# Editorial

It wouldn't surprise me in the least if you thought Swiss Sound had quietly passed away since the last issue (no.30). After all, almost four turbulent seasons have passed since then. Some anxious readers inquired whether they had been crossed off the mailing list. Even the much beleaguered post office was (unjustly) thought to be the culprit. Well, what you are now holding in your hands is No. 31 - nothing happened in between.

Actually this is not quite correct. In the meantime, in line with the restructuring of the new STUDER REVOX AG, our advertising / PR / editor's office has been reorganized. The editor's office has produced a few editions of our internal company newspaper STUDER REVOX PRINT; at the same time the in-house print shop was liquidated, as a result of which we have been burdened with additional work.

However, we have not remained totally inactive. With this and the next edition we have embarked on a special mission. One of the declared objectives of SWISS SOUND is to introduce our technological developments to a broad circle of readers. The most obvious aim is to describe in detail the most complex development Studer has ever undertaken, the D820-MCH multichannel tape recorder. No less than eight different authors will contribute to this topic.

You may also have noticed that the layout of SWISS SOUND has slightly altered. Since the graphical layout of our magazine has remained virtually unchanged since 1982, minor corrections now seem appropriate. In particular we have given greater attention to the improved rendition of pictures. The new layout with 2½ columns gives us greater flexibility for different picture sizes. In this way we hope to serve our readers better.

Marcel Siegenthaler

# The digi-saur?

Leading article by Dr. Joachim Fasbender

**Dr. J. Fasbender**

The D820-MCH:
...our new "flagship"...
...most feature-packed multichannel
tape recorder...
...most powerful digital format
of all times (DASH-F)...
..."State of the art"...

High-tech superlatives are appropriate. However, top premium technology of this type quickly becomes impersonal, almost untouchable. High-gloss brochures all too quickly detract from the people who stand behind the product and the performance they have jointly created. It shouldn't be this way. This is why we would like to introduce in words and pictures some of the development team members in this edition of SWISS SOUND. But before we get too technical, a few words about the appearance of the D820-MCH:

### Nickname "Dino"

Nicknames are often based on external appearance. This applies to our 24/48 channel digital tape recorder D820-MCH. Because of its considerable size it soon received the nickname "Dinosaur". If we take a ruler and scale to compare the D820-MCH with its only (Japanese) competitor, we find that "Dino-West" is indeed some inches larger and a few kilograms heavier than "Dino-Far-East". It is the heftiest tape recorder on the market today.

However, the D820-MCH is more than a tape recorder in the conventional sense. The DASH recording format combines the precision and reproducibility of digital technology with the highest currently available number of channels (48) and recording density, which over the past few years has been able to establish itself as the dominating digital format.

In order to exploit this precision, analog/digital recorders of matching quality are required, and Dino's "good sound" has in the meantime become proverbial.

In addition to straightforward recording, the D820-MCH supports supplementary functions such as built-in synchronizer, a "sound memory" for creative post-processing of the audio material, and comprehensive automation facilities for alignment and calibration. Some of these features are currently unique; particularly among the 24-channel machines the D820-MCH has no peer.

It therefore comes as no surprise that "Dino's" guts are filled with technology that is anything but primordial. The 66 installed "intelligent" processor chips would easily suffice for a medium-size computer center. And the precision of the tape deck is widely known because it has been adopted from its analog siblings.

As you can see, Dino's physical size is not its only superlative, and it consequently is not surprising that the process of its making was also quite unique; 30 "fathers" (i.e. engineers) with an international flair have contributed to its development over many years. Needless to say this was the greatest engineering and development feat in the company's history.

There were many hard nuts to crack and clever ideas were needed. Of course, we can only mention a few in this edition:

**STUDER D820-MCH - a general overview**
by Dr.Daniele Pelloni
**System control**
by Daniel Senn
**Digital audio processing**
by Dr.Christoph Musialik

And to keep you in the picture here is a list of topics wich will be discussed in the next edition of Swiss Sound:

**Antialiasing filter**
by Paul Zwicky
**Equalizer**
by Matthias Zbinden
**Internal synchronizer**
by Kurt Schwendener
**Purpose and function of the sound memory**
by Marc Biver

PLATEOSAURUS

# STUDER D820-MCH – a general overview

by Dr.Daniele Pelloni

**Studer D820-MCH is the name for a con-glomeration of highly sophisticated electronics and super precision mechanics. The objective of this article is to give you a framework for understanding the interaction of the different sections of the most complex recorder Studer has ever produced.**

The D820-MCH is based on the well-known A820 tape transport. The tape transport control in PLAY mode is slightly different from the analog version: The capstan motor is basically controlled by the tacho pulses to obtain the desired nominal speed, as is the case in the analog machines. But additionally there is a small correction which depends on the difference between the internal 1 kHz sync signal (reference sector) and the corresponding signal from the reference track (RT, sometimes called CTL ), which is written only during the first recording of the tape. The nominal speed is directly proportional to the selected sampling rate and is exactly 30 ips at 48 kHz.

The precision achieved by the tape transports of today's analog recorders is also well suited to the digital world. The correct punch in/out operation (Insert) requires a tolerance of one block. A block is the smallest unit on a DASH tape and contains 12 audio samples. Its length is 0.1905 mm. Four parameters influence the tolerance of writing data within a block interval: The tolerance of the tape speed, the variation in the tape tension, the error in the distance between playback and record head, and the "overshooting" in the capstan control algorithm. The precision tape transport also increases the reliability of the machine, e.g. if the RT track is lost, the machine is still able to read the data correctly for up to 2 minutes in free running mode, maintaining the old tape speed when RT still was present.

## Head block and head electronics

The headblock contains four heads. The first one on the left is an erase head for the 4 auxiliary tracks RT, TC (SMPTE/EBU time code), CUE 1 (left channel) and CUE 2 (right channel). The second one is the NEW REC (or ADVANCED REC) head. It is a thin-film record head for all 52 tracks of the DASH format (48 channels and 4 auxiliary tracks) and is used for formatting a virgin tape or overwriting an old tape completely. The third head is a ferrite playback (PB) head for all 52 tracks. The last head (SYNC head) is identical with the NEW REC head and is used for SYNC recording (Insert).

In order to work in sync recording mode, the RT track at least must have been written before. In NEW REC mode, read-after-write is possible. For digital audio data recording no erase head is necessary. The HDM-1 code used in the DASH format has its main spectral energy between 128 and 384 kHz, so previous data is totally erased by overwriting it with new data.

### Erasing of CUE and TC tracks

There are two possible recording methods allowed in the DASH format for the auxiliary tracks CUE and TC: with a wide-gap head using bias recording, and with a narrow-gap head using pulse width modulation (CUE) and
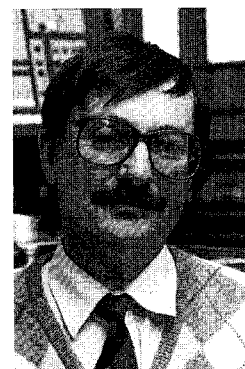
*Dr. D. Pelloni*
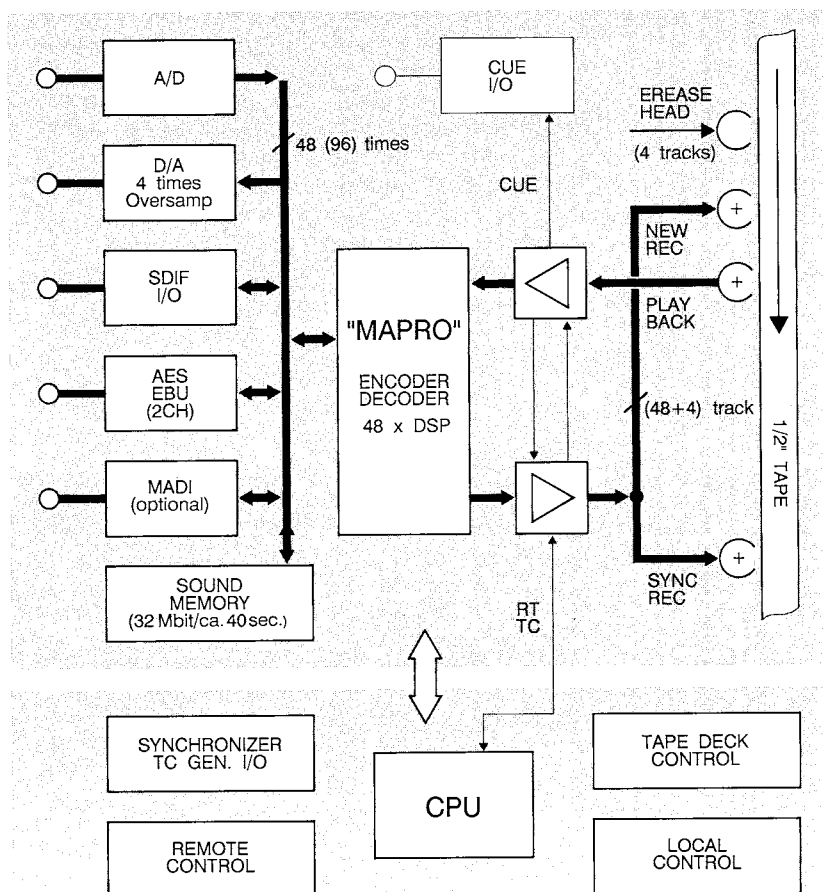


**STUDER D820-MCH**
*Block diagram*

biphase modulation (TC). It is impossible to overwrite a bias recording with a narrow-gap thin-film head, because of the very deep magnetization on the tape. In order to assure full compatibility between tapes, a special erase head for CUE and TC has been introduced.

The erase head is always active in NEW REC mode and can be activated by the user (only CUE and TC tracks) in SYNC REC mode.

Pulse recording

The record electronics is based on pulse train recording. Instead of writing the HDM-1 coded digital data (audio, biphase coded RT and TC, PWM coded cue signals) directly as square wave signals (which could destroy the thin-film head by overheating), these signals are resampled with the run frequency of 2.3 MHz with small pulses (length about 100 ns) and then transferred to the head as a pulse train.

The second edge of the PWM modulated cue signal is not on the HDM-1 data grid, because it is determined by the level of the analog cue signal. For this reason the last pulse of the pulse train is shifted according to a special Studer patent. This results in a very high quality of the CUE signals.

Pulse recording amplitude

The amplitude of the pulses to be recorded, the so-called record current, is a very important parameter for digital audio recorders. If it is too low, the CRC error rate may increase rapidly during playback, and if it is too high, tape overwriting could be a problem. The DASH document specifies the rules for setting the optimum value, and also the tolerances (which are very strict: +/- 5% for digital audio data, and +/- 10% for auxiliary data) in order to guarantee tape compatibility.

In order to adjust the record current exactly and efficiently (104 values for 52 tracks and 2 record heads should be determined) an automatic method has been developed. This method allows to optimize all of the 104 parameters with a calibration tape, and finally to keep them memorised in the RAM of the machine.

The playback electronics consist mainly of the PB-preamplifier, the analog equalization circuitry and the digital phase lock loop (DPLL) circuitry.

**Digital audio path**

The head electronics is bidirectionally connected to the six MAPRO (**M**ultichannel **A**udio **PRO**cessor) boards which together represent the heart of the machine. They perform all digital audio processing functions, which are required in a DASH recorder:

a) DASH encoding
b) DPLL circuitry for clock recovery
c) Time base correction and Sync Separation
d) DASH decoding
e) Splice & error concealment; configuration of the data paths according to the different modes; crossfades between different signals, and generation of special test signals.

The digital audio processing of the machine is based on a modular 8-channel concept. The input source for each channel is determined by the user. There are five possibilities: A/D converter, MADI, SDIF, AES/EBU 2-channel interface, and the internal sound memory. The output of each channel simultaneously supplies the D/A converters, each of the digital interfaces, and the sound memory.

The digital audio data from all of the 8-channel groups is collected on the so-called ping-pong board. The inputs and outputs of all 48 channels are available on this board simultaneously. The routing of each channel to each other can take place here. Sources and destinations are controlled by software. Local routings on the 2 AES/EBU and the sound memory board are also implemented in order to assign 2 resp. 4 sources and destinations to/from the 48 channels. The communication between the 8-channel groups and the ping-pong board is assured via the six I/O boards, controlled by the six MAPROs.

The D820-MCH recorder supports all common forms of audio signals. Because of today's typical studio environment the analog inputs and outputs are still the most popular ones. For this reason the quality of the A/D and D/A converters is critical for the sound impression. We spent a lot of time developing converters with a good sound.

The AES/EBU 2-channel digital audio interface has emerged as the worldwide accepted standard, even though many implementations

*Digital audio electronics*
*Block diagram*
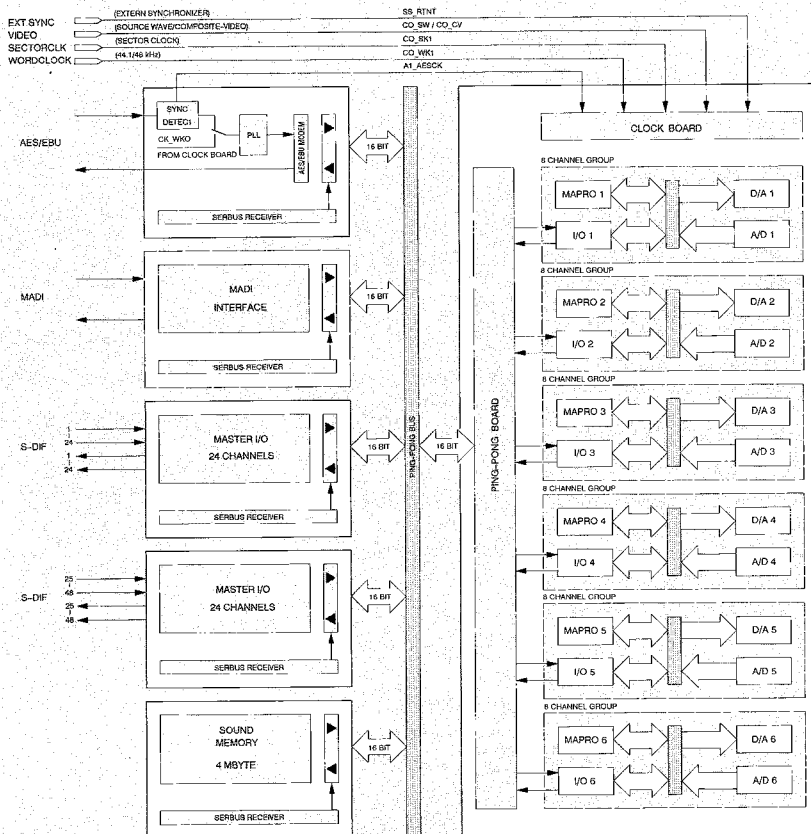
exist on the market, depending on the supported format (consumer or professional) and the channel status implementation. The AES/EBU input circuitry we have chosen is very tolerant and should accept all of today's existing implementations. The output corresponds to the professional standard. In the AES/EBU definition two channels are supported. The user is allowed to choose any two of the 48 channels.

The recorder supports the balanced SDIF I/O format, which is presently one of the few possibilities to transfer multichannel digital audio data in a compact way. The D820-MCH is already prepared to install the MADI interface. MADI is a kind of multichannel AES/EBU format, and will probably become the worldwide standard for digital audio multichannel I/O .

## Synchronization and interfacing to other systems

In many applications the digital recorder is not a stand-alone unit, but is integrated in a complex system with an editor or controller, many different audio recorders, and also video machines. To have a working system, not only the digital audio data format interface but also the synchronization signals interface as well as the control interface(s) should be carefully defined.

Synchronization

In the D820-MCH the synchronization is performed by the internal synchronizer board and by means of the internal clock board, which allows the machine to work with several external clock sources.

The available external clock references are: **Word clock** (48/44.1/44.056 kHz square wave or extracted from the AES/EBU signal), **video clock** (square wave or composite video/composite sync signal with 25/29.97/30 frames/s) and **external varispeed signal** (9600 Hz corresponding to 30 ips).

If no clock reference is available, the "TC LOCK MODE" can be selected: The machine never switches to the preselected clock reference, but remains under the control of the external timecode all the time.

The machine also contains a timecode generator which is always locked to the external video signal (if present). If "SYNCHR VIDEO" has been selected, the timecode is generated in such a way, that it is exactly in phase with the incoming video clock.

Interfaces

In order to control the D820-MCH, several interfaces are available. The most important are:

**a) parallel interfaces** "tape deck" and "synchronizer". The two 25-line parallel interfaces, "tape deck" and "synchronizer", have a long
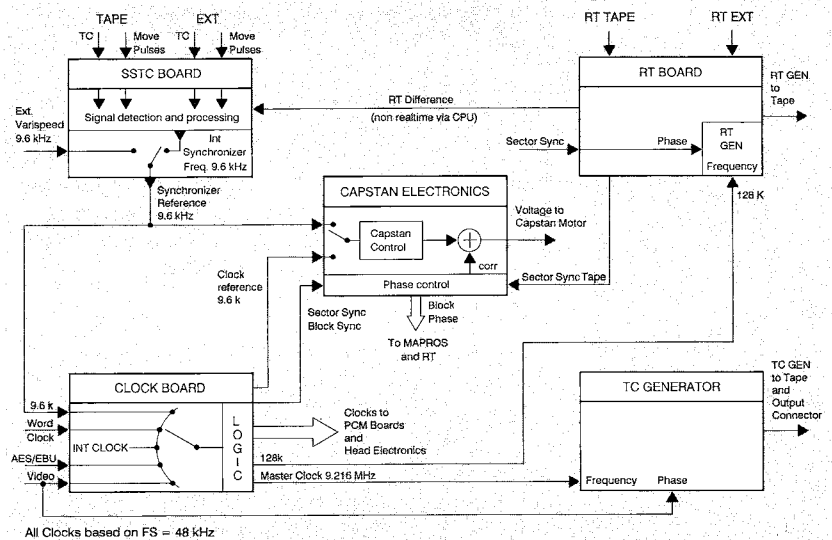


*Clock control*
*Block diagram*

tradition in analog tape recorders and are well established in the audio world.

**b) REMBUS interface.** The REMBUS is a new development and has been designed to interface the machine to its remote control units on a unique cable. Currently three control units are available: the **tape deck controller** (consisting of the autolocator, a control unit for the internal synchronizer, and the control module for the sound memory), the **audio remote controller,** (which allows to set each channel individually to on/off, ready/safe, input/repro, emphasis on/off), and the **parallel audio interface (PAI,** which concentrates the many LED/key information going/coming to/from a mixing console and transmits them through the REMBUS cable).

## "Sound memory" board
The D820-MCH contains an optional 40 sec sound memory board, which offers the following features:

*a) Sound memory:* Up to 4 tracks can be copied from tape or from the external inputs into a memory of 10 seconds, 13 seconds for 3 tracks, 20 seconds for 2 tracks and 40 seconds for 1 track. There are 3 different ways of filling the memory: "Instant", "Continuous" and "Trigger". The memory can be edited and played back or looped indefinitely.

*b) Track slipping:* Any track n between 1 and 24 can be delayed by an adjustable amount of time during playback and sent to output n+24 or copied onto track n+24 on the tape. The maximum delay is 40 sec for 1 channel and 1.8 sec for 24 channels simultaneously. All channels are delayed by the same (adjustable) amount. ◉

# System control

by Daniel Senn

**D. Senn**

**A salient feature of all Studer equipment has always been utmost convenience of operation. In the design of the D820-MCH this tradition has been faithfully continued.**

On the D820-MCH all parameters such as input and output level, recording currents, tape tension settings, emphasis filters, digital/analog input combinations, interface selection, entire channel configurations or even system configurations can be entered with the push of a button or be modified and stored via a personal computer. Also the assignment of the two AES/EBU inputs and outputs to any two digital audio channels or direct digital copying from the digital audio tracks to one or several other tracks (digital routing, PING PONG) is possible by pressing only a few keys. The software provides the D820-MCH with a few additional attributes, which are hard to find in competitive products.

## Architecture and system control

The system control is based on a Motorola 68000 processor (CPU) and is the nerve center in which all incoming information is processed and from which corresponding control signals are output. The CPU is supported by 6 IPCs (so-called intelligent peripheral controller). These IPCs are used for pre- and post-processing of the incoming and outgoing control and data flows, i.e. they take care of the scanning of a keyboard or control of a display. In the D820-MCH the IPCs are used exclusively as I/O drivers. They optimize the data signals for the traffic between the CPU and the peripherals. This is particularly important in serial communication links where the interrupt rate can be very high, because an interrupt is generated for almost every received character of a message. The IPC can greatly relieve the CPU by recombining the individual characters into a complete message. As a result, the CPU receives only one interrupt for each transmitted message and this only when the complete message has arrived. In the opposite data flow direction the CPU can transfer the message to be transmitted to the IPC which processes it character by character. As an interface to the hardware the IPC also performs translation functions, i.e. the CPU does not have to concern itself with the hardware control. It receives information that is already translated, or it can output an IPC control command in the form of simple macro commands to the hardware.

The IPCs make no independent decisions, except in an emergency, for example if the communication with the CPU is interrupted or in the event of a power failure. The IPCs respond to such a situation by executing safety programs (switch machine to stop, disable the record currents, etc.). The IPCs contain "only" control functions (programs) for their own hardware environment (periphery), i.e. *no* operation dependent functions. As a consequence the software of the IPCs needs to be changed only if the corresponding hardware changes.
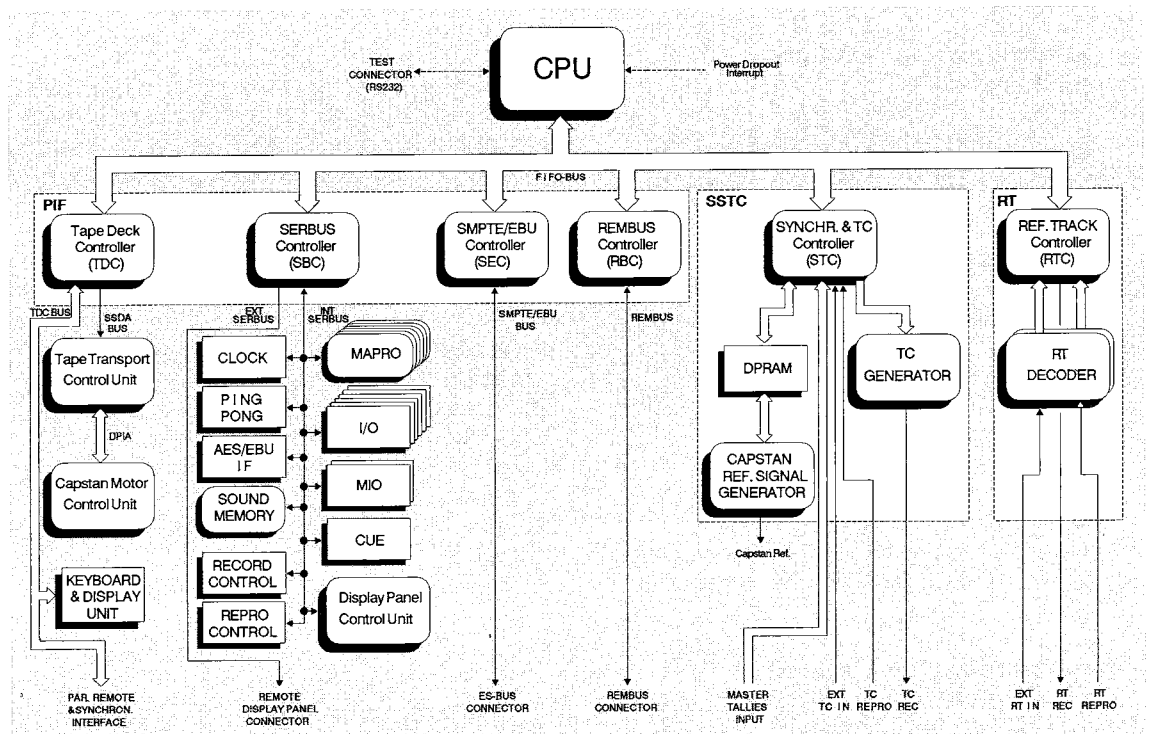


**Fig. 1:**
*System control architecture*

Fig. 1 illustrates the system control architecture of the D820-MCH and the hierarchical break-down into several command levels.

## Processors of the D820-MCH

On the top level of the hierarchy there is the CPU, followed by the 6 IPCs (TDC, SBC, SEC, RBC, STC and RTC). These IPCs communicate directly via the so-called FIFO bus with the CPU and perform the following functions:

**TDC:** The *tape deck controller* decodes the events from the tape deck keypad, the cue and shuttle wheel as well as the parallel remote and the external synchronizer interface, and transmits these to the CPU. It checks the display units on the tape deck (LCD, LEDs and time display). Via the well-known SSDA interface (from the analog A820 family) it establishes the connection to the tape deck control (TAPE DECK MPU) and thereby relieves the CPU from the SSDA interface handling.

**SBC:** The *serbus controller* prepares the data for transmission between the CPU and the SERBUS transmitter/receiver (port master). In addition it transmits the audio level data from the I/O boards to the display units (bargraph).

**SEC:** The *SMPTE/EBU controller* is responsible for processing the data from or to the ES-bus. It thereby relieves the CPU from processing the protocols of all ISO layers up to the USER level.

**RBC:** The *rembus controller* establishes the connection to the remote units. Like the SEC it handles the protocols of all ISO layers up to the USER level.

**STC:** The *synchronizer & TC controller* is one of three processors on the SSTC board. On the one hand it transmits the control commands of the CPU to the time code generator (TMS 320), on the other hand it decodes an external time code and the time code from tape. In addition it monitors the capstan synchronization. In this task it is supported by the reference signal generator.

**RTC:** The *reference track controller* is one of the three signal processors (TMS 320) on the RT board. Aside from its function as an internal RT generator it transmits to the CPU the decoded external information as well as the RT information from the tape of the two other signal processors.

The other processors are:
 TD **MPU/capstan MPU:** These processors which are also found in the analog tape machines control the tape deck and capstan unit.

**MAPRO/SOUND memory:** these controllers are discussed in separate reports and consequently are not covered here.

**Display panel controller/remote DPC:** These perform the periodic scanning of the keyboard and the control of the display units of the display panel(s).

The remote controllers connected to the REMBUS perform the same task for the corresponding remote control units.
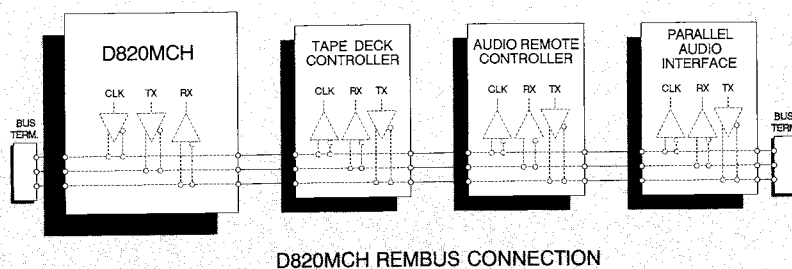


**D820MCH REMBUS CONNECTION**

## Processor communication

The processor communication takes place via different bus systems. These are partially new developments (FIFO bus, SERBUS, REM-BUS), while others have been adopted from the A820 (SSDA bus, TDC bus, ES bus).

**FIFO BUS:** The CPU board consists of two parts: The FORCE/SYS68k board and the CIF board (with FIFO bus interface). The FIFO bus is a parallel bus with a transmission rate of approx. 700 Kbyte/s (simplified VME bus). The CPU communicates point-to-point with each IPC. Because the largest part of the peripherals are character oriented, a FIFO has been selected as the ideal connecting link. This is why it has been named "FIFO bus".

**TDC bus:** The TDC (*tape deck controller*) bus is basically the externally terminated microprocessor bus of the TDC. It corresponds to the bus of the Series 820 master MPU. Via this bus the TDC controls the display driver for scanning the tape deck keys and for controlling the LEDs and the LCD. Also connected to the TDC bus is the parallel remote and synchronizer interface.

**SSDA bus:** The SSDA bus (*synchronous serial data adapter*) is a synchronous, bit-serial point-to-point transmission interface (38.4 Kbaud). It interconnects the TDC on the PIF board and the tape deck MPU in the tape.

**DPIA bus:** The DPIA bus (*dual peripheral interface adapter* bus) is a parallel 8-bit interface between the tape deck and the capstan MPU which communicate with each other via two separate "ports" (bidirectional).

**SERBUS:** The SERBUS is a synchronous, bit-serial bus with a transmission rate of 1.25 Mbit/s. This is an enhanced SYSBUS of the D820X. In addition a custom chip (PORTMAS-TER) for receiving and transmitting the data has been developed. The data transfer is controlled by the SBC (SERBUS controller).

**ES bus:** The ES bus is a bus system (48.4 Kbaud) with network capability. It will not be discussed further here because comprehensive documentation on the ES bus is already available.

**REMBUS:** The REMBUS is based on the Intel high serial communication controller (HSCC 82520). The master-slave bus principle or point-to-multipoint principle is used. The transmission format is similar to the X.25 LAP B specifications, which are particularly well supported by the HSCC. The REMOTES can transmit messages to the D820-MCH independently under control of the HSCC which controls the bus access. The D820-MCH has priority over the REMOTES because there is no contention on its transmission line. This is particularly advantageous because the D820-MCH has to send data to other device ports (remotes) by using so-called global messages which can be received simultaneously by all remotes.

The REMBUS supports the connection of several remote control units (see Fig. 2). This means that in the future additional units (such as a PC interface) can be connected to the D820-MCH.

### Example of a process sequence: REC [A] key

The keyboard and display controller 8279 scans the tape deck keypads every 2 ms. It is programmed by the TDC in such a way that any key depression triggers an interrupt [B]. This interrupt suspends the momentary activity of the TCD. The interrupt task KEYINT is activated and reads in the matrix organized state of the 8279, translates it to a keyboard code and transmits it to the KEY QUEUE buffer. Through this entry the KEY task is synchronized (activated) by the real-time operating system and it fetches the code from the QUEUE. A QUEUE is necessary because the interrupt task KEYINT reacts much more quickly than the KEY task. In this way more code keys can be buffered in a short period of time than can be processed by the system. The KEY task now calls the FPUT function which writes the message into the FIFO. This write operation triggers a MESSAGE interrupt [C] which is generated by the FIFO and signals a new message to the CPU.



The CPU interrupts its momentary activity when the TDC FIFO interrupt is received. The FIO interrupt task is activated and reads the message code from the FIFO which states that a message is located in the FIFO buffer. The FIFO itself signals this with an entry in a MAILBOX. The effect of this entry is that the real-time operating system activates the TDS task which has been waiting for this event ever since the FGET function has been called. The FGET function then supplies the TDS task with the message read from the FIFO buffer. TDS now decodes the message and calls the corresponding KEY function. The KEY function now determines, for example, whether or not the PLAY key is already pressed or whether the recorder is ready to execute the RECORD command. If this is not the case the message will be ignored. If the message is acceptable, the KEY function calls the main function RECORD which is now transmitted to COMMAND. The latter contains a facility, which by means of semaphores protects certain functions against multiple activation.

The RECORD function then first checks whether all record conditions are satisfied (e.g. in NEW RECORD all channels must be switched to READY). If everything is in order, it initiates the RECORD phase by calling a special sequencer. This sequencer makes sure that the now initiated operations are executed in the right sequence according to a specific timetable. For the purpose of simplification let us assume that the recorder is already in PLAY mode. The sequencer detects this and calls the output driver for the record head electronics. This output driver now enables the record current for all channels in READY mode by generating the corresponding command for the SERBUS interface and transmitting it via the FPUT function to the SBC [D].

The SERBUS controller then receives a MESSAGE interrupt [E] from the FIFO. The SBC immediately interrupts the current task and switches to the COMMAND interrupt task, which reads out the FIFO via the FGET function. The message is decoded by COMMAND, which in turn calls the corresponding function. This function converts the command with the parameters into a SERBUS command string and transmits it via the SBPUT function and the SERBUS to the corresponding PORTMASTER chip [F].

The PORTMASTER transmits the signals to the head electronics, which in turn enables the record current.

The record sequencer in the CPU now calls additional output drivers, which ensure that the MAPROs are switched to record and the status LEDs are updated. After the record sequencer has been started, the TDS task returns to its starting position and waits for the next event from the TDC.

### Tasks, events, mailboxes, queues & semaphores

To ensure that the overall system can process the external events in a timely manner, the processors must be coordinated with each other. This coordination has been optimized by using real-time operating systems in the individual processors. This means that the software is internally structured in such a way that the control processes are split into several individual processes, so-called tasks. These tasks are synchronized to the externally arriving events. The communication takes place via mailboxes, a sort of electronic mail box with built-in synchronization; in other words in the D820-MCH practically all processes are event controlled.

The principle is explained by a small example in the box opposite. Certain processes, which are initiated when the "REC" key is pressed, are traced.

To ensure that the tasks respond quickly to these events, the real-time operating systems have to satisfy very demanding requirements (e.g. very short task switching times). These operating systems have been partially developed in-house and are continually being improved, which means that we now have a high-quality operating system. For the CPU we use the operating system VRTX32 [1] which has also proven itself.

### Summary

Because of the modular structure of the CPU software and the decoupling of the hardware control through the utilization of IPCs, the system is easier to service and permits changes and enhancements for future software releases. The different control interfaces (which can still be partially extended) and the capability of accessing all parameters and functions via software across these interfaces make the D820-MCH a trend setting studio recorder which can be integrated into practically any environment. ◉

# Digital Audio Signal Processing

by Dr.Christoph Musialik

**Audio signal processing functions such as fade-in and fade-out going into stop, play or record, or crossfade while switching between input and repro, are relatively simple to implement in an analog recorder. Similar functions in a digital recorder, however, require very fast and complex hardware with mathematical capabilities. Usually, the same hardware is also used for various error processing functions, such as interpolation, mute and splice.**

Considering the large number of boards already contained in 2-channel digital recorders, the development of a multichannel version was a true challenge for Studer engineers. Of course, it was clear that a 48-channel recorder should not be 24 times larger than a two-channel one and its development should not take several times longer. Such market conditions forced us to consider new concepts. The two most important measures taken here were the prolific use of SMD and ASIC technologies to increase component density, and the use of programmable components to ensure a good margin of flexibility.

These rules, enhanced by some unique circuit solutions, resulted in a very reliable, high-quality audio electronics of a size comparable to a typical audio box of an analog 24-channel recorder with Dolby SR.

### 8-channel group concept for digital audio processing

The digital audio circuitry of the 48-channel DASH recorder D820-MCH consists of six identical 8-channel groups. Each group covers four 8-channel boards: analog-to-digital converter, digital-to-analog converter, digital input/output interface and the so-called MAPRO (**M**ultichannel **A**udio **PRO**cessor).

### A/D and D/A Converters

The 8-channel A/D converter board is implemented with a single high-speed, high-quality 16-bit hybrid converter. In order to achieve absolute phase coincidence, 8 separate sample & hold circuits are used. They are sequentially switched to the A/D module via an analog multiplexer. The converter needs only $1.5\,\mu sec$ to output a digital result with 16-bit accuracy. Because there is no oversampling, the overall delay time (analog and digital) of the A/D board is kept at a very low level (approx. $100\,\mu sec$).

The antialiasing filters on the A/D board were designed to achieve a good compromise between the attenuation value at half of the sampling frequency and the "musicality" of the

converter. In order to save supply current (48 channels!) the analog filters are based on a new passive structure (paper by Paul Zwicky in the next Swiss Sound).

The input stage in each channel is implemented with a special low-level distortion transformer developed by Studer. The input gain is independently adjustable with a multiplying converter (MDAC) via a serial control bus.

The D/A converter board is implemented with high quality standard components. A 4-time oversampling filter and a 16-bit low-distortion D/A converter are used. The reconstruction filter is phase-linear and very gentle. The output stage is balanced and transformerless. Special circuit design allows also asymmetrical loads without any output level penalty. The output level can be set by the MDAC to any typical value independently for each channel. The A/D and D/A converters have an optional switchable pre-emphasis respectively de-emphasis circuitry.

During the development of the A/D and D/A converters we paid much attention to their "musicality". Intensive subjective tests with sound engineers, musicians and producers in a well known recording studio showed that there was statistically no difference between direct sound from a reference quality CD-Player and the same sound that was additionally "processed" by our A/D and D/A converters connect-



*Dr. Ch. Musialik*
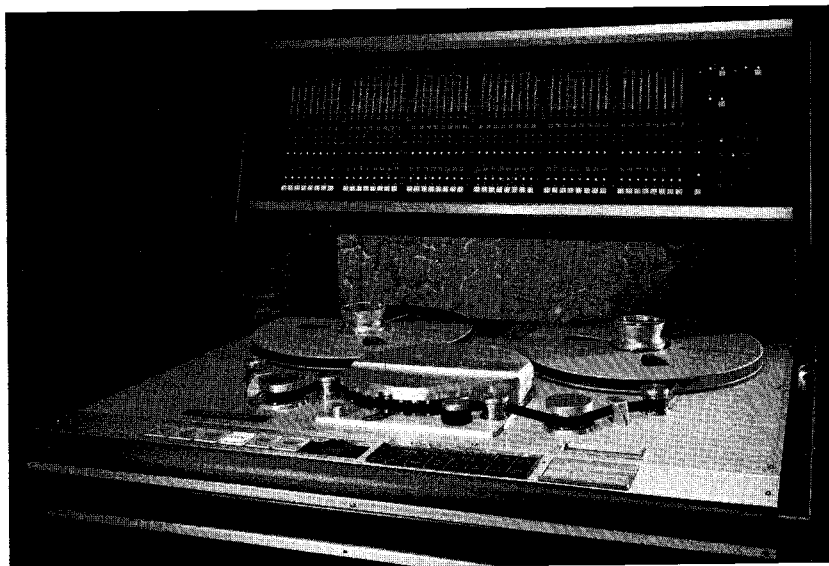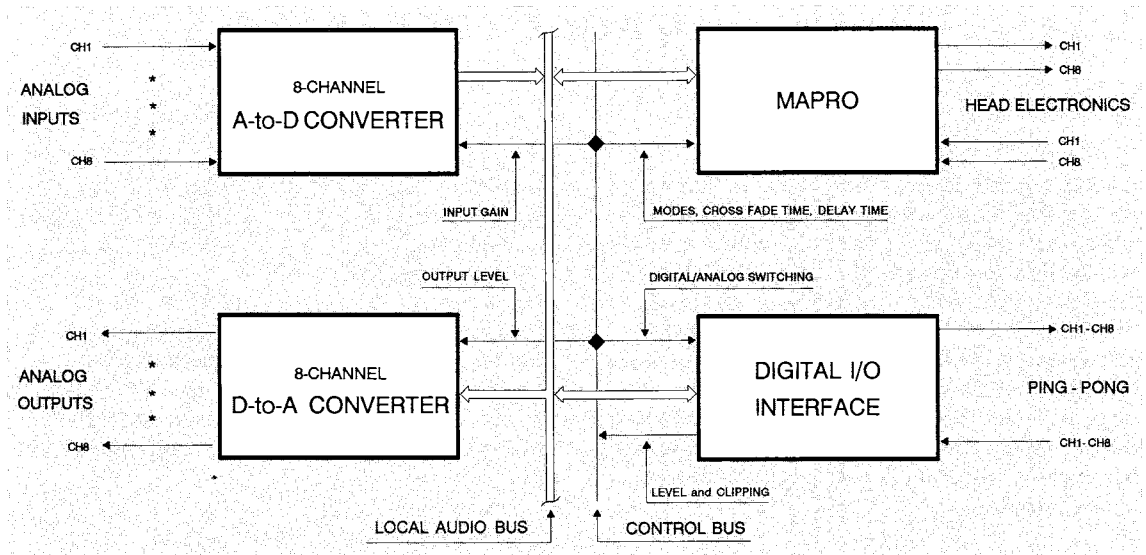
**Fig. 1:** *Block diagram of the 8-channel audio processing group.*

ed serially to the CD-player output. In spite of these two additional conversions there was no penalty in the sound quality. Even with three consecutively connected pairs of our converters the differences were almost imperceptible.

**MAPRO Board Concept**

The MAPRO board can be described as the brain of each 8-channel group. It receives digital audio from either the A/D converter or the digital interface and supplies digital audio to both the D/A converter and the digital audio interface. Furthermore it produces the DASH coded and HDM1 modulated signals for the record electronics and receives HDM1 modulated signals from the playback path (Fig. 1).

The concept of the MAPRO board is different in comparison to other solutions on the market. In today's multichannel recorders a function-based board concept is applied, i.e. separate boards are used for the main digital processing functions such as coding, decoding, signal processing etc. If one of these boards malfunctions, the whole recorder is also out of order.

We employed the channel-based board concept, i.e. the whole digital signal processing circuitry belonging to a particular channel is implemented on the same board (in our case 8 complete channels per board). If one of these boards is out of order, the remaining 40 channels are still working. Another advantage is that the channel based board concept achieves far greater reliability because there are fewer different boards in the system; we need six times the same four boards.

Audio Data Interconnections

The audio data exchange inside the 8-channel group is established via the parallel LOCAL AUDIO BUS (Fig. 1). The inputs of the A/D converter and the outputs of the D/A converter are directly terminated on external XLR connectors. The digital audio I/O signals for the eight channels are available through a serial audio data link between the I/O interface board and the Ping-Pong board which concentrates digital audio signals for all 48 channels (Fig. 2). The four 8-channel boards receive control data and send status data from/to the main CPU via the SERBUS, a special Studer-developed serial link for transmitting control data. The SERBUS link is part of the CONTROL BUS that also supplies timing signals needed for the synchronization of audio data to the particular time slots incorporated with each 8-channel group.

**The MAPRO board**
Functional Description

For clarifying the central functions of the MAPRO board, the 8 main blocks (Fig. 2) must first be briefly explained:

1. *DASH ENCODER* (ENC): formats the incoming sequence of digital audio samples into error-protected blocks and modulates them into HDM1 code (record path);

2. *DPLL* (Digital Phase Locked Loop): Performs the edge synchronization of the incoming HDM1 signals from the playback head path;

3. *TBC/SYNC.S* (Time Base Correction and Sync Separation): For HDM1 demodulation, block detection, CRC error detection and time base correction (playback path);
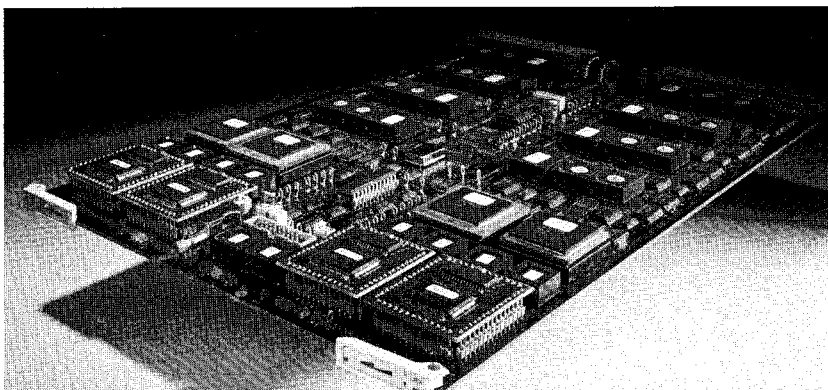
4. *DASH DECODER* (DEC): Deformats the blocks coming from the TBC, performs the error correction and generates the Splice, Word Error and Mute Request flags for the digital signal processor (playback path);

5. *DSP:* Concentrates all the signal paths, performs the different fading operations and splice & concealment routines and distributes the signals according to the chosen mode (stop/play/ record, input/repro, newrec/sync-rec/ advsyncrec);

6. *TIMING block* (CLOCK): Generates all required clocks;

7. *SERBUS* Chips: Performs the serial communication with the Master-CPU (Mode, Cross-fade Time, Delay Time etc);

8. *LED DISPLAY:* For verification of the total MAPRO performance (CRC Error, Word Error, Splice, Mute, DPLL Lock, Test Mode).

## DSP Functions

Each of the eight signal processors on the MAPRO board concentrates all audio signals pertaining to one channel and manages them depending on the CONTROL WORD which contains the channel status information. Specifically, the signal processors perform the following functions:

1. *FADER OPERATIONS* according to the selected mode:
- fade-in and fade-out in the record and playback path,
- crossfade for input/repro switching,
- crossfade for sync record mode,
- fade-in and fade-out for mute condition.

2. *SPLICE and CONCEALMENT OPERATIONS* when the incoming samples are in error:
- selection of the proper error concealment routines, depending on error pattern,
- odd/even masking and crossfade between data before and after the splice point.

3. *DELAY COMPENSATION* in Advance Output Sync Mode to compensate internal and external delays in the audio signal path.

4. *DIGITAL DITHER* computing.

5. *TEST SIGNAL* generation.

Fig.3 explains the three main modes of the digital multichannel recorder D820-MCH. The audio signal routing and all faders are shown. All required states of the recorder can be derived by appropriate controlling of the faders and switchers.

NEW REC Mode (Fig. 3a) is used only for virgin tape recording or tape formatting. Monitoring (read after write) is possible in this mode via the playback head.



**Fig. 2:** *Block diagram of the MAPRO Board* ▲



*Fig 3:* The main modes of the STUDER D820-MCH digital recorder: a) New Rec Mode
▼ b) Normal Sync Mode
c) Advanced Output Sync

a) New Record Mode

b) Normal Sync Mode

c) Advanced Output Sync Mode
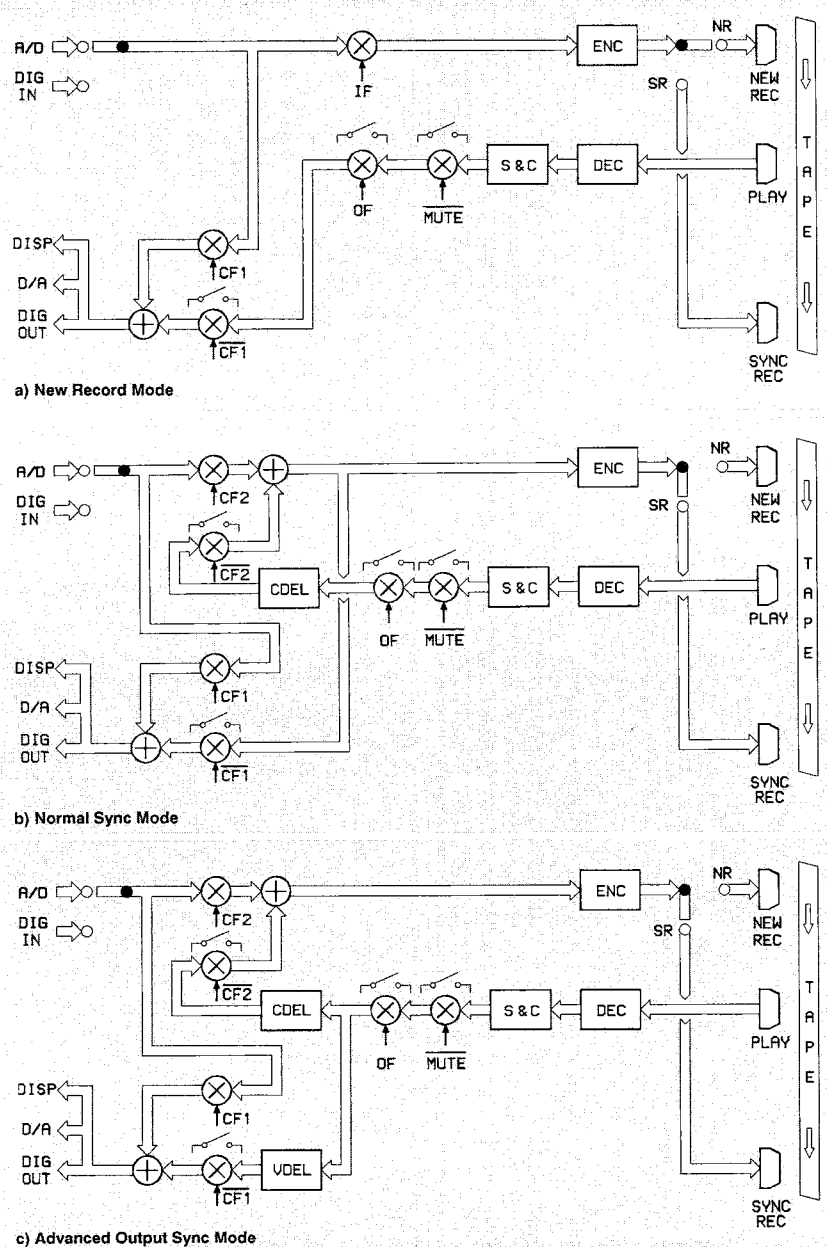
In Normal SYNC Mode (Fig. 3b), insert and assembly recording can be performed with the help of the so-called Sync Record Head which is placed after the playback head. Parts of the old tracks can be overwritten by new information and takes can be added to the end of existing tracks without any adverse audible effects. The only condition is that the delay on the tape between the playback and sync record
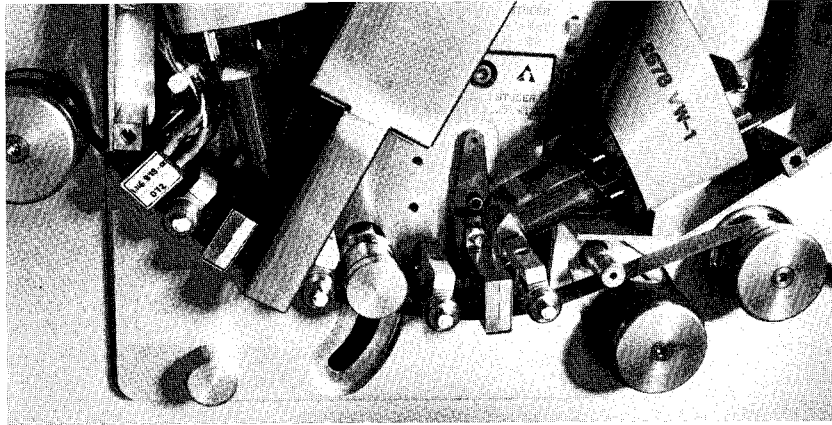


heads must be exactly the same as the electronic delay between the output from the reproduce head and the input to the sync record head. The crossfade time between old and new audio data (crossfader CF1) is selectable between 0 and 683 ms.

The third mode (Fig. 3c) called ADVanced Output SYNC Mode, is used for synchronized recording of additional tracks to already existing tracks or for track copying, especially if an external digital mixing console is used. Digital consoles are characterized by relatively long delay times. In such cases the Advanced Output Sync Mode allows absolutely exact alignment of new tracks with the existing tracks. The delay compensation time for external

nal delays is selectable between 0 and 1.7 ms. Of course, also internal delays incorporated with the digital as well as analog inputs and outputs are compensated in this mode. Also track-to-track copying with the ping-pong function is performed without any delay.

All fade operations are dithered by the same DSP 15-bit bipolar dither with rounding is employed.

## The global states

To avoid excess rounding and truncation operations after multiplying and accumulating and hence to preserve computational power, the number of faders has been reduced in a special way. The serial faders in one path have been combined into one. The remaining faders, now called GLOBAL FADERS, allow the implementation of all the desired functions, provided that certain rules concerning the choice of the increment and decrement steps are observed:

– When one fader in a chain is turned down, the global fader must be turned down. The step size of the global fader is equal to the decrement of this fader.
– When several faders are turned down, the step size of the global fader is equal to the biggest decrement of the chain.
– The global fader is only turned up when all faders in a chain are turned up. In this case, the step size of the global fader is equal to the smallest increment in the chain.

In addition to the above described fader reduction algorithm, all possible states of the recorder have been analyzed to determine the distinguishable states. For example, "new record safe" for the recorder is for the MAPRO board the same as "new record stop". The states which remain after this logical consideration and after the addition of the Global Fader rules are called GLOBAL STATES.

These reduction algorithms lowered the software costs and increased the overall system reliability.

## Splice & Error Concealment

In order to protect the recorded material against drop-outs, the audio samples must be coded. The DASH coding, for example, allows the complete correction of a few thousand erroneous samples per million, if they occur randomly. This means that the lost samples are exactly calculated from the code redundance.

If the destruction of data on the tape exceeds the decoder's correction capability, the only thing that the decoder can do is to set the word error flag of the erroneous samples. The lost samples can then be approximated with the help of interpolation.

**Fig. 4:** *General concept of the Splice & Error Concealment Algorithm.*
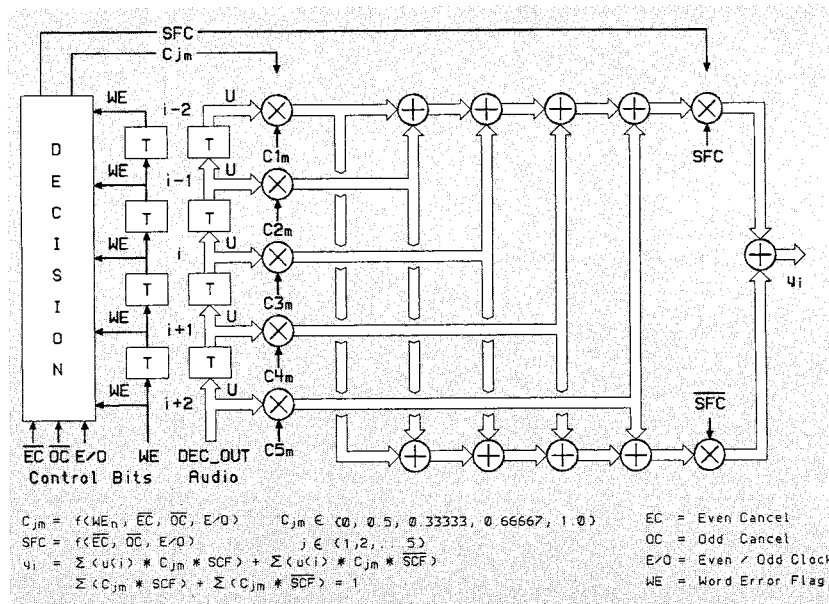
The general idea of the interpolation mechanism in the recorder is shown in Fig.4. A window of 5 samples is continuously observed. Depending on a combination of good and bad samples, the best interpolation algorithm is selected. The 32 different error situations can be reduced to 10 distinguished modes. Four of them are shown in Fig.5. Of course, if the sample u(i) is good, then y(i)=u(i). If all five samples are bad, the last output sample will be held (see Mute Request further below).

A special situation occurs if the tape has been cut, if an insert recording has been made without proper use of the sync mode or if a longer drop-out has happened. To avoid any noisy audio transients, a splice mechanism is available in the DASH format. Using this feature, old data can be crossfaded with new data.

The splice operation, controlled by two splice flags from the DASH decoder, is also performed by the signal processor. In the first splice phase even samples are masked, then odd data is faded out and even data is faded in, and at last, the odd data is masked. Of course, the splice operation is continuously accompanied by error concealment activities. During the splice condition, 128 modes are theoreti-

**Fig. 5:** Examples of Error Concealment Modes (no splice).
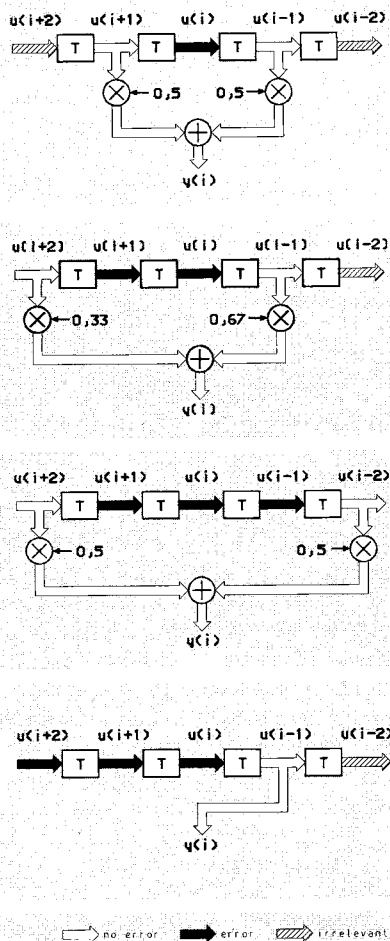


## Spontaneous contribution from the development team

# A matter of course?

**Of course,** the largest development project in the company's history was the most costly, involving a multimillion (2-digit) budget (regardless of whether you are thinking in Swiss Francs or in Dollars).
**And of course,** such an investment takes many years to pay off.
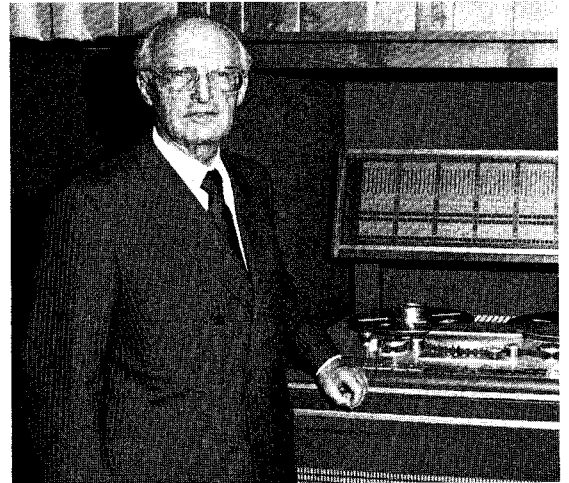


*A proud Dr.W.Studer at the inauguration of the first STUDER D820-MCH on March 21, 1990, in the Powerplay Studios, Pfäffikon-Zurich.*

However it was not a matter of course, rather an **unselfish** and far-sighted, courageous decision of Dr.W.Studer's to start the D820-MCH project just a few years before his retirement and well past his 70th birthday. It was an investment into the company and its employees.

**Thank you.**

cally possible. By using some rules, the number of modes can be reduced to 32 distinguished modes. Some of them are shown in Fig. 6.

If the data destruction on the tape is so great that after decoding more than 5 consecutive samples are erroneous, no reconstruction is possible. To protect our ears as well as the equipment connected to the recorder's analog output, a MUTE REQUEST signal is generated before the bad cluster reaches the mute crossfader. This flag activates the mute crossfader in the signal processor (Fig.3).
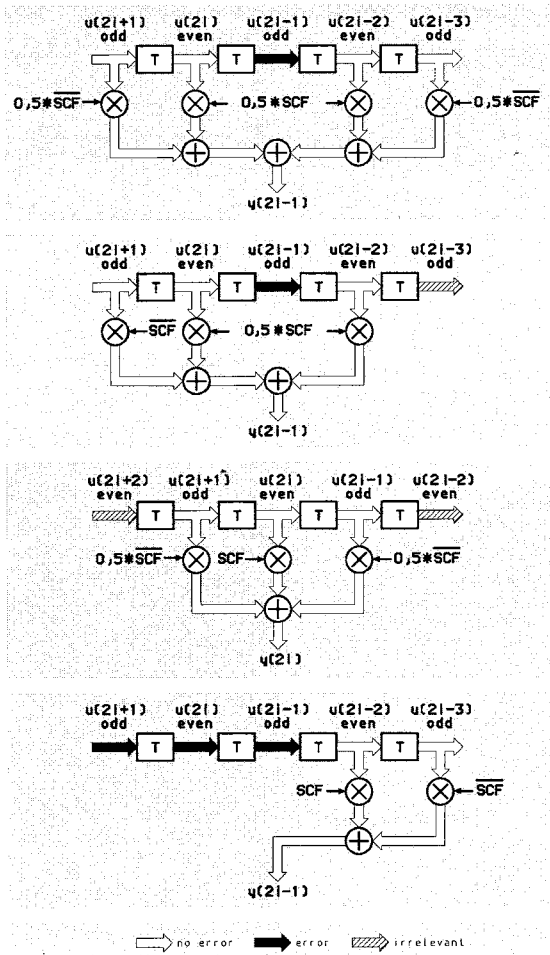
*Fig. 6:* Examples of Error Concealments Modes in splice condition.

## Software and Hardware "Tricks"

The number of operations to be performed by the signal processor during one period of the sampling frequency is quite large (Fig. 7). The DSP applied to the MAPRO board, a TMS320C10(15) with a 25 MHz clock at 54 kHz sampling frequency (48 kHz + varispeed margin) runs 115 cycles per sample period. This is not enough to perform all the functions described before, if using standard programming methods in a typical application environment, even on the condition that the above-mentioned optimized algorithms (see Global States) are applied. The decision process needed to select the right operation mode from 8192 possible combinations requires at least half of the available cycles if implemented as a tree structure with branch instructions.

A major issue was to reduce the decision expense by hardware generation of the specially composed MAPRO CONTROL WORD. It is a vector which causes the processor to select the proper configuration of the routines using only a small number of instructions. The required hardware consist only of a few simple standard logic IC's.

In addition to the hardware support, some special DSP-specific software design rules were strictly observed, thus reducing the overall number of required instructions during one sample period.
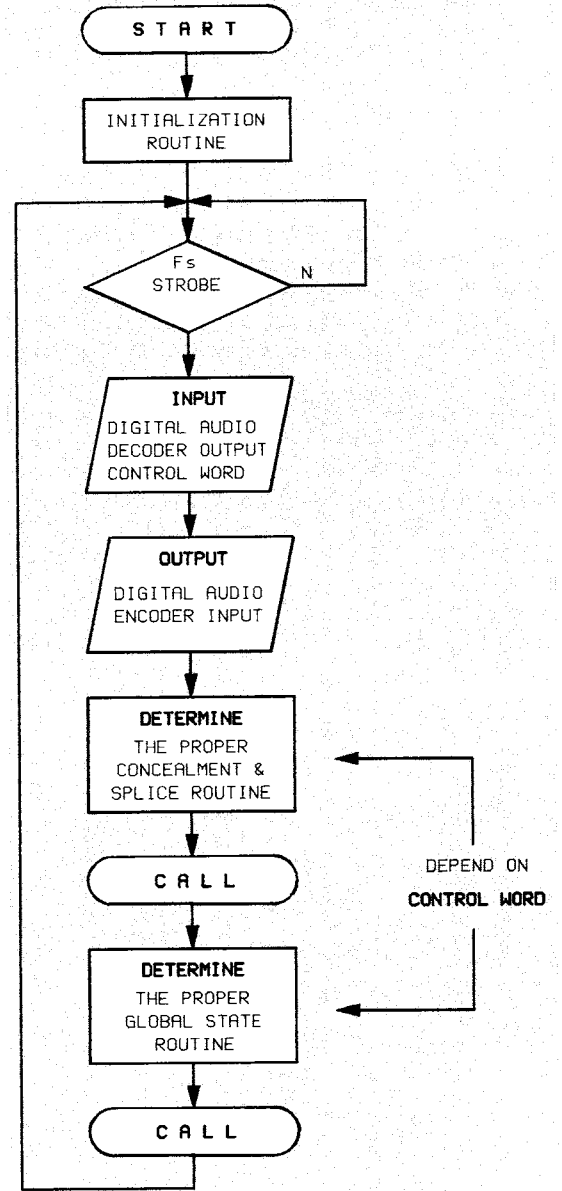


*Fig. 7:* Flowchart of the DSP main program.

## Conclusions

The MAPRO board design shows that the complex digital audio processing functions, which are typical for any digital recorder, could be efficiently implemented with general-purpose digital signal processors, resulting in a cost-efficient alternative to today's discrete hardware solutions. Thus, the development time was markedly reduced and the design flexibility preserved, opening up the possibility of adapting the system to future requirements. New or improved functionality may be still expected, and quick redesign may be implemented in response to user feedbacks. Because of the extensive use of the SMD, ASIC and multilayer technologies, the component density was greatly increased in comparison

with previous solutions. Due to CMOS technology a low power dissipation was achieved.

However, excellent digital electronics is not enough to produce an outstanding digital tape recorder. The overall sound quality is still strongly dependent on the interfaces to the analog world: A/D and D/A converters. It is not easy to achieve an unadulterated analog sound on the output of a digital recorder. Any high-quality analog circuitry design in a largely digital environment needs a lot of experience and a sure instinct. In addition, the analog circuitry itself, i.e. filters and input/output stages, should fulfill the much more stringent specifications with respect to noise and distortion than is required for analog recorders, so that the possible converter resolution is not impaired by the analog circuitry.

Keeping these principles continually in mind, we tried to design cost-effective (48 channels!) and nevertheless good neutral sounding converters. I believe that we have been successful; ever since the introduction of our recorder on the market, its sound quality has been appreciated by producers and sound engineers. Thus, our traditional striving for "musical" converters and top audio quality has been once again recognized and rewarded. ●

---

New CD software from Studer

# "Wheels"

In this section we have reported on various size extensions to our CD sound libraries on several occasions. We are now proud to be able to introduce our latest library "WHEELS" comprising 24 CDs.

As the name implies this library contains sounds produced by cars, trucks, buses and motorcycles. Not only the latest types of cars from American, Japanese and European manufacturers are represented but also vintage cars, such as the legendary Ford model T. All sequences of the library, logically structured with all imaginable internal and external sounds, have been newly recorded from digital masters.

"Wheels" is now in stock: the order number is 10.241.113.00. The number of the new comprehensive catalog is 10.241.189.01 and should be ordered separately.
With "Wheels" the Studer sound library now

comprises 130 CDs. Excluding the libraries for "production music" the selection is as follows:

| | | |
|---|---|---|
| Series 1000 | Sound library | 28 CDs |
| Series 2000 | Sound library | 22 CDs DDD |
| Series 3000 | Atmosphere | 12 CDs DDD |
| Series 4000 | Hollywood | 5 CDs DDD |
| Series 5000 | Wheels | 24 CDs DDD |
| | "Lucas" library | . 6 CDs DDD |
| | "Sampler" library | 6 CDs DDD |

The production music library comprises 25 CDs and is the ideal base for audio dubbing and commercials for which original music is desired. As with all Studer libraries, the purchase price includes all rights, except those of public performance.

The music library is continually being expanded. These extension sets (of which there are already 2 comprising 5 CDs each) are also well suited to new users to whom the complete library appears too large.

| Order number: | Set I | 10.241.122.00 |
|---|---|---|
| | Set II | 10.241.123.00 . |

For additional information please contact our local distributors, who will be glad to assist you.

Jan van Nes
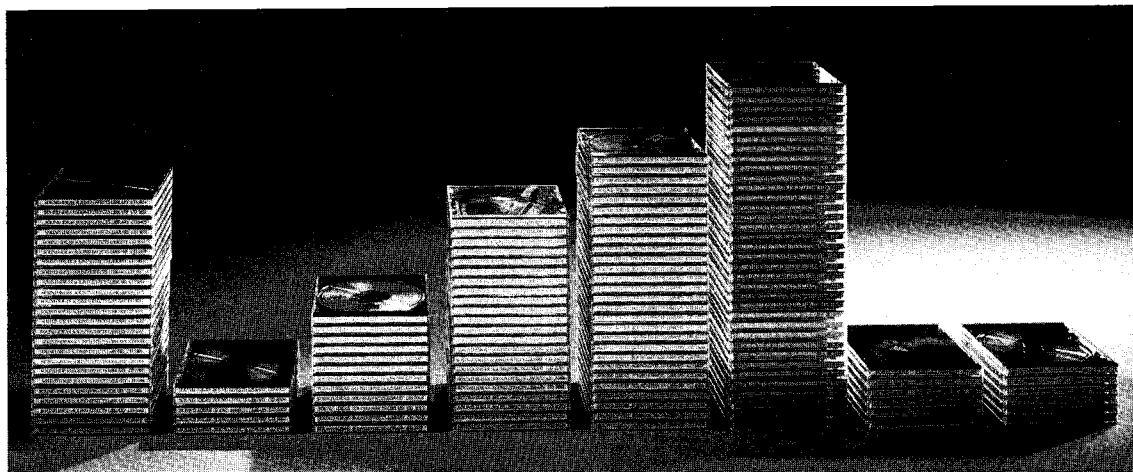
STUDER A827 Multichannel

# Fascinating rationale

by Lorenz L.Sauter



**L. Sauter**

**Those who thought that the development of the analog multichannel machine had reached the limit of the technically feasible when the top model A820-MCH was introduced, will be in for a surprise. Although the A827 belongs to the A820 family, it is characterized by some interesting differences. Some of these will be outlined below by the product manager for professional audio recorders.**

The A827 multichannel fulfills a distinct market requirement: Unlike the A820-MCH, the "Rolls Royce" which can be equipped with every conceivable option, the A827 is a machine that concentrates on essentials but without sacrificing the traditional STUDER quality.

**The differences at a glance**

Modern studio productions are controlled exclusively from the mixing console. Frequently there is not even a direct line of sight to the tape recorder. Local indicators and controls are consequently needed only during calibration and service, whereas the productive work is performed via the remote control. For this reason the versatile bargraph panel of the Series 820 has been replaced by easy-to-read VU-meters which, by the way, are preferred by

many users. There are no other controls on the meter panel, but there is space for the local control unit of the TLS 4000 synchronizer. Optionally a small amplifier with a monitor speaker (e.g. for controlling the TC track) can be integrated.

Essential, of course, are the popular large-surface command keys on the tape deck. The tape timer display is located in the usual position. There is no shuttle wheel next to the tape deck command keys but such a wheel is available on the autolocator or on the serial remote control. The other elements of the control panel are located in logically arranged key fields. An alphanumeric LC display provides information on the machine state, guides the user through the calibration procedure and the alignment of the tape deck and audio parameters. The soft keys can be reprogrammed by the user as desired.

You will fail to find the controls of the 24 audio channels on the machine itself because they are now located on the audio remote panel, which can be positioned ergonomically in the control room next to the mixing console. For alignment and test work that require local channel operation on the machine, corresponding keys (the only non-programmable ones) have been provided below the LC display.

The front of the A827 features only two hinged covers. The upper cover provides access to the tape deck and audio control components, the lower, larger compartment contains the entire audio electronics. Underneath, next to the power supply, there is room for two 19" modules (one unit of vertical rack space each, e.g. for TLS 4000 or EMULATOR).

Conveniently accessible on the rear panel are the connectors for the peripheral devices such as audio remote, parallel remote control, serial ports for SSL interface, remote control and synchronizer, ES bus as well as the control for an NR system (Dolby A, SR, Telcom; the NRS cards cannot be installed in the machine itself).

In addition there are three D-type connectors with hardwired, aux equalized sync output for all 24 channel (e.g. control of noise gates). A special synchronizer panel permits convenient access to the connectors of an optionally installed TLS 4000.

## Proven tape deck

The tape deck, including the control logic, has been adopted from the A820-MCH without change. A reel size of 14" (356 mm) and a maximum spooling speed of 15 m/s are important for work efficiency. Heavy duty disc rotor DC motors, together with processor-controlled inertia measurement of the tape pancakes ensure very fast reaction. The tape tension sensors on both sides supply the input variables for calculating the optimum tape tension and for very gentle processing of the valuable audio tapes. Particularly in synchronizer mode, the rapid search of cue and edit points with maximum acceleration and deceleration even for short locate distances is astonishing. The switched spooling motor control is designed for high efficiency and minimal heat dissipation.

## High-precision capstan drive

A dedicated processor controls the brushless DC capstan motor. It supports record and playback in both tape directions at three different nominal speeds and with overlapping varispeed ranges. The low rotor mass and a wide varispeed range are key factors that contribute to the excellent synchronization characteristics of the machine. As a result it is eminently suited for "audio follows video" applications.
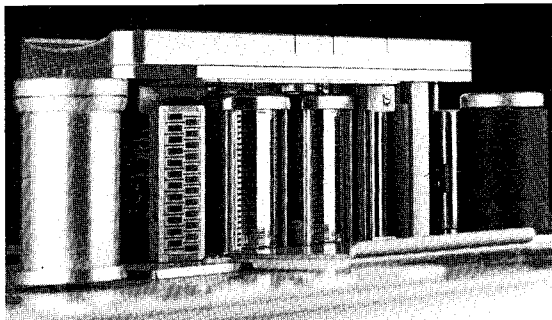


## Glass metal heads

The headblock containing the erase, record and reproduce heads (the last 2 in glass metal technology), can be exchanged without mechanical readjustment. 16-channel 2" tapes from the archive can be processed without re-alignment because the microprocessor automatically identifies the installed headblock and accesses the previously stored calibration data. Very little time is required to change the tape guide rollers to the 1" format and to install an 8-channel headblock. The audio alignment parameters for 3 headblocks and 2 tape types each are stored. The changeover between tape types as well as NAB or CCIR equalization can be accomplished simply by pressing a button.

## Simple, easy cabling

The complete deck can be tilted upward for service access. The concentration on essential features becomes readily apparent in the cabling. It comprises only three harnesses for the audio connection between the heads and the audio electronics, and flat control cables that

are clearly separated from the audio sections and routed in cable ducts. In addition to the easy maintenance, this neat arrangement is also beneficial to the audio quality.

## Audio electronics

The audio section has been completely re-designed. The objective was to drastically reduce the number of connectors, cables and boards. This is not only beneficial for the audio quality and reliability but also lowers the manufacturing costs. The result of this development is highly successful: whereas 47 individual boards were previously required for each group of 8 audio channels, the A827 now contains only 5!

A quick glance under the lower hinged cover on the front panel shows immediately how this was accomplished. Two complete audio channels, including HF electronics, are arranged on a single circuit board. Four of these are plugged into a so-called audio base board and constitute an 8-channel module that requires no further cabling. The line in/out and the sync out connectors are located directly on the back of the audio base board. The D-connector on the front of the dual audio board conducts the head signals (HF and audio) and is connected to the only audio cable that leads to the headblock. The line inputs are transformer coupled, the outputs are coreless; both can be adjusted to a studio level of -6 dB to +10 dB. The audio electronics of the A827 is fully phase compensated; all parameters such as level, equalizer, bias, bass and treble equalization are control-
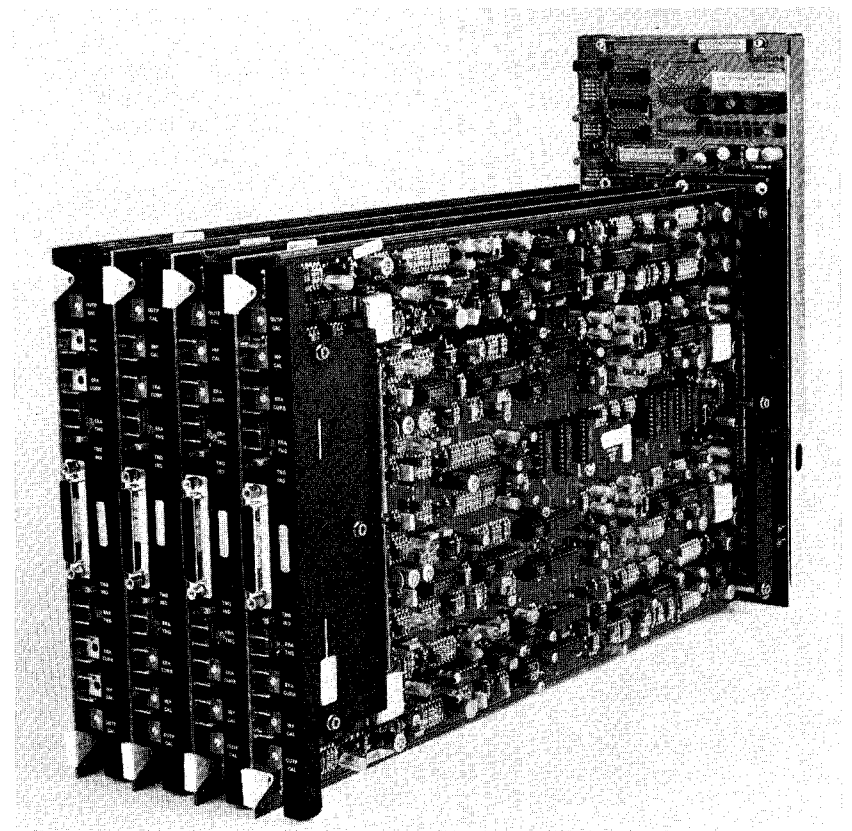
Studer A807 for the blind

# Talking Books

Björn Elfvin, Technical Manager of PHIL-STRÖM & ELFVIN, Stockholm sent us the following letter:

«We have just installed our fifth Studer A807. When we started producing talking books and newspapers for the blind in October 88 we chose the A807. Now we are producing 4000 hours of books and 1100 hours of newspapers per year. We use our Studer machines 84 hours a week and we have never regretted our choice.

The only maintenance required is regular cleaning. We have to give them a little love and care but that is no problem, they just work day in and day out».

Thank you for this nice compliment.

led by digital/analog converters. For aligning the repro side with a reference tape a level shift of +10 dB and +20 dB is implemented.

### Practical application
In practice the audio remote control will be the most frequently used operator surface. The status of all 24 channels is visible at a glance and can be set to SAFE, REPRO, SYNC, INPUT or READY with the corresponding keys. Six complete channel states can be stored and immediately recalled with the corresponding keys. Like all setup parameters, the memory content is preserved even when the machine is switched off. With MONITOR MODE, automatic muting during the spooling, capstan acceleration and deceleration phase, and automatic changeover to INPUT during tape stand still or spooling can be selected at the push of a button. The keys assigned to RECORD MODE activate MASTER SAFE, REHEARSE and switch off the record delay (time compensation between the erase and record head).

If the tape is to be saturated higher than the calibrated standard flux defined as 0 dB, the machine can be set to 0 dB(250nWb/m), +2dB (320nWb/m) or +6dB(510nWb/m) above studio level. This setting is shown on the audio remote control under FLUX. HX ON signals that the built-in DOLBY HX function is active, which at lower tape speeds achieves a greatly improved treble dynamic range. In addition to the tape deck functions, several locator functions can be recalled: Spooling to the tape address "0", spooling to the tape address at which the last STOP-PLAY or STOP-REC command was performed (LOC START), tape timer reset, freely selectable locate address and a LOOP

function (INSTANT LOOP). The varispeed functions are controlled from a separate keypad with status LEDs.

The audio functions of the A827 can also be controlled completely from mixing console pushbuttons and indicators with the aid of an optional serial interface.

### Audio alignment with microprocessor support
The tape deck alignment and the audio parameter setup require a minimum of keying sequences and consultation of the manual because the user is guided via the alphanumeric LC display. Although the A827 is not equipped with an alignment computer, this time-consuming activity is nevertheless greatly reduced through the processor intelligence. With the arrow keys you simply navigate to the ALIGNMENT AUDIO submenu from where you are prompted in the logical sequence to key in the values for LEVEL, TREBLE, BASS and BIAS. Only the internal level and the erase current must be adjusted with the screwdriver on the audio board. However, this task is only required after board repair or exchange. Because of the glass metal heads, whose characteristic remains stable throughout their extremely long life, repetition of the repro alignment with a reference tape is hardly ever necessary.

Recalibration with tape will only take place if in addition to the two tape types stored in the machine a third make needs to be processed. Subsequently the new parameters can be saved on tape and reloaded when required. In this way any number of tape types can be efficiently used with their exact characteristic.

Of course, all mechanical settings (tape tensions for all operating modes and tape types, spooling speeds, rollback time), can be conveniently set via menus and keys (ALIGNMENT DECK).

The collective term MODE SETTING refers to a multitude of adjustments with which the A827 can be adapted to any national and user preferred mode. This includes, for example, the function assignment of the user programmable keys, definition of an external noise reduction system, mute, input and tape deck behavior of the machine, spot erase, definition of remote controls and many others.

And finally a special programming lock ensures that "sensitive" settings are not inadvertently changed.
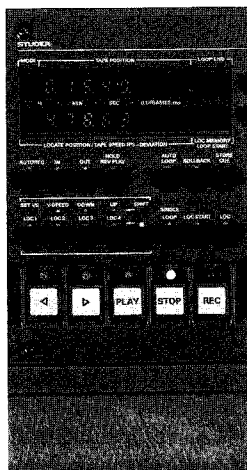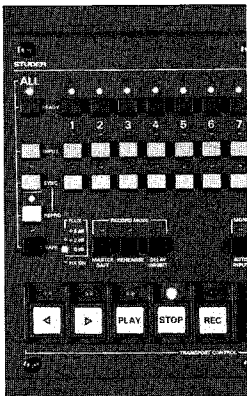
Another practical option is the audio generator with external feed possibility which acts on all audio channels. Its controls are conveniently located behind the upper hinged cover on the front of the machine. ◉

News from Studer Editech

# STUDER DYAXIS 2+2:
# a powerful and versatile workstation

Studer Dyaxis has evolved into an exciting set of products since it was added to the Studer product line 2 years ago. From that moment on when Studer took over former company IMS, R+D activities were stepped up considerably, and only a few months later the Dyaxis 2+2 product was announced. This product is a compatible extension to the stereo Dyaxis and allows 4 channel operation. Like with any other Studer product, it was also considered important that all the existing stereo Dyaxis user would be able to upgrade their system at minimal cost. With that, it does not come as a surprise that Dyaxis 2+2 sales so far have been better than projected.

And now, a new upward compatible software version of MacMix with new features can be announced:

## MacMix 3.2 - an exciting software release
For post production in TV and film, but also for premastering CDs, new functions have been made available which make work more efficient and creative. This release represents well over a 1 man year of development. MacMix 3.2 is the most comprehensive software release yet for Dyaxis.

*The new EQ (5 band equalizer) event editor* is a snap shot automation package for our popular digital EQ. The Event Editor allows you to set up different EQ and/or gain changes and have them reproduce automatically in real time. This is an extremely powerful feature that will greatly enhance the use of Dyaxis in virtually all its applications.

*"Save Selection As" with EQ* allows you to rewrite any file with an EQ setting applied to it. This means that you can have different EQ settings for each file in a mix or tracklist.

*Simultaneous EQ and meters.* It is now possible to have the meter and EQ windows up and working at the same time.

In MacMix 3.2 our *synced to timecode performance* has been improved to an amazing 300 micro seconds.

*Faster play list response time* (increases speed up to 50 %). The new user interface package is a new look for MacMix. It includes a scrolling cursor during playback for the view, mix and tracklist, highlighted elements during playback in a mix or track and a whole new graphics scheme. The result of all these changes is that it is far easier for the user to see what audio is being played back at any given time.
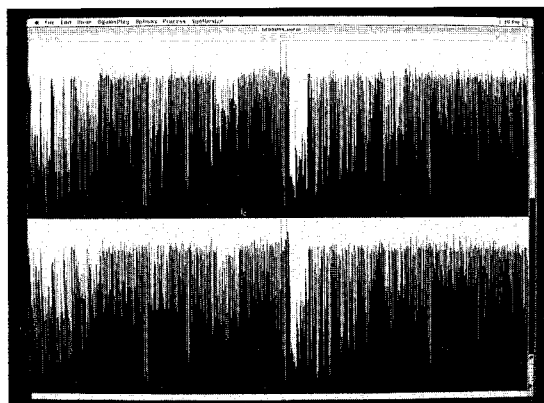


*Stereo signal, ready for editing*
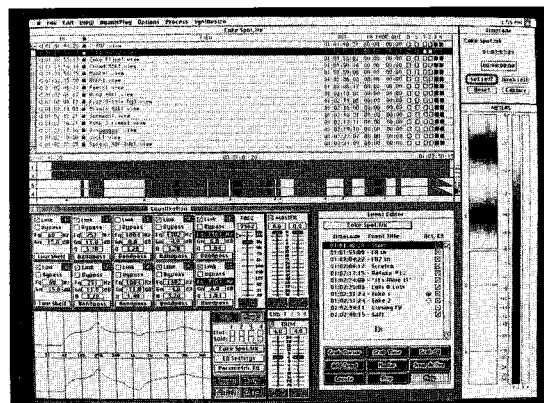


*Track list, 5-band equalizer, Event editor, Level control and Time code window*
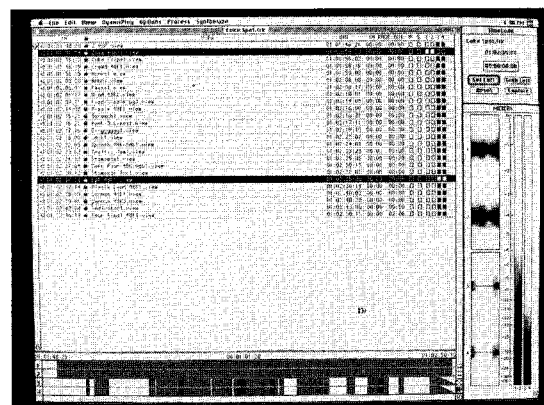


*DYAXIS in playback – on a TV commercial*

## Announcing a new Optical Drive
The introduction of the Magneto Optical has created a lot of excitement for Dyaxis users. Now, for the first time, a user can record directly to, and play directly from, a removable optical disk in the Dyaxis system. Limited editing functionality is also available. The optical disk can be used in conjunction with hard disk drives as a powerful new tool in the workstation.

This Magneto Optical drive is a 600 meg drive that yields about 28 minutes of recording time per side. Although the optical drive is

quite a bit slower than our hard drives, a system configured with both an optical and a hard disk will yield the same performance as a hard drive, yet no back up is needed.

### DYAXIS in daily use
### Video / Audio Post Production:
*Pinewood Sound Tracks* in Vancouver just received 2 Dyaxis 2+2 systems which will be used to edit sound effects for a US television series.

*HDC Communications* in Vancouver uses their 2 channel Dyaxis to edit the audio for hi-definition video programming. They selected Dyaxis because its high audio quality matched the quality of the high definition video.

### Radio Commercials:
*KWMX 101 FM.* According to Production Director Jerry Hill of KWMX-FM, the new "MIX 101" is the first radio station in Seattle, Washington, to install and use a digital audio workstation for their day-to-day production tasks. And the digital audio workstation, they chose was Studer Dyaxis. Hill comments, "Dyaxis is perfect for the daily demands of radio production".

### Premastering:
*Studio Nauta* in Tokyo uses their Dyaxis for Music mixing and editing as well as audio post production. The president of Studio Nauta, Mr. Shinkawa, tells us that he choses Dyaxis because of its high sound quality and editing versatility.

*Mix/Keysotne Studio* in Tokyo uses Dyaxis for CD Mastering. They work with a variety of digital formats and Dyaxis support of these different formats is most useful.

Sandra Hale / Peter Joss

# Studer
# Training courses 1992
**Course costs:** sFr. 250.— per day/participant

**A827/A820 MCH** (English)
**2 March - 10 March 92** (6½ days) M.Berner/L.Tejero
**Attention: on the first day we start at 14 h** (2 pm)
Tape deck features, ports, disassembling/assembling and alignment of the tape deck and audio section, explanation of various circuits, trouble shooting.
**Attention:** This is a combined course therefore participation during the whole time is required.

**A727/A730/D740/D780** (English)
**6 April - 15 April 92** R.Rüesch/S.Egli
**6 April - 8 April** A727/A730 2 days
**Attention: start time on the first day 14 h** (2 pm)
Features, ports, explanation of circuits, transport alignment

>> 8 April afternoon (from 12 h on) 1/2 day **OFF**
9 April - 10 April **D740** 2 days
Explanations to the CD-R medium, operating, explanation of circuits, disassembling/assembling and servicing

13 April - 15 April **D780** 3 days
Operating, DAT format, AES/EBU format, alignment, servicing, explanation of circuits, trouble shooting.

**Dyaxis** (English)
**27 April - 29 April 92** (3 days) G. White
Operation, applications, on site service, trouble shooting, working with the Apple Macintosh

**Dyaxis** (English)
**4 May- 6 May 92** (3 days) G. White
Operation, applications, on site service, trouble shooting, working with the Apple Macintosh

**D820 MCH** (English)
**6 May - 14 May 92** T.Knäple/S.Egli
6 May - 8 May 1st part 3 days
**Operation and Application**
– Introduction in digital technology / digital interfacing
– Setup
– Detailed operational aspects
– System synchronization
– Overall block diagram
– Trouble shooting from the operators point-of-view

11 May - 14 May 2nd part 4 days
**Electronics, Trouble shooting and Service**
– Review of setup and operation
– General block diagram
– Explanation of audio electronics, head electronics and tape deck
– Alignment and adjustments
– Disassembling and assembling
– Trouble shooting and built-in test features

**A807** (English)
**7 Sept. - 10 Sept. 92** (4 days) M.Berner/L.Tejero
Tape deck features, ports, disassembling/assembling and alignment of tape deck, explanation of various circuits, trouble shooting

**990** (English)
**23 Sept. - 25 Sept. 92** (3 days) T.Tamas/A.Ackermann
Graphic display, static automation, dynamic automation, features, operating

**961-963/970** (English)
**28 Sept. - 1 Oct. 92** (3½ days) A.Ackermann/T.Tamas
Description of basic modules 961/962, circuit explanation, alignment, differences between 963/970, trouble shooting.
**Attention: on the last day coure finishes at 12 h** (12 am)

**D740/D780** (English)
**12 Oct. - 16 Oct. 92** R.Rüesch/S.Egli
12 Oct. - 13 Oct. **D740** 2 days
Explanations to the CD-R medium, operating, explanation of circuits, disassembling/assembling and servicing
14 Oct.- 16 Oct **D780** 3 days
Operating, DAT format, AES/EBU format, alignment, servicing, explanation of circuits, trouble shooting.