mixer amplifier (16).

A white noise generator was selected as air sound source, supplying a "white" noise in a low frequency range (audible range). The white noise generator is dimensioned in such a way that its lower limit frequency lies at 27 Hz and its upper limit frequency at 10 kHz.

Determination of the lower frequency range stems from the consideration that by means of subsequent filters bumping noises may be derived from these frequencies, occurring, for example, during rolling on the runway or when landing gear is retracted or extended. The

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[1] The numbers in ( ) refer to the function components of figures 55, 56 and 57.

lower limit frequency is determined according to the performance of the electronic-acoustic device applied to producing sounds in the cockpit.

The upper limit frequency results directly from the analysis of characteristic air intake sounds of an engine. Using this limit frequency the wind sound may also be derived from the signal of this noise generator (see section 4.2.2).

4.2.1.3 Exhaust Noise

Via suitable filters for "air sounds" in the white noise generator (7) the output of the first filter (8) supplies a low frequency noise, muffling frequencies exceeding two kHz. Due to this frequencies of /77 less than 2 kHz seem to be especially brought out subjectively; this corresponds to the exhaust noise of a jet engine. Audibility of this sound portion also decreases with increasing velocity. Since exhaust sounds up to 50% RPM (idleling) are hardly audible, while the air intake sounds dominate in the RPM range exceeding 50%, the dependence of the amplitude of this sound portion on number of rotations is altered by the function generator (14). Function sequence was analyzed subjectively and is indicated in figure 42 by the dotted line, which only assumes noticeable values after 50% thrust. Subtraction of the input factor "IAS" is carried out in the summation instrument (15), with an output signal regulating the amplification of VCA (12), so that those voltage controlled amplifiers supplies the "exhaust noise" as a function of the input factors "RPM" and "IAS".

The sound portions engine howling, air intake and exhaust "composed" individually in this manner are mixed in (16) and result in the engine sound of one engine.

4.2.2 Wind Sound Simulation

From a velocity of approx. 60 knots on wind sounds are audible in the cockpit and increase with increasing velocity. No substantial difference in audibility of the wind are perceived in the case of speed alterations in the range of 250 knots to 300 knots. As may be seen in figure 41, engine noise dominates in the range up to 60 knots. The input factor "velocity" rates the voltage controlled amplifier (17), containing the white noise generator signal "air sounds" from the input (figure 56). In addition to this wind sound created by the fuselage of the aircraft large amplitude changes in wind sound results due to retracting and expanding landing gear, which also has to be taken into account.

The discreet operation function "retract landing gear" or "extend landing gear" reaches a delay element (21) as a step function, which simulates the time period necessary for retracting or extending landing gear. The sum of the input factor "velocity" and "retract/extend landing gear" (20) represents the landing gear geometrical portion in the wind /78 flow. This signal voltage rates the amplitude of noise generator "air sound" via a VCA (18) and supplies a "air sound" of the landing gear dependent on the velocity at the exit of the voltage controlled amplifier (18). The wind sound of the fuselage and landing gear are put together in the mixer step (19) for the sound portion wind.

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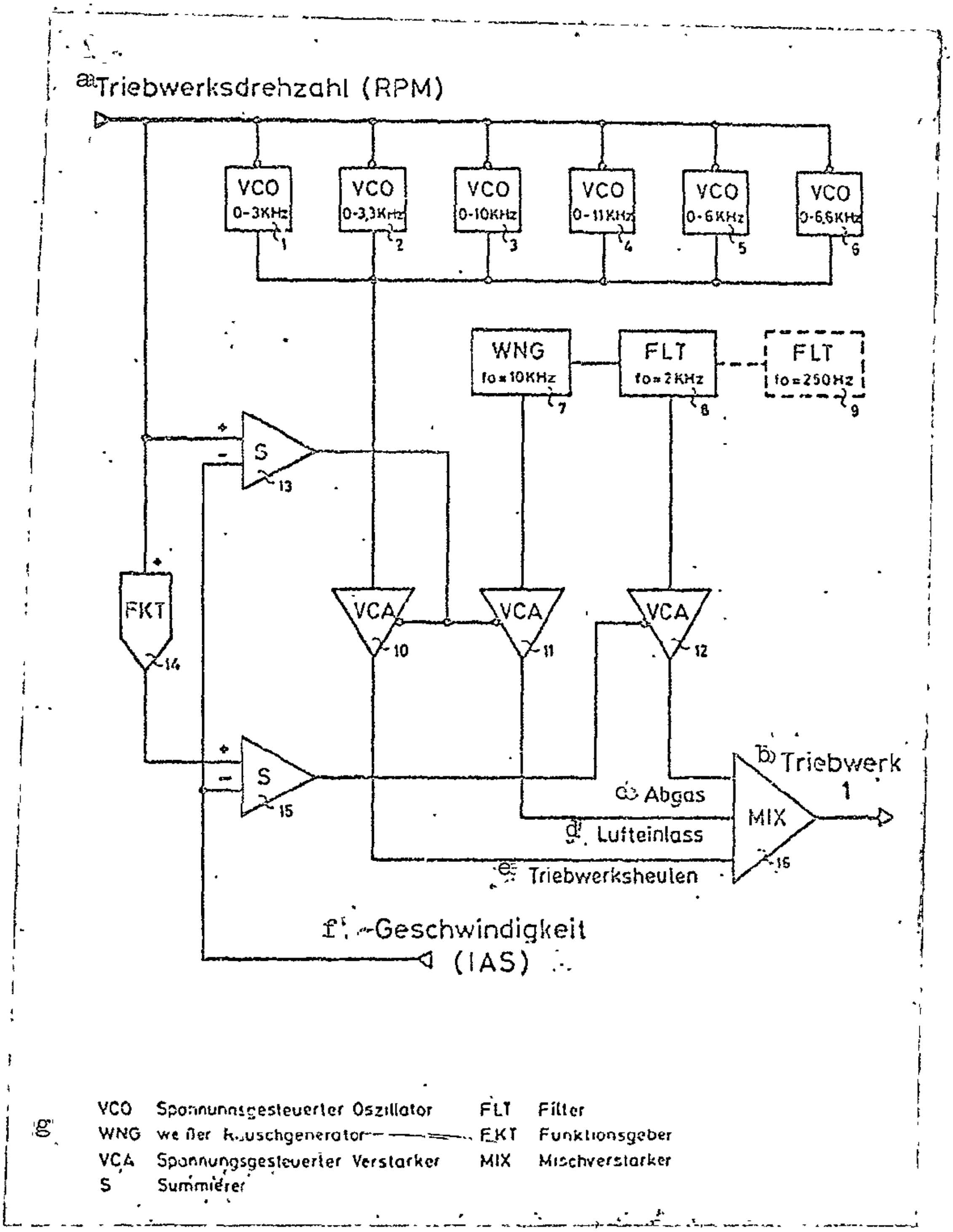

Figure 55: Block Diagram of Sound Synthesis of an Engine

Key a. Engine RPM
b. Engine
c. Exhaust
d. Air intake
e. Engine velocity
f. Velocity

g. VCO Voltage Controlled Oscillator
WNG White Noise Generator
VCA Voltage Controlled Amplifier
S Summation Instrument
FLT Filter
FKT Function Generator
MIX Mixer Amplifier

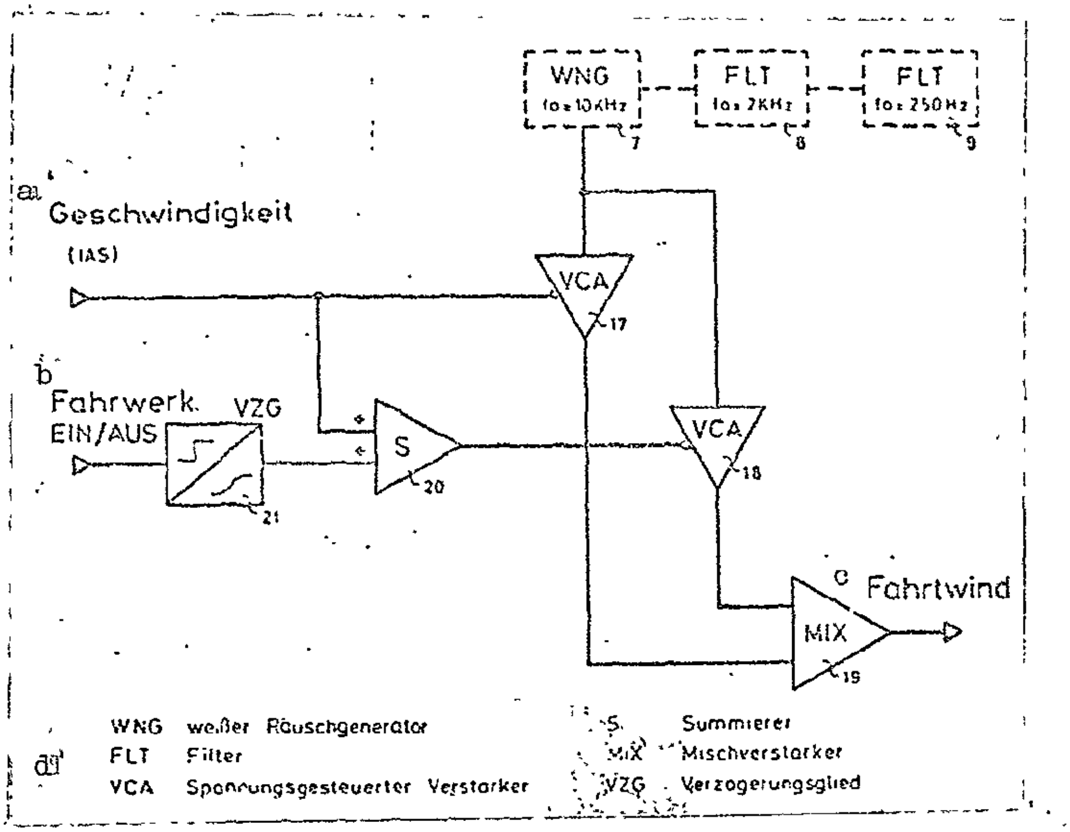

Figure 56: Block Diagram of Sound Synthesis Wind

Key a. Velocity
b. Retract/Extend landing gear
c. Wind
d. WNG White Noise Generator

FLT	Filter
VCA	Voltage Controlled Amplifier
S	Summation Instrument
MIX	Mixer Amplifier
VZG	Delay element

4.2.3 Simulation of the Rolling, Landing Gear and Landing Sound

During rolling, retraction and extension of landing gear and landing low frequency sound portions result, designated here as "bumping sounds". The output of a second filter (9)[1] with an upper cutoff frequency of 250 Hz feeds the signal inputs of three voltage controlled amplifiers (24) (28) (30) which have outputs supplying the sound portions "rolling", "landing gear" and "landing" (figure 57).

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[1] The numbers in the ( ) refer to the function components of the figures 55, 56, and 57.

/79

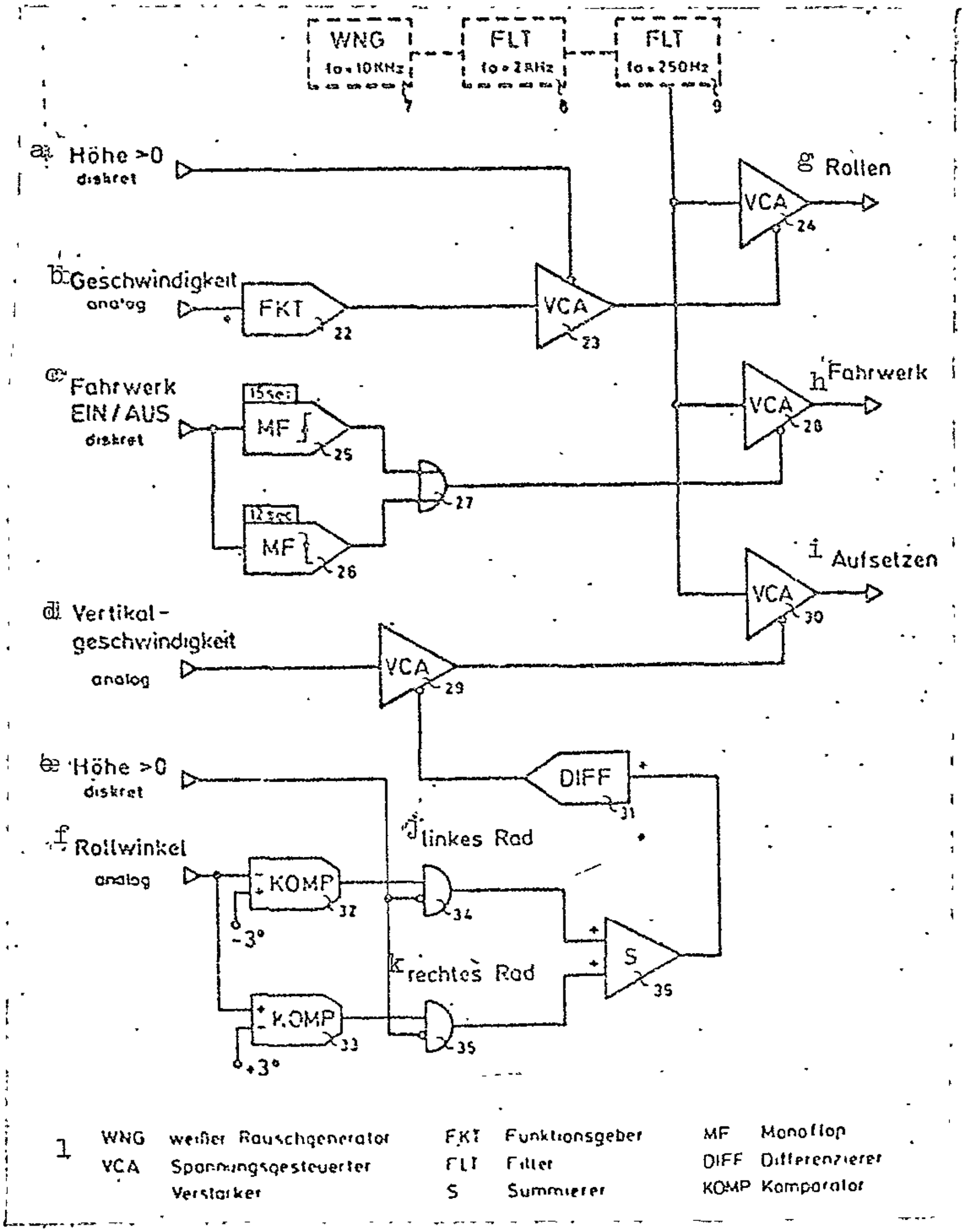


Figure 57: Block Diagram of Synthesis of Rolling, Landing Gear and Landing Sounds

Key a. Altitude discrete
b. Velocity
c. Landing Gear retract/extend discrete
d. Vertical velocity
e. Altitude discrete
f. Angle of roll
g. Rolling
h. Landing gear
i. Landing
j. Left wheel
k. Right wheel

l.

WNG	White noise generator
VCA	Voltage Controlled Amplifier
FKT	Function Generator
FLT	Filter
S	Summation Instrument
MF	Monoflop
DIFF	Differentiator
KOMP	Comparator

4.2.3.1 Rolling Sound /80

The input factors "velocity" and "altitude" are employed for rating the rolling sound, in which the non-linear velocity dependency of rolling sound is reproduced in the function generator (22). Altitude determines whether the rolling sound is audible or not (figure 56). The non-linearity is demonstrated in the fact that on the one hand smallest velocities already produced a rolling sound and that on the other hand the influence of velocity in the range of 80 to 100 knots on a volume increase is small. The voltage controlled amplifier (23) operates via the discrete information that control input like a switch, heating the control voltage 0 (no rolling sound) at the altitude H 0 to the control input of the VCA (24) and the rated velocity signal at the altitude $H = 0$ to this control input, which then supplies a rolling sound dependent on velocity at the output.

4.2.3.2 Landing Gear Sounds

When the landing gear switch is activated either a monoflop (25) with the time constant of 15 sec. for "extend landing gear" or a second monoflop (26) with the time constant of 12 sec. for "retract landing gear" is released, simulating the time span for the moving landing gear. Via an OR connection (27) this signal reaches the control input of the VCA (28) and causes the "bumping sound" positioned at the input of this VCA to occur for the duration of the landing gear motion, while the amplifier (28) blocks through a corresponding control voltage at the defined landing gear condition retracted or extended.

4.2.3.3 Landing Sounds

The audible sound portion of landing consists in a short, rapidly disappearing bumping, caused by a momentary load on the landing gear. Tire squealing during landing is not audible in the cockpit in the case of an aircraft of this type and is therefore disregarded in a simulation. Audibility of landing sounds is dependent on vertical velocity of the aircraft at the time of ground contact. Further input factors for simulation of landing sounds are the roll angle φ and the altitude /85 $H > 0$. The comparators (32) and (33) determine whether the left or right wheel first makes contact with the ground using the analog information of roll angle φ, where

$\varphi > +3^\circ$	the right wheel makes contact first
$+3^\circ > \varphi > -3^\circ$	Both wheels make contact simultaneously
$-3^\circ > \varphi$	the left wheel makes contact first

These comparator signals, however, are not switched through from the AND Gates (34) and (35) to the summation instrument (36), until the aircraft makes contact with the ground, i.e. when the determination signal altitude has switched from logic "1" to logic "0". The differentiater (31) supplies the voltage controlled amplifier (29) with a rapidly decreasing landing signal, with an amplitude at the output of the VCA (29) proportional to the vertical velocity. This signal controls the portion of bumping noises in the voltage controlled amplifier (30), contributing the sound portion landing.

4.2.4 Total Synthesis of Sound Portions, Shown Using the Example of the Take-off Phase

The engine sound was composed from three sound portions, as may be seen from Figure 58 (compare section 4.2.1). Factors to be differentiated are a) engine howling produced by voltage controlled generators, represented by the corresponding loudness diagram, b) the /82 air intake found at the compressor intake of the engine, synthesized by means of a noise generator with an upper limit frequency of 10 kHz, indicated in the figure by a loudness diagram and c) the exhaust sound corresponding to the exit of exhaust flow at the thrust nozzle, synthesized by means of a noise generator with an upper limit frequency of approx. 2 kHz also further explained by a loudness diagram.

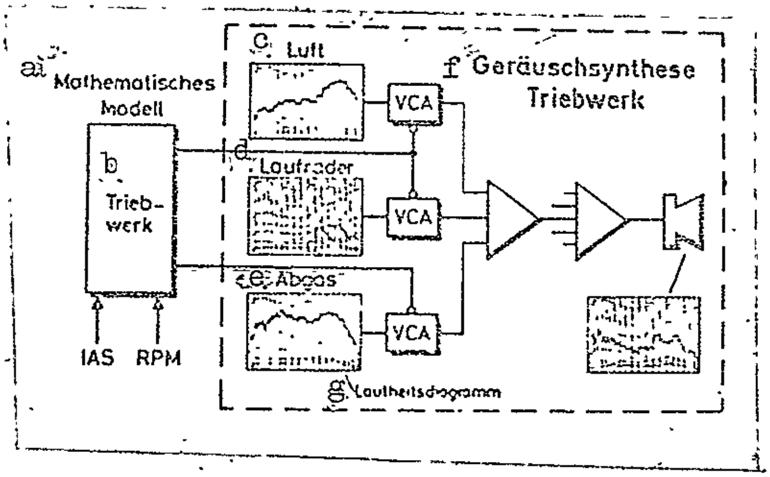


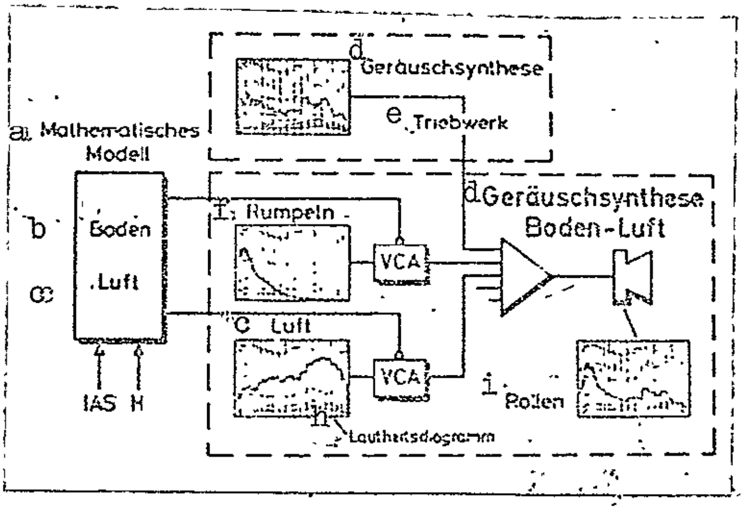
Figure 58: Sound Synthesis Engine

Key a. Mathematical model
b. Engine
c. Air
d. Impellers
e. Exhaust
f. Sound synthesis engine
g. Loudness diagram

Rating of the voltage controlled amplifier arranged subsequently occurs via the mathematical model "engine" with the input factors "velocity" and "RPM". The subsequent mixing combines the sound portions of the engine. In a performance mixing step further sound portions may be fed in. In figure 58 at the lower right the loudness diagram is indicated for a simulated engine performance of 100% thrust.

The sound synthesis "ground-air" is shown in figure 59. Noise generators again serve as the sound sources: for the bumping of rolling sound a generator with an upper limit frequency of 250 Hz and for the air sound of the wind a generator with an upper limit frequency of 10 kHz.

The voltage controlled amplifiers are rated here by the mathematical /83 model "ground-air" with the input factors "velocity" and "altitude".



- Figure 59: Sound Synthesis Ground-Air

Key a. Mathematical model
b. Ground
c. Air
d. Sound synthesis
e. e. Engine

f. Bumping
g. Sound synthesis ground-air
h. Loudness diagram
i. rolling

The velocity is rescaled for regulation of bumping sounds into a "rolling velocity range" and the altitude determines the audibility of rolling sound. These are then fed into the performance mixer amplifier already mentioned. This loudness diagram shows one moment of the take-off phase at 90 knots rolling speed. The two next figures will utilize the situation as an example.

Figure 60 shows the relationships between the individual components such as engine, wind, rolling, landing gear and landing without taking into account mathematical model design in a very simplified block diagram of the total sound simulator. As may be seen in this figure and in the figures 58 and 59 the generators for white noise and the VCO are the actual components for producing noise, regulated in volume by means of the VCA. Mathematical model design is shown in the figures 55, 56 and 57, specifically via signal conversion of the factors of the mathematical models of aviation mechanics such as "RPM, altitude, velocity, vertical velocity and landing gear motion in the signal voltages regulated by the VCA. At the left side of the figure simulation of an engine is shown, operating with an engine performance of 100% thrust as indicated in the loud mesh diagram. In the right figure portion the VCA for wind and rolling are active. The sound portion set together in this manner are fed to a performance mixer amplifier and the total sound is radiated via six loud speakers. The loudness diagram shows the momentary recording of take-off phase at 90 knots rolling velocity.

In figure 61 the corresponding momentary record of the already explained subjective analysis (figure sequence 42 - 47) is compared to this synthetic sound spectrum (black lines in the inset). The contribution of engine is easily seen in the periodic portions of the first compressor (1), the second compressor and the turbine. Further the rolling portion (4) may be seen. The portion air, exhaust, and wind are only noticeable in a loudness increase of the mean and higher frequencies.

/85

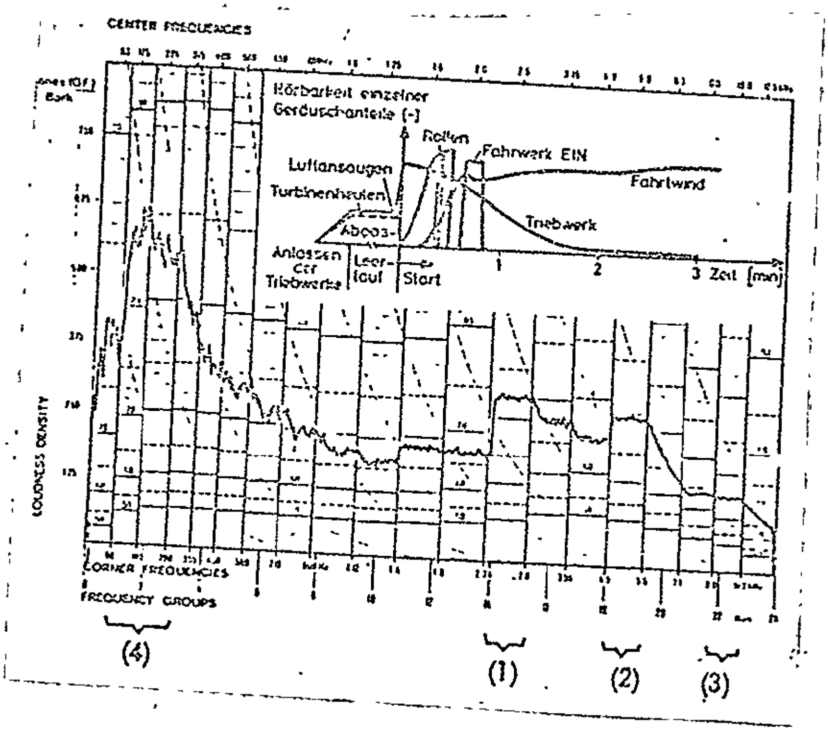


Figure 61: Loudness Diagram: Rolling at 90 Knots
Inset Figure Corresponds to Figure 47

Note: Please take the key from figure 47 on page 57

4.3 Mixing the Sound Portions

Sound presentation is carried out by means of six loud speakers, as shown in figure 62a, supplied via six regulators from a performance mixer amplifier (38). This is supplied with the sound portions wind, landing gear, rolling, landing and engine. The two engines of the HFB 320 are combined via the amplifier (37) to an engine sound, since in the cockpit sounds from the left and right cannot be differentiated. Through the positioning of loud speakers in the cockpit a sufficiently good diffuse spacial sound field is created (figure 62b). The most favorable regulator adjustment is determined subjectively from the judgement of trial persons.

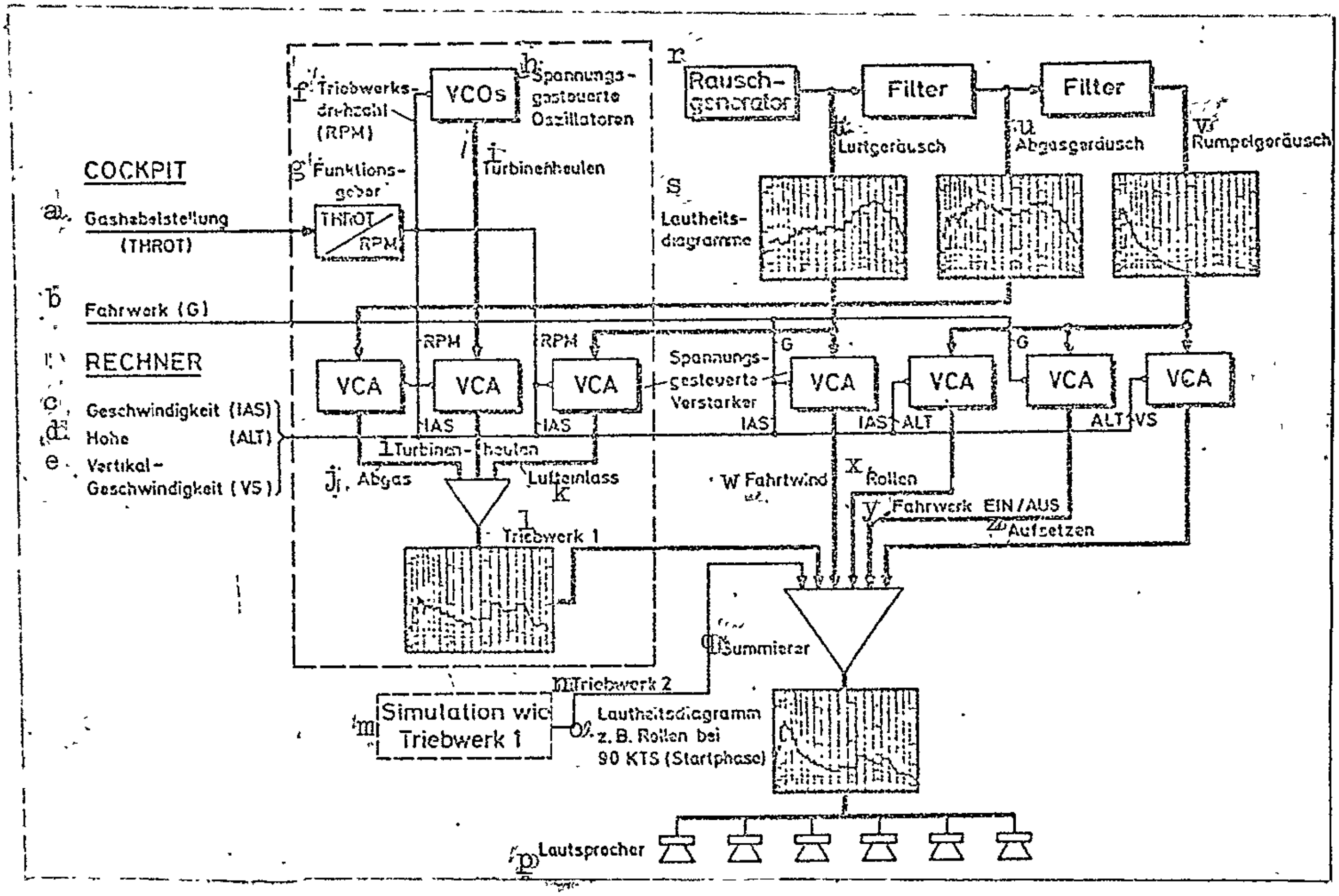


Figure 60: Block Diagram of Sound Simulation of a Two-Jet Aircraft Using the Example of the Hansa Jet HFB 320

Key a. Throttle Position
b. Landing gear
c. Velocity
d. Altitude
e. Vertical velocity
f. RPM
g. Function generator
h. Voltage controlled oscillators
i. Turbine howling
j. Exhaust
k. Air intake
l. Engine 1
m. Simulation as in engine 1
n. Engine 2
o. Loudness diagram, for example rolling at 90 knots (take-off phase)
p. Loud speakers
q. Summation instrument
r. Noise generator
s. Loudness diagrams
t. Air sounds
u. Exhaust sounds
v. Bumping sounds
w. Wind
x. Rolling
y. Landing gear retracted/extended
z. Landing

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4.4 Technical Description

The sound simulator supplies the sound portions, heard in an aircraft of the type HFB 320 under normal operating conditions during

Engine warm-up
Engine idling
Take-off /87
Flight
Landing.

The simulator consists of an 19 inch push-in unit, a performance mixer amplifier and six loud speakers.

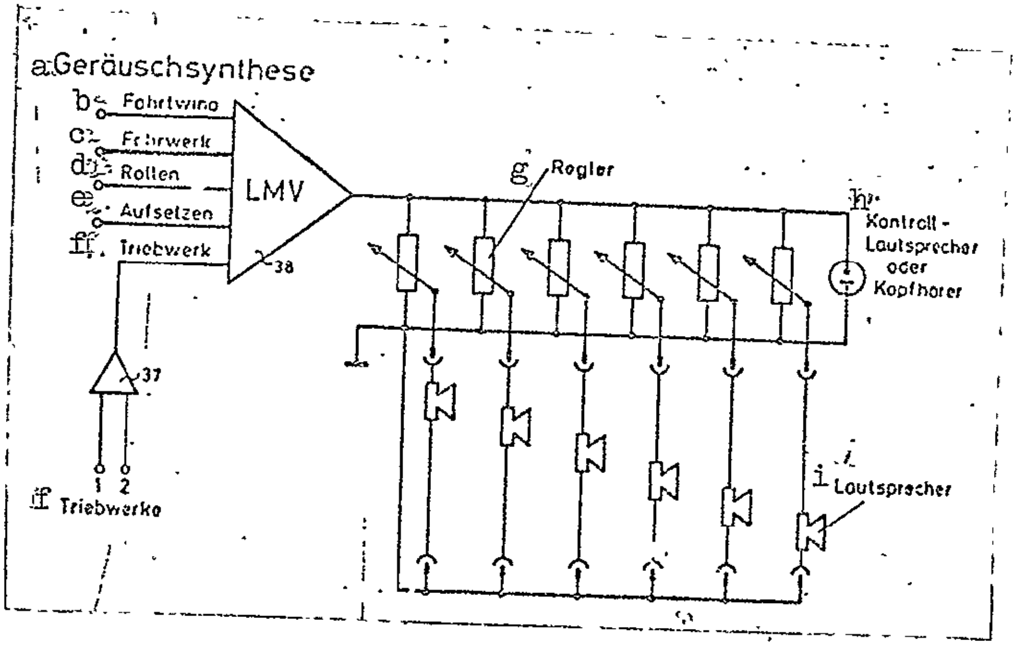

Figure 62a: Block Diagram of Sound Presentation

Key a. Sound synthesis
b. Wind
c. Landing gear
d. Rolling
e. Landing
f. Engine
g. Regulator
h. Control loud speakers or ear phones
i. Loud speaker

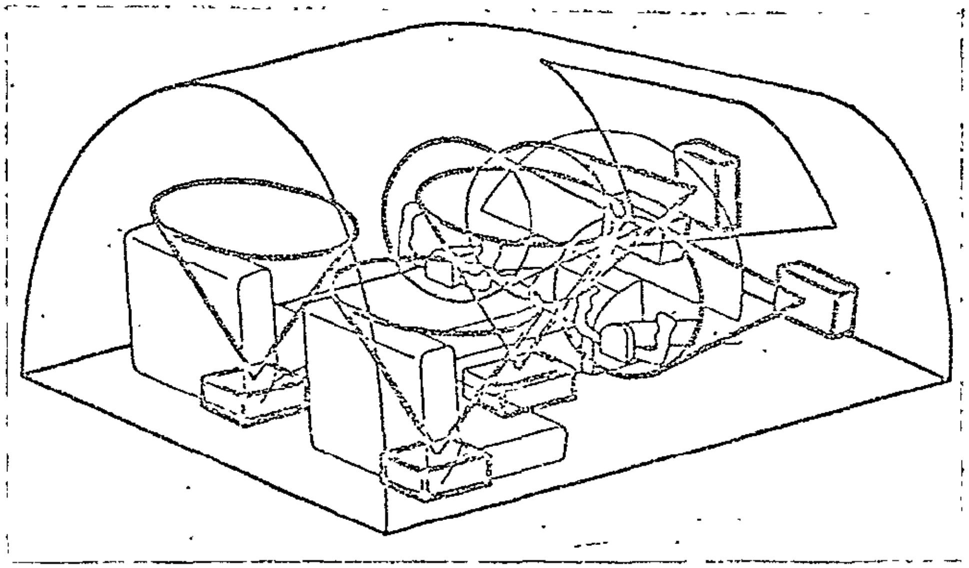

Figure 62b: Arrangement of the Six Loud Speakers in the Cockpit for Achieving a Diffuse Sound Field

The components of the 19 inch push-in unit are the operating field, the chassis provided for the printed circuits and the necessary power supplies for voltage supply. The operating field contains all switches and regulators for the manual or automatic operation of the sound simulator. In the operational mode "automatic" the simulator is controlled by the computer. The sound of two jet engines, wind, landing gear, rolling and landing are reproduced. The employment of six loud speaker systems and a power mixer amplifier of high quality make possible the representation of a sound meeting sufficient standards of genuineness and tone color.

4.4.1 Description of the 19" Push-In Units /88

4.4.1.1 The Operating Field

The operating field contains all the necessary switches and regulators for operating the sound simulator. Via a common power unit switch the various power units are activated. Eight operational type converters (AUTO-MAN) distribute the control signals desired in each case to the analog as well as the discrete inputs of sound simulator.

In the position "manual" the four analog inputs

Engine RPM 1
Engine RPM 2
Velocity
Vertical velocity

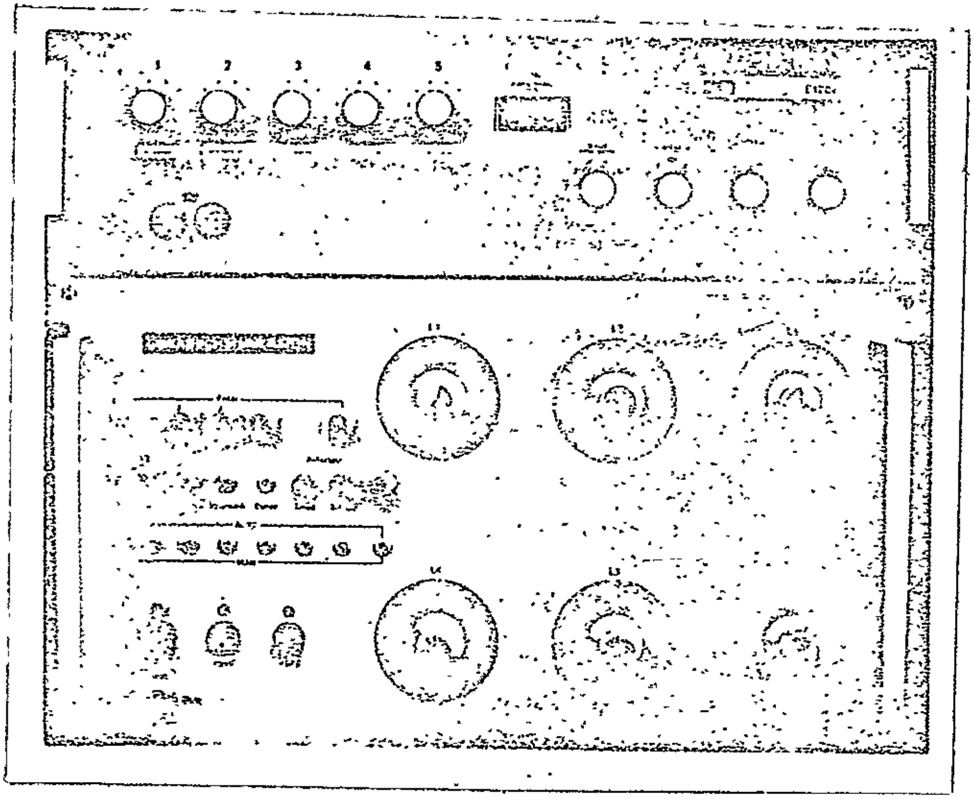

Figure 63: A View of the HFB 320 Sound Simulator with the Performance Mixer Amplifier

main text are connected internally with a continuously variable controlled voltage of 0-20 V by means of the corresponding potentiometer (engine 1 and 2, IAS, VS).

The remaining four discrete inputs such as landing gear, rolling and landing "left" and "right" are positioned via two change-over switches and two frequency shifts either on ground potential or a fixed control voltage of + 20 V.

When an operation mode change-over switch is in the position "AUTO", the input of the sound simulator involved receives the analog or discrete control signal directly from the computer or cockpit of the flight simulator. Above the potentiometers and switches for manual operation of the sound simulator the potentiometers of 5 RC filters are arranged. With their aid the tone coloring of the five different sound portions engine, wind, landing gear, rolling and landing may be adjusted independent of one another.

On the right side of the control field six L regulators and a control loud speaker output are arranged. The L regulators make an independent volume regulation of the six loud speaker systems in the cockpit possible. The control loud speaker output serve for the connection of a loud speaker or ear phones during repair or service work which may become necessary.

4.4.1.2 Chassis Provided for the Printed Circuits /89

A 19" chassis provides space for 21 circuit boards. These are classified as follows:

2 Circuit Boards	with 6 voltage controlled oscillators each for synthesis of the engine howling of two engines
1 Circuit Board	with 14 voltage controlled amplifiers and 8 analog inverters or scaled amplifiers
1 Circuit Board	with 1 noise generator, 2 active filters and 12 computer amplifiers
1 Circuit Board	with 10 input impedence convertors and 5 mixer steps
1 Circuit Board	for the synthesis of the avionics sounds (see chapter 5)
1 Circuit Board	for signal processing between computer/cockpit and the sound simulator

For suppression of undesired low frequency connections six shield circuit boards are installed between the printed circuits of the sound simulator. The slider of the L regulators are carried out on six output loud speaker jacks, in order to create an easy and standardized connection for the loud speaker systems.

4.4.1.3 The Power Units

For operation of the sound simulator DC voltages of ± 24 V, ± 15 V and + 5 V are necessary. All voltages must be regulated to ± 0.5 %. For the DC voltage supply of the HFB 320 sound simulator the following power units were employed:

a. 2 x Zentro-Coutant ATC 100/24 (± 24 V, 1A)
b. 2 x Zentro-Coutant ATC 50/14 (± 15 V, 500 mA)
c. 1 x Zentro-Coutant GP 500/6 (+ 5 V, 5A)

4.4.2 The Performance Mixer Amplifier /90

In order to supply the cockpit with sufficient sound power, an output amplifier with sufficient power reserve is employed. The power reserve must be selected in relation to distortion factor of the amplifier and losses in the regulator. The mixer amplifier of the E 120 e of the "K + H Telewatt" company was selected as a suitable amplifier. This amplifier has five imputs which may be mixed with an input sensitivity of 100 mW and an input impedence of 100 kΩ. The distortion factor amounts to approx. 0.2% at nominal power and a frequency of 1 kHz.

The frequency spectrums of the sound portions to be transmitted permit a very low upper limit frequency. Even the over waves of the high frequency wind and engine sounds hardly exceed 15 kHz. These frequencies may be transmitted by any hi fi amplifier. The lower limit limit frequency of the amplifier, however, considerably influences the sound quality of the sound presented, since the sound portion of landing gear during retracting and extending as well as during rolling on the ground contain frequencies of as low as 15 Hz. The frequency range of the Telewatt Company amplifier, type 120e, reaches low enough in the 1 dB point at 20 Hz.

4.4.3 The Loud Speakers

The selection of loud speakers to be employed is conducted according to various aspects. On the one hand the amplication of commercially available loud speakers provides advantages which should not be underestimated with respect to the sound image. On the other hand the spacial relationships in the cockpit of a flight simulator must be taken into account in the case of the necessary arrangement of at least six spacially distributed sound sources (see also figure 62b) and should be taken into consideration early enough in the design of the cockpit.

Since the design of a simulator cockpit often is undertaken without a thought about sound simulation, the positioning and dimensioning of sound sources is then carried out later during installation of a sound simulator using the space still free. When proceeding in such a manner a satisfying sound field in the cockpit may be achieved only in a few cases in spite of application of a good sound simulator.

For the cockpit of the HFB 320 simulator loud speakers of the type LE 400 of the Canton Company were selected. These loud speakers are advantageous on the one hand because of their small dimensions. In addition the loud speaker has a frequency range of 27 Hz to 18 kHz in 1 dB points. The sound performance amounts to approx. 30 watts sustained tone per loud speaker. /91

5. Avionics Sound Simulation

Avionics sounds are understood as the information transmitted in the cockpit via board loud speakers and ear phones, including morse signals, station identification, warning signals and radio messages. These signals may be produced today in computer control with the exception of the two way radio. There are two types of radio connection: in the one-way communication information is requested by the pilot such as weather forecasts. These may be called up from a tape, controlled by a computer. The normal type of radio communication is the two-way communication. In this case the second communication partner would have to be the computer. A simulation of this type is not yet possible and may have to be disregarded even if the problem can be solved, for reasons of cost.

5.1 Electronically Generated Acoustical Information

In this category morse signals, station identification and warning signals are generated electronically and called for by the computer or pilot. Figure 64 shows as an example the simulation of station identification. A radio beacon may be selected in the cockpit via the operating device. A corresponding coded identification key releases the morse cycle, modulated according to the prescribed frequency. In the lower portion of the figure a determination is made that the aircraft is flying over the position of the come-in signal of a runway stored in digital coordinates by means of the local coordinates and is made audible via the system on board correspondingly frequency modulated.

In figure 65 a modern warning system is represented which gives off acoustical warning signals in addition to the visual warning signals, not corresponding to conventional emergency signals. In this system the pilot receives a warning signal calls such as "WHOOP WHOOP PULL UP", "WHOOP WHOOP TERRAIN" and "GLIDESLOPE" directly via ear phones or the loud speaker on board [31]. It can be seen from this example that the sound simulation as subsystem also finds application in an actual aircraft, as has been the case for quite some time in steering power simulation, which simulates the steering sensation in larger aircraft. /92

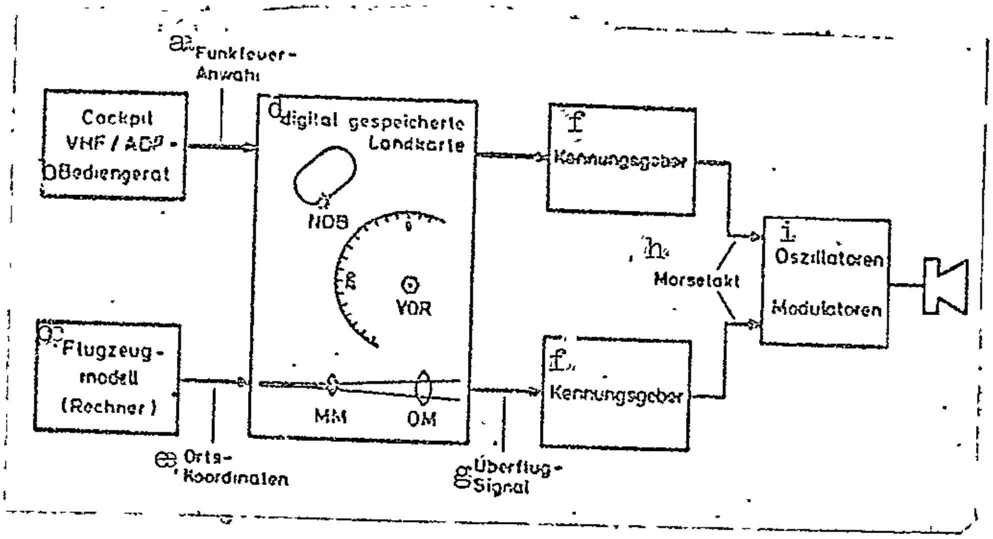

Figure 64: Block Diagram of the Acoustic Simulation of Station Identification

Key a. Radio beam selection b. Control device c. Aircraft model (computer)
d. Digital stored map e. Local coordinates
f. Identification key g. Flying over signal h. Morse cycle
i. Oscillators
Modulators

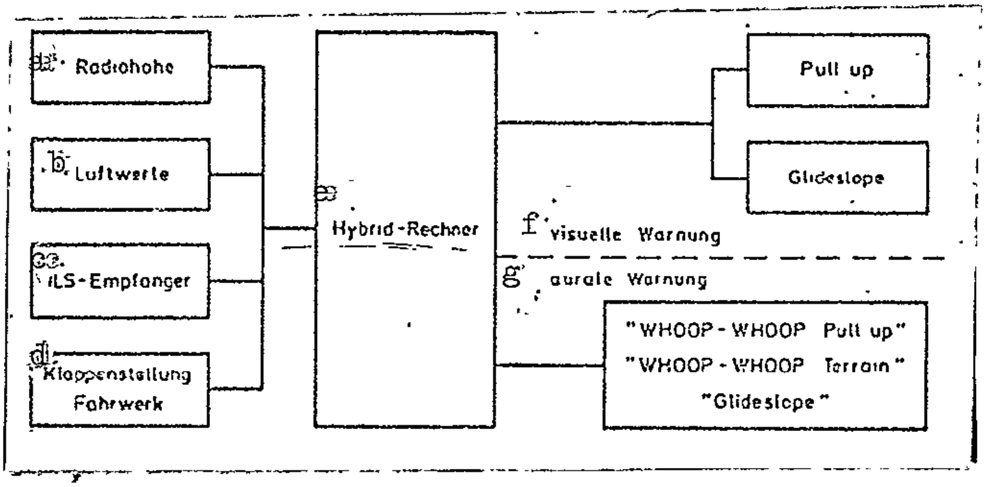

Figure 65: Computer Controlled Voice Articulated or Warnings [31]

Key a. Radio altitude b. Air values c. ILS Receiver
d. Flap adjustment e. Hybrid Computer f. Visual warning
Landing gear g. Audio warning

5.2 One-Way Communication /93

For one-way communication a tape is put into operation via a call from the pilot. In addition to this passive method there is also an active one-way communication. This is put into effective use, for example, when the plane is being talked down to a slope path. After the pilot has called for a landing approach guided from the ground (GCA), the information of the instrument landing system (ILS) are communicated to the computer, i.e. the deviations from the gliding path, after which the computer searches the tape for the corresponding correction instructions. The landing approach or glide path instructions are accommodated in each case on separate tapes.

Approaches are known for synthesizing human speech in a computer controlled method. In the case of limited vocal phrases it would be possible to replace the tape by such a synthesizer [32].

5.3 Two-Way Communication

In the past few years various approaches to computer-controlled voice synthesis have become known. The requirements made of a two-way communication system will probably not be met in the near future. This system would have to be capable of not only synthesizing human speech, but also of analyzing it. In this connection it must be asked, whether the financial demand is not too great for radio simulation, even if the solution to the basic problem is found.

5.4 The HFB 320 Avionic Sound Simulation

The simulation of avionics sound is limited to radio beam and come-in identification keys. The identification have the character of acoustic morse signal, consisting of tones of fixed frequency and amplitude. The time cycle is carried out according to the international agreement on morse alphabet. Since this cycling is conducted in a digital manner, a digital sound simulation could be used. The digital stored "map" described in section 5.1 of an airport with radio beam coordinates and identification keys as well as ILS come-in signals may be easily programmed in software. The digitally generated identi- /94 fication signal modulates an analog sinus wave and is fed into the loud speakers or ear phones in the cockpit as identification.

In addition to the computer control generation of avionics sounds the run way 25 of the air port Cologne-Bonn including the radio beacons whipper (WYP), cola (COL), LW and LV and the come-in signals OM and MM are constructed in hardware for tests and service work.

5.4.1 Synthesis of Radio Beacon Signals

A substantial part of this synthesis of radio beacon signals is, as seen in the figure 66, a ROM[1] capable of being programmed with a storage capacity of 256 one bit words, which may be called for via /95 a resetable counter. The storage spaces contain the sequence of

---

[1] ROM = Read only memory, a fixed value storage programmed a single time

logical conditions in each case corresponding to the sequence of morse signals, in which a morse-dash is three times as long as a morse dot. The storage capacity for the letters SOS (...---...) including letter and word pauses would for example occupy 32 one bit storage places.

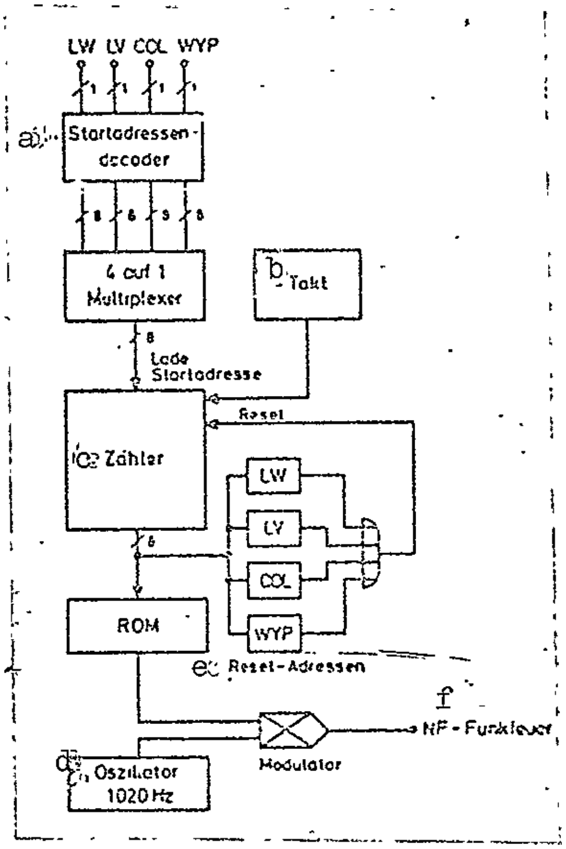

Figure 66: Block Diagram of Sound Synthesis of Radio Beacon

Key a. Start address decoder
b. Cycle
c. Counter
d. Oscillator
e. Reset address
f. NF-Radio beacon

Via a starter address decoder and a multiplexer a cycle counter is set at the necessary starter address for each individual indentification. The output of the counter calls for the storage content of the programmed ROM's and serves in addition for the generation of the reset impulse. The reset impulse returns the counter to the starter address and generates in this manner the periodically returning identification signal. The morse cycle at the output of the ROM modulates a sinus voltage of 1020 Hz and is fed to a low frequency amplifier.

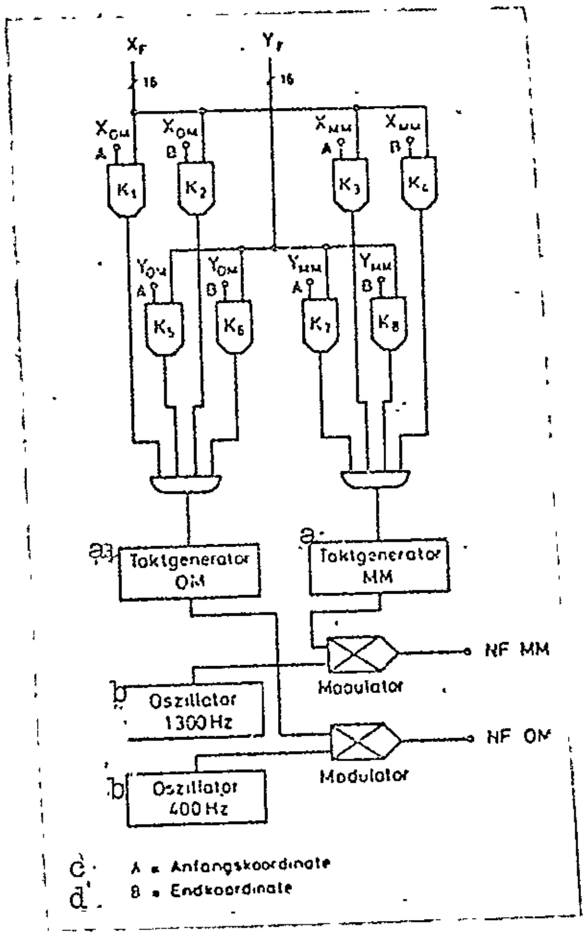

Figure 67: Block Diagram of the Sound Synthesis of Approach Marker

Key a. Cycle generator b. Oscillator c. A = beginning coordinates
d. B = final coordinates

5.4.2 Synthesis of Approach Marker /96

The synthesis of this identification is much simplier, since the signal of the OM (Outer Marker) consists of 2 dashes per second and the signal of the MM (Middle Marker) consists of alternating dashes and dots. The such cycle generators are easily produced from astabile multivibrator.

As can be seen from figure 67, the outputs of the cycle generators modulate sinus waves of 400 Hz for OM and of 1300 Hz for MM which are fed to a low frequency amplifier. The comparator network shown in the figure insures that the cycle generators are only triggered when flying over the corresponding approach marker. The natural receiving range of approach marker, requiring elaborate equipment for simulation, is assumed rectangular for reasons of simplification and is determined per signal via four comparators.

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