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SWISS SOUND

NEWS AND VIEWS FROM SWITZERLAND

Editorial

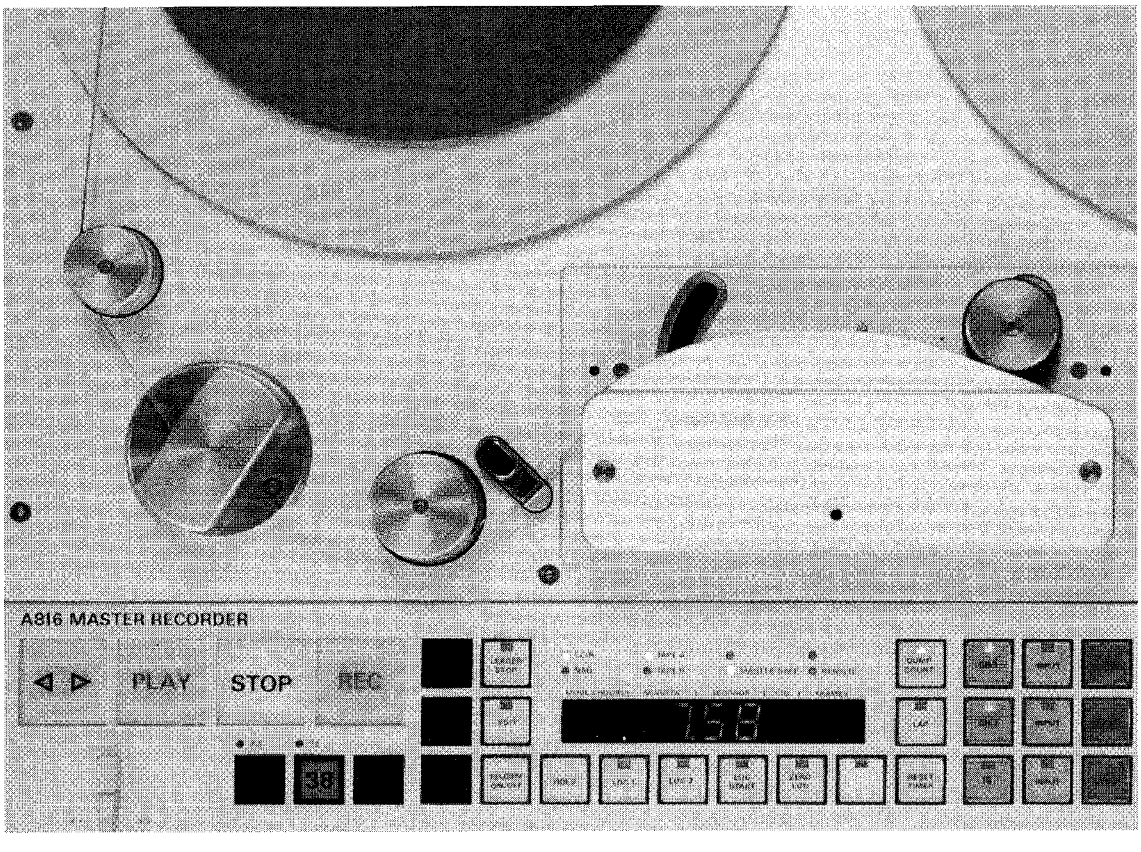
Ten years ago there were intensive discussions at Studer International on the idea of publishing a magazine "to provide information to our increasing number of customers worldwide based on what is happening inside the Studer Revox Group" ... as Eugen Spörri formulated his thoughts in his editorial to the first edition. All employees from sales and advertising/PR were enthusiastic about this project. But it soon became apparent that there is a significant difference between formulating an idea and the regular publication of a technical/product oriented magazine. Stamina was needed, because it was not always easy to set aside the required time in addition to the regular work. This applied and still applies in particular to the authors in research and product engineering, all of whom have contributed voluntarily and without any special remuneration. In the name of the management of Studer Revox AG, the publisher of SWISS SOUND, I would like to express my sincere thanks to all authors. The cooperation has remained excellent throughout these years.

I would also like to thank the many readers who in the form of appreciative letters, questions and suggestions have helped us establish a firm place for Swiss Sound among the plethora of professional magazines. As a particularly nice example, this edition contains a letter that we received from Albania.

In addition to other customer contributions this edition contains the previously announced second part of the comprehensive report on the D820 MCH digital multichannel tape recorder.

Thank you, dear readers, for your attentiveness and we hope that you will continue to obtain much inspiration by reading SWISS SOUND.

Your Marcel Siegenthaler



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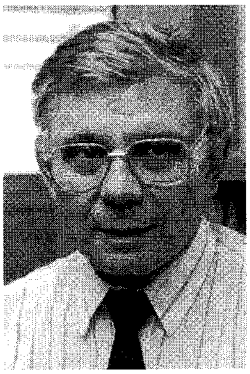
STUDER D820 multichannel

Antialiasing filter

by Paul Zwicky



Paul Zwicky

If analog audio signals are to be converted to digital signals, the momentary deflections of the analog signal must be measured in a sufficiently fast rhythm. In DASH machines, for example, the "snapshots" are taken at intervals of 20.83 µs, which corresponds to a sampling frequency f_S of 48 kHz.

The sampling operation does not produce a unique result. The sampling process is similar to the mixing operation we know from superheterodyne receivers. At both sides of the sampling frequency, so-called sidebands occur that are as wide as the low-frequency band.

Fig. 1 clearly shows that the frequency range of the audio band should not go beyond $f_S/2$, otherwise the bands would cross. Aliasing would occur during playback. For this reason the audio range has to be accurately clipped. This is the purpose of the antialiasing filter.

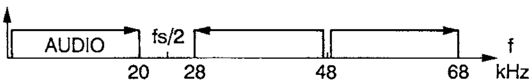


Fig. 1: *Sampled frequency spectrum.*

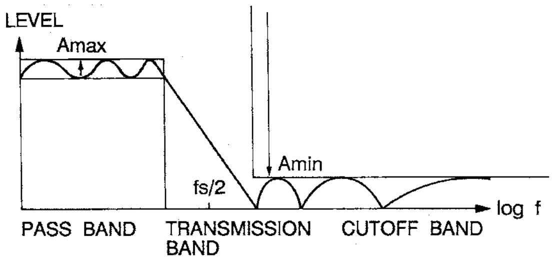


Fig. 2: *Filter bands.*

Requirements

A filter design is characterized by three bands (Fig. 2). The *pass band* should extend to max. 20 kHz, and the deviation from the linear frequency response should typically not exceed 0.1 dB.

The *cutoff band* starts at 28 kHz. Since the human ear does not perceive parasitic signals below –60 dB (masking), a cutoff attenuation A_{min} of 60 dB is adequate.

In the *transmission band* the antialiasing filters of the record and reproduce circuits share the work. It suffices if the attenuation at $f_S/2$ is = 24 kHz is approx. 30 dB. The side effects of the filter should be negligible, i.e. noise and unwanted spectral lines should not impair the system. S/N and THD should be better than –92 dB.

Every filter has a signal delay. Care must be taken that the delay is the same for all frequencies. For analog filters this is only approximately possible. The compromise must be selected in such a way that the filter sounds "good".

Filter type

The relatively stringent requirements with respect to a clearly defined transition between pass band and cutoff band, can be satisfied with little effort if so-called elliptical filters are used. According to the results of a preliminary study, a filter of the 7th order suffices. According to Saal [1] a type C 0715/48 would be suitable.

Passive filters?

The table values for passive filters always assume that the reactive elements are loss-free. This may largely apply to capacitors but not to coils. Some basic rules specify how high the coil quality has to be. If ferrite cores with low shearing are used, the achievable coil quality is sufficiently high so that the deviation form the theoretical frequency response can be tolerated. However, such filters produce inadmissibly high non-linear distortions. This is caused by the non-linear behavior of the ferrite. The situation can only be improved by giving the core a large air gap (shearing). As a result, more windings are needed and the resulting higher coil resistance reduces the quality to inadmissible values. The situation can be salvaged by using large cores. Excellent values can be achieved with the size RM12. Three coils of this type weigh approx. 300 g (Fig. 3). For a 48-

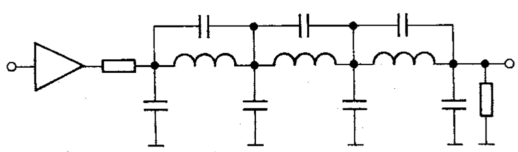


Fig. 3: *A passive filter leads to voluminous coils.*

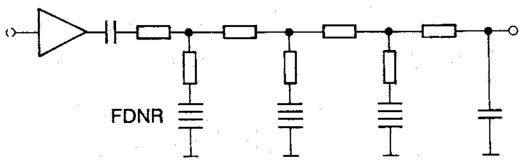


Fig. 4: *Topology with FDNR - consumes much power and is inaccurate.*

track machine these coils would weigh 14 kg. No wonder that an active solution was sought.

Active filters?

The above "passive" solution comprising ladder filters has one significant advantage: The sensitivity of the frequency response towards manufacturing tolerances of the components is very low. The obvious solution is to adopt the structure and to implement it in an active form. The division of all impedances by $j\omega$ leads to filters with FDNR (frequency dependent negative resistor). The design was based on [5] and [4] respectively (Fig. 4).

The results with respect to frequency response, distortion and S/N ratio are good. In addition this results in a compact design. However, certain disadvantages induced us to search for other solutions:

– *Power consumption*
11 opamps (including phase correction) of 6 mA on ± 18 V result in a consumption of 114 W for 48 channels. If we rate the power supply efficiency as ≈ 50%, a machine would draw 228 W from the mains just for the filters and convert this energy to heat.

– *Reproducibility*
An FDNR filter features a large number of components that are available only in limited precision. If expensive alignment work is to be avoided, these tolerances will have to be accepted. A Monte Carlo analysis demonstrates that with commercially available components a reproducibility within 0.2 dB would not be feasible.

Final solution: Return to the passive filter

Let us recall why the implementation of the passive filter was initially abandoned: The high distortion forced the core to shear, which meant a high number of turns, loss through copper, and ultimately intolerable frequency response errors.

Can these errors be equalized? Nai-T Ming [7] demonstrated the filter size required for reactive elements with a uniform loss factor. Engineers who are familiar with this filter design know that the poles and zero points of the transmission functions can be represented as locations in the complex frequency level. If each element has a loss factor of tan δ, the coordinate cross appears to rotate by the angle δ (Fig. 5). Through a coordinate transformation the new pole and zero positions can be calculated. From this we can determine the values of the elements. In [6] this work has already been performed and the results presented in the form of tables.

The implementation according to [6] with a structure according to Fig. 3 is strictly routine work. In addition to the three coils, 7 precision capacitors are needed. This can be prevented by a transformation (described in [1]), see Fig. 6.

By taking the coil losses into consideration for the filter design, we are able to use coils with relatively high loss factors. In our solution we use three RM5 cores that weigh only approx. 5 g each. For 48 channels the total weight is only 720 g. However, we should not conceal the fact that many versions had to be calculated and built before the distortions were sufficiently low and the frequency response error was negligible.

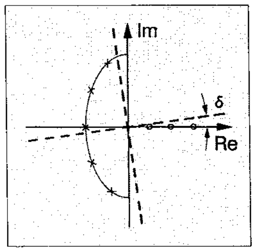


Fig. 5: *Gyration of the coordinate axes takes into consideration the loss factor tan δ.*

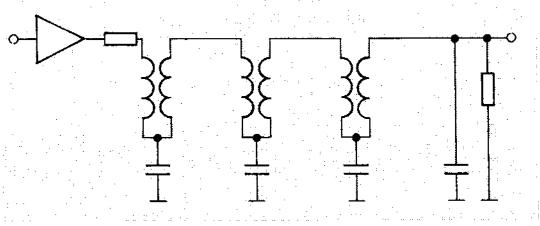


Fig. 6: *Transformation requires three additional windings and eliminates three capacitors.*

Phase correction

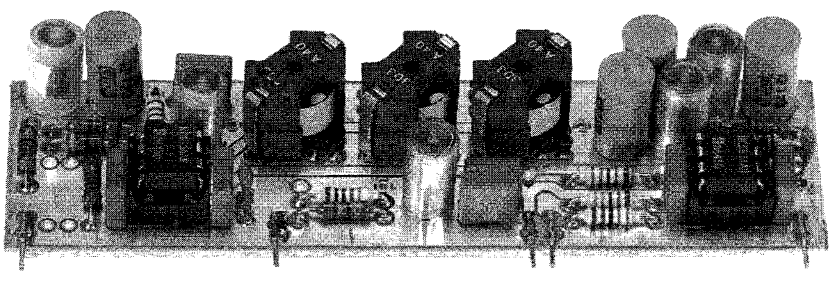
Filters of the type discussed so far do not have an ideal group delay. Frequencies near the response limit are delayed more than other frequencies. This phenomenon is called dispersion. It is well-known that the human ear is less sensitive to lagging oscillations than to leading oscillations (masking). There are methods for reversing such dispersions. However, before such a costly solution is chosen, it is important to investigate to what the human ear actually responds. Is it the phase response, the group delay or the pulse response? Which quantity is to be optimized? The optimization of the pulse response is described in [8]. The authors see their findings applicable to video signals. Based on our experience in the development and design of studio monitors we know that the pulse response is a critical factor also for the human ear. With the aid of a 2-pole allpass we have optimized the pulse response in such a way that the overshoot of square-wave signals is symmetrical.

Results

The circuit (Fig. 7) has been implemented as a module. It exhibits excellent characteristics. Highly acclaimed is the "sound" of the filter. The module is not difficult to manufacture.

The typical frequency response for the passband and the cutoff band is shown in Fig. 8.

Antialiasing filter including delay compensation.



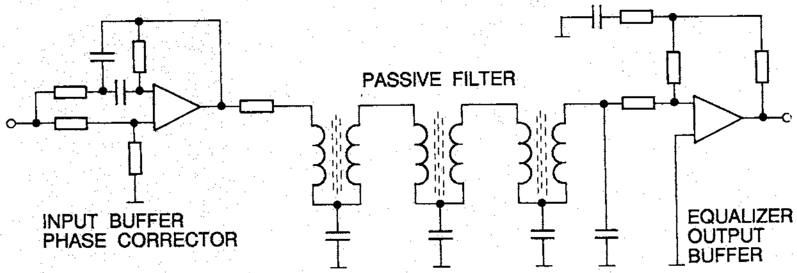


Fig. 7: Schematic circuit diagram of the implemented filter.

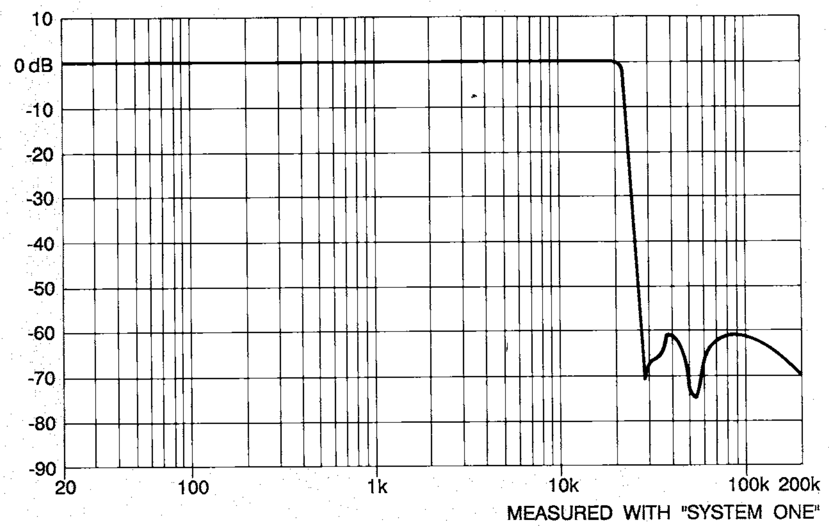


Fig. 8: Frequency response of the filter, from the analog input to the A/D converter.

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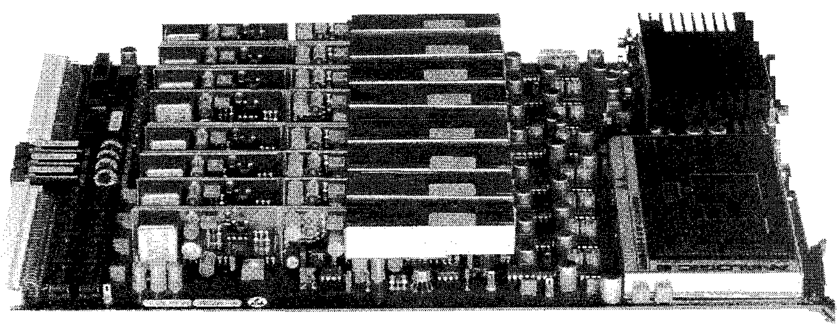
8-Channel circuit board, analog input to A/D converter.



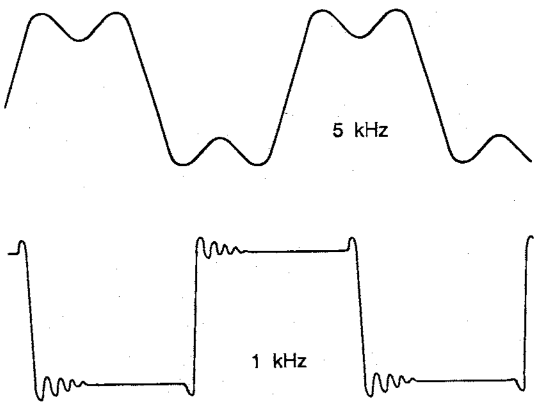


Fig. 9: Filter response to square-wave signals.

Fig. 9 illustrates the responses to 1 kHz and 5 kHz square-wave signals. Striking is the coincidence of the maximum overshoots. The ringing has a frequency of approx. 22 kHz and is consequently outside the audible range. In addition they are masked by the leading edge.

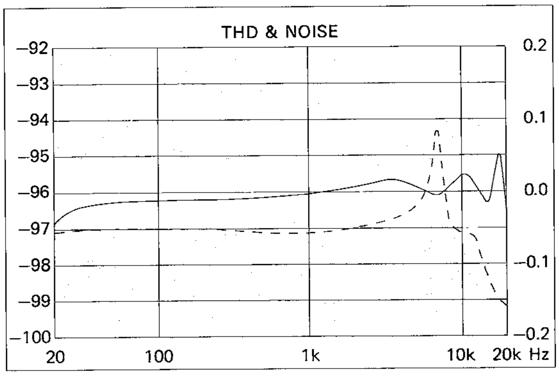


Fig. 10: Frequency response tolerance, THD and noise.

Fig. 10 illustrates the frequency response in the passband again. The manufacturing tolerances are also specified. This high accuracy is a precondition for achieving an overall tolerance of ±0.3 dB. It also shows that the filter has a signal-to-noise ratio of 97 dB. The maximum distortion occurs at 7 kHz and is better than –94 dB. The guaranteed value for noise and THD on the D820 MCH specification sheet is –85 dB.

Summary

Passive filters are well suited as antialiasing filters. The sensitivity to component tolerances is low. Power consumption and background noise are low. High weight and high distortion factors have been avoided by a design that takes into consideration the influence of low coil qualities. The dispersion has been optimized with an allpass to the sensitivity of the human ear. ●

STUDER D820 Multichannel

The equalizer

by Matthias Zbinden

When recording and reproducing digital audio signals, we are confronted with the problem that a relatively large amount of data must be transmitted in a short period of time. With a sampling frequency f_S of 48 kHz, the analog audio signal is scanned every 20.8 µs and represented as a 16-bit digital signal. This means that within 20.8 µs all 16 bits have to be recorded on tape. For each bit this gives a time slice of 1.3 µs which corresponds to a maximum frequency of 384 kHz.

By using digital tape and narrow-gap heads, a maximum frequency of approx. 200 kHz can be achieved at 30 ips tape speed. Although this is a remarkable value, it does not suffice for transmitting audio data.

For this reason the frequency must be increased by electronic means. Actually this is nothing new since similar circuits have been used in analog tape recorders for many years. But in digital audio technology the requirements are more stringent because the data are stored as a sequence of polarity changes, and the information is embedded in the interval between adjacent edges. For this reason the frequency response correction should be as linear as possible, otherwise time errors occur that will impair the correct reproduction of the recorded data.

In the digital machine the frequency response correction stages are referred to as equalizers because they compensate the amplitude loss above 200 kHz by boosting the higher frequencies.

This is normally accomplished with a delay line, but for the D820 MCH a totally different approach was chosen. The solution is based on the fact that an ideal phase-linear treble boost can be achieved with a straightforward series resonant circuit.

In order to explain the operating principle of the equalizer we shall briefly digress into the theory of filters. A series resonant circuit comprises 3 elements, namely an inductor (L), a capacitor (C), and a resistor (R). When a signal is applied to a series resonant circuit, a frequency-dependent voltage drop occurs at each of these elements. As can be seen in Fig. 1, the voltage across the capacitor (UC) is a low pass, the voltage across the resistor (UR) is a band pass, and the voltage across the inductor (UL) is a high pass.

The corresponding frequency response shows that all components basically have the same phasing. They differ only in their constant phase shift of 90° and 180° respectively. This constant phase difference is the key to the implementation of the equalizer. The components UC and UL have a phase difference of exactly 180°. With an invertor it is now possible to rotate the phase of one of these two signals by 180°. In this way both components have the same phasing. Two signals with identical phasing are now available that differ considerably with respect to their amplitude response. When UL is amplified and added to UC, we obtain the desired phase-linear treble boost (see Fig. 2).

Matthias Zbinden

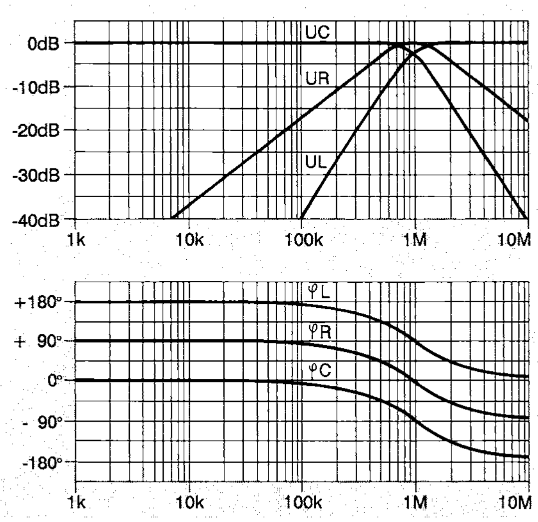


Fig. 1: *Amplitude and phase of the components in the series resonant circuit.*

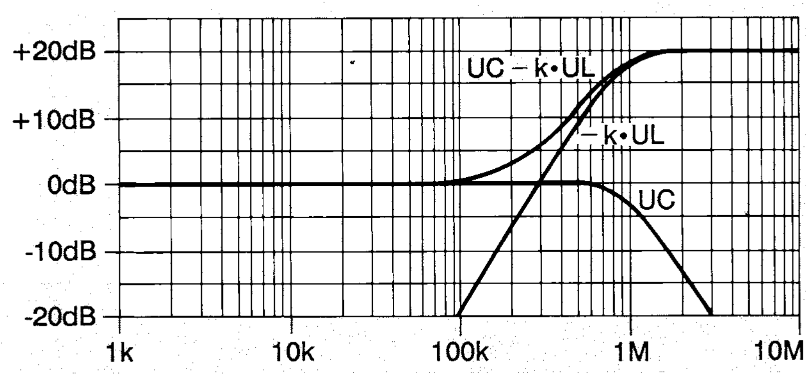


Fig. 2: *Addition of UC and -k*UL.*

Fortunately the steepness of the amplitude rise corresponds almost exactly to the drop to be compensated. It can be demonstrated that through this method the resulting frequency response can be extended by approximately one octave. In this way it is possible to transmit frequencies of up to approx. 400 kHz via tape. The bandwidth required for a digital recording is thus achieved.

Practical implementation

The practical implementation of the principles discussed above is relatively simple. The input signal is applied to a series resonant circuit that effects the split into the two components (Fig. 3).

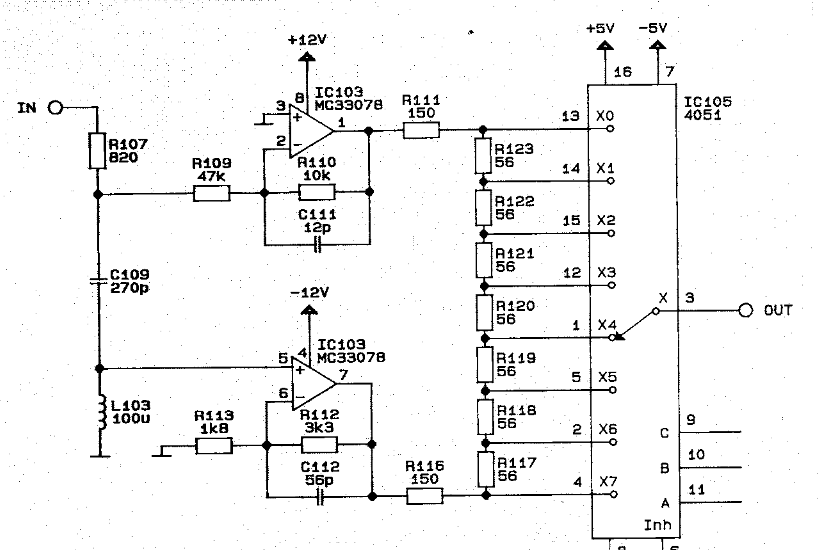


Fig. 3: *Equalizer circuit diagram.*

The voltage UL is tapped across the inductor by an opamp and amplified. Somewhat more difficult is the situation with the voltage UC because neither capacitor terminal is connected to ground potential. This means that a differential amplifier would have to be used. In practice, however, this is not necessary, because the voltage at the upper capacitor terminal can also be used. In the frequency range of interest (below the resonant frequency of the series resonant circuit), it hardly differs from UC. This voltage is tapped with a second opamp, inverted, and attenuated.

Since opamps cause a signal delay at frequencies of 400 kHz, both stages have to be adjusted to the same group delay by means of feedback capacitors.

The output signals of the two opamps can now be added. The simplest solution is to use a potentiometer through which the attack point of the treble boost can be adjusted. If the gain of the opamps is set to the same peak amplitude, the equalizer output voltage remains the same for each potentiometer setting. This is highly advantageous for the subsequent data detection.

In the D820 MCH, however, the equalizer is not adjusted with a potentiometer but with a resistor network that is tapped with an analog switch. In this way the treble boost can be software controlled in eight steps.

The strong treble boost of the frequencies up to 1 MHz has the disadvantage that noise and high-frequency parasitic signals are also amplified. This is the reason why a lowpass of the 4th order is connected to the input of the equalizer. This lowpass eliminates the treble boost above the required bandwidth. Fig. 4 shows the measured frequency responses of the equalizer module (including lowpass filter) for all eight switch settings. The phase response is exactly the same for all eight steps. This shows that in this implementation the phase response is not influenced by the treble boost.

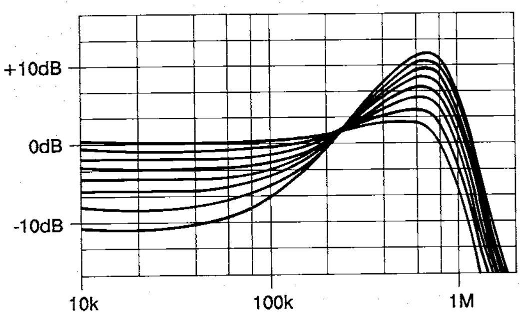
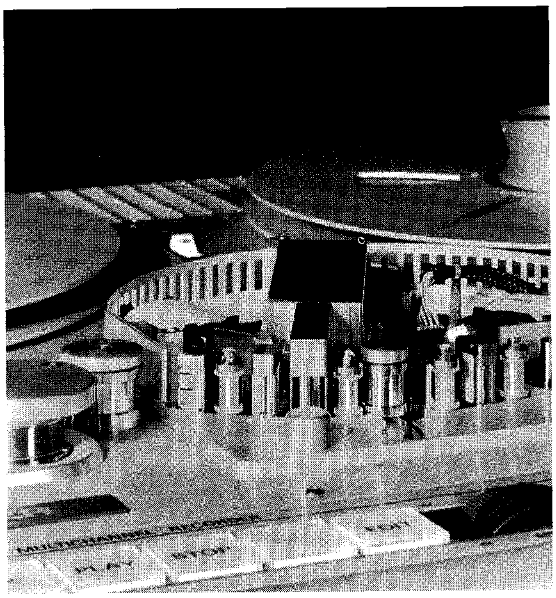


Fig. 4: *Frequency response of the equalizer module*

In conclusion we should mention that the implementation of circuits with wideband analog stages and subsequent digital electronics also requires an optimum circuit board layout. At this point I would like to thank Mr R. Greutmann for his invaluable help in the development and implementation of the D820 MCH reproduce electronics. ●



STUDER D820 multichannel

Internal synchronizer

by Kurt Schwendener

The D820 MCH is the first STUDER machine to have been equipped with an internal synchronizer. This device is used for coupling the tape deck with an external reference by means of SMPTE/ EBU time code or RT signals.

In contrast to an external synchronizer, this concept achieves a much more efficient allocation of the functions to be performed. For example the existing tape deck and capstan modules perform those functions that fall within their own sphere. The master CPU coordinates the synchronization process.

Design of the synchronizer board

The synchronizer board (SSTC board) is designed as a modular plug-in unit and is integrated in the machine structure as a device on the FIFO bus. A TC generator is piggy-back mounted on the SSTC board as an additional function unit.

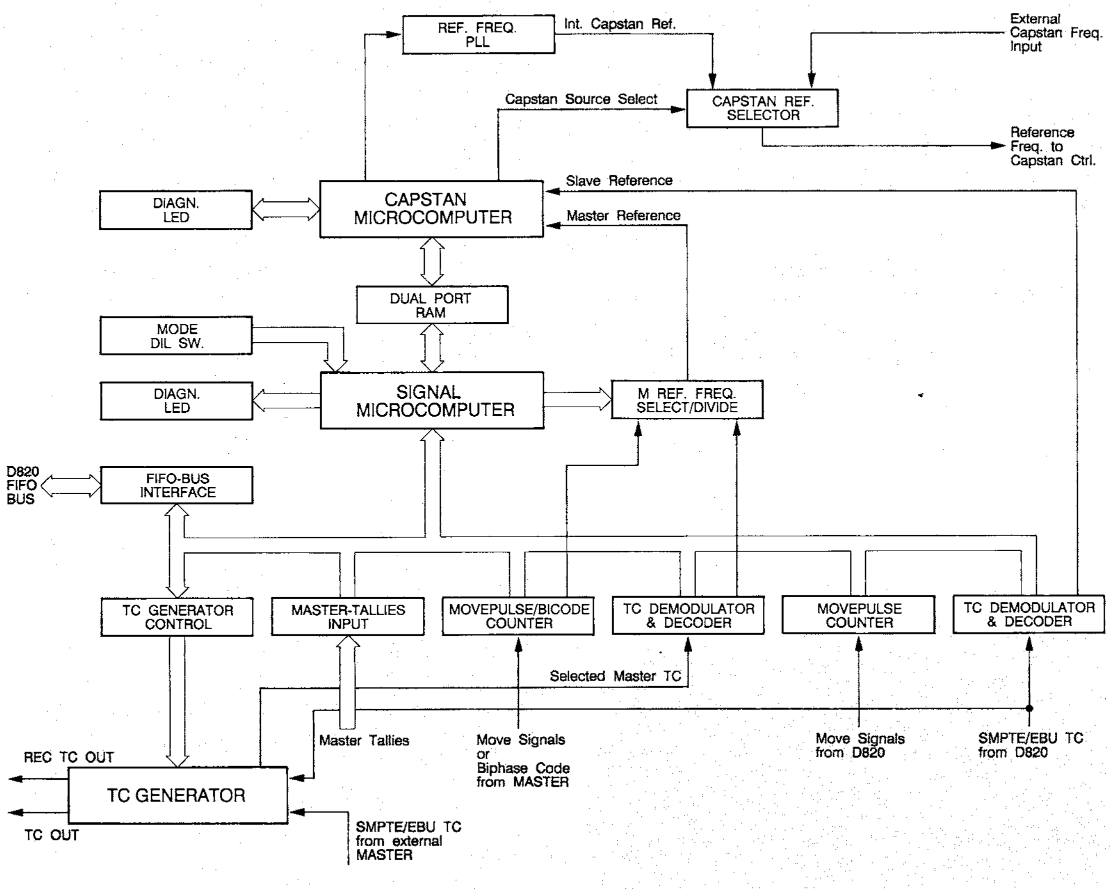
The block diagram shows the breakdown of the synchronizer module. It contains two main blocks, each of which is equipped with an 8-bit microprocessor (Motorola 6803). One of these processors is responsible for the various code signals (signal microprocessor). It also controls the interfaces to the CPU via the FIFO bus and to the TC generator. It is connected to the other processor (capstan microprocessor) via a dual port RAM. This microprocessor contains the algorithm for the capstan phase control. For this purpose it produces a control frequency.

Code, reference signals

For the synchronization code signals are required from master and slave (D820 MCH). The signal microprocessor can process SMPTE/ EBU time code signals of any code type from both sides. If available, also move pulse information is analyzed. This information is useful if the code is missing (spooling mode) or corrupted. The hardware is also designed for decoding biphase code from perfo machines.

The analyzed information is available to the CPU at any time and when required, it is requested from the signal microprocessor via the FIFO bus. For synchronizing digitally recording tape machines, the information of the RT signal

Kurt Schwendener



Structure of the synchronizer board (SSTC board).

can be used instead of the SMPTE/EBU time code. However, the signal is evaluated by the RT board, the SSTC board already receives from the CPU the momentary difference.

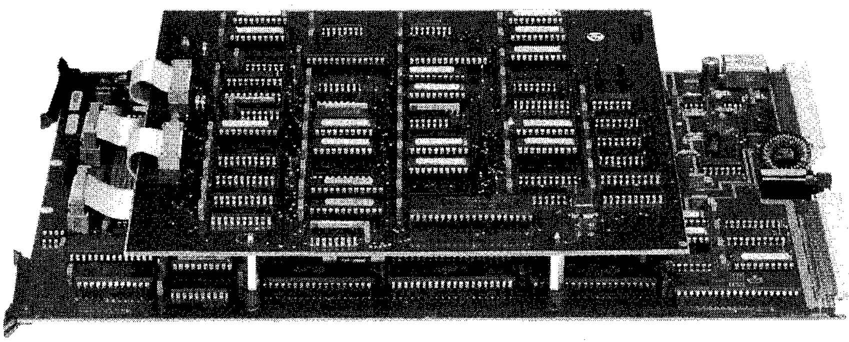
The two main modes are therefore:

TC synchronization
The slave references are the SMPTE/EBU-TC and move pulses. As the master reference at least a SMPTE/EBU time code is available. The move pulse information and tape deck status tallies from the master can also be taken into consideration when available.

RT Synchronization
The slave and master reference are derived from the RT signal. The RT board supplies the CPU and the SSTC board with the necessary data.

Synchronization procedure
I. CHASE SYNC: During this initial phase the CPU controls the tape deck with the aid of the position, difference and speed information from the synchronizer or the RT board, until the conditions for capstan synchronization are satisfied (speed in PLAY range, the difference can be eliminated by the capstan). The CPU then switches the tape deck to "external vari-speed PLAY" and starts the next phase.

Synchronizer board (SSTC board) with piggyback mounted TC generator.



II. PLAY SYNC: During this second phase the capstan microprocessor controls the D820 MCH with a control frequency. The momentary difference is calculated as a control variable from the signals and data supplied by the signal microprocessor or by the RT board (via CPU and signal microprocessor). As soon as the difference is small enough and the machine does not operate in TC LOCK mode, the CPU switches the synchronizer off. The capstan controller receives the command to continue controlling with the aid of the block information from the digital audio data.

In TC LOCK mode the synchronization is closed with this phase. This mode is used if no common PLAY reference exists between the master and slave (e.g. analog tape machine).

III. BLOCK SYNC: In this concluding phase the information from the SSTC board is required only for monitoring functions because coupling of the BLOCK SYNC signal of the D820 MCH with the master is performed by the capstan controller. The precondition is that the master and slave use the same master reference (e.g. audio data when coupling two digital audio machines, composite video in mixed video/audio configurations).

If the difference increases due to jumps in the recorded code or change in the desired offset, the CPU may switch back to CHASE or PLAY SYNC, depending on the magnitude.

Event controller
The time code is suited for more than just synchronization and positioning. For implementing electronic editing systems, time-critical record commands must be executed at a specific tape location. To this purpose the CPU can load several registers on the SSTC board. If there is coincidence between the register and the selected code (master or slave), the signal microprocessor transmits a trigger pulse to the CPU by means of an event message.

Time code generator
Another function of the signal microprocessor is the transmission of control commands to the TC generator. The CPU is able to load the generator with any value (time, user bits), to start and to stop it. Various modes (format and reference clock) and source changeover for the master code (generator or external), TC output (TC from tape, generator) and TC recording (generator, external, with delay compensation) are feasible. Whereas the foregoing commands are forwarded directly without any evaluation, the JAM command causes a one-time synchronization of the TC generator with the slave or external master code. ●

STUDER D820 multichannel

Tasks and functions of the sound memory

by Marc Biver

The sound memory of the D820 MCH is a hardware facility for storing and replaying digital audio data or for delaying up to 24 channels. The total sound memory capacity is 38 to 52 seconds, depending on the sampling frequency. The sound memory can be partitioned into several independent blocks and is consequently able to process several channels simultaneously.

The sound memory supports three different functions: *Memory – Track slipping – Track bouncing.*

With the **Memory** function the data of up to four audio channels can be stored in the dynamic memory and played back as required. Any channel of the D820 MCH can be assigned as a source. Different operating modes are available for recording and reproducing digital audio data:

- The *"Instant mode"* records data until the memory is completely filled.
- In *"Continuous mode"* the recording continues up to the stop command. The oldest data are continually overwritten by new data (like an endless tape of finite duration).
- In *"Trigger mode"* the sound memory records the data only when a trigger command is received. However, the audio data are recorded already 0.5 seconds before the trigger command (so-called pretrigger function). This ensures that in the event of a slightly delayed recording start due to the slow reaction time of the operator, the desired passages of the audio track are still recorded. When the sound memory is full, the recording is automatically stopped.

For playback the user can select between continuous and break mode. In the first case the end of the recording is crossfaded with the start. In the second mode, a 0.5 second pause is inserted between the end and the start. The start and the end are both crossfaded.

With the **track slipping** function it is possible to delay up to 24 channels by a specific amount. Depending on the number of selected channels the maximum delay is 1.8 to 10.9 seconds. Channels 1 to 24 can be specified as sources. These channels are delayed by the track slipping and transferred to the channels 25 to 48.

In **track bouncing** mode up to four channels are simultaneously copied to four other channels. The sources can be selected from among any of the 48 channels. This extends the ping-pong function of the D820 MCH.

Architecture of the sound memory

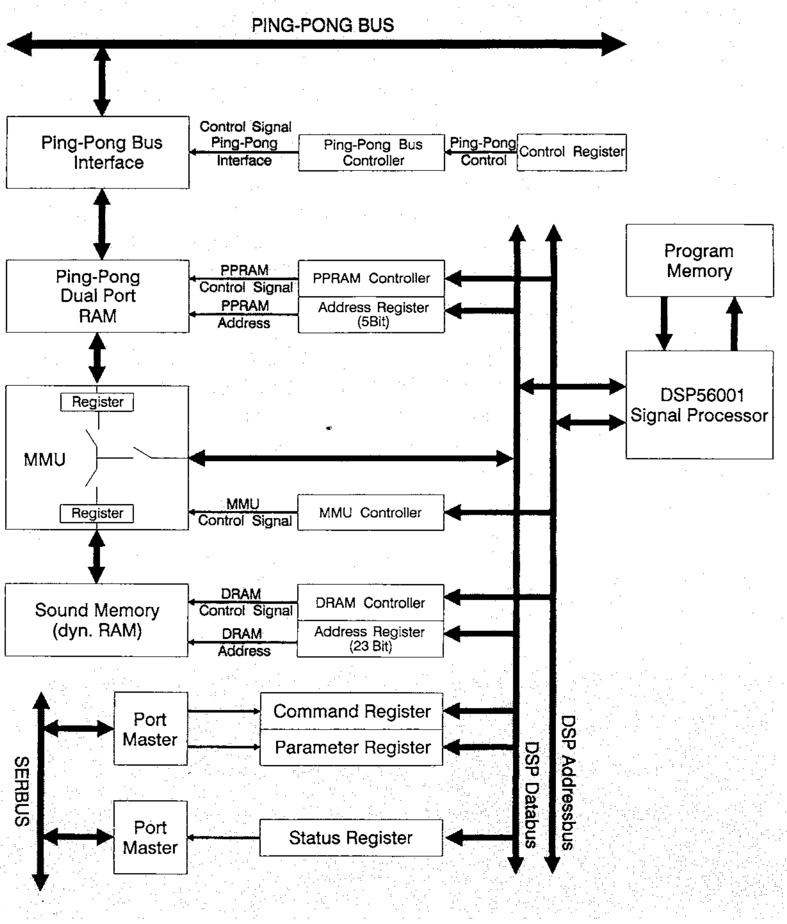
The heart of the sound memory is the Motorola signal processor DSP56001 that operates at a clock frequency of 20 MHz. This processor performs all signal processing activities such as crossfades during the start and stop of the play function, and management of the memory functions.

The memory consists of 32 dynamic 1-Mbit standard RAMs that are distributed across two memory banks with a word width of 16 bits. These chips are also used in larger computers as storage devices.

The dynamic memory is addressed and controlled by a memory manager that is implemented with programmable logic devices (GALs). This unit is also responsible for refreshing the dynamic memory chips and relieves the signal processor from simple data transfer functions.

Marc Biver

Block diagram of the sound memory architecture



The audio data are input/output from the ping-pong bus of the D820 MCH via an interface with a dual port RAM. Because of this special memory design the DSP can access the audio information at any time.

The control commands and the operating parameters are supplied to the DSP of the *sound memory* via the *Serbus* of the D820 MCH. These commands are interpreted and subsequently executed. An additional 24-bit Serbus channel is available for transmitting various time values. On request of the Serbus, the DSP supplies status information or the time capacity of the memory.

Due to the modularity of the design and the utilization of a signal processor, high flexibility with respect to enhancements and supplementary functions is assured.

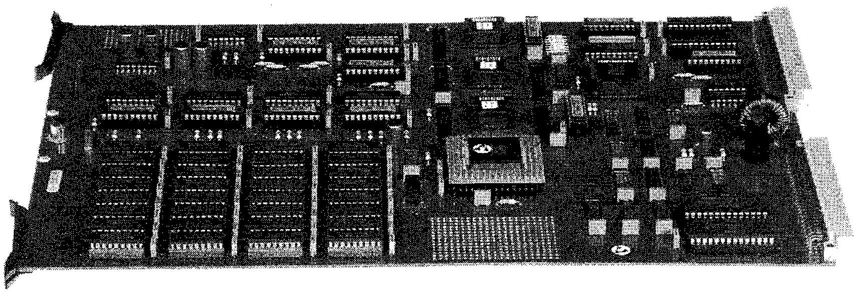
Operation of the sound memory

The sound memory is operated via a control section located in the remote console. All functions of the sound memory can be performed via this control section.

The three functions (memory mode, track slipping and track bouncing) are mutually exclusive. After a function has been activated, a configuration process is offered (specification of the input and output channels). This is indicated by a flashing LED in the function key. The completion of the configuration process is confirmed by pressing the function key again. The corresponding sound memory function is subsequently performed. The function can be canceled with the corresponding function key.

The time data that are important for the various operations are shown on a time display. During the configuration phase the display shows the memory capacity per channel in seconds. In memory mode the duration of the recorded sequence is continually displayed, whereas in track slipping mode the delay setting is shown.

In the memory and track bouncing functions the numbers of the input and output channels are shown in the four display fields. If fewer than 4 channels are needed, the unused channels can be deleted by entering CLR as the source. The corresponding display remains empty. The maximum time allocated per channel is calculated (and shown on the time display) as a function of the number of channels.



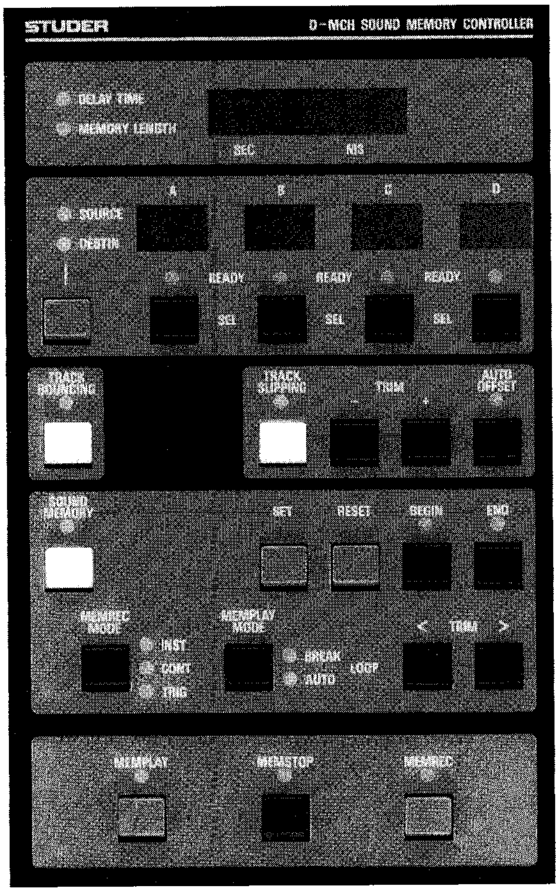
The sound memory hardware is part of the D820 MCH audio electronics.

In track slipping mode the configuration phase is followed by the input of the delay with the TRIM+ and TRIM– keys. The delay value is shown on the time display.

With the memory function the start and end of the recorded sequence can be shifted later if desired. The new start or end is entered with SET BEGIN or SET END during playback. These new addresses are taken into consideration during the next playback. With RESET BEGIN and RESET END the original addresses are restored. The addresses can be fine-adjusted with the TRIM+ and TRIM– keys.

Summary

Together with the control panel the sound memory represents an important function group of the D820 MCH that offers a number of interesting functions for postproduction and generation of special effects. Moreover, within the



The operator controls for the sound memory are integrated in the remote control console.

D820 MCH itself, the sound memory offers the ability of sample-accurate copying of passages on tape without any additional synchronization equipment.

STUDER REVOX in Albania

Our woman in Tirana

The editor's office receives letters and telephone calls from all parts of the world. We are particularly pleased to learn that there is also a thoroughly practical application for our magazine. Mrs. Flutura Myftiu writes to us from Tirana:

"I work as an electrical engineer in the Management of the Albanian Broadcasting Company and was recently awarded the title 'Candidate of technical sciences' by the Albanian Education Ministry. For writing my thesis 'Stereo recordings in the Studio of the Albanian Radio and Television Broadcasting Company', the studio equipment which we had purchased a few years ago from Studer Revox was of great help. ***And also the SWISS SOUND magazine was most helpful. I have translated many articles into Albanian and discussed the topics with my colleagues."***

Mrs. Myftiu's letter was accompanied by a brief contribution to SWISS SOUND, which we would like to pass on to our readers.

STUDER REVOX in Albania

Many years ago the radio and television studios of the Albanian Broadcasting Company were modernized with Studer equipment. First the music studio was equipped, with a 369 console (24 inputs) and two A80RC tape recorders. A few years later we bought additional Studer equipment, a 269 mixing console (16 inputs) and 6 additional A80RC tape recorders for the assembly room (programs and radio dramas) as well as for studio 3, from which the programs of the first and second channel are broadcast.

At Radio Shkodra, which covers the Northern Albanian region, two studios also operate with Studer Revox equipment. In 1983 a 369 mixing console and three A80RC tape recorders were installed in Studio 1 of Albanian Television; this equipment still functions very well. In 1985 the A80RC mono tape recorders were upgraded to stereo. Since then, music has only been recorded in stereo mode.

Radio Tirana, music recording studio

The latest acquisitions were made in 1989, at which time we installed Studer 962-14/3 mixing consoles and some A68 amplifiers for combined radio / television applications. We also use some PR99 tape machines in our Albanian music cassette production.

Currently we are faced with enormous problems but we do the best we can. The articles in SWISS SOUND and other publications of your company are very helpful to us.

Flutura Myftiu, Tirana

Studer Automatic Alignment Program

SAAP 1.2

The automatic alignment procedure is now even better and faster. The new, significantly revised version SAAP 1.2 permits the automatic calibration, testing and logging of all mono and 2-channel tape recorders A807, A810, A812 and A820. With the Performance Check function it is now possible to completely test and log the performance of a machine. The logs can be printed, displayed on the screen and stored on diskettes. A new feature is that two tape types can be stored for each machine. New configuration files for the title, the printer and the filters are included. The user can now assign an individual title to the log, adapt the printer, and freely assign the internal filter slots of SYSTEM ONE. Calibration, testing and logging with this system is up to 5 times faster than if this task were performed by a skilled service engineer. Moreover, even inexperienced users can now calibrate a tape recorder without difficulty.

Marcel Cattani

A turnkey realisation of STUDER DIGITEC

The Nerve Center

By Sylvie Casteel

Sylvie Casteel
Product Marketing Engineer at Studer Digitec

Since its entry in the STUDER group in August 1991, STUDER DIGITEC is happy to make its first appearance today in the SWISS SOUND. By way of introduction, I will present you one of our prize achievements: the R.T.L. Master Control Room. Equipped in 1989, it still runs flawless without having suffered from any breakdown.

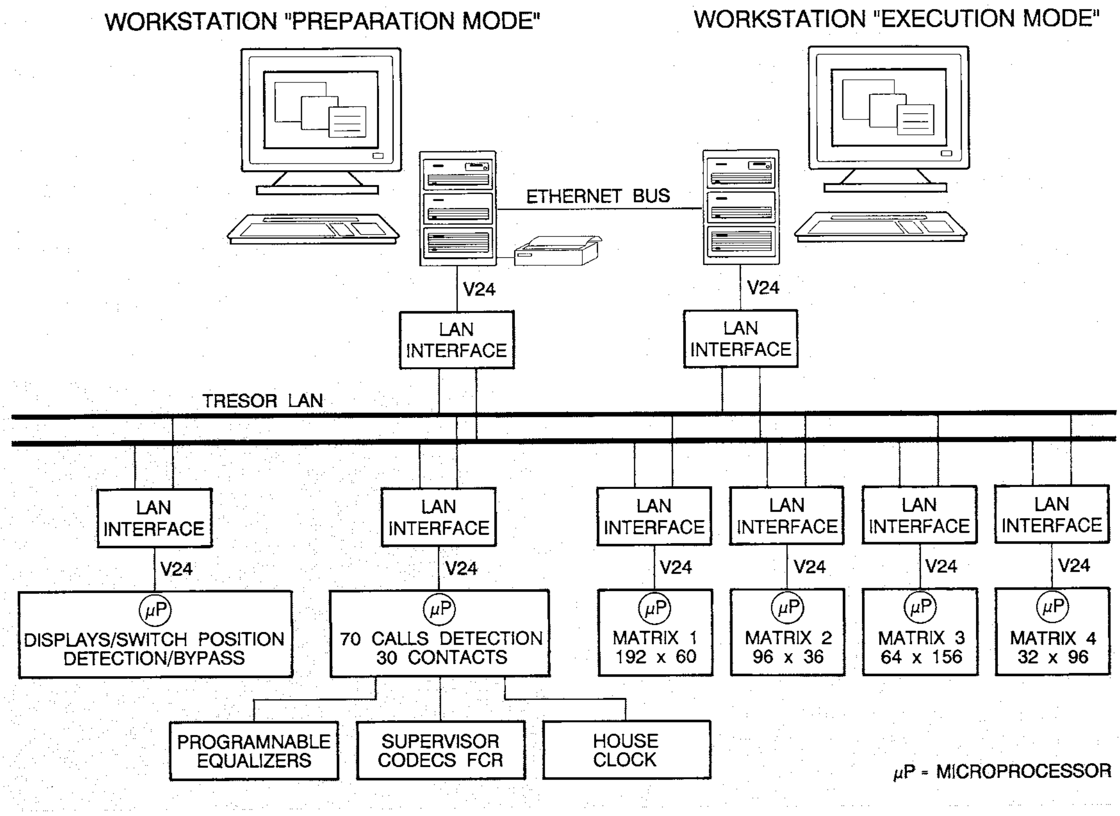
Located in Paris, Radio Télévision Luxembourgeoise (R.T.L.) is the largest broadcasting station in Europe with the largest listening audience in the French-speaking countries. The radio station receives and transmits a great deal of signals daily and operates around the clock. Also the Chief Engineers pay great attention to the quality and reliability of their Master Control Room, the nerve center of the system. Upon their recommendation, R.T.L. awarded the contract for the turnkey-project to STUDER DIGITEC (DIGITEC at that time).

The system is composed of the audio part on the one hand and the control systems on the other.

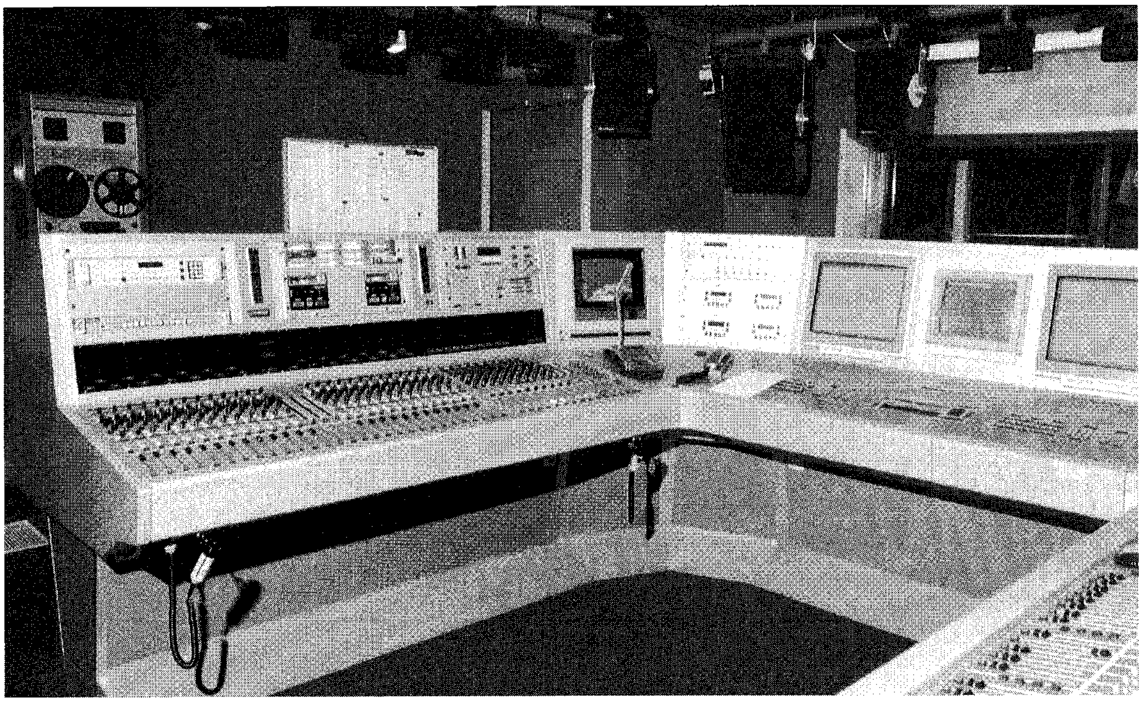
For processing signals, STUDER DIGITEC has implemented 4 distribution matrices of large capacity (32x96, 96x36, 64x156 and 192x64), a specific "4-wire desk", a mixing console specialized in the simultaneous correction of received and transmitted external signals (on 4 wires), a VHF mixer for the communication with OB-vans, a programmable multi-channel equaliser used for manual or automatic recall of 30 pre-programmed telephone line equalisations, and a TRANSCOM 64 Kbits codec, a piece of equipment sold by FRANCE TELECOM for the transmission of high quality speech signals (7 kHz) on the French digital telephone network.

STUDER DIGITEC has organized the control system around TRESOR™, a very reliable piece of LAN architecture whose primary advantage is quick access time for users due to intelligence distribution. Two central operating positions (HP series 9000), running under UNIX, control the equipment, and a coupler, playing the role of an interface, normalizes the exchanges between the workstations and the other audio peripherals.

Besides the quality of the equipment, STUDER DIGITEC's fame lies in its software know-how. In order to optimize the man-machine interface, a keyboard has been specially designed for the R.T.L. routing center and allows direct



R.T.L. Master Control Room: Control system block diagram



R.T.L. Master Control Room.

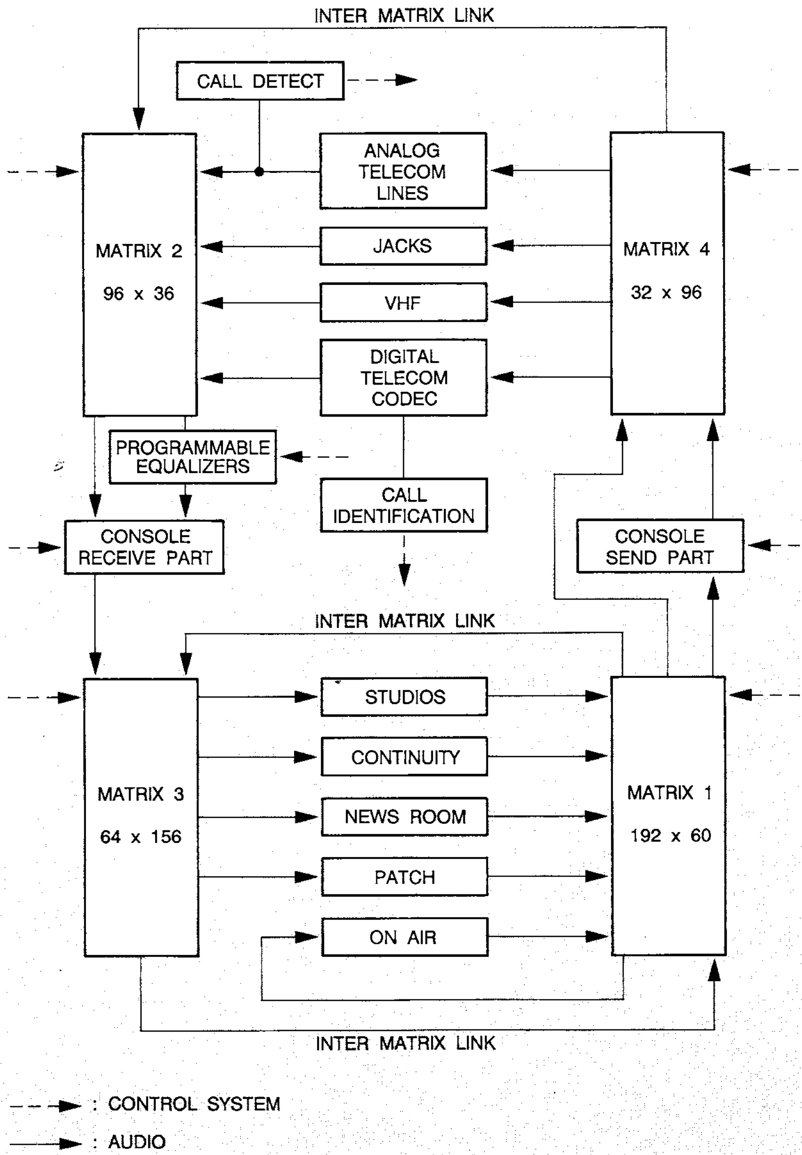
access to the system. The control system is divided into 3 operational modes: the *direct operational mode,* the "normal one" – for direct access to the routing system –, the *preparation mode* enabling to make in advance and recall crosspoints preparation, and the *supervisory mode* for administration and configuration.

The control system presents the following key-facilities:

- Direct switching and display of the XY physical view of the system,
- Detection of calls on telecom lines,
- Automatic routing of the signals between the four matrices,
- Labelling any source with display on the workstations and the console,
- Time scheduling for automated recall of preparations,
- Printing of preparations, crosspoints, and history files,
- Automatic testing of the matrices and input / output monitoring of the matrices,
- Management of a database of PTT lines and associated taxes.

The R.T.L. routing system is a good example of a high-tech turnkey project that STUDER DIGITEC has been able to implement.The success of this project was also made possible by the close collaboration between the customer and the STUDER DIGITEC team on the basis of a strict project development methodology. Confident in the reliability of its equipment, STUDER DIGITEC has granted an extended guarantee of two years at no additional cost. After two years of perfect functioning, R.T.L. will soon be asking for the establishment of a maintenance contract with STUDER DIGITEC for software evolution and hardware support. ●

R.T.L. Master Control Room: Audio block diagram



Studer A816: 1/4" broadcast studio machine

The workhorse

by Hans R. Hässig

Hans R. Hässig

The A816, developed as a "workhorse" for demanding and continuous duty in radio broadcasting, is a real multitalent that offers excellent stability as well as outstanding flexibility and operator convenience.

The Studer A816 is equipped with a processor-controlled tape deck and amplifier logic, and excellent facilities for modern peripheral systems. This machine can be installed without modification into the standard ARD chest (same installation dimensions as the AEG Telefunken M15A). The electronics bay swings forward and the connectors are located in front. When the electronics bay is swung backward, the seated operator obtains maximum knee space.

The tape deck is mounted on a ribbed, extremely rigid, die-cast aluminum alloy chassis and is isolated from the supporting surfaces by a three-point suspension so that no bending forces can be transmitted. The tape transport is designed for 12.5" (317 mm) pancakes with the oxide coating on the outside. Particular attention has been given to the ease of tape threading. Preloaded jeweled bearings are responsible for smooth tape guidance across the headblock. This technology ensures excellent phase and amplitude stability which is particularly important for high audio frequencies. No headblock shield is necessary because the so-called zone concept greatly reduces noise fields through strict separation of the subassemblies.

The tacho roller for generating the control and counter pulses is arranged to the left of the headblock, which means that correct tape length measurement is ensured also in dump edit mode. A light barrier detects leaders (according to DIN), and pre-head scissors can be installed as an option.

Gradual listening in is possible with an editing lever, and for tape marking the outer right guide roller can be briefly lifted off.

Drive concept

The tape is transported by a low-inertia, maintenance-free, brushless DC capstan motor equipped with its own microprocessor control (acceleration time better than 0.5 s at 15 ips). This design achieves extremely low wow-and-flutter values within a wide temperature range (guaranteed wow-and-flutter performance between +5...40°C, start from -5°C). Additional advantages: fast stabilization at all four tape speeds of 3.75, 7½, 15 and 30 ips, in varispeed and reverse play mode.

The AC spooling motors – also brushless and maintenance-free – are supplied by an internal 3-phase generator and produce a steady, vibration-free torque. The mains frequency and the control signals superposed by the utility company have no influence on the tape tension and spooling characteristics. The power stage is switched. This results in high efficiency, low heat dissipation and very short acceleration and shuttling times.

Quiet tape transport

The tape is guided almost exclusively across wear-resistant jeweled surfaces (guide rollers and pins) and glass metal heads (for excellent long-term frequency response). The pinch roller arm movement is mechanically damped.

The entire tape deck is equipped with wear-resistant motors, actuators and sensors which means that the maintenance is limited to cleaning the tape transport elements.

The two pancake diameters are continually calculated based on the spooling motor tacho signal and the tape speed, i.e. independently of the actual tape length. This information is also used by the system for detecting an approaching tape end so that, regardless of the reel size, it can reduce the spooling speed or stop the tape, depending on the programmed function.

Power supply

The power supply is designed in such a way that it produces minimal ripple and low heat dissipation. Even when the machine is installed in a chest, no fan is required. A current limiter protects against high peak inrush currents.

The A816 has demonstrated its EMC performance (according to VDE871, category B, EMC R2) together with peripheral devices in various tests (RBT certificate).

Additional protection is provided by secondary fuses with selective line voltage monitoring and a short-circuit-proof stabilizer. The individual supply voltages are indicated visually.

Audio concept

The amplifiers achieve optimum pulse transmission through equalization with group delay compensation. Other advantages are:

- Powerful HF output stages can also cope with high coercion tapes.
- 2 calibration parameter memories for each speed and equalization.
- Outputs either with or without transformers
- High-quality electronic output stages for up to 28 dBu line level (balanced +24 dBu) and excellent common mode rejection.
- RF oscillator adapted to the erase heads, or inhibited for reproduce-only mode.
- Audio parameter settings different or same for NAB and IEC (CCIR).
- Type selection "TAPE A" or "TAPE B", combined with tape speeds.
- Drop-in/drop-out compensation .
- Bandwidth changeover for the sync channel.
- Line level (operating level) programmable for reference level +6, +10, +14, +16 dBu (or 0, +4, +8, +10 dBu).
- Mono-stereo changeover electronics with/without audio generator (option).
- Automatic alignment via external PC through serial interface (option).

A new circuit design feature is the parameterized microprocessor control: This eliminates the need for conventional potentiometers, and the calibration process can be automated. Digital/analog converters (DAC) convert the control signal to analog control variables. The parameters are set by the processor (and shown on the LC display). The record/reproduce amplifiers are phase compensated so that PCM-like audio recordings can be produced.

The inputs and outputs of the A816 can be balanced with transformers or electronically, depending on the selected input/output boards. The nominal level is program controlled as described above, four different values 0/6 dBu, 4/10 dBu, 8/14 dBu and 10/16 dBu are stored in the audio function memory. The first value relates to the nominal level for VU instruments, the second value for PPM instruments according to DIN.

For optimum calibration the output levels can be fine-adjusted by means of a multiturn helical potentiometer within the range of ±1.5 dB. In contrast to other 2-track machines, the A816 does not need an additional sync reproduce amplifier because amplifiers are recalibrated by the microprocessor control when the machine is switched to sync mode (only on machines with audio channel control panel).

The record electronics has two other special features: Firstly, a delay can be added for optimizing the drop-in. This delay corrects the activation of the bias as a function of the erase head distance and the selected tape speed. Secondly, a switch controlled Dolby HX PRO™ circuit is implemented for optimum treble response.

The record and reproduce heads are made of abrasion resistant glass metal. A 2-channel preamplifier located directly below the headblock assures high-quality amplification of the reproduce signal from tape. Furthermore, each headblock features an electronic index based on which all tape deck and audio parameters set for this configuration are automatically reestablished.

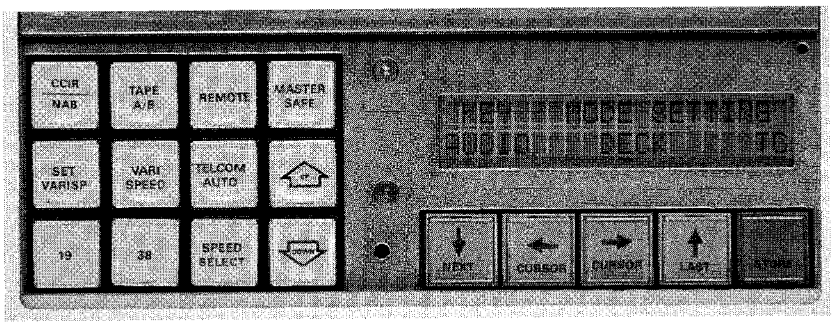
Operating concept

The operating concept is geared to maximum efficiency and reliability. The control panel is mechanically decoupled from the tape deck. The spooling speed and direction are determined by a shuttling lever (left). An editing lever (right) permits gradual listening-in during spooling and activation of the dump edit mode.

The microprocessor control stores not only the audio parameters but also the counter content, the selected tape speed, and the locator addresses. The locator addresses are based on the tacho roller pulses; for this reason the locator functions are independent of the selected tape speed and the addresses are always indicated in real time (as is the case for the tape timer).

The tape deck command keys are grouped in an elementary keypad, function keypad as well as secondary keypad. All keys can be user-programmed with the functions stored in the program memory (soft keys).

The function field which is normally programmed with the locate and optional tape deck commands, also contains the large tape timer LED display with 1/10 second resolution. There is also a status LED display for the most important functions: speed selection, varispeed mode, Telcom™ on/auto, broadcast, CCIR or



NAB equalization, tape type A or B, remote control/fader start enabling, opto trigger and master safe.

The secondary keypad located below a hinged cover comprises 10 additional function keys that can, for example, be assigned to presettings (speed, varispeed, etc.) or infrequently used input facilities such as remote control enable, etc. Five other keys are available for paging through the menu and for programming the keys and functions. The factory assigns certain standard operations to all function keys and these are correspondingly labeled. However, the user is free to reprogram these keys with a repertory of 40 functions, according to his personal requirements.

An alphanumeric LC display in the secondary keypad is used for programming the audio and tape deck data, for displaying the software status, the varispeed data, and error messages. The error message also provides a rough description of the cause, such as the loss of audio or tape tension data, etc. From the nature of the error message the operator can also determine whether or not downgraded operation with the machine is still possible.

Programming concept

When we examine the program facilities, the flexibility of the A816 becomes readily apparent. The individual menus are selected via a so-called status tree.

In the normal operating state the input and output level settings are shown on the LC display. When the "Next" key is pressed, the "User Setup" with the partitioning into "Alignment" and "Mode" is displayed. The alignment menu is used for presetting various tape deck parameters such as the maximum spooling speed, the tape tension, the interface formats, etc.

In "Tape Deck Keys/only", the spooling functions can be programmed to the desired keys, whereas in the alignment deck menu the maximum forward and reverse spooling speed as well as the library wind speed can be individually defined. This is highly advantageous if the tape to be processed does not allow very high spooling speeds, e.g. if the uncoated side has a very rough matt surface.

The two record functions A and B define whether switching to record mode requires that the play and record key be pressed simultaneously or whether the record key alone suffices.

An interesting facility is offered by the rollback function: When this function is activated, the tape backspaces by the amount defined in the alignment deck menu (range 1 sec. to 59 sec.). Depending on the programming, the machine then switches from rollback to stop, play or record.

In addition to the simple "Locate Zero" function, five additional programmable locator memories are implemented in the tape deck.

In the functions "Loc Start Play, Stop and Rec" the tape searches the address at which the last play command was entered and then automatically initiates either play, stop or record, depending on the preselection.

Also the tape counter can be set to an individual time by means of a "Set Timer" function. The "Lap/Watch Display" function switches the display from the main timer to an independent auxiliary timer, for example, for determining the playing time of a given selection.

New "Theater" function

A completely new feature of the A816 is the so-called theater mode in which the following functions are available:

- NEXT TAKE
- PREVIOUS TAKE
- PREPARE
- START CONTROL

These commands function correctly only if the "theater leader tape" has previously been defined. If the same leader is always used, programming needs to be performed only once because the A816 stores this value. A brief introduction to these new functions is given below:

– *NEXT TAKE / PREVIOUS TAKE:*
The tape deck automatically searches the next or preceding marking and positions the start of the oxide coating in front of the reproduce head, taking into consideration the programmed acceleration time. When this key is pressed in more than one location, the tape deck counts the number of key depressions and then searches the corresponding location.

– *PREPARE:*
The tape deck searches the first marking in play mode and then positions the start of the magnetic tape at the defined address.

– *START CONTROL:*
Allows "listening-in", the machine switches to play. As soon as the key is released the machine returns to the starting position.

Leader recognition

The A816 is equipped with a leader recognition facility that supports the following functions:

– *TELCOM AUTO:*
When a TELCOM leader is used and this function is preselected, the noise reduction system is activated and disabled only when the tape tension levers detects "tape out".

– *LEADER STOP:*
In play mode, the tape is spooled to the programmed position when the leader is detected (set leader offset).

Automatic calibration

The machine can be automatically calibrated via the RS232 interface (SAAP option).

Test program:

For the service engineer a standard test program is available for testing the electronics. This program tests the LC display, decoding of all keys, and the values read by the microprocessor from the tape shuttling lever, the tape tension sensors, the leader detection, and the supply voltage. The rotation direction and servo control of the two spooling motors and the capstan motor can also be checked.

For checking the individual solenoids (pinch roller, brakes, etc.), special tests are performed. In the lamp test the program activates all LEDs. Also the individual inputs/outputs of the various interfaces can be checked with corresponding tests. ●

"German coating geometry"
or
The A816 has an interesting background

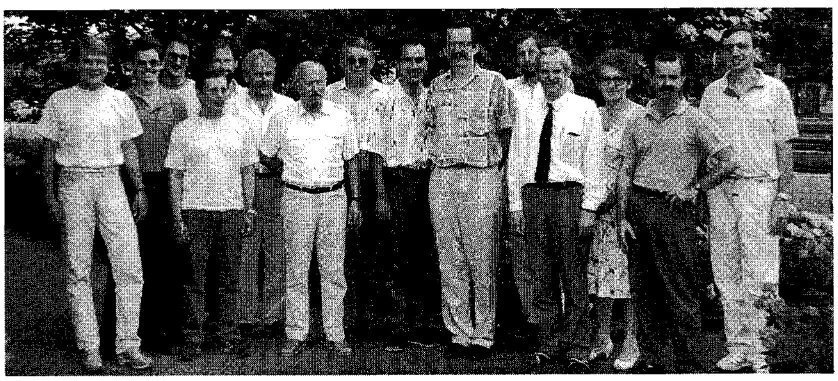
With the introduction of the tape recorder at the 1935 radio fair in Berlin where AEG TELEFUNKEN exhibited the "Magnetophon" that operated with the plastic tape developed by BASF, the world of radio broadcasting was changed dramatically and our music culture later revolutionized. This pioneering achievement resulted not only in lasting fame for the German audio industry, but also in a peculiar coating geometry. German archives hold some 60 million reels of tape with the oxide coating facing outward.

In 1989, when AEG TELEFUNKEN transferred its magnetic recording products division to Willi Studer, the uniqueness of the oxide coating facing outward immediately attained a new dimension. The "German market" was awaiting the M16 (successor to the AEG M15A). The question as to whether Studer would develop this prototype to production maturity had to be answered negatively due to logistic and service reasons.

After thorough investigations, Dr. Studer pledged to the "German market" to develop his own compatible machine for the ARD console (standard chest). Two ad-hoc work teams of this public institution were founded: The radio broadcasting managers (HFBL), headed by Dr. Roth, and the measurement engineering group, headed by M. Schneider. Both teams developed requirements catalogs. When they first arrived in Regensdorf in October 1989, the existing conceptions where translated into concrete specifications and the development scope was defined.

At the Tonmeister Convention 1990 in Karlsruhe, the first prototype of the Studer A816 was introduced to the public. Intensive discussions and measurements followed. In the spring of 1991 the first preproduction series machine was made available to the independent Radio Broadcasting Engineering (RBT) Institute for detailed tests. After the necessary design changes had been implemented, the first production machine was sent to RBT in Nuremberg for rechecking. The machine was released for production at the beginning of December.

The Studer A816 development group is justly proud of its achievement.



Elektroacoustics

Air, the nonlinear medium

by Paul Zwicky

Paul Zwicky

Overspecialization is said to make blind. Maybe, because frequently even experts overlook obvious facts. This is also one of the reasons why factors of minor import are adduced to underpin sophistical arguments. The field of electroacoustics is particularly susceptible to this, because also emotional moments need to be considered.
By providing a few glimpses at the underlying physical principles, the following report hopes to give some food for thought.

When considering the quality of the hi-fi chain we talk about the components and materials involved. We discuss whether transistors provide a better sound than tubes, whether an electrolytic capacitor should be banned from the hi-fi technology altogether, whether or not metal resistors should be coiled, whether paper or polypropylene should be preferred as the diaphragm material, or whether the purity of the copper wire should be 99.99% or even 99.9999%. Only the air which serves as the mechanical filter element or transmission medium is frequently forgotten. Because it is part of nature, we may possibly feel that we cannot change it and therefore thrust aside all thoughts in this direction.

In this report we shall talk about the air as a gas in the physical sense and investigate its involvement in electroacoustics. When sound is reproduced by speakers, the existing static air pressure must be changed in some way. As we remember the atmospheric pressure depends on the weather and the elevation above sea level. The normal air pressure is approx. 10^5 N/m². A variation of only ± 2 N/m² corresponds already to a sound pressure level of 100 dB. A speaker diaphragm that moves faster than the displacement velocity of the surrounding air, creates such a pressure variation. In fact the diaphragm displacement produces a change in volume which, coupled via the gas law, causes a change in pressure. All speaker designs attempt to make the diaphragm follow the input signal as closely as possible. Of interest here is what the pressure change created through the linear volume change by a diaphragm looks like. Is it afflicted with distortion?

We assume that the sound pressure at a distance of 1 meter is known. We know the relationship:

$$p_1 = 2 * 10^{-5} * 10^{\left(\mathrm{SPL}\,\frac{1}{20}\right)}$$

This is an RMS value. Because the maximum pressure is created on the diaphragm surface, we convert the pressure:

$$p_m = p_1 \frac{4}{d}$$ where d is the diaphragm diameter.

We assume that the signal is sinusoidal, which means that the peak pressure on the diaphragm surface is

$$\hat{p} = p_m \sqrt{2} = p_1 \frac{4}{d} \sqrt{2} = 8 * \sqrt{2} * 10^{-5} \frac{10^{\left(\mathrm{SPL}\,\frac{1}{20}\right)}}{d}$$

We also know the adiabatic gas law

$$p * \mathrm{Vol}^{(\gamma)} = \text{const} \qquad (\gamma = 1.40 \text{ for air})$$

We convert: $\mathrm{Vol} = \text{const} * p^{\frac{-1}{1.40}}$

According to Taylor this function is represented as a power function of the third order in the position p = 10^5 N/m².

$$\mathrm{Vol} = \text{const}\left(388 * 10^{-6} - 8.08 * 10^{-9}\,\Delta p + 102 * 10^{-15} * \Delta p^2 - 746 * 10^{-21}\,\Delta p^3\right)$$

Δp is replaced by $\hat{p} * \sin \omega t$

Δp^2 is replaced by $\hat{p}^2 \left(\frac{1}{2} - \frac{1}{2}\cos 2\,\omega t\right)$

Δp^3 is replaced by $\hat{p}^3 \left(\frac{3}{4}\sin \omega t - \frac{1}{4}\sin 3\,\omega t\right)$

$$\mathrm{Vol} = \text{const}\left(388 * 10^{-6} - 8.08 * 10^{-9}\,\hat{p} * \sin \omega t + 51 * 10^{-15}\,\hat{p}^2 \cos 2\,\omega t - 186 * 10^{-21}\,\hat{p}^3 \sin 3\,\omega t\right)$$

Also known are the formulas for the distortion factors:

$$K_2 \approx \frac{\text{amplitude from } 2\,\omega t}{\text{amplitude from } \omega t} \qquad K_3 \approx \frac{\text{amplitude from } 3\,\omega t}{\text{amplitude from } \omega t}$$

We search the amplitude from the last volume equation, insert the calculated value of $\hat{p}$ and obtain:

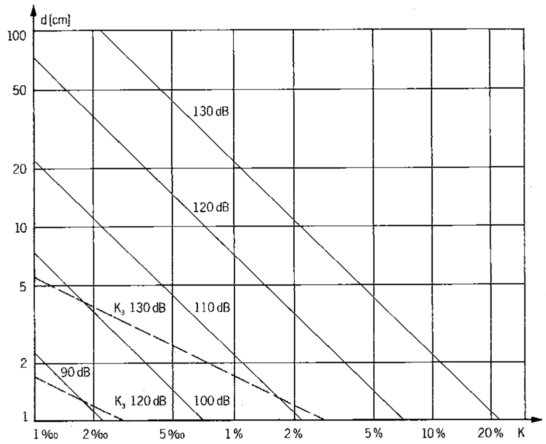
$$\boxed{\begin{aligned} K_2 &\approx \frac{715 * 10^{-12} * 10^{\left(\mathrm{SPL}\,\frac{1}{20}\right)}}{d} \\ K_3 &\approx \frac{295 * 10^{-21} * 10^{\left(\mathrm{SPL}\,\frac{1}{10}\right)}}{d^2} \end{aligned}}$$

What do these results mean in practice.
We shall demonstrate this with a few examples.

a) A speaker designer would like to achieve a high quality standard. He defines the harmonic distortion as ≤ 0.5%. How large can the sound pressure be at a distance of 1 meter for a 25 mm diameter tweeter?
→ 105 dB

b) How large would K_2 be if at 100 dB SPL a horn-type tweeter with a throat diameter of 1 cm were used?
→ 2.26%

c) How large would the throat diameter of a horn-type tweeter have to be if a distortion factor of 0.5% is to be achieved at 130 dB SPL?
→ 45 cm



Interrelation between diaphragm diameter, sound pressure and distortion factor (K_2 and K_3).

As we can see, high sound pressure levels and low distortion factors are difficult to achieve in practice. The situation is particularly difficult in the case of the tweeter, even if the specified maximum sound pressure level in the treble range is not as high as in the midrange. Horn type tweeters achieve their superior efficiency only if a correspondingly thin throat is selected. But the resulting distortion factor will hardly satisfy demanding requirements.

Being aware that a speaker inherently has a distortion factor in the magnitude of 0.3% to 20% prevents us from wasting too much energy on reducing the distortion in other places from 0.03% to 0.01%. The speaker designer is aware that the distortion source discussed in this report is not the only one. ●

Relationship between pressure and volume

Remember the general gas state law according to Boyle-Mariotte-Gay-Lussac?

$\frac{p * V}{T}$ = constant

p = pressure
V = volume
T = absolute temperature

If we ensure that the temperature remains constant, we obtain the *isothermal gas equation*

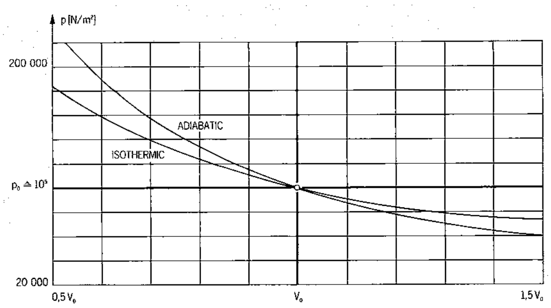
$p * V$ = constant

If we allow the gas to heat up during compression (bicycle pump effect), we must make a correction:

$p * V\gamma$ = constant, $\gamma = 1.40$ for air

This formula is known under the designation: *adiabatic gas equation.*

Both gas equations are nonlinear. A linear change in volume results in a *nonlinear* pressure change.



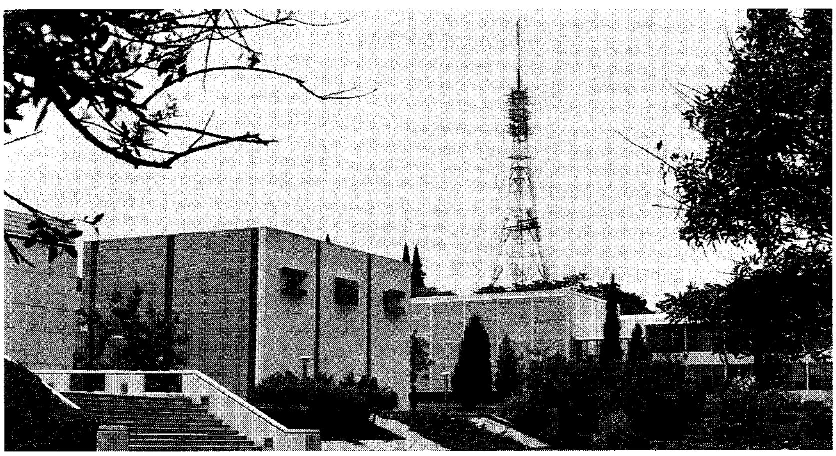
Large-Scale Project of Zimbabwe Broadcasting Corporation

Studer International has been able to offer ZBC the advantage of utilizing a Swiss soft-loan credit that consists of a 50% interest-free Swiss government portion and a Swiss bank portion of another 50% at a firm interest rate for the complete renewal of three radio production centres in Montrose (Bulawayo), Mbare and Pockets Hill (Harare).

ZBC and Studer have negotiated a contract for the supply of professional audio equipment, engineering and installation work of totally Swiss Francs 11.8 million to be spent on this project.

More material for the technical installation of the three master control rooms will come from ALCATEL STR, contract partners and sub-suppliers of Studer International, also responsible for the entire installation work involved. Reconstruction work will last until March 1993, as the completion of each studio has to be effected successively in order to guarantee the uninterrupted transmission of ZBC's radio programmes. Studer supplies latest technology; the music and radio studio of Pockets Hill centre will use a 16 channel Studer A827 tape recorder and CD Recording/Reproducing equipment D740 as well as R-Dat recorders.

Rolf Breitschmid



STUDER REVOX at the Tell plays

1991 was Switzerland's seven hundredth anniversary, and in the autumn of that year my colleague, Benno Germann, who works in the studio planning division SRAG, recommended me to attend the Tell plays in Altdorf; these plays are renowned far beyond our national borders.

In the theatre, built in 1925 specifically for this festival, 130 lay actors gave a notable performance of Schiller's freedom drama with much enthusiasm and idealism under the professional guidance of Franziska Kolund. Within the framework of the long overdue refurbishing of the PA system, Benno Germann had installed new Studer Revox equipment himself, investing countless hours without assistance in order to achieve perfection in sound. Already six years ago a Studer 901 mixing console was installed, and three years later two A807 tape recorders, an A710 cassette recorder, an A725 CD player, an EMT 930 turntable, an A68 power amplifier and new PA speakers were added. Benno Germann is also responsible for the smooth operation of the audio equipment during the approximately 30 performances which take place every three years between July and September.

The audio quality of the installation and also the performance of the lay actors was superb and provided the visitors with utmost pleasure.

Eugen Spörri

Control room with Studer 901, 2x A807, A710, A725 and A68.

SWISS SOUND

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New patents

Procedure for erasing magnetic recordings on magnetic tape.

As we all know, magnetized areas on a tape can be erased by exposing them to a decaying alternating field of sufficient amplitude to saturate all particles. The same process is also used for recording information by adding a signal component to the alternating field. If this component is missing, this is a null recording or erasure.

The erase process is influenced by magnetizations that originate from the unerased tape immediately before the recording zone. This magnetization is picked up by the pole shoe and added to the erase field. As a result we obtain a weak rerecording of the signal that has just been erased. This rerecording occurs when two conditions are met:

1. The left-hand pole shoe must be large enough to pick up the signal.
2. The recording zone must be small enough to allow rerecording.

The present invention defines a freely selectable cut-off wave length at which neither condition 1 or condition 2 is satisfied. The result is that none of the wave lengths included in the spectrum is rerecorded. A head design that takes these findings into consideration correctly also erases low frequencies. A problem that has plagued us for 30 years has finally been solved.

This patent of Paul Zwicky was published by the European Patent Office on November 21, 1991 under the number 166 210 B1.