A PUBLICATION OF STUDER REVOX
April 1989, Nr. 26
SWISS SOUND
NEWS AND VIEWS FROM SWITZERLAND
STUDER REVOX

86th AES convention, Hamburg

Window to the audio world

The Studer Revox products were demonstrated to the interested audio experts at the 153 m² stand.

From March 7–10, 1989 the world convention of audio specialists AES (Audio Engineering Society) was held in Hamburg for the fourth time.

With over 250 exhibitors and more than 80 papers the AES convention in Hamburg is one of the world's largest events of this type. In comparison with last year's European exhibition in Paris, there has been another growth of 10%. If we make the comparison with the previous AES convention held in Hamburg in 1985, the growth was 45%. No wonder that the CCH (Congress Center Hamburg) was crammed to bursting point. On the visitor side, however, the participation tended to stagnate in comparison with last year's convention in Paris. One of the reasons could possibly be the relatively recent «Tonmeistertagung» held in November 1988.

AES Currently has a worldwide membership of over 10,000 experts from the industry, above all sound recordists, audio engineers from radio and television broadcasting stations, and from sound recording studios. The European membership is about 2,200. Worldwide there are 44 branches in 71 countries.

Studer Revox stand at the exhibition
Professional Studer products and Revox products for professional use were exhibited at the 153 m² stand. Equipment and systems for PCM/PQ editing, synchronization/postproduction, tape recorders, mixing consoles, and CD products/peripherals were arranged in five different sectors. In an additional sector, various units of the Revox C270 and PR99 MKIII series were exhibited in conjunction with Studer products.

Studer Revox papers
Two papers were presented by Studer employees: K. O. Bäder reported on «A new subjective speaker test experiment» and Dr. C. Musialik discussed «Audio signal processing and error concealment in digital recorders using general purpose DSPs». AES preprints on this paper are available.

Studer Revox sales meeting
The opportunity of the «concentrated audio atmosphere» in Hamburg was also exploited directly for presenting to the sales meeting the new Studer Revox products, with emphasis on sales arguments and strategy. Over 50 employees from 26 subsidiaries and distributors from all over the world participated at this heavily scheduled meeting. Seven reports were given on new developments, viewpoints, and structures.

The Studer Revox sales meeting provided an excellent opportunity for bringing the participants up-to-date and for mutual communication.

The annoucement that the management of STUDER REVOX AMERICA Inc. is now in the hands of Tore B. Nordahl, and the news that AEG Olympia has discontinued its activities in the field of studio tape recorders and transferred it to the Studer Group, also fell within the scope of this sales meeting (see separate press release in this issue).

Marcel Siegenthaler

This Studer 900 mixing console is part of the infrastructure of the CCH (Congress Center Hamburg). This was incidentally one of the first Series 900 mixing console ever delivered.

SWISS 26 SOUND

Read in this issue:

Press-Information

In conjunction with the convention/exhibition on professional audio engineering held by the worldwide Audio Engineering Society (AES) in Hamburg (March 7–10), the AEG Group of Frankfurt/Germany released the following publication concerning STUDER REVOX of Regensdorf/Switzerland:

AEG Olympia terminates its activities in the sector of studio tape recorders.

On April 1th 1989, AEG Olympia will terminate its business in the sector of magnetic tape recorders in its facilities in Konstanz, Germany.

Production of analog tape recorders of the current range will continue to remain in Konstanz, whereas, all marketing activities including repair service and spare parts supply will change over to the STUDER Group of Companies. For customers in Germany this will become the responsibility of STUDER REVOX GmbH Studiotechnik in Löffingen, Germany. All other countries will be served by STUDER INTERNATIONAL AG based in Regensdorf/Zurich, Switzerland.

Repair services and parts supply for digital recorders will be maintained by the Division of Operations and Services of AEG Hamburg in Germany.

In the segment of magnetic recording technology, AEG Olympia has an annual turnover figure of DM 20 Mio. and it employs a staff of 60.

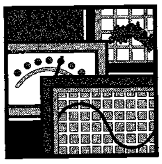
From AEG to STUDER:

AEG was a pioneer of magnetic tape technology

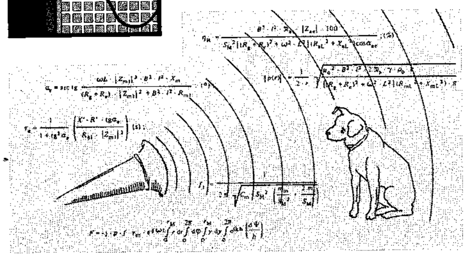
In the field of magnetic audio signals recording, AEG achieved the breakthrough to modern technology already in 1936 in cooperation with I. G Ludwigshafen (BASF). As an experiment, a public concert was recorded and reproduced «with surprising clarity and purity», as the company magazine reported. The tape transport mechanism and the electronics were developed by AEG Berlin while the tape originated in the laboratories of BASF. With the «Magnetophon», as the unit was then called, a major milestone in the evolution to modern tape recorder technology was reached. The high-frequency bias required for high-quality recording was detected by chance by Braunmühl and Weber in 1940. But developments continued only after the war, in the United States (which explains why the unit of measure for this system is based on inches), before they returned to Europe.

In 1948 Willi Studer founded his factory for electronic apparatus, and in 1949 he started production by initially converting American tape recorders for the European market.

Marcel Siegenthaler



Speaker or acoustical transducer?



This question is provocative. But not exclusively so. The question is of immediate interest in view of the latest developments, even though the author raises a topic that has often been discussed. However, it still remaines a hotly debated issue.

It is a long way from the composer's inspiration to the playback of a work. I would like objectively to recall certain principles related to the individual phases of this process.

The lyrics writer, composer, and arranger are the first artists involved in the creation of a work. Each artist is a master of his particular trade and uses it as a tool to express his artistic inspirations. The musicians and vocalists translate abstract values into acoustical ones. They interpret and model the original, but they hesitate to make any changes to the fixed specifications such as text or tone sequence. In many cases a conductor intervenes in this process. The acoustics of the hall also has to be taken into consideration which means that even the architect is indirectly involved. The concert can start.

If, however, the performance is to be recorded and later reproduced in a living room major problems occur. The complex experience at the concert must be greatly simplified so that it can be reproduced in a living room. This is the task of the audio engineer. He is familiar with the objectives of the art and with his training in engineering and physics he can capture the sound experience in such a way that it can be reproduced through two channels in the living room and offer the listener a satisfactory sound image of the concert performance. Since the result cannot be accurate for physical reasons, the audio engineer must influence the sound impression. He thus becomes almost a co-interpreter. He assumes considerable artistic responsibility.

Hi-fi- chain

The audio engineer is the last member of a chain of people who make artistic contributions. Thereafter the sole objective is to ensure by means of the hi-fi chain that the sound impression in the living room corresponds to the one the audio engineer subjectively felt to be correct in the con room. This is only possible if he know what a living room is and how the speakers are arranged there.

The characteristics of an average living room are known and defined in standards. The arrangement of the speakers in relation to the listener is also defined. This standard, applied also to the control room, enables the audio engineer to hear exactly the same thing as the listener at home. Ideally, two conditions are satisfied:

1. The electrical transmission takes place without noticeable loss of quality.
2. The sound reproduction by the speaker pairs in the control room and at home takes place without noticeable loss of quality or the speakers have at least identical deficiencies.

Characteristics of the hi-fi chain

Let us exclude the speakers for the time being. We know that the transmission characteristics should be such that possible deficiencies are inaudible. But where is the audibility threshold? This is difficult to answer. Let us therefore try to define what is generally found to be inadequate. A preamplifier, for example, with a total harmonic distortion greater than 0.1 % should be taken out of service. Or a CD player with inbalances in the reproduction of needle pulse of up to 25 µs should at best be used as a secondary unit.

An FM tuner with a frequency response drop of 0.5 dB at 15 kHz should be offered as a second-hand unit. An output stage that deforms a 10 kHz squarewave pulse under a complex load should be bequeathed.

Now back to the speakers. Nobody would dare to apply the same standards to speakers as to the other members of the hi-fi chain. The result would be catastrophic. Frequency response errors of up to 10 dB are common in speaker systems. Harmonic distortion of several % are tolerated. Pulses are modified so strongly that they become unrecognizable. No wonder that in view of this situation, objective measurements are being substituted for subjective terms taken from the world of senses such as we commonly find in gastronomy and visual arts. Speakers are said to have a «pleasant, warm, spacious, fresh, dominating» or at the other end of the scale «compressed, aggressive, nervy, isolated, annoying, or even ugly» sound characteristics.

An answer to the title question
Strictly subjective assessment of speakers has only become necessary because the industry has been unable to build speakers that have the same transmission characteristics as other members of the hi-fi chain. This fact has encouraged countless inventors to build their own speaker systems. Each of them puts the emphasis on a different characteristic and neglects others. Even unexperienced amateurs are able to produce reasonably passable products. Many speakers offered on the market today are unfortunately at this level.

More stringent demands like those traditionally imposed on other members of the hi-fi chain can only be satisfied by so-called acoustical transducers. Their technical characteristics can be specified, which means that it is not necessary to qualify them with abstract and non quantifiable terms. However, acoustical transducers can only be produced successfully by employing vast technical and financial resources.

Paul Zwicky

Studer A723 – professional acoustical transducer in monitor quality

The acoustical transducer

Each system in an audio chain, except effect machines and the correction of known distortions (e.g. RIAA phono equalization), has the function to transmit to the output a signal without modifying the information content. This may seem to be a clear and easy task but its implementation can be rather difficult. Our project manager responsible for the professional A723 has the following to contribute on this topic:

In order to achieve this objective, the following conditions, listed in the order of their significance, must be satisfied:

- No linear distortions
  (frequency and direction independent amplitude response)
- Unmodified spectral content
  (no harmonic distortions, intermodulation distortions, etc.)
- No dispersion
  (constant group delay)

The fact that a speaker is one of these elements and consequently subject to the same rules, is generally overlooked.

The design objective of the A723 was not to build a speaker and then to give it a pleasant sound, but rather to construct an acoustical tansducer that approaches the aforementioned ideal as closely as possible.

General
The aim of this report is to point out the peculiarities of the A723. In order to keep this report short, it is assumed that the reader is familiar with certain basic concepts:

- Due to the bundling phenomena and partial oscillations of the diaphragm, the audio spectrum must be split into different paths. In the A723, three such paths exist with crossovers at 300 Hz and 3 kHz.
- The box must work actively so that physical errors can be systematically combatted.
- The housing should be stable and have a low resonance.

As a professional monitor unit, the A723 is also equipped with a balanced input, various trimmer potentiometers, and protective circuits in order to achieve the best possible results in operation.

The acoustical transducer in practice
When we examine such a multiway system more closely, we discover four problem complexes:

1) Crossover networks.
2) Acoustic delay difference between the chassis, caused by their mechanical design.
3) Amplitude and phase response of the chassis.
4) Nonlinearities of the chassis and the box.

In order to achieve the objective, these four problem complexes must be optimized jointly because the effect of partial correction is lost due to the remaining errors (except in the case of nonlinearities). For this reason there is little sense in physically recessing the tweeter in order to compensate the delay difference in the signals relative to the midrange speaker, unless the signal delay of the crossover networks and the chassis is also in order.

Based on the requirements outlined at the beginning, we shall now discuss the consequences of these four error sources.

Distortions

Although they are less important than the frequency response, we shall first examine the distortions because at least one of the remedies also influences the amplitude and phase of the chassis. This is very important for subsequent improvements.

Masking effects and the high harmonic content of music are the reason why the human ear is not very sensitive to nonlinear distortions. However, this errors are of a magnitude that far exceed the audibility threshold of 0.1%.

In addition to the harmonic distortions there are also intermodulations. These are far more annoying because also subharmonic components are involved which are practically not masked and which contain frequencies that are not in a natural relation to the original spectrum. Such distortions can manifest themselves for example in the form of strong but unclean bass reproduction.

The principal source of nonlinear distortions is the chassis itself (Fig. 1):

- Magnet
  - Inhomogenous magnetic field: different magnetic fields flow through the voice coil, depending on the latter's position.
  - Magnetic field modulation by the voice coil current: The windings immersed in the magnetic circuit generate a field in the core which, depending on the polarity of the current, attenuates or boosts the common field excitation. The result is a force vector that is independent of the current direction and results in k_2 distortions.
- Nonlinearities of the restoring forces: these include the spring forces of the diaphragm suspension and the air cushion in the housing (adiabatic).

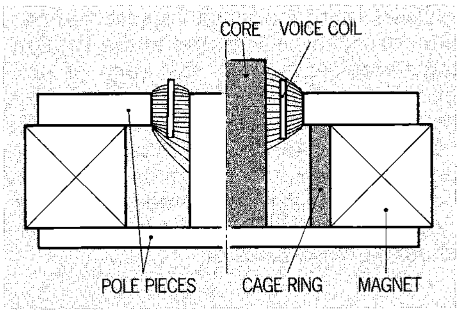


Fig. 1:
Ways of linearizing the voice coil motion.

Errors in the motors can only be alleviated by design measures. A core extension, for example, symmetrizes the magnet field, and a cage ring significantly reduces the modulation of the fields.

The influence of the restoring forces could be eliminated if it were possible to impart a speed to the diaphragm that is proportional to the signal. This can actually be done by giving the amplifier a negative output impedance, which in magnitude and phase ideally corresponds to the voice coil impedance. In practice compensation levels of 80 to 90% are feasible.

The principle of negative impedance is based on measuring the load current. A major problem is the temperature dependence of the voice coil resistivity. Only after extensive test series did we find a current measuring resistor that reproduces the behavior with sufficient accuracy. As the IM measurements (Fig. 2) show, the effort was worthwhile because the signal-to-intermodulation ratio of the A723 is approx. 10 dB higher than in conventional designs. Additional advantages of the negative output impedance are the absolute insensitivity of the diaphragms to external excitation and the linearization of the amplitude and phase response of the chassis.

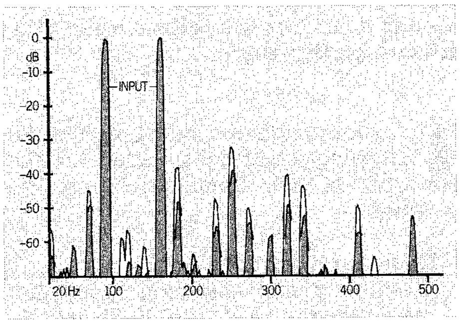


Fig. 2:
IM distorsions.

Amplitude response

The amplitude response, in our case the sound pressure level as a function of the frequency, should have a linear pattern in all parts of the room. Since there is no spherical source that handles the entire frequency spectrum, this otherwise ideal solution runs into physical difficulties: the chassis are in different locations, they bundle the sound dependently of the frequency; the acoustic delays to the listener differ, etc.

This means that acceptable conditions can be achieved only within a limited environment of the main radiation axis. As a typical case we shall examine the situation at ± 30° horizontally at ± 10% vertically. If the individual systems are arranged vertically, as it is the case in the A723, the horizontal radiation characteristic depends practically only on the quality of the systems (Fig. 3a, b).

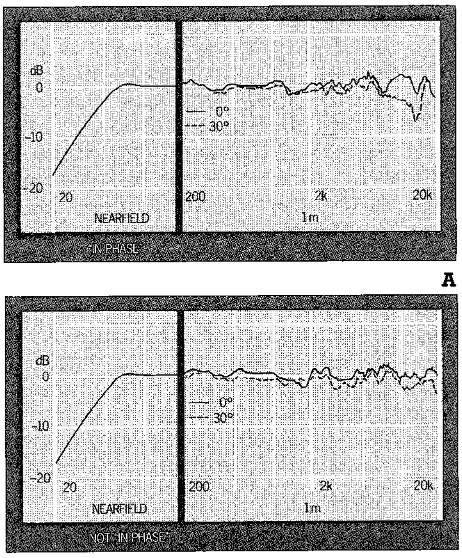


Fig. 3:
Frequency response 0° and 30° horizontal.

In the vertical direction, however, a bundled aggregate signal is obtained at the transition between the frequency ranges (dipole, Fig. 4). This effect cannot be prevented but it can be minimized and we can make sure that the direction of the lobe does not change with the frequency. When the beam bounces around in the room, the sound pattern is adversely affected, and it is difficult to locate the sound sources.

There are two possible remedies which can be employed individually or preferably in combination:

- The distance between adjacent chassis should be kept as small as possible. If the wave lengths (lambda) in the crossover range are much larger than the distance between the sound centers, no lobe at all is produced.
- The signal components of the low-pass and high-pass must be in phase. This should not be confused with phase linearity!

The second requirement is that the sum of all phase errors between two systems has to be brought to zero. Unfortunately it does not suffice to build an in-phase crossover network (e.g. Linkwitz-Riley filter) because additional errors in the signal path must be eliminated. These include delays of the sound center located nearer to the listener (Fig. 4, Tweeter) and the linearization of the individual chassis. Of course these statements apply only to the selected main radiation direction.

Correction of the chassis is achieved with negative impedance, as mentioned previously. The required delay times are generated by precision cascades of all-passes of the 4th order which must be very accurate (Figs. 5a,b). The bass/midrange crossover shows light deviations from the ideal, although the chassis spacing of approx. 21 cm is sufficiently small (at 300 Hz approx. 1.1 m). But the in-phase requirement has been sacrificed for phase linearity. On the other hand at these low frequencies we have to deal only with a lightly deformed spherical field rather than a narrow lobe. The influence on the sound coverage in the room is consequently small.

The situation in the midrange/treble area is far less problematic even though the wave lengths (approx. 11 cm at 3 kHz) are already very short. Despite the phase-linear filters it was possible to satisfy the in-phase requirement.

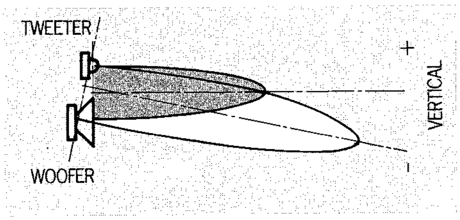


Fig. 4:
Radiation lobe.

Dispersion

The third prerequisite, dispersion, goes one step further and also requires that frequencies of the audible range arrive simultaneously. In other words the group delay should be constant.

Our $\sin^2$ pulse measurements on conventional speakers showed that pulses can be dispersed by up to 10 times their original duration and that totally different (new) amplitude conditions are created. This means that transient signals are difficult to recognize (Fig. 6a, b)! A large problem concerning all non phase-linearized multiway speaker systems is the leading harmonics which are readily perceivable as early echoes. As we shall soon see, the reason for this behavior can be found in the crossover network.

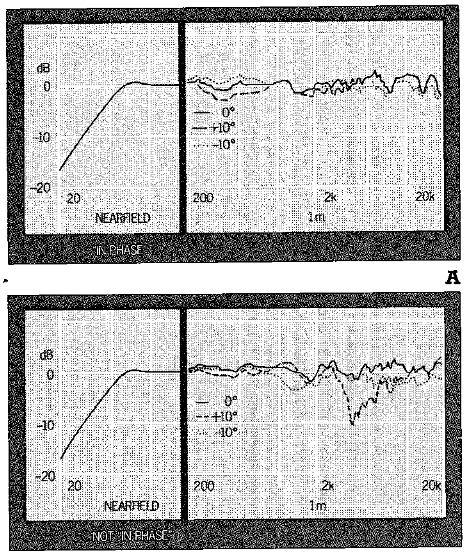


Fig. 5: Frequency response 0°, +10° and -10° vertical.

Phase-linear crossover networks are basically easy to build. However, problems occur if the edges of the crossover networks should also be steep (≥ 12 dB/oct) so that the chassis are effectively protected from frequencies they cannot transmit.

Generally all filters have a group delay that decreases at higher frequencies. At lower frequencies this delay is proportional to the filter cutoff frequency. In the case of a low-pass this means a delay in the pass range; in the high-pass, however, the delay in the pass range is nearly zero. This is the reason for the early echo.

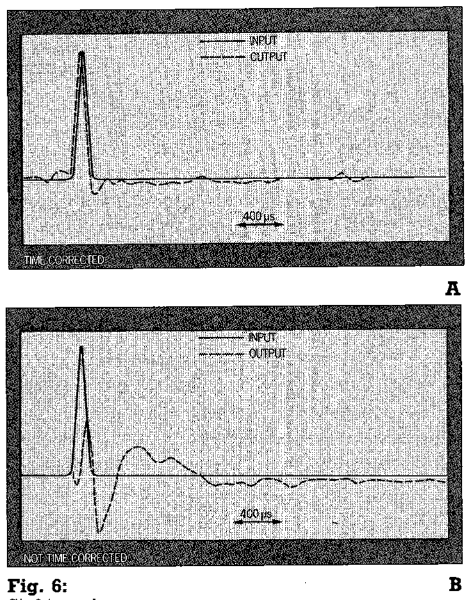
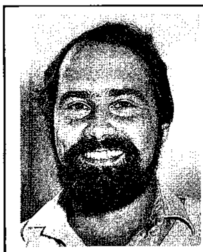


Fig. 6:
$\sin^2$ impulse.

The high frequency components must consequently be delayed by the delay of the low-pass so that they arrive at the listener at the same time as the bass frequencies. But this soon leads to an elaborate circuit design.

Example: a Butterworth low-pass of the 2nd order with a cutoff frequency of 300 Hz has a group delay of approx. 750 µs at 0 Hz. An all-pass of the 4th order with 0.05° ripple in an otherwise linear phase response achieves a delay of only approx. 66 µs at a bandwidth of 20 kHz. In order to compensate the low-pass, an all-pass of the 44th order would be required! However, only one of the crossover networks would now be corrected. Such an elaborate design is principally out of the question because its noise level is much too high. Unfortunately, equalization up to about 20 kHz is required, otherwise audible early echoes will occur.

After extensive tests we have found structures based on Lipschitz-Vanderkooy crossover networks (subtraction of the low-pass signal from the correspondingly delayed original) which despite the achievable delay times are phase linear and sufficiently steep.



Roger Schultheiss (35) studied electronics at the Swiss Institute of Technology (ETH) in Zurich. After his graduation in 1981 he joined Willi Studer AG. Since 1985 he has been working in the application laboratory where he is responsible for finding solutions to speaker problems. Latest assignment, project manager for the electronics development of the professional active monitor A723.

Conclusion

Based on the experience with conventional speakers in which the frequency response alignment is a mixture of trial and error, experience, and many listening sessions, we were uncertain for a long time as to how the severe constraints of the previously mentioned prerequisites would influence the actual sound experience. When the engineering work was finally completed, the sound fidelity, excellent locatability of the sound sources and pulse fidelity of the A723 convinced right from the start. This acoustical transducer will brutally reveal any recording flaws.

However, this is a characteristic much appreciated in a professional tool.

Roger Schultheiss

First digital IC from Studer

Portmaster

This is the name we have given to our first digital IC. Its function is to establish serial bus connections for fast and reliable data communication, particularly tape machines, or other similar control functions.

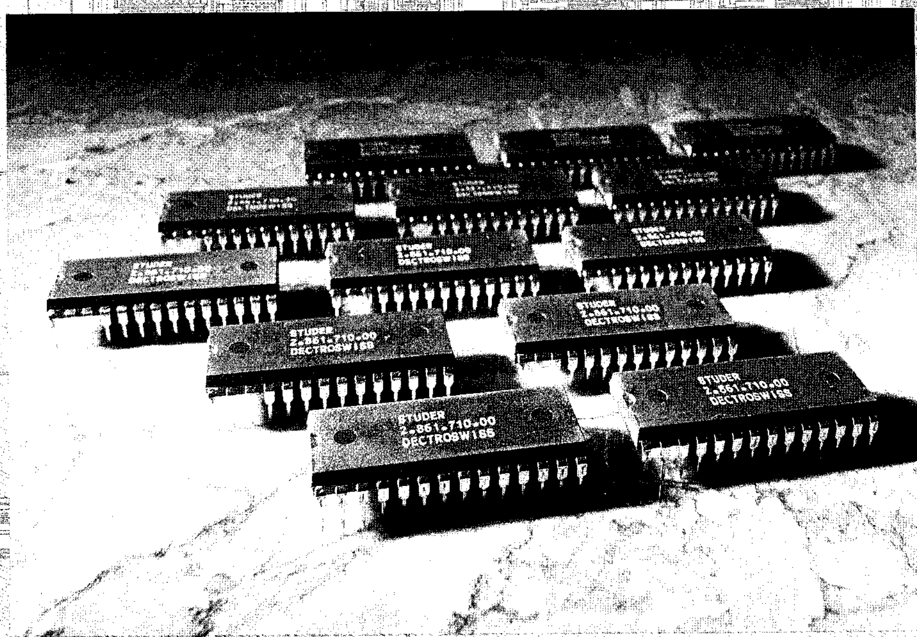
Already during the planning phase of the 48-track digital tape recorder it became clear that particular attention had to be given not only to the processing of digital audio data but also to the transmission of the complex control information. Large quantities of control commands, status indications, error messages, system parameters, etc. have to be read from or supplied to almost all circuit boards. In order to transport these data, a simple and reliable system had to be developed.

SERBUS

Although the serial bus was developed for the D-820 multichannel tape recorder, it is suited also for other larger systems. Its main advantage is that it can serve up to 64 device ports with only four differential signals (a device is normally a circuit board that receives or supplies control data). The bus can transmit data with a frequency of up to 2 MHz. It is operating by a central bus controller.

The PORTMASTER

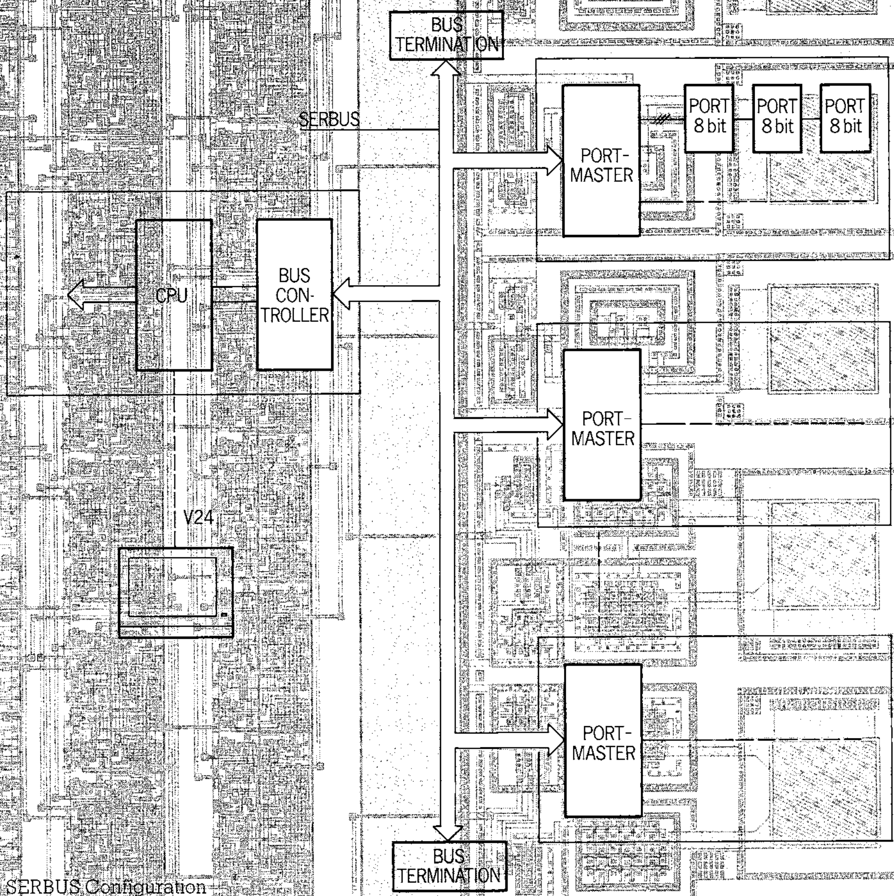
The Portmaster is a circuit that exists on each bus component as an interface. On the one side it is the communication partner for the bus controller and on the other it controls the fine distribution of the data on the corresponding circuit boards (see Fig. 1). When data communication is desired, the bus controller calls the corresponding Portmaster by its address, and in response the ports on the circuit board are through-connected to the bus.



These insignificant looking Studer CMOS-ICs for bus systems have been given the prosaic name 2.861.710.00.

First custom IC

Because of the expected frequent use of the PORTMASTER circuit (40 to 50 units per machine) and the potential use in other systems, integration of the circuit appeared to be justifiable. The plan was to produce a gate array or a standard cell IC. Without integration the same circuitry can be built with 7 ICs (of which 3 are PALs). The cost estimates in 1987 showed that for a planned quantity of 25,000 units a gate array would cost about the same as a standard cell IC. However, because the required bus drivers and receivers were not available in gate arrays, the standard cell solution was selected. In comparison to a non-integrated circuit, a cost reduction of 33% was projected. Not only has space been saved on the circuit boards, but also a greater reliability has been achieved.



SERBUS Configuration

Cooperation with DECTROSWISS SA

This young company domiciled in Neuchâtel / Switzerland specializes in the integration of complex circuits, and produces ICs as a general contractor. They promised us extensive support for the realization of our first IC project. This meant that we could carry out the circuit design and the complex logic simulation with the aid of HILO software ourselves in Neuchâtel and at the same time count on the patient support of the DECTROSWISS engineers. A special comparator with hysteresis and a powerful bus driver were newly developed to our specifications at no charge.

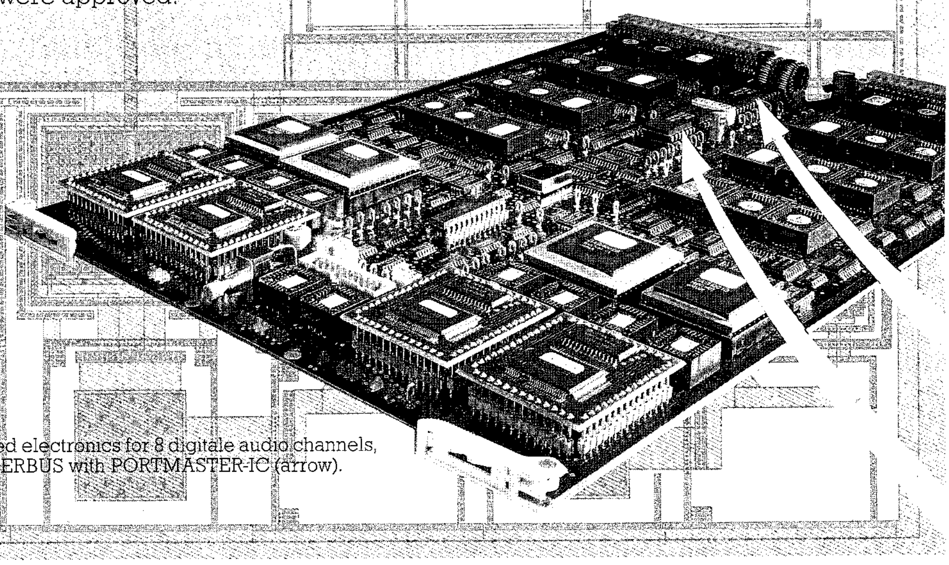
DECTROSWISS planned three months for this first phase. The layout, the mask preparation, the wafer production at Intermetall, packaging, and final testing of the 20 prototypes were scheduled for the second phase to which also three months were allocated. The ICs were produced in 2.2 μm n-trough CMOS technology.

All parties involved were most eager to discover whether the schedule could be met and whether the circuit would actually function.

The logic simulation revealed two problems that necessitated a minor circuit change. The simulations then of course had to be repeated. This «mishap» was one of the reasons why the 20 prototypes arrived in our factory with a delay of only 2 weeks relative to the 6-month time schedule. They subsequently underwent thorough testing and were approved.

This IC has since proven itself in many applications.

Hans-Peter Girsberger



Highly integrated electronics for 8 digitale audio channels, controlled via SERBUS with PORTMASTER-IC (arrow).

Robust method for data detection in the D820X

Run Processing

When Mr. Reeves filed his Pulse Code Modulation (PCM) patent in 1939, his aim was to achieve the most robust signal transmission possible. He proposed first to convert an analog signal into a suitable intermediate format, e.g. pulse-time or pulse-amplitude modulation, and subsequently to convert the time or amplitude steps into numbers. Numbers can be represented with only two signal states. They can be manipulated easily and protected for example with redundant information. Both approaches are certainly correct at the PCM system level. But this does not solve the problem of digital signal transmission as demonstrated in a report by the project manager of the D820X.

If we examine the microstructure of an individual information unit (bit) closely we again find a pulse-time modulation (pulses of identical height but different lengths). Numbers are consequently transmitted as a sequence of time-limited amplitude changes (Fig. 1). The intermediate shape described above which is supplemented by the PCM technique and consequently made more robust, reappears with all its problems.

What is a run?

The data bits generated by the A/D converter are mapped by a simple conversion process in such a way that they can easily transmitted or stored. This is accomplished by means of the so-called modulation code which takes the bandwidth of the corresponding channel (e.g. the tape) into consideration (channel coding). As can be seen from Fig. 1a, the information is contained in the intervals between two transitions. The DASH format defines seven different time intervals. These intervals are referred to as «runs» having the interval T, but only the discrete intervals from 3T to 9T occur. T itself is defined by the duration of a half bit of the original information (cells). The channel code prescribed by the DASH format has the designation HDM-1 (high-density modulation).

During the transmission the intervals are influenced by linear and nonlinear effects. Fig. 1b illustrates a signal that has evidently been modified by such influences after double differentiation. The effect is that the intervals between the transitions change. Fig. 1c shows that the transition timing no longer coincides with the one in Fig. 1. This effect is also referred to as peak shift. With the flux densities (1000 flux changes per mm) that occur in the DASH format, the intervals can be shifted so strongly that a run for example of 6T is received as 5T or 7T. It ist the responsibility of the receiver (in the tape recorder this is the reproduce section) to reconstruct the intervals between the transitions as accurately as required.

Run Processing

We shall first give a brief description of the principle behind the Run Processor (Fig. 4): With the aid of the detecor and the run counter the intervals between the signal transitions are measured on a representative system which in our case consists of tape, heads, record and reproduce electronics. The run lengths are expressed by numbers (Fig. 2). The ideal run length 3T is assigned to the value 36, 4T to the value 48, etc. The resolution is 12 units.

If we now plot the number of runs in the form of a histogram, we obtain the picture shown in Fig. 3. From this illustration we can see how many runs of a given quantized run length have arrived per unit of time.

One of the most important assumptions on which run processing is based is that the system to be corrected behaves in the same manner as most natural systems do, i. e. that a momentary event is influenced by a limited number of future and past events. For this reason we have further broken down the analysis of arriving runs and analyzed fifth order system. All combinations of pre-preceding, preceding, present and two future runs are represented in histograms. A system of the fifth order results in $7 \times 7 \times 6 \times 7 \times 7 = 14406$ combinations with the HDM-1 code.

Once the effects of the transmission system on each possible combination are known, it is possible to generate a correction factor by means of a look-up table in which the corresponding values are stored. This correction is added to the current run so that it can be restored to the original length.

The simplified block diagram of the Run Processor is illustrated in Fig. 5. Zero crossings of incoming signals are detected and converted by the counter from a time duration into a number. By means of suitable delay units («Z» blocks) it is possible to have the future, the present, and the preceding run simultaneously abailable at the input of a memory (decision table). This results in a specific address in memory (decision table in Fig. 5) where based on the analyses described above a run is stored that corresponds to the original one. We now have to deal with a structure of the third order. In the memory $7 \times 6 \times 7$ combinations are stored. The values in memory have been determined off-line with a structure of the fifth order to achieve greater accuracy.

(Patented process, European patent office, No. 0 148 413.)

Adaptive Run Processing

As mentioned above, the correction table contains only values of a typical system. In practice the tolerances to be expected can be large, for example for different tape types, heads, and variable tape speed. For this reason we decided to connect a similarly working but adaptive processor in series to a non-adaptive processor.

The principal task of the adaptive Run Processor is to set the thresholds between the individual run distributions exactly at the minimums (Fig. 3) in order to optimize the interval margin from distribution to distribution and consequently from run to run. The threshold values are updated adaptively. They are based on signal statistics which we shall explain. The criterion for updating is extracted from normal data. i.e. no test phase with special data is needed in order to obtain information on the quality of the adaptation and for correcting the latter.

Fig. 1:
a) Write signal.
b) Read signal after double differentiation.
c) Restored read signal showing peak shift.
Fig. 2:
Assignment of runs to counts.
Fig. 3:
Histogram of run distribution.
Fig. 4:
Non-adaptive run processing, simplified structure.
Fig.5:
Non-adaptive run processing, simplified block diagram.
Fig. 6:
Adaptive run processing, structure.
Fig. 7:
Adaptive run processing, simplified block diagram.

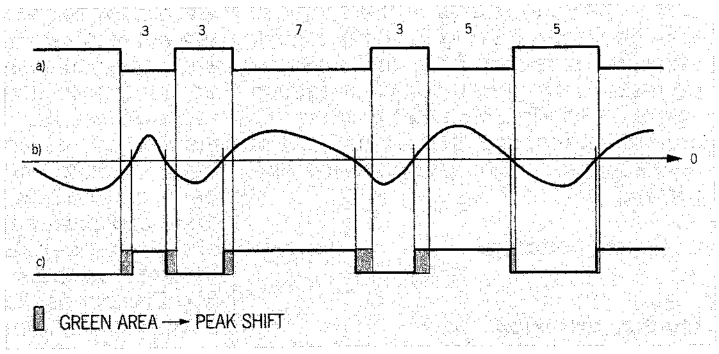


Fig. 1:

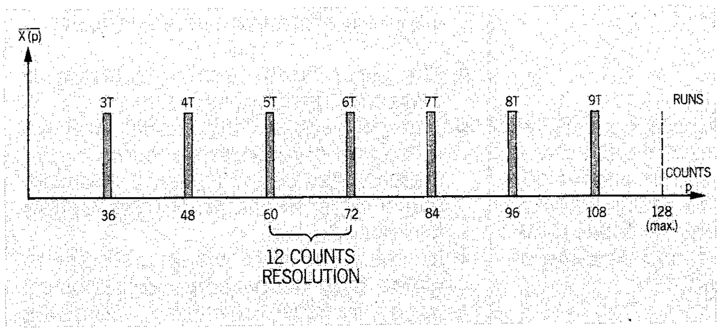


Fig. 2:

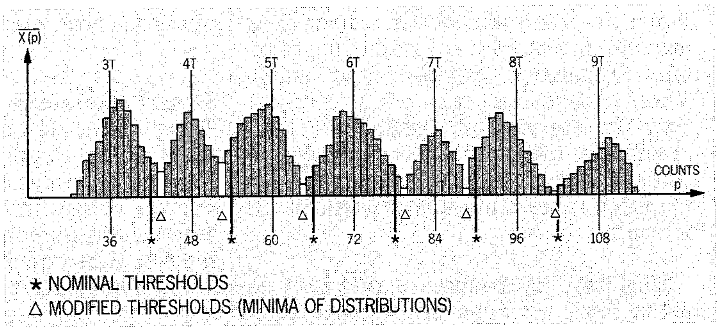


Fig. 3:

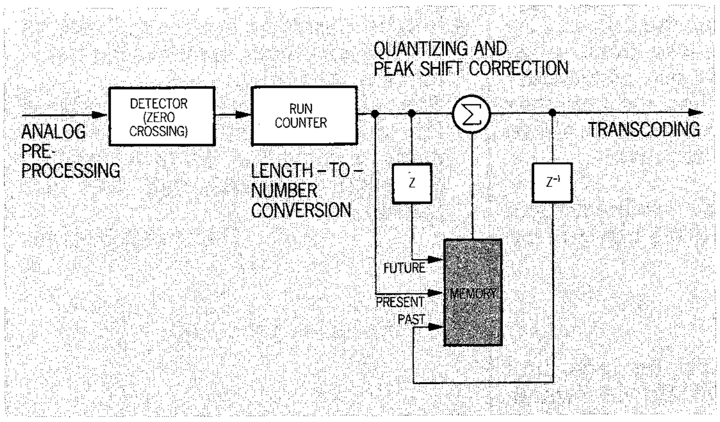


Fig. 4:

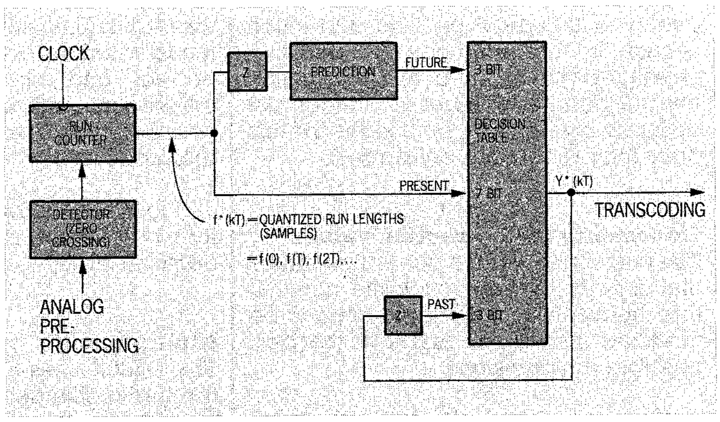


Fig. 5:

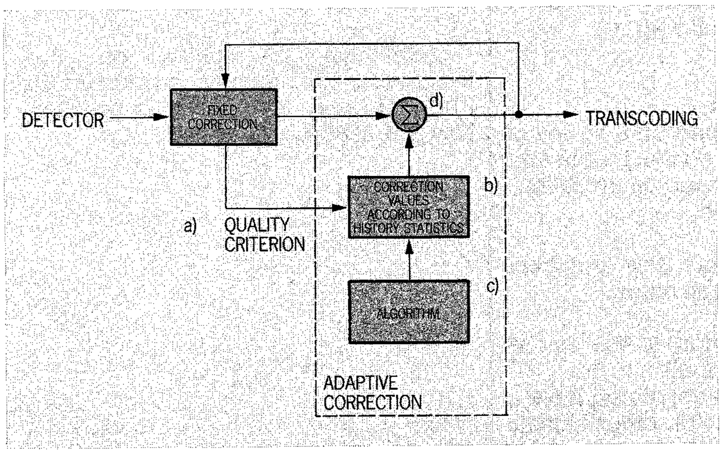


Fig. 6:

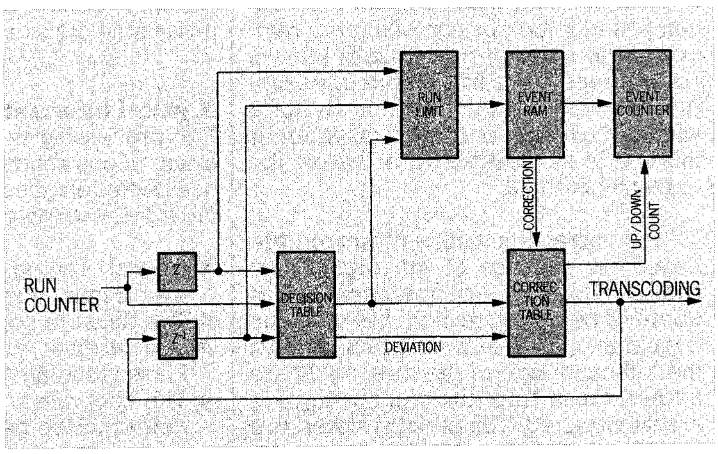


Fig. 7:

The basic elements of the adaptive Run Processor are (Fig. 6): a) the quality criterion extracted from the previous process, b) the generation of correction values, c) an algorithm that controls this generation function, and d) that part of the processor which performs the correction.

(Patent pending)

Quality criterion
Adaptive run processing is based on analyzing the statistical occurence of all events defined as «dangerous» runs. These are runs that occur adjacent to decision thresholds (± 1 step of the maximum resolution between the runs). When «dangerous» runs occur increasingly on one side of existing decision thresholds, the processor continues to shift its threshold until the same number of «dangerous» runs are measured on both sides.

The fixed correction or decision table in Fig. 7 contains predefined decision thresholds and supplies a provisional decision on the currently available run, as well as information on the quality of this decision. The correction section in turn modifies the decision threshold with the aid of this information until an identical number of «dangerous» events is obtained on both sides. At this point the process has converged.

Generating the correction values
The correction section generates values that specify by how much the nominal thresholds of the fixed table need to be modified in order to arrive at the best possible run decisions.

To perform this task it is necessary to convert the runs from the fixed table into threshold values. This can only be accomplised if the position of the run relative to the existing threshold is known. For this reason the fixed table must supply two items of information: a) that a «dangerous» run exists and b) whether this run is located above or below the current threshold.

The correction values produced are based on statistics of the digital sum values (DSV) of «dangerous» events above or below a threshold between run combinations. Events originating from the left-hand side of the thresholds are added, those from the right-hand side are substracted. The result is stored in a RAM (event RAM in Fig. 6).

Algorithm for controlling the correction values
The correction values described above do not immediately influence the correction table but only after a sufficient number of «dangerous» events have been counted and a clear trend has been determined. The modification is, therefore, based on a statistical analysis of past «dangerous» runs. If the DSV of «dangerous» runs in the event counter (Fig. 7) exceeds 128, a specific and possibly already modified decision threshold is shifted by one unit. The shift direction corresponds to the opposite side from which the excess number of «dangerous» runs have originated.

Signal processor
This function is performed by a ROM, used as an adder (correction table in Fig. 7). The inputs are the threshold correction values from the event RAM as well as information from the fixed table, the run to be corrected, and its distance to the nominal threshold.

From the addition of this distance to the correction factor we obtain the distance of the run to the momentary threshold. Only now can the decision be made whether the run was «dangerous» or not and should be statistically processed, and whether the run may possibly have to be modified if the above distance has become negative.

Fig. 7 also shows the feedback path for modified past runs to the inputs of the decision table.

Multiplexing
The D820X uses 8 tracks for recording two audio channels (DASH twin format). The fixed and the adaptive Run Processors are implemented only once in hardware and can be used independently in 8-way time division multiplexing.

A priori information
Run processing is based on a series of assumptions concerning the system and has been designed based on the following a priori knowledge:

- Causal system, only few adjacent events influence each other.
- The data are binary.
- The original duration of the run is known (coding / format).
- The intervals to be corrected have a deterministic behavior, other signals (e.g. noise, interference) cannot be corrected.
- Error propagation: limited to one block by the RLL code.
- Convergence must be achievable. This means that only one maximum per run distribution occurs and that the probability distribution of runs on both sides of the maximum progresses steadily, and that the correction range suffices in order to influence the actually occurring peak shift effectively.

Advantages of adaptive Run Processing
- The adaptation is performed continually, no learning phase is required.
- The error signal (quality criterion) is extracted from normal data.
- The process is fully digital and requires no service settings and alignments; it can be multiplexed (multiple utilization of the same hardware) and integrated.
- In comparison to other methods in which high frequencies are boosted (zero forcing equalizer, decision feedback equalizer), run processing achieves a superior carrier-to-noise ratio.
- Run processing can correct linear as well as non-linear errors of the transmission channel.
- The reference table can be modified for specific channels and can be made switchable so that strongly deviating channels can be processed.

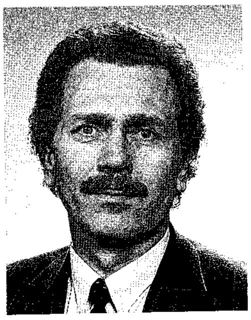
Marcel Schneider

Additional information on Run Processing can be found in preprint No. 2709 of the 85th AES convention, Nov. 3–6, 1988, Los Angeles.

The Studer-Group

«Who's who»

This column has been reserved for introduction of personalities of our companies and representatives in Europe and Overseas.



Tore B. Nordahl

President of Studer Revox America Inc., Nashville, Tennessee ● born 1944 and grew up in Oslo, Norway ● after high school graduation studied electronics at the Royal Norwegian Air Force Communications and Electronics College and graduated 1963 ● immigrated to North America in 1966 ● with Studer since October 1988.

In his career, Tore Nordahl has gained particularly wide experience in the audio industry of the U.S. market. From 1967 to 1971, he worked as design engineer and chief development engineer in the CATV industry, before engaging in sales and marketing activities. He became president of Neve North America in 1975, and also held the position of a board director of the Neve UK Holding Company. In 1983, Tore Nordahl initiated the Pro Audio Group Division of Mitsubishi Electric and served as division president and CEO.

In his new position as head of Studer Revox America, Inc., he sees his foremost challenge in associating Studer to an increasing degree with the requirements of the U.S. professional audio market, to guarantee a steady flow of information to Studer on today's and future market trends as well as making the U.S. professional audio market recognize even more Studer's capabilities of bringing superior solutions to the clients' needs.

Going back to the early days of SRA's existence, it was rough going for the young organization. The products introduced to the North American market were of exclusive European make, produced in limited numbers and consequently more expensive than others; in addition, certain products in the accessory field - a must in the American market - were often not available. The break-through came when organisational changes produced their effect and a complete range of studio products was introduced, including the Studer A800 multichannel machine. In spite of severe international competition, STUDER REVOX attained a major position in the U.S. market - not only a challenge, but also a reward for extra hard work, good strategy and service.

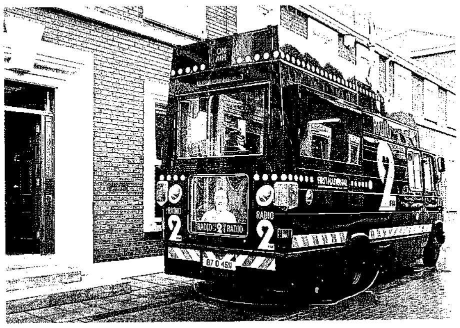
There is still a lot to be done to develop existing market opportunities. The special priority of SRA's new president is to increase the market share of Studer Revox in the U.S.A., by making the sales force more responsive to market requirements, and to fine tune the entire organization to support sales staff and customers. The mobilization of all existing staff capacity makes SRA successful against a strong competitive background.

Tore Nordahl's engagement in business does not leave much time for hobbies, however, from time to time, he gets involved in home building projects - his favourite pastime activity.

Asked about his business principles, Tore Nordahl emphasizes that «hard work will be rewarded in many ways; don't expect your staff to work more than you do; stand behind commitments and promises; tell it the way it is.»

Renate Ziemann

Studer DJ Console «on air» in Eire



A Studer 970 series has scored a notable first in Irish Broadcasting. The equipment was recently the centrepiece of an experiment by Radio Telefis Eireann to transmit a regular schedule from the office of the Studer supplier in Dublin. To facilitate the on-site use of the equipment, the Studer representative office in Dublin effectively became a studio for three hours of national transmission on Radio 2, the «pop» channel of the Irish Republic.

Chosen to field-test the Studer mixing console was «The Larry Gogan Show», broadcast every afternoon, which has a high following among fans of popular music. It was Gogan's first time to use the Studer 970 series, with its many refined features. Ireland's top DJ expressed himself pleased with his new «pilot seat». Also pleased with the exercise was Joseph Buckley, Managing Director of Leatech Limited, sole distributors of Studer Revox in the Irish Republic, who worked closely with broadcast technicians in the preparation and supervision of the transmission. Lines were run directly from the 970 to the O.B. van parked outside; offices became broadcast studio for the day.

Considering the present changes in broadcasting where a network of commercial radio stations are prepared to compete with the established State services (four channels for 4 million listeners), it is assumed that by the end of 1989, about twenty new regional services will be on air, catering for local communities. «Most of the new commercial stations will have a major music output», says Buckley. «With the Studer 970 Series, we have shown broadcasters - and listeners - what can be done.»

Joe Buckley

New Patents

Method and equipment for synchronizing sampling frequencies.

Analogous signals are sampled with a predefined frequency and converted to digital signals. When such signals are reprocessed, such as in a mixdown operation, there is a problem that the two signals have approximately the same sampling frequency but come from different sources. For processing, these signals should be in absolute synchronism but this can only be achieved when the signal generators can be synchronized with each other. In most situations this is not the case.

Synchronism can be forced by declaring approximately equal sampling frequencies as synchronous and by dropping excess signals or making up missig signals through duplication. As a result of this process the reproduction quality declines.

The patent method eliminates this disadvantage. A special algorithm shifts and interpolates the values in such a way that no quality loss occurs. The effort to implement this process is minimal.

This patent award to Dr. Roger Lagadec was registered with the U.S. Patent Office under the number 4,780,892 on October 25, 1988.

Method and equipment for reproducing digitized signals that are transmitted as binary signals in the form of pulses.

Recording and reproduction of signals are analog processes. In contrast to the digital/analog conversion which is trivial, the complementary process (analog/digital) is much more difficult.

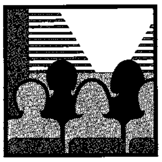
An analog signal with a more or less well recognizable pulse shape must be scanned (digitized) both with respect to amplitude as well as time. The amplitude normalization can be accomplished with a simple method, e.g. limitation, but the time moments of the zero crossings must be compared with a flywheel cycle. Minor deviations are rounded.

In the processing of analog signals, typical time errors can creep in. For example a zero crossing can be quasi shifted by the immediate surrounding. To prevent such errors is a difficult task for the development engineer. The corresponding circuitry is sophistcated. In addition a different implementation is required for each tape speed.

The patent method maintains statistics on how the zero crossings occur, taking into consideration the signal environment. Because the nominal times are known, a correction variable can be determined with which the shift in zero crossings can be compensated.

This invention by Julien Piot and Dr. Roger Lagadec was registered with the European Patent Office under the number 0148 413, on June 22, 1988.

Paul Zwicky



Studer Training courses

04. 09. - 06. 09. 89 German
Mo 09.00 h - Mi 12.30 h
A807 Tonbandmaschine
Laufwerkfunktionen, Demontage/Montage des Laufwerkes, Geräteeinstellungen, Schnittstellen, Schaltungserklärungen, Fehlerbehebung.

06. 09. - 08. 09. 89 German
Mi 13.45 h - Fr 16.00 h
A810 Tonbandmaschine
Laufwerkfunktionen, Demontage/Montage des Laufwerkes, Geräteeinstellungen, Schnittstellen, Schaltungserklärungen, Fehlerbehebung.

18. 09. - 20. 09. 89 German
Mo 09.00 h - Mi 16.00 h
Digital Editor 4003 / PQ Editor LHH 3050 / 3055
Anwendung und Bedienung, Serviceaspekte.

02. 10. - 06. 10. 89 German
Mo 09.00 h - Fr 16.00 h
Kombinierter Kurs über:
A812 ¼" / A820 ¼" Tonbandmaschinen
Laufwerkfunktionen, Demontage/Montage des Laufwerkes, Geräteeinstellungen, Schnittstellen, Schaltungserklärungen, Fehlerbehebung.
Hinweis: Auch die Schulung nur eines Gerätetypes erfordert die Teilnahme während der ganzen Kurswoche.

09. 10. - 13. 10. 89 German
Mo 09.00 h - Fr 16.00 h
D820X Tonbandmaschine
Bedienung, Anwendung, Schaltungs- und Schnittstellenerklärungen.
Wichtig
Kenntnisse des Laufwerkes A820 ¼" / D820X werden vorausgesetzt.

23. 10. - 25. 10. 89 German
Mo 09.00 h - Mi 16.00 h
TLS 4000 Synchronizer
Funktionen und Bedienung, Anwendungen, Schaltungserklärungen, Interfaces, Fehlerbehebung.

26. 10. 89 German
Do 09.00 h - 16.45 h
SC 4008 Controller
Funktion und Bedienung, Anwendung, Schaltungserklärungen.

27. 10. 89
Fr 09.00 h - 16.45 h
SC 4016 Controller
Funktion und Bedienung, Anwendung, Schaltungshinweise.

06. 11. - 08. 11. 89 German
Mo 09.00 h - Mi 16.00 h
963 / 970 Mischpulte
Anwendung, Bedienung der Module, Schaltungserklärungen, Einmessvorgang, Fehlerbehebung.

13. 11. - 17. 11. 89 German
Mo 09.00 h - Fr 16.00 h
A820 Mehrkanal-Tonbandmaschine
Laufwerkfunktionen, Demontage/Montage des Laufwerkes, Geräteeinstellung, Schnittstellen, Erklärung der einzelnen Platinen, Fehlerbehebung.

04. 12. - 05. 12. 89 German
Mo 09.00 h - Di 12.30 h
A727 CD-Spieler
Anwendung, Schaltungserklärungen, Laufwerkfunktionen, Fehlerbehebung.

05. 12. - 06. 12. 89 German
Di 13.45 h - Mi 16.45 h
A730 CD-Spieler
Anwendung, Schaltungserklärungen, Laufwerkfunktionen, Fehlerbehebung.

Die Kurse werden nur bei einer Mindest-Teilnehmerzahl von fünf Personen durchgeführt. Alle Kurse setzen gute Grundkenntnisse in Elektronik voraus.

Editor:
Marcel Siegenthaler
Contributors of this issue:
Joe Buckley, Hans-Peter Girsberger, Marcel Schneider, Roger Schultheiss, Renate Ziemann, Paul Zwicky
Please mail your letters to:
SWISS SOUND, STUDER INTERNATIONAL AG
Althardstr. 10, CH-8105 Regensdorf
Telephone (+411) 840 29 60 · Telex 58 489 stui ch
Telefax (+411) 840 47 37 (CCITT 3/2)
Artwork: Lorenz Schneider
Publisher: Willi Studer AG, Althardstr. 30
CH-8105 Regensdorf

Printed in Switzerland by WILLI STUDER AG
10.23.8210 (Ed. 0489)