SWISS SOUND

NEWS AND VIEWS FROM STUDER

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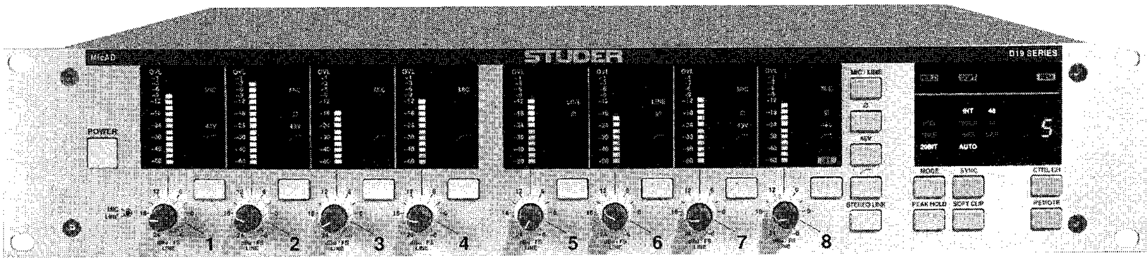
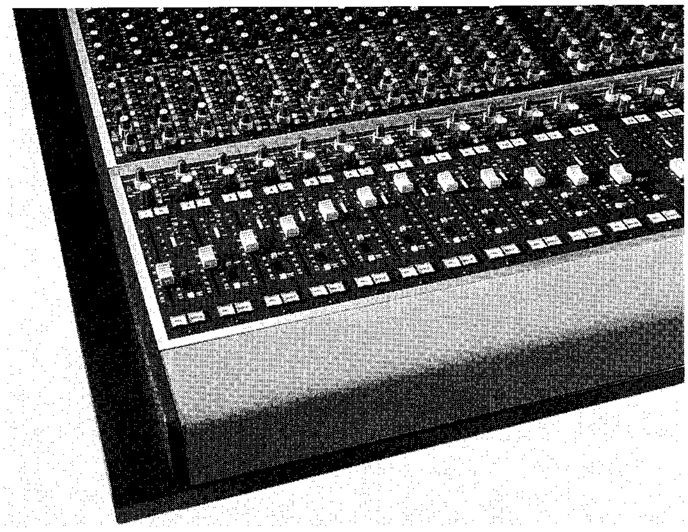
Page 2: D19 MicAD

The start of the new STUDER D19 series.
A simple, autonomous 8-channel MIC/LINE preamplifier with digital outputs in a compact 19"/2U housing.





Page 7: 980 Mixing console

The analog mixing console still has a future. Our answer to this challenge is the STUDER 980, a new mixing console that is based on the concept of the proven Series 900.

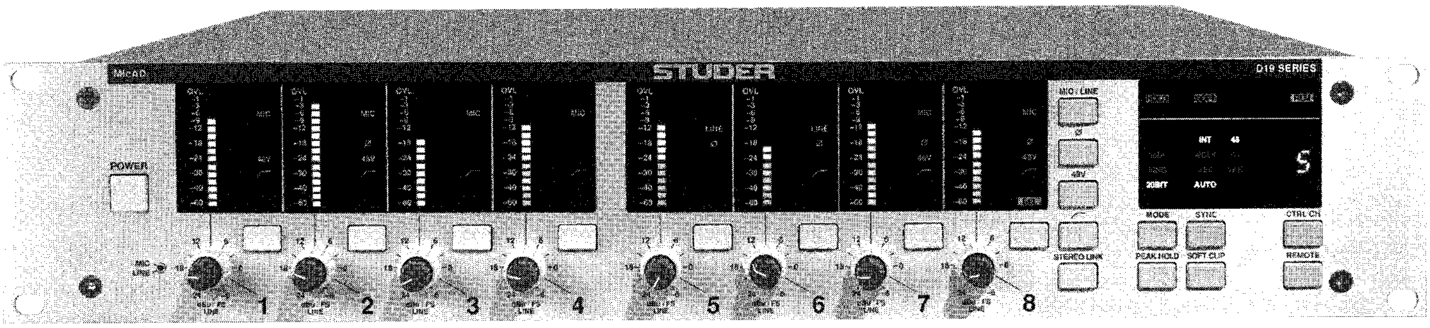
Page 20: NISKO

Trend-setting STUDER technology in the Radio Studio Zurich where the new, fully digital broadcast system NISKO has been in operation since March 10, 1995. The decisive step for Radio DRS into the digital age.

In this issue:

New 8-Channel Microphone Preamplifier

D19 MicAD - Highest Quality Mic Front End for digital studios

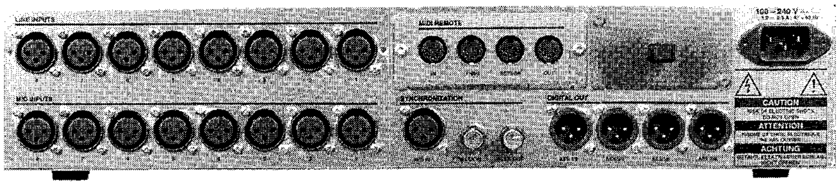


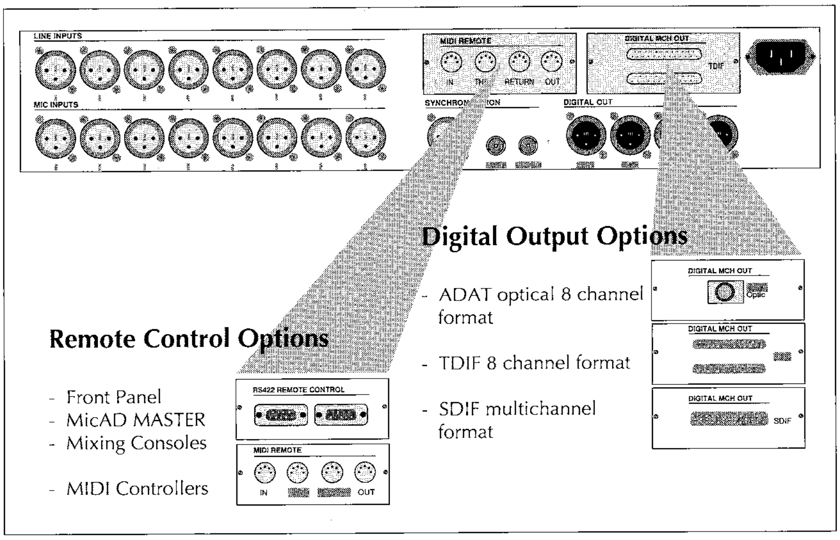
D19 MicAD and MicAD MASTER Front Panel


Rudolf Kiseljak

by Rudolf Kiseljak

Studer recently introduced a new series of rackmount products named *The D19 Series.* The intention behind this launch was to utilize the renowned Studer know-how in building «classical» analogue circuitry and the newer digital technology, thus creating products that unify the best of both worlds. On the other hand, a definite lack of high quality but affordable products to cater for the need of the digital studio has been detected. Thus, the D19 Series is aimed at the professional and the project studio alike.


D19 MicAD and MicAD STAGE Back Panel



D19 MicAD Remote Control Options and Digital Output Options

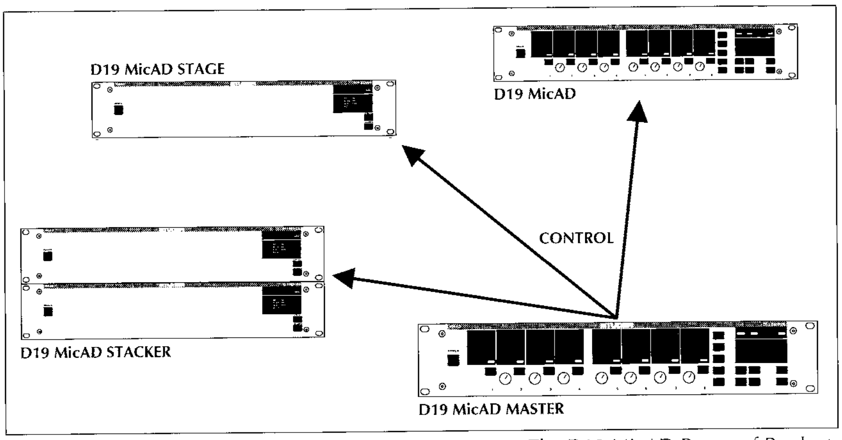
The new **STUDER D19 Series** consists of a number of 19" units, performing many useful functions such as amplifying, digital processing and transport of the audio signal. The D19 MicAD is the first product in the D19 Series.

The **D19 MicAD** is a simple, self-contained 8-channel MIC/LINE preamplifier with digital outputs in a compact 19"/2U housing.
As all other units of the D19 Series, the D19 MicAD features highest audio quality and support of industry standard interfaces such as AES/EBU, ADAT™, TDIF™ and SDIF.

The D19 MicAD features:

- 8 high-quality MIC/LINE switchable preamps with separate XLR inputs.
- 4 AES/EBU outputs on XLR plus a choice of optional ADAT™ optical or TDIF™ or SDIF output.
- 20-bit high performance A/D converters and switchable DSP Dithering and Noise Shaping.

- Local/remote control of individual GAIN, MIC/LINE, Phase, Highpass Filter and Phantom.
- Local/remote control of global Softclip, Mode, Sync and Sampling Frequency settings as well as the Peak Hold mode of the meters.
- 8 Channel meters with 16-segment PPM indication.
- Optional dedicated D19 MicAD MASTER Remote Controller.
- Optional RS422 or MIDI control port for other remote controllers.



The D19 MicAD Range of Products

20 bit A/D converter / noise shaper

The dream of a 16 bit medium

by Silvio Gehri

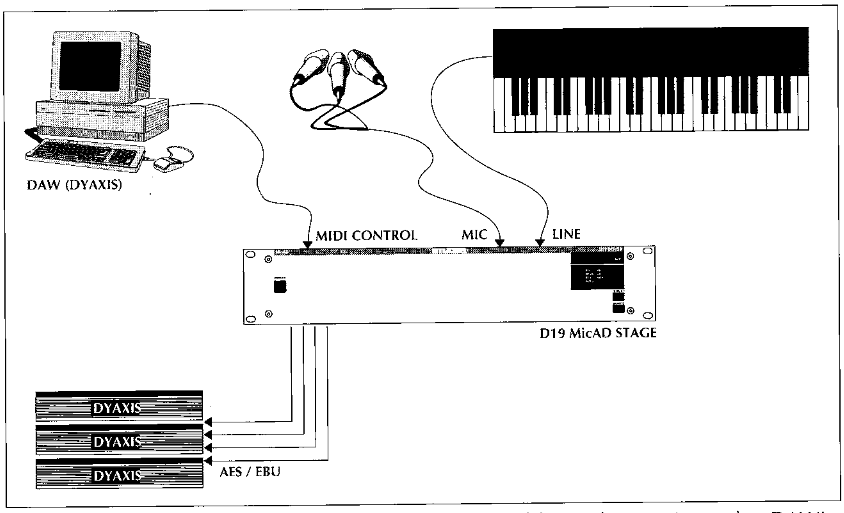
Once upon a time there was a 16 bit medium called CD which affectionately was called DASH or DAT. CD was very sad because it had to live with a dynamic range of only 98 dB. Then a fairy godmother appeared and kindly said: «Don't be sad. There are more than 16 bits inside of you.» And with her magic wand she put a *20 bit A/D converter* of the most exquisite quality into its hands. Joyfully the CD medium traveled far and wide and presented its new achievement. But few were willing to listen, they only shook their heads. The CD become very despondent.

Then the fairy godmother again appeared to the lonely medium and said: «Hello, CD, just imagine, through absent-mindedness I had forgotten to supply you with a *Noise Shaper*. No wonder everyone dreaded your sound. Without noise shaper your superb 20 bit converter looses all its qualities. Even worse, you even produce additional interference such as noise modulation or distortions. People don't like this at all.»

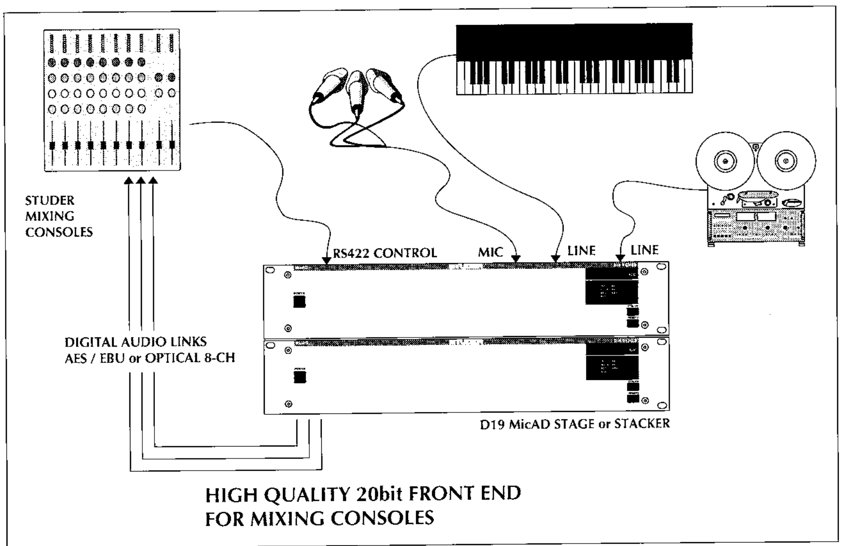
CD listened attentively. «With noise shaping», the fairy continued, «you can finally pack 20 bit quality into your 16 bits». In response CD asked: «And how will the people respond to me?» «They will listen and find it marvelous. But be honest and tell them right away that they should not be frightened by the measurements. The noise has not disappeared but rather been shifted toward the higher frequencies. Your S/N will be lower but you will sound better. And if you happen to meet a sound engineer, be sure to warn him. He may enjoy you, but with one restriction: Do not allow him to process you without crossfades. Forbid all punch ins/outs or switchovers without crossfades, is this a promise? If they don't follow this principle something bad will happen and their products will have clicks even if the switch during quiet passages or even pauses.»

The CD medium promised, thanked the fairy godmother, and went on its way. Since then many were amazed by the intricate details that were audible on this CD.

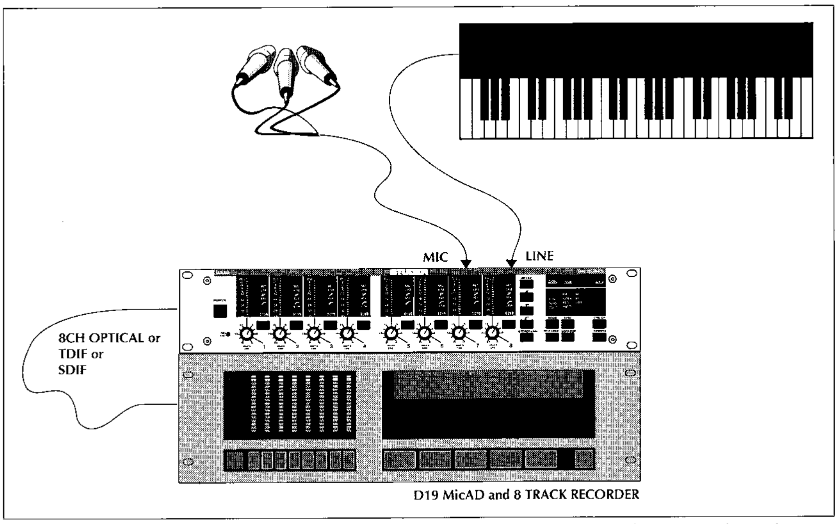
The moral of the story: **Do not forget the noise shaper!** ■



Studio-in-a-box front end for Studer Dyaxis or other DAW's.



Mic/Line front end, remotely controlled from a digital mixing console or a router.



Portable studio with an 8-CH Recorder (ADAT™ or DA88™), enabling direct cuts from the mics to the digital track.

D19 MicAD Remote Control Options

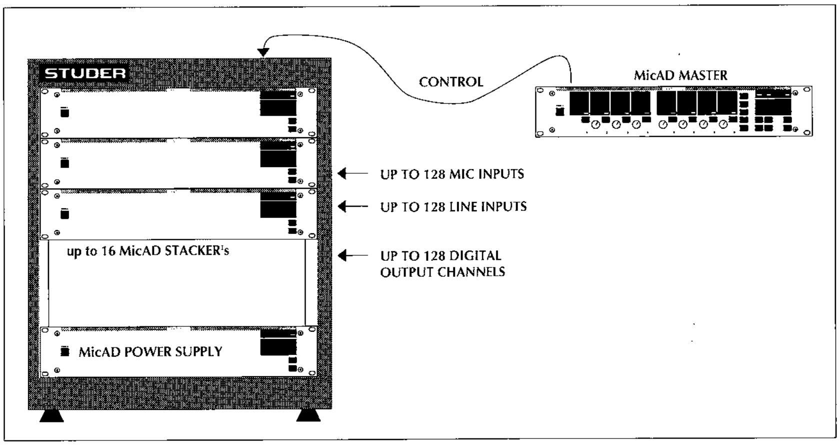
The D19 MicAD is fully controlled from:

- The front panel, locally
- The rackmount MicAD MASTER Remote Controller, utilizing the optional RS422 Remote Control port
- Studer Mixing Consoles and other Studer products, or indeed any other controlling device that supports the D19 Control Protocol. For this, the optional RS422 Remote Control port has to be fitted
- generally available MIDI controllers or DAW's utilizing MIDI protocol, via the optional MIDI Remote port

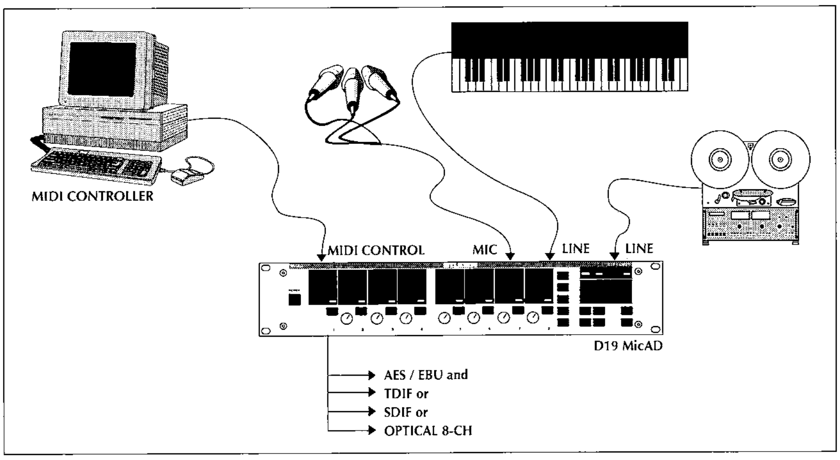
D19 MicAD Digital Output Options

The D19 MicAD features standard AES/EBU outputs. Optionally, further output ports can be added to cover the following formats:

- Alesis ADAT™ optical 8-channel format, or
- Tascam TDIF™ 8-channel format, or
- SDIF multichannel format



Stacked Mic/Line studio interface system, controlled via the dedicated remote controller MicAD MASTER.



Universal studio tool, controlled via a MIDI controller.

D19 MicAD Basic Technical Information

Analog Inputs:

- 8 separate Mic and Line inputs on XLR's, transformer balanced and floating
- 8 dual concentric potentiometers for MIC/LINE gain control
- Mic full scale = -50 ... +20dBu
- Line full scale = -6 ... +24dBu
- Functions selectable per channel: HPF, Phase, Phantom (Mic only)
- Stereo link control for adjacent channel pairs
- Selectable for all channels simultaneously: Soft Clip

Digital Audio Processing:

- 20 bit A/D conversion, 105dB S/N unweighted
- DC reject, always on
- Output wordlength 20Bit, 16bit Dither or 16 bit Noise Shaper modes

Metering:

- Peak level meter with switchable peak hold function
- 16 segment bargraphs, range -60dBFS ... OVL

Synchronization:

- Internal: 44.1 or 48 kHz
- External wordclock range: 32 ... 48 kHz ±12.5%
- External AES/EBU range: 32 ... 48 kHz ±12.5%
- Automatic, hierarchy: AES, WCK, INT
- Wordclock output always available

Digital Outputs:

- 4 AES/EBU on XLR's, fitted as standard, transformer balanced and floating
- Optional audio ports: ADAT™, TDIF™, SDIF

Remote Control:

- Optional control ports: MIDI or MIDI-RS422
- Device number (1...16), settable on front
- Remote On/Off function

ADAT is a Trademark of ALESIS Corp. USA
TDIF and DA88 are Trademarks of TASCAM Corp. Japan

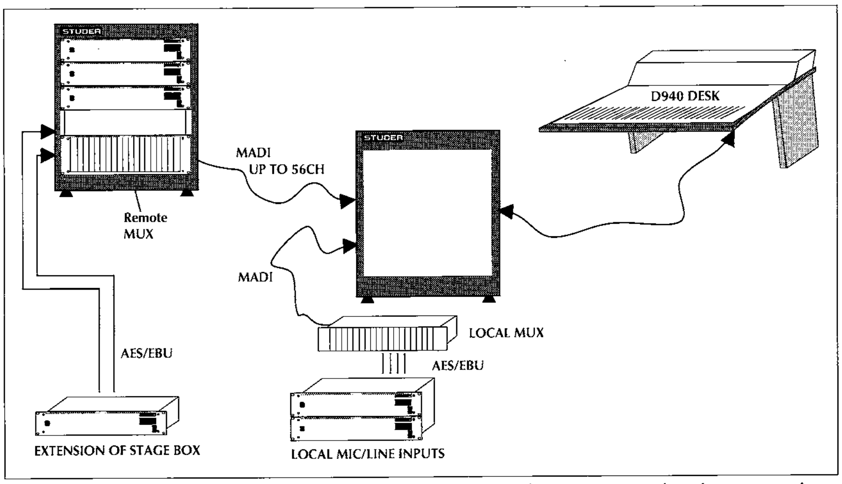
The D19 MicAD Range
The D19 MicAD is not only one product! Several versions are available to best suit specific needs:

D19 MicAD
Selfcontained fully featured 8-channel unit with integrated local control panel. For standalone applications.

D19 MicAD MASTER
Dedicated Remote Controller for up to 16 MicAD units. Controls all functions of the MicAD.
Can be used with all versions of the MicAD.

D19 MicAD STAGE
8-channel MicAD unit without the local controls, intended for use in remote applications. Up to 16 STAGE units can be controlled from one controller.

D19 MicAD STACKER
As MicAD STAGE, but with a 24V DC power input, to be used in larger installations with external redundant power supplies for greater reliability. Up to 16 STACKER units can be controlled from one controller.



Front end for STUDER D940 Digital Mixing Console, also as stagebox.

MicAD Applications
The MicAD being such a versatile product, it lends itself to a variety of applications. In the examples we try to hint at several possible ways to use the MicAD. For this, we use graphics rather than words.

Together with forthcoming new products in the D19 Series, more and more applications shall be made possible. In all cases, it is our intention to supply highest audio quality, good engineering and simple operating at affordable prices. ■

D19 MicAD project team

Big teamwork for small units

by Hans R. Hässig

The ideas for the D19 series were raised in various discussions in R&D and together with Dr. T. Frohn, the area sales manager for Germany. After the ad hoc team has been put together, the prototyping was done in a concurrent approach within two months, just in time for the first presentation at the AES in Paris and Frankfurt Musikmesse.
Using existing technology like sophisticated MIC input circuits as well as A/D converter technology was the basic idea, offering a solution with an optimum of features for a wide range of applications. This means high audio quality (20 bit), big system-connectivity, remote controllable device with an easy to operate human interface, best design, and last but not least a competitive price performance ratio.


The MicAD project team (sitting from left to right):

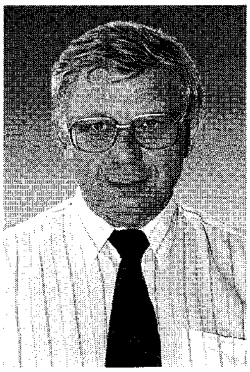
Andreas von OW, (34), El. Ing. HTL, 5 years Studer. Processor system architecture & software system design. **Patrick Zoller**, (31), technician, 14 years Studer, audio & layout specialist (low noise), CAD implementation & realisation. **Paul Grenacher**, (47), technician, 14 years Studer, analog & power supply specialist, measuring & integration. **Daniel Specker**, (37), El. Ing. HTL, 5 years Studer, controller software & MIDI merge harware. **Paul Zwicky**, (59), El. Ing. ETHZ, 34 years Studer, analog input basics & scematics, low noise transformer design.
Dominik Tarqua, (26), sound engineer, 8 months Studer, concept proposal, human interface.

(standing from left to right):

Silvio Gehri, (30), El. Ing. ETHZ, 5 years Studer. Department leader Digital Signal Processing, DSP specialist. Project leader, concept & DSP software (noise shaping). **Werner Stauffacher**, (46), Mech. Ing., 17 years Studer. Department leader Mechanical Constructions. Mech. construction & realisation, design. ■

MIC input with defined input impedance

by Paul Zwicky



Paul Zwicky

Microphone inputs should put a low load on the microphone. But an input transformer operates better when it is terminated with low impedance. For this reason it makes sense to terminate the input with a resistor. The input impedance may be 1 kohm.

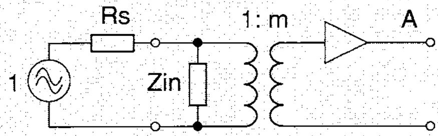


Fig. 1

The transformer now «sees» a lower signal voltage and a lower termination resistance. Distortions become smaller and the lower response frequency limit drops. However the noise figure increases by approx. 0.8 dB. Somewhat better results are achieved with the arrangement illustrated in Fig. 2.

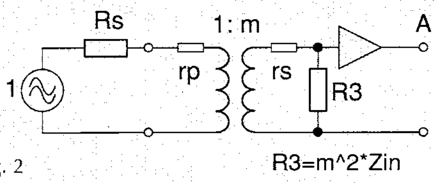


Fig. 2

In contrast to Fig. 1 the common-mode signals are also attenuated. Small, economical transformers have relative high values for r_p and r_s which are principally responsible for the noise figure. Resistor R_3 further impairs the result. This can be prevented by connecting R_3 not to a voltage of zero but to a value below zero, that is, a negative value (Fig. 3).

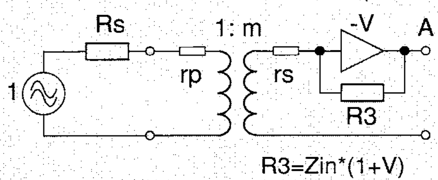


Fig. 3

The attenuation effect has remained but the noise figure is improved. If the level diagram does not allow large gain factors, the output voltage can be stepped up in the transformer.

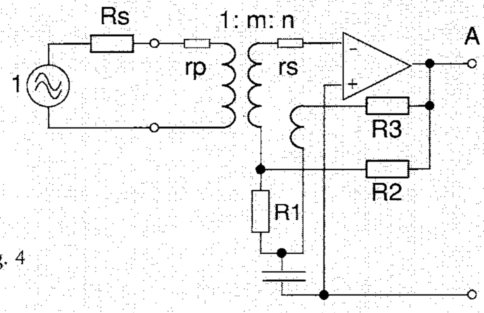


Fig. 4

An analysis of the circuit leads to the two formulas:

$$\frac{R_2}{R_1} = \frac{A\,(R_s + Z_{in})}{m\,(Z_{in} - r_p)} - 1 \quad \text{sowie} \quad R_3 = A * n * (Z_{in} + R_s)$$

To achieve the best noise figure «m» must be matched to the amplifier. «n» is selected so low that the amplifier can just handle the load R_3 or that the increase in the noise figure remains negligible.

The STUDER microphone input has a noise figure of 3.6 dB. With a source level of 0 dBu and a frequency of 40 Hz the distortions are less than 0.25% which in view of the size and price of the transformer must be regarded as sensational. Of course, the common-mode rejection of 70 dB at 20 kHz and 120 dB at 50 Hz is also good.

If even higher levels are offered, the circuit is automatically adapted to the changed situation. The negative series feedback with R_2 and R_1 are eliminated. Instead two series resistors are inserted on the primary side.

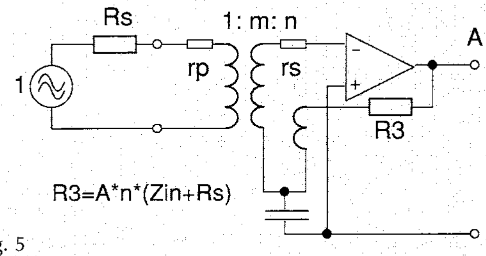


Fig. 5

The transformer works with a voltage of zero and practically without distortion. The gain is determined with R_3 and «n». Our circuit can handle levels of +20 dBu with distortions of 0.03% throughout the entire audio frequency range. ■

Right on Success

Regional Radio Studios in Egypt

by Rolf Breitschmid

Our customer, ERTU in Cairo, selected our offer for the supply of equipment for three Regional Radio Studios, Tanta, Elmenia, and Aswan. Each of the center are equally equipped.
The contract amounts to CHF 2,2 Mio., and includes 16 Series 900 Mixing Consoles, 3 complete Master Control Rooms, 19 pcs Monitor Loudspeaker A623, and accessories, spare parts and training. ■

980 mixing console

Why analog?

by Hermann Stierli

The first mixing consoles of the new STUDER 980 series have already been delivered. The purpose of this report is to briefly introduce this mixing console, its capabilities and implemented solutions. But first I would like to discuss some basic questions that were posed even before the development was started. The principal question was: Does an analog mixing console still have a future today?

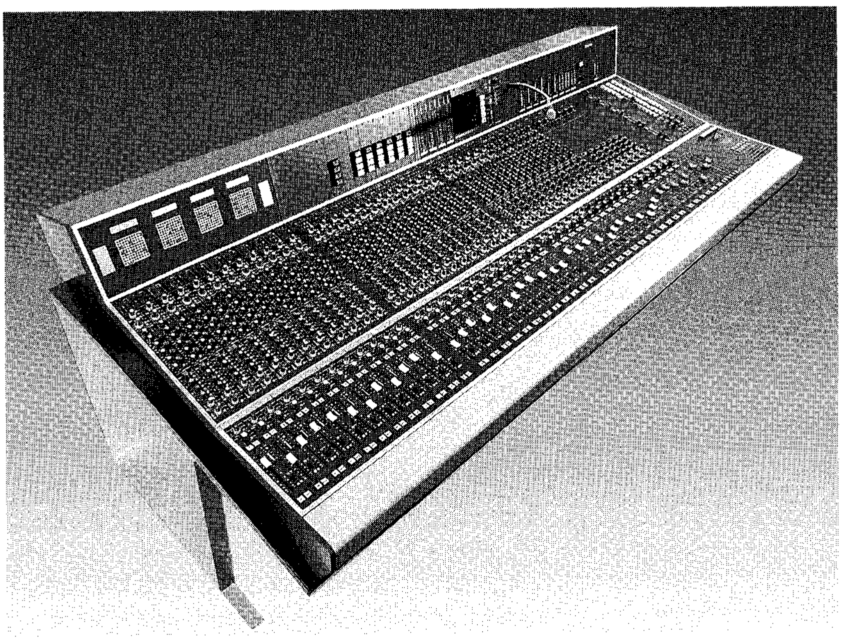
The audio technology is increasingly becoming digital, nothing can change that. But also undeniable is that all natural sounds originate in analog form and that ultimately they must be reproduced to analog form so that they can be perceived by our ears. To bypass the ear with a direct fiber-optic input is impossible even with genetic manipulation and the most advanced surgical technology. Even if this were possible, can you envision the complications involving the standardization of this connection? Shall it be male or female (or based on a quota system?), operate with red, green, yellow or blue light (politicians!), which modulation method shall be preferred (technicians!), may and should the data be transmitted in compressed format (matter of principle!). The socio-political complications would be difficult to fathom and the different parties would probably never achieve a consensus.

Fundamental questions

Some of the reasons that speak for analog mixing consoles are easier to justify but nevertheless are weighty. These reasons are also accepted by our customers and can greatly influence the decision between analog and digital.

Price: The price of a digital mixing console is still considerably higher than for an analog mixing console with similar features.
Adaptation to customer requirements: With a hardware-oriented, analog mixing console, special requirements are easier to satisfy than with a software-controlled digital mixing console. This contradicts the prevailing thinking that «only» software adaptation is required in order to satisfy special requirements. But in practice the implementation of hardware adaptations is less costly and less fraught with risks.
Conversion: Each conversion from analog to digital and back results in losses which should only be accepted if the conversion to a digital environment offers other advantages, for example, low-noise recording or digital post-processing.
Direct broadcasts and stereo recordings: If a sound event is mixed down directly to stereo and input to the transmitters or recording devices, any necessary conversion can be performed at the mixing console output. Only one converter is required at the mixing console output rather than separate A/D converters for each microphone.
Technical data: The transmission quality of today's analog mixing console is practically determined only by the quality of the microphone input stage. This is where optimization pays off.
Transit times: In coupled recording of video and sound and wherever sound events are played back to the musicians, signal delays can cause problems. During the conversion as well as signal processing delays in digital mixing consoles are unavoidable. In analog signal processing this effect is always negligible.
Direct access to the user interface: Most digital mixing consoles have a central operating unit. Channel-related operation such as we know it from most analog mixing consoles is restricted because only a reduced number of controls per channel is available. In direct broadcasts and recording of non-repeatable events, direct intervention through channel-related operation is highly appreciated.
Maintenance: Analog mixing consoles are still more stable and less problematic when impro-

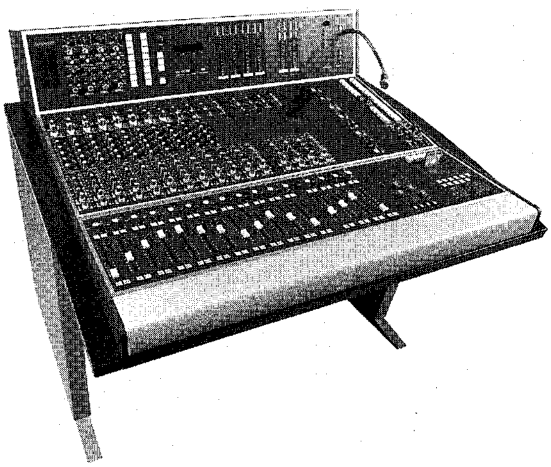
Hermann Stierli



vised installations are necessary, such as in OB applications. Analog mixing consoles are easier to repair and system crashes are even rarer than on digital mixing consoles. In the future this is likely to change when specialized personnel, better safety concepts and more stable platforms are available.

For this reason our answer to the question posed at the outset is unequivocally: **Yes, an analog mixing console still has its place.**

The new mixing console must satisfy the more demanding technical requirements which arise from the capabilities of high-quality digital audio signal transmission and storage. At the same time it must support new operational procedures such as four, five and six channel audio dubbing of video and film productions. Simple storage of certain console setups and fader automation must also be possible.



The answers with a new mixing console
Our answer to this challenge is the **STUDER 980**, a new mixing console that is based on the proven series 900. The principal characteristics are:

- The scope of functions and technical features, particularly of the input units and the fader modules have been thoroughly re-engineered. By staying with the 40 mm format it is possible to use all series 900 special units (of which there are about 200) which from the start gives the new concept an enormous degree of flexibility and adaptability to application requirements.

- The control concept is designed in such a way that all switching functions can be stored and re-established. The solution is on-top control which is superposed on the inherent functionality. This means that mixing consoles, for example in smaller configurations are fully functional also without the central control unit, or that they can still be operated in the event that the central control fails.

- Snapshots and sequences can be stored on memory cards.

- A new line of input units, linear faders, and aux. master units offers significantly improved processing and control facilities.

- Specially designed monitor, input and fader modules support the extended function requirements for film, video and HDTV applications with four, five and six master outputs and monitor channels.

- New STUDER design.

- Excellent technical data, optimum electromagnetic compatibility, and high reliability taken for granted.

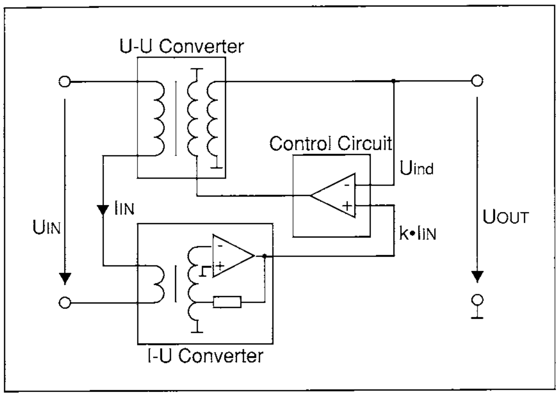
Flexibility
A modular system as used for the series 900 that offers great flexibility for tailored configurations and specific customer requirements, was one of the main challenges to the designers of the 980 mixing console. No less than eight different input units and 12 different linear faders were created. If we count only the most sensible combinations, the customer already has a choice of 24 different input channel units. Additional channel parameters can be modified by setting internal jumpers, or through global switchover on the central control panel. For additional combinations with the input units there is a large choice of group and master units, 22 different output meters, approx. 30 additional modules for incorporation into the instrument panel, and approx. 35 different EU-standard PC boards and 30 matching submodules are available. This suffices for even the most exotic customer requirements.

I shall now discuss some of the details that went into the design of the new units.

Input circuit
Each input unit features two or three different inputs.
The microphone input circuit takes into consideration all the insights discussed in P. Zwicky's report (MIC input, page 6), but in addition the output level is reduced by a patented negative feedback via a second transformer (also for medium levels) to such a degree that distortions remain in the range of –80 dB.

Due to the selected circuit design microphone inputs can also be used as high level inputs.

Output levels from condenser microphones positioned close to wind instruments, as well as electronic music instruments that are connected directly to microphone inputs, no longer present any problems.



At the high level inputs electronically balanced and transformer balanced input stages are available. Due to the advanced circuit design practically no differences can be measured or heard. But in OB applications and in polluted environments transformer inputs with common-mode rejection values of over 120 dB at 50Hz are still preferable.

For the development engineers not only high common-mode rejection but also noise immunity and electromagnetic compatibility was important. Each input and output is equipped with appropriate filters, tested, measured, and optimized.

Pre-fader listening, P. Solo
Should the pre-fader listening function be disabled when the fader is opened, or should the pre-fader listening signal still be audible? Should the pre-fader listening keys be interlocked or should the signals be added? Should the solo function not influence the main signal path or should the solo be destructive? All these functions can be set with jumpers. Subsequent changes and adaptation to new operating environments are consequently easy to implement.

Auxiliary
Each input and group channel has eight auxiliary outputs (Aux) of which four are treated as mono signals and the other four can be combined into two stereo pairs. The changeover is performed on the aux master unit. The function of the two volume potentiometers per input channel changes to a volume control and a panorama potentiometer. Stereo and mono outputs are thus no longer a question of the basic configuration but can be adapted to the production requirements.
If not enough channels are available for a mixdown, this does not pose a problem because in addition to the prefader and postfader signal aux channels 7+8 can handle also a channel processing independent signal that is fed via the second line input.

Direct output and N-1 output
All input channels are equipped with a direct output . Its signals can be tapped either before or after the fader, and in the broadcast version even the level can be controlled.
To feedback the current broadcast to one or several outside commentators - without their own input - only the N-1 key needs to be pressed on all input units to which external sources are assigned. In the central key pad the user can select whether only the channels with pressed N-1 key or all existing inputs are to be combined into the N master.
The TALK key switches the command microphone to the corresponding output. This simplifies rehearsals and allows fast intervention if a correspondent should miss his entry. If all outputs are to be accessed at the same time, the «TALK TO ALL» key can be pressed on the central control panel.

Level meters
Each linear fader features a small level meter that can be switched internally to VU or PPM characteristics. The instrument always indicates the output level of the direct or N-1 output, but with a key on the central control panel it is possible to switch to the input signal of the channel (before the linear fader). Due to the assignment to the channel feeders, monitoring of the input level does not present a problem even for complex and improvised recordings.

Panorama and balance control
Whereas for mono inputs the panorama potentiometer is used only for setting the mono source on the axis between the stereo speakers, the same potentiometer has two different functions in conjunction with stereo modules. It can be used for correcting volume errors between the left and right channel, and it is also possible to shift the entire stereo sound image to one side or the other. This involves two radically different processes. In the case of the correction only one channel is to be attenuated or boosted relative to the other (balance), whereas in the shift (panorama) function the information content of the two channels must be retained also in the extreme positions (left or right limit position), however, without side information, that is, mono.
This changeover between panorama and balance function is possible on all stereo channels.

Signaling

Signaling and faderstart are used rather differently by the various user groups. Often a normally open contact suffices which is activated when the linear fader is opened. But some users require that the light signal in the studio is activated only when the audio signal from the master is actually through-connected via the group and the input channel unit. In this case three fader contacts, the master and group selection, mute switch and input selection are enabled. With internal jumpers a wide range of custom configurations can be set. ■

980 mixing console

980 Mixing console with multichannel sound for film, video and HDTV

by Hermann Stierli

Hermann Stierli

When Peter Frigo, managing development engineer for analog STUDER mixing consoles turns on his home TV he preferably chooses a film with the Dolby Surround® logo in the leader. The rest is simple: Switch on the Dolby Decoder and the sound instantaneously changes from the usual stereo to space surrounding four-channel sound.

Even though the stereo base has become wider, the dialog remains anchored at the picture through the center speaker. But the space has been opened also toward the back because sounds and music are fed via the surround speakers arranged behind the viewer. The illusion of being in the middle, of being part of the action - a concept already realized for cinemas - is not yet fully implemented for home viewers but at least they don't have to wait for the intermission to get a snack! Not just Peter, our sound guru, but more and more viewers with a fine ear no longer want to miss the cinema multichannel sound when watching TV.

If HDTV should make the expected breakthrough in a few years, a fifth sound track will also be transmitted. Sounds, effects and music will be locatable even on the back because a stereo pair will be used instead of the surround speaker.

Even if HDTV has faded somewhat into the future and the super picture will not be introduced until the next century, many television producers seriously play with the thought of using multichannel sound for their latest productions. The format to be used is still debated, but because the new digital video recorders feature four audio tracks, the four-channel format appears to be the logical solution. Until the quantum jump to HDTV is achieved, compatibility with all existing television sets is important because not every television viewer is willing to install a «home theater». As a result only the Dolby Surround Format remains a viable option which according to my personal and not decisive opinion will prevail.

Concept for multichannel productions

One goal pursued in the design of the STUDER 980 mixing console was to allow HDTV compatible multichannel productions. The summary of the principal film and television formats (see box) shows that except for the sparingly used SDDS format, systems with maximum six channels have established themselves. For this reason also our new mixing console had to be equipped with six channels but support also mono, stereo, four-channel and five-channel productions. The adaptation to these requirements required significant deviations from a normal broadcast mixing console in three places:

- *Bus selection*
- *Output distribution* of the monophone audio signal to the buses, and
- *Monitoring unit* which must allow mono, stereo, four, five and six channel monitoring.

Bus selection

Instead of the eight group and four master buses used in the broadcast version the film/TV mixing consoles have six groups and six masters.

In film dubbing the mixing console is often subdivided into three separate workstations for dialog, music and effects. Each mixing console section works on separate recording tracks. Only in the last pass are the three track groups mixed down to the desired format. Mixing consoles for this application can be built by making the groups for each sector available at external terminals whereas the masters can be selected by all sectors.

Output distribution
More complex was the task of allocating the mono signal of an input unit to any of the six masters, and groups.

For mono outputs direct selection via the bus selector keys suffices.
In stereo distribution a PAN key activates a panorama potentiometer through, which the mono information can be set to any point between the two speakers.

Formats with four to six monitoring channels require an additional potentiometer that controls the distribution between the front speakers and the back speakers. In this process it does not suffice to arrange the mono sources in a line in front of the listener but it must be possible to position the sound event anywhere on the surface of the room. This Front-Back potentiometer is activated via the FILM key.

In the Dolby Surround method, sound coverage for the space in back of the viewer is reduced to mono information that is opposed to the three front information signals left-center-right. This «Surround» information can be inserted via a single speaker, or as in the case of cinemas, via an array of speakers.

In the TV 3-2 standard also the back signal is transmitted separately for the left and right speaker. This means that the panorama potentiometer must remain effective whereas for the four channel technique only the front back distribution is relevant.

All multichannel techniques for film and video require more than the additional front back dimension. Also the front imaging has changed from the standard stereo method. In addition to the left and right speaker there is a center speaker. If the viewer were positioned ideally in an ideal listening room, this center speaker would be unnecessary because the optimally positioned listener does not detect any difference between three channel and stereophonic playback in which the center information has been added to the stereo pair. Unfortunately normal living room architectures are practically never optimal for this type of sound reproduction and also the listener or viewer is rarely in the ideal listening position.

This has the effect that the voice of the person seen in the center of the picture is heard from a side speaker that is located far away from the picture. By contrast, the additional speaker to which usually the entire dialog is allocated establishes a fixed point and ensures that the sound belonging to the picture remains stable in the picture. The side speakers and back channels reproduce additional information such as music, effects, and room atmosphere,

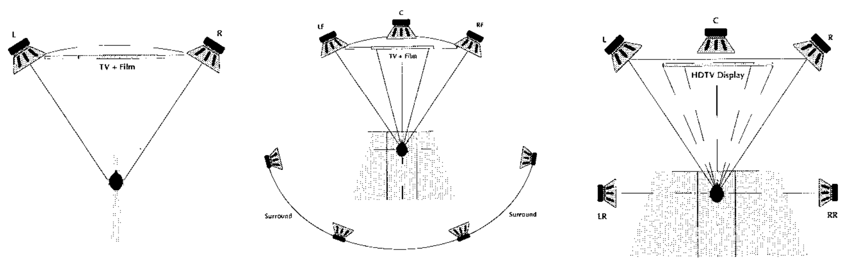


Fig. 2
Multichannel playback of TV and film productions

which are not relevant to the picture itself.
In the switchover to film the characteristic of the normal panorama potentiometer (Fig. 2) is changed in such a way that the three output channels (left, center, right) can be controlled with different levels, depending on the angle of rotation (Fig. 3).

Also the functions of the DIV key and the divergence potentiometer are superfluous under ideal listening conditions because these controls can only reduce the strict separation of the front speakers. In the center position of the panorama potentiometer a signal component is added not only to the center speaker but also to the two side speakers (Figs. 4). For a listener

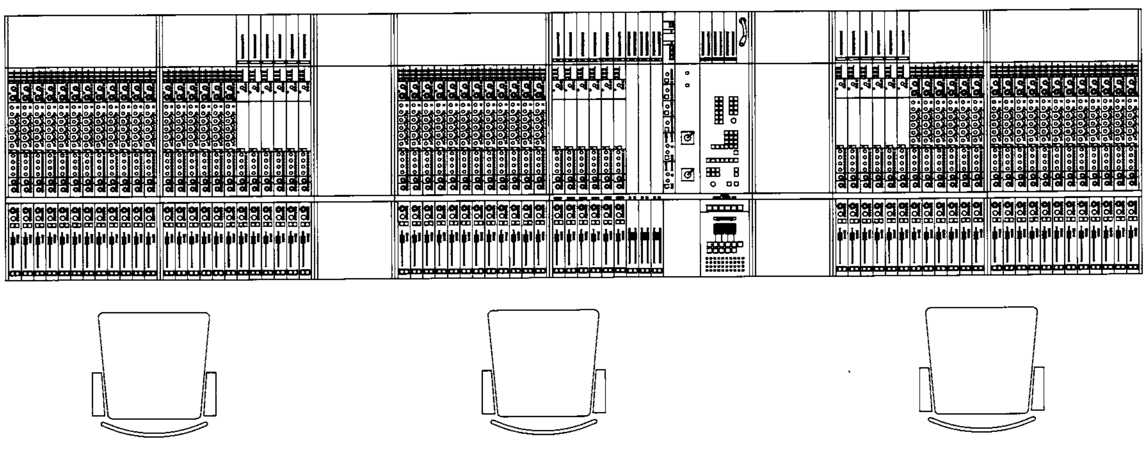


Fig.1 Film dubbing with isolated console sections for dialog, music and effect

who is not ideally positioned the imaging of the point-type mono source is broadened.

Also feasible would be the real broadening of the sound image by using phase shift filters in the adjacent channels but here we still are in an experimental phase.

HDTV / Film monitoring

Of course it is useless to equip the input units with all adjustment facilities for multichannel tone if the result can only be monitored stereophonically. Of prime importance was, therefore, the integration of a monitoring system through which mono sources, stereo productions but also Dolby Surround, TV 3-2 and true 6-channel recordings can be monitored. Particular emphasis was placed on the possibility of replaying the signals before and after coding, to make the decoded form audible, but also the ability to check how a recording in one format sounds when it is played back in a different format.

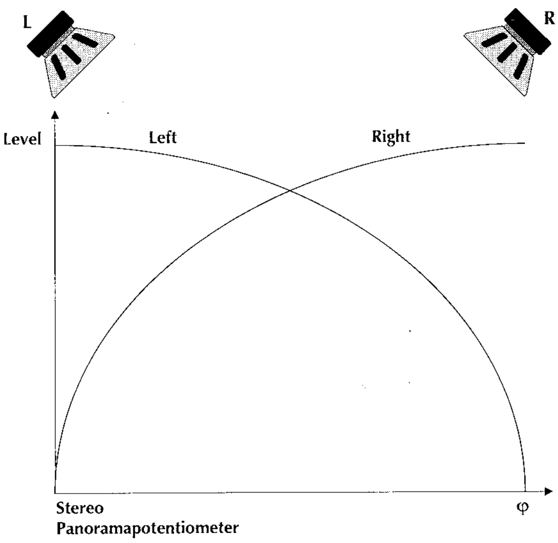
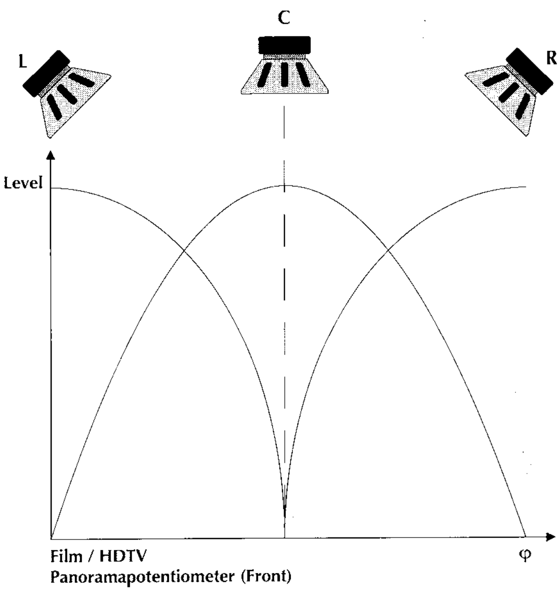


Fig. 3
Characteristic of the stereo-panorama potentiometer



Characteristic of the film-panorama potentiometer

For a convenient representation of all possibilities, the monitoring keys (MONITOR SELECTION) were arranged by channel number and production format and the MONITOR MODE row with six keys was introduced through, which the mode can be selected.

For representing the signal paths a display field is arranged between these two groups of keys. The display shows how and when the decoders and encoders are inserted.

The MONITOR MODE keys control also the speaker control keys located immediately below them. The pilot lamps identify the active monitor speaker. Manual intervention is possible at any time and speakers can be enabled and disabled (Fig. 5).

We shall now give a brief description of the different monitoring modes:

Stereo production

The desired stereo source is selected via the source selector. The monitor mode is automatically switched to STEREO and the L + R speakers are activated.

The monitor mode can now be switched to MONO CENTER or MONO L+R for checking the mono compatibility. In the first case the monitor mix of the selected stereo channel is reproduced through the center front speaker, whereas in MONO L+R the audio signal is reproduced in parallel via the left and right front speaker.

If the selected stereo source is a coded Dolby 4-2-4 signal, the DOLBY 4-2-4 monitor mode can be selected to achieve decoded 4-channel reproduction.

Multichannel sources

After the selection of a multichannel source and selection of the DIRECT monitor mode all four or six source channels are switched to the speakers in the sequence defined above. Channels that are not wanted can be suppressed by pressing the LOUDSPEAKER ON keys.

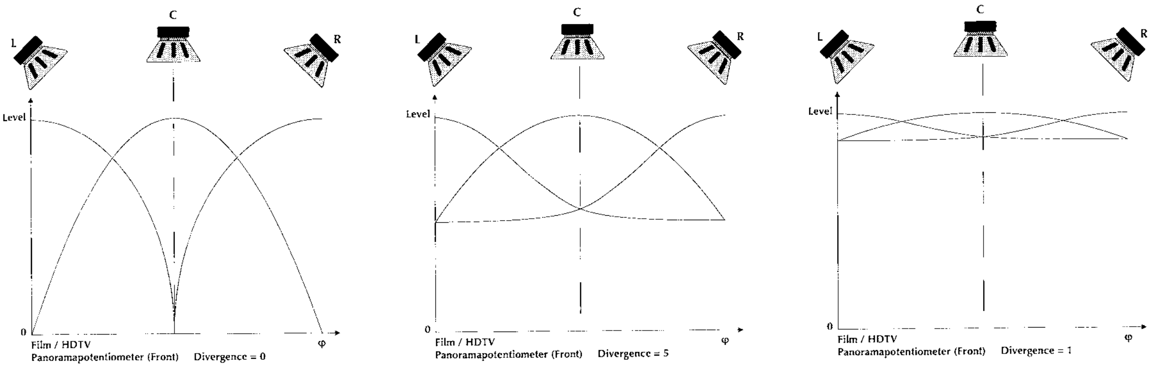


Fig. 4
Divergence = 0 / 0.5 / 1

Changeover of the monitor mode to TV 3-2 connects the sources 1, 2, 3, 5 and 6 without change to the monitor speakers and adds channel 4 to speakers 5 and 6.

Monitor mode DOLBY 4-2-4 connects the sources 1, 2, 3 and 4 without change and adds channels 5 and 6 to channel 4.

In *STEREO mode* channels 1 and 2 are connected without change to speakers 1 and 2, channels 3 and 4 are connected in parallel to outputs 1 and 2, and channel 5 to channel 1, and channel 6 to channel 2.

In *MONO mode* the above stereo signal is mixed down to a mono signal and reproduced via speakers 1+2 or 3.

DOLBY 4-2-4
Selection of the DOLBY 4-2-4 source and DIRECT monitor mode reproduces the uncoded signal available on the input of the Dolby encoder. Speakers 1, 2, 3 and 4 are active.

In *STEREO monitor mode* the coded Dolby signals can be heard via speakers 1 and 2. In this way the compatibility of the signal can be checked for all listeners whose receivers are not equipped with a Dolby decoder.

In *MONO CENTER and MONO L+R mode* the coded signal is mixed down into a mono signal and output to speaker 3 or 1+2 respectively.

When *DOLBY monitor mode* is selected the signals travel through the full encoder / decoder path. The final result of a Dolby surround production can be monitored with speakers 1, 2, 3 and 4.

In *TV 3-2 mode* the decoded Dolby signal is output without change to the front speakers 1, 2 and 3 whereas the surround channel 4 is output in parallel to the TV side channels 5 and 6.

TV 3-2
Since the utilization and type of the TV 3-2 encoder and decoder are not yet defined by binding standards, no definitive assignment can be made yet. However, the hardware contains all inputs and outputs for connecting this equipment. The necessary switching and configuration commands can be retrofitted through software. ■

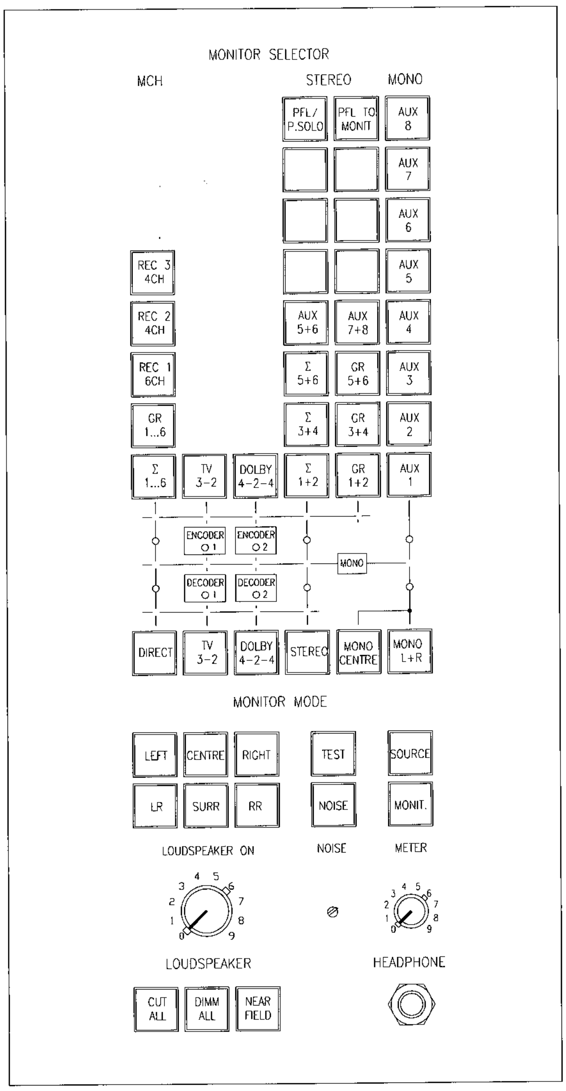


Fig. 5
Monitor module

SURROUND is a Trademark of DOLBY Corp. USA

Multichannel tone for film, video and HDTV

Summary of the principal audio formats for film, video and HDTV

	Coding	No. of channels coded / decoded	Reproduction (channel assignment)
Optical sound / TV mono	None	1	Front center
Optical sound stereo	None	2	Front left / front right
TV stereo or two-channel		2	Front left / front right mono reproduction for 2-channel sources
Dolby Surround	Yes	2/4	Front left (1) / front center (3) / front right (2), back surround (4)
HDTV 3-2	?	5	Front left (1) / front center (3) / front right (2) Surround left (5) / surround right (6)
Dolby SR*D	None	6	Front left (1) / front center (3) / front right (2) Surround left (5) / surround right (6) subwoofer channel (4)
DTS-6	on CD ROM	6	Front left (1) / front center (3) / front right (2) Surround left (5) / surround right (6) subwoofer channel (4)
Sony SDDS	None	8	Front left / left center / front center / right center / front right Surround left / surround right subwoofer channel

Right on Success

All African Games, Zimbabwe

by Rolf Breitschmid

Harare, the capital of Zimbabwe geographically located south of the equator in Africa, will be responsible for the this year sport competition named ALL AFRICAN GAMES.

This event requires a number of audio and video equipment. Studer discussed the technical demands with Zimbabwe Broadcasting Corporation (ZBC) in 1994. Our offer was approved by the Swiss Government Organisation BAWI, which will support the delivery under the Swiss Aid Agreement.

The project includes for the audio part: 30 pcs 069 Reporter Mixer, 40 pcs A807-0.75 VUK Tape Recorders, 12 pcs 961 Mixers, 2 pcs 970 Mixers, 2 pcs 963 Mixers and two OB Vans constructed and delivered from Studer England.

The total contract amount is CHF 2,6 Mio. Delivery must be completed by July 15th 1995. One of our engineer will carry out the installation, the training and the final handing over. ■

Magneto-optical recording

STUDER MO vision

by Kurt Schwendener

In the computer industry the progress in storage technology is so rapid that already a few months after we have bought a computer the prices have dropped again and the CPU performance has taken another quantum leap. However for our recording equipment the situation is not yet so dramatic.

Users of computer aided systems such as the *Dyaxis* workstation from *Studer Editech* have been familiar with hard disk and optical disc recording for a long time and consequently have also experienced the breathtaking developments in storage media. In the field of Studer recording and reproduction equipment the big technology leap is coming soon. Although the step from analog to digital recording has been made some time ago, tape (RDAT and DASH multitrack tape) is still widely used in two-channel and multi-channel applications. CD players and recorders are devices where a new optical medium has established itself, although the CD can be written only once (WORM, write once read many).

If we want to profit from the enormous development investments made by the computer industry we must look beyond our own sphere in order to find a medium that can replace the magnetic tape and supplement the compact disk. RAM chips that have the required capacity are not yet available at an acceptable price whereas a hard disk must be transported with the complete drive. Magneto-optical storage is increasingly becoming a viable compromise between wear-free storage in library quality, available capacity, rugged medium, and acceptable price.

For this reason Studer is supplementing the Dyaxis DAW products and the CD line with a new family of recording devices that are based on the MO disk as a versatile, new storage medium.

Magneto-optical recording

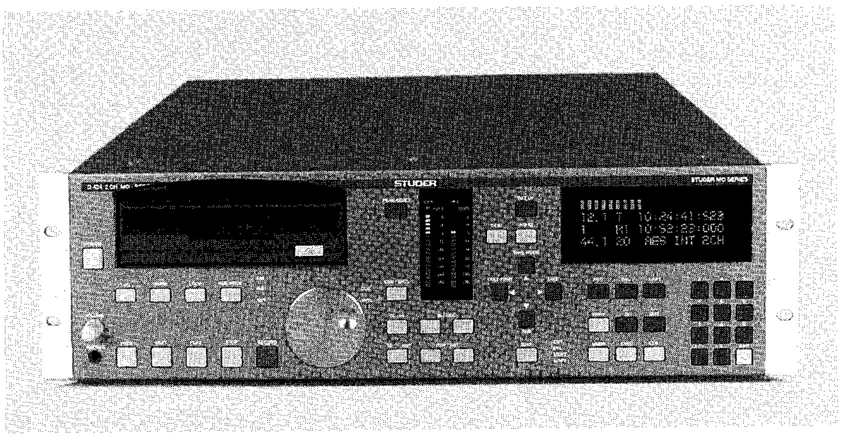
The playback principle is the same as for the CD: A laser reads the data based on the different reflection of the beam from the surface of an removable disk. For recording the corresponding position on the disk is locally heated with the laser beam and the recording layer is polarized with a weak magnetic field (Fig. 1). This poarization remains after normal temperature has been reattained, which causes the laser beam to be optically polarized during the read operation. Suitable filters convert this polarization into intensity fluctuations.

Kurt Schwendener

MO disks are now available in 3.5″ and 5.25″ sizes. Some of them can be recorded on both sides. Capacities ranging from 128 MB (3.5″) to over 2 GB (5.25″, double sided) are offered. Like hard disks, MO disks are formatted into tracks and sectors. Various ISO standards exist that ensure the interchangeability of disks on equipment of different makes.

Important specifications for audio recording are the access time and the data throughput. Although hard disks are still faster due to the magnetic recording process, today's MO drives already fulfill the requirements which have to be met for data recording and editing (Example: the 2.2 GB MO drive from Pioneer has an access time of 20 msec and a transfer rate of 2.5 MB/sec).
Audio information can be processed in non-destructive editing mode, that is, edits and crossfades can be performed in real-time in accordance with a «playback list».

As in the case of hard disks the following recording capacities can be achieved (stereo, at 16 bit, f_s = 48 kHz) per side: 28 min (650 MB), 54 min (1.3 GB) and 90 min (2.2 GB)



MO in the studio

Also in the studio there is a general trend toward computer oriented applications with fully digital environments and automated processes. Analog equipment is difficult to integrate into such systems, users are no longer prepared to invest in technologies that are not future-oriented.

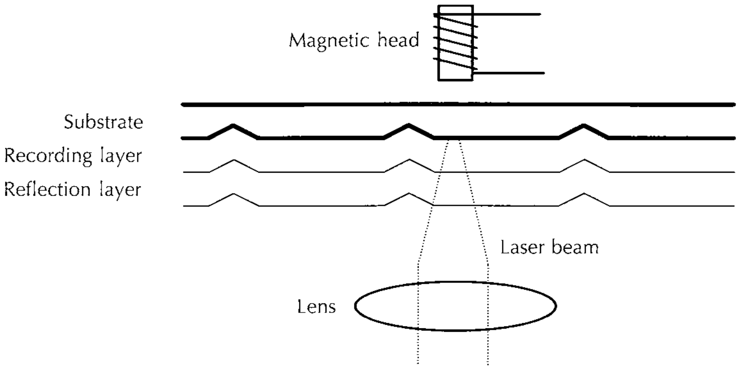
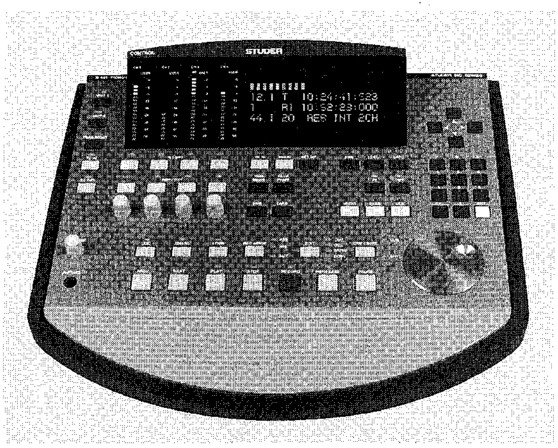


Fig. 1

The equipment that is put onto the market today must satisfy not only the present requirements but also be open to foreseeable future developments. A recording device that is based on an SCSI drive is an ideal platform. The MO technology is now able to fulfill these specifications. At the same time alternatives (CDR, hard disk) or new media can be integrated without modification to the remaining components such as the user interface and control interfaces.



Today's situation in the studio is characterized by a multitude of old and new media: Analog and digital tapes, compact cassettes, RDAT, perfo tapes, CDs, hard disks, and networked systems. Certain applications have specific media but no single media is able to cover the entire range. The result are significantly higher production costs, caused by the necessary copying operations and maintenance for a broad range of equipment.

Studer's MO strategy is to offer a solution that combines cost effectiveness, reliability and open architecture. It is geared to the few remaining work media: SCSI drives such as MO and CD, as well as locally networked systems. The strength of MOs ranges from recording to editing and utilization as players. The CD/CDR is still viable as an economical library and effect medium.

Due to the open architecture of the new Studer products, smooth integration into existing system environments will be possible. But also the utilization of media from specific application niches (for example, PCMCIA cards with data compression for portable recorders used in news applications) will be feasible.
Enhancement and compatibility with other Studer product lines such as Dyaxis and Numisys is taken for granted.

MO core

The recording equipment market is not the only one that can profit from this technological advancement. In all product areas the future is in digital signal processing, automation, and system integration. The division between individual segments such as recording, mixing consoles, and post production will disappear even further. A common platform has a positive effect on the price/performance ratio of all products.

For this reason a joint project in cooperation with STUDER Regensdorf, Studer Editech (USA), and Studer Digitec (France) was started. The result is a digital audio recording core, the basic electronics for recording digital audio data on SCSI drives (Fig. 2). This core is powerful enough for simultaneous recording and playback of up to 10 audio tracks. As a result it can be used for a variety of applications.

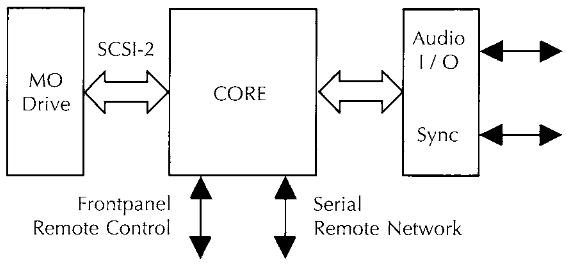


Fig. 2

Product strategy

This core which provides a significantly higher functionality than is required for a simple two-channel recorder can serve as the basis for many products. Because of its extremely versatile design it is suitable for a variety of products and other economical SCSI drives can be adapted very easy.

Our first product will be a two-channel MO recorder that is pre-engineered for 4 tracks. It will continue the Studer tradition in two-channel to 4-channel recorders (see below).

Expansions with various channel and operating versions (for example, editing controller) will follow so that a bridge to the Dyaxis DAW area can be established.
Modular enhancements, for example, with high-speed SCSI copying to other CDR equipment, are planned as well.

With respect to the recording medium, future enhancements and developments the optical technology can be adopted without any change to the equipment concepts.
Other SCSI media will be supported as required. Multitrack applications become accessible by expanding the capacity through RAID solutions (Redundant Array of Independent Disks).

D424
The first product of the new generation will be the two-channel MO recorder D424. It gives the user a reliable new tool that is economical in demanding everyday duty and can be integrated in existing and future applications. It can replace the existing workhorses like analog and DAT recorders and at the same time assume new functions. The user interface has been largely adapted to existing work practices.

The standard version of the D424 comes with a high-capacity 5.25" MO drive (2.2 GB). The recorder has the following characteristics:

- Two-channel MO recorder with proven user-interface
- Storage capacity over 70 minutes with linear 20-bit recording on one side
- Full media compatibility with Dyaxis II DAWs and Numisys applications, support of OMF standards.
- Front operation with basic recording functions, direct title selection, composition of programs and edititng facilities (edit, copy, delete).
- Can be integrated into a network for control and audio data transmission.
- Playback with crossfade and level calibration.
- Digital monitoring in shuttle and cue mode
- Recording and playback selection individually for both tracks
- Selectable recording format: 16, 20 or 24 bits, with all commonly used sampling frequencies
- Digital audio interfaces: AES-3 and SPDIF
- SCSI-2 port for additional storage media
- Synchronization to different external clock signals

- SMPTE/EBU time code interface in all standards
- Control via 9-pin RS422 or Ethernet

The following options will be available:

- Ergonomically designed desktop remote control
- Analog audio interface
- Data compression
- Network interface (Ethernet)

Further information on the new STUDER MO product range will follow in subsequent reports. ■

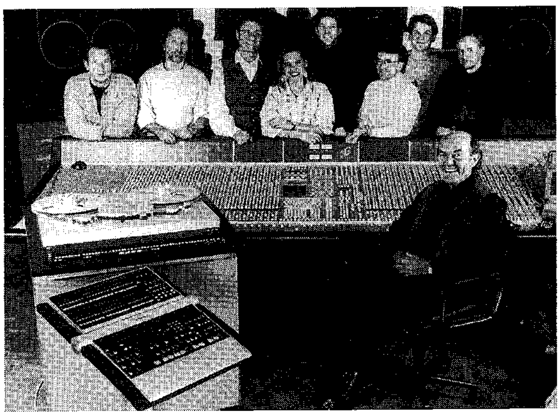
Right on Success

«Sound Studio N» produces with Studer DASH D827-48 MCH

by Thomas Frohn

Sound Studio N in Cologne, member of The World Studio Group, has celebrated its 25th anniversary by giving itself a present. The well-known studio has upgraded its recording equipment to the highest quality level through the purchase of a 48-channel STUDER D827-MCH.

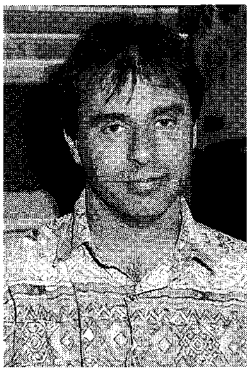
The first production on the new machine was initiated with the latest album of the Kelly Family «Over The Hump». Success was not long in coming, the album received «platinum» for the third time. ■



Broadcast mixing console 916

The ideal equipment for local studios

by Ivo Bischof



Ivo Bischof

With the model 916 Studer offers a very rugged and flexible mixing console that is especially designed for demanding duty in regional / local radio stations. It is the direct professional successor to the proven Revox MB 16.

The STUDER 916 mixing console features a functional and well-engineered design. Large illuminated keys for channel selection and prefader listening enhance the ease of operation. Even a document storage space is provided so that this console is also highly suitable for DJ mode. Of course each channel is equipped with its own faderstart output. All 6 microphone/line inputs feature a balanced mono insert point (6.3 mm module) so that limiters, compressors, effect machines, etc. can be used. All balanced inputs and outputs are terminated on XLR connectors.

The telephone and microphone inputs can be programmed with DIP switches either for DJ mode (muting of control room monitor) or normal studio mode (muting of studio monitor).

Telephone inputs...
Two inputs are reserved for telephone lines. With an external STUDER telephone hybrid telephone lines can be coupled via a telephone filter (2 kHz ±15 dB) in balanced mode. All telephone functions of the hybrid can be controlled with the command keys on the instrument panel. For example conference circuits with two external callers can be set up. The DJ can connect himself into the conference via the talk-back function.

Input channels...
On the 916 mixing console the following inputs are available

- *6 balanced microphone/line inputs.*

The line input can be programmed as an unbalanced stereo input with cinch connectors. Three inputs are equipped with treble and bass equalization filters. Each input is also equipped with an auxiliary output (AUX) which can be used either before or after the linear fader. All input channels are equipped with voltage controlled amplifiers (VCA). And with the remote control option video editors can also be connected.

- *8 balanced stereo inputs,* 4 of which are equipped with equalizers and the same functions as the mono inputs.

Outputs...
The following master outputs are available on the 916:

- *Balanced to transmitter*
- *Balanced for recording* (master to logging)
- *Unbalanced*

The recording output (master to logging) can also be remote controlled with a VCA. The master fader of the auxiliary output (AUX) is also located in the master module. In addition, an auxiliary input (AUX Return) is available which can be mixed to the master via a potentiometer.

Instrument panel...
For monitoring the stereo master signal and the control room monitor signal two integrated PPM bargraphs are provided. A stop watch with fader start is standard equipment; it can be programmed to the corresponding input unit through an external DIP switch. With the built-in speaker either the master or the control room monitor signal can be monitored.

Options...
Installable in the instrument panel:

- *High-resolution* STUDER *gas plasma bargraphs with PPM or VU characteristic,* as an alternative to the standard bargraphs with PPM or VU characteristics
- *Moving coil instruments*
- *Various correlation indicators*
- *Tone generator* with pink and white noise
- *Red/green signaling* for controlling the access lamps
- *Additional input selectors* for connecting several external sources to the desired stereo inputs
- *Remote control units* for CD players and tape machines.

Monitoring...
With mutually interlocking keys the corresponding signals such as master, auxiliary, on-air, etc. can be selected for the control room monitor. The output is terminated on balanced XLR connectors. With a selector key the prefader listening function (PFL) of the input channels can be switched to the control room monitor. The built-in talk back microphone can be connected either to the studio outputs or auxiliary outputs. To prevent feedback the output level of the control room monitor is decreased by 20 dB.

The studio monitor signal is also terminated on balanced XLR connectors. Either the master or the control room monitor output can be connected to the studio monitor output. For DJ mode a so-called «Guest phone» jack is provided so that guests can listen into the program. ■

Worldwide sales successes

Outstanding references

by Robert Müller

One of the surest signs for the success of the D827-MCH is certainly our sales success over the last months. In the USA alone Masterfonics in Nashville Tennessee, Right Track in New York, Clinton Recorders in New York, Elysian Recorders in Florida and Toy Specialists in New York have decided in favor of the D827-MCH. Also a superstar like Whitney Houston has obtained a D827-MCH. ■

Worldwide sales success

Numisys project in North Rhine-Westphalia, Germany

by Eberhard Kaulbach

An outline agreement valued at over DM 6.5 million for two broadcasting centers of Radio Westfunk in Essen and Radio Bielefeld, as well as 22 additional broadcasting studios networked via ISDN in 22 different cities, has been signed.

This project is unique for Germany, if not for all of Europe. For the future of Numisys and Studer Germany this order will have a major market impact. Today's version of the Numisys system has been decisively formed by the excellent cooperation between the ordering parties, Studer Germany, and Studer Digitec. For the first time also the D.A.V.I.D system is integrated in such an order.

The preparations for this project required over 18 months. A total of 26 meetings in Essen, Düsseldorf, Oberhausen, Paris-Chatou, Regensdorf-Zurich and Cologne took place. In addition to at least 5 product demonstrations, each of which was conducted over 4 to 5 days in the corresponding location, the system was also demonstrated at the NAB in Montreux and the Tonmeistertagung in Karlsruhe as well as Radio-Bit in Oberhausen. In this connection we would like to thank Klaus Ramoser, product specialist for Numisys and branch manager SD in Munich for his competent demonstrations, without which we probably would never have been able to obtain the order.

The outcome of long negotiations: A contract for 24 Numisys systems with WESTFUNK.

From this large project we have reaped an unexpected side benefit: an additional Numisys order valued at approx. DM 300,000 was received from Radio 7. ■

Digital system technology

NISKO - fully digital information and broadcasting complex

Swiss Radio DRS, Radio Studio Zurich

by Hermann Stierli

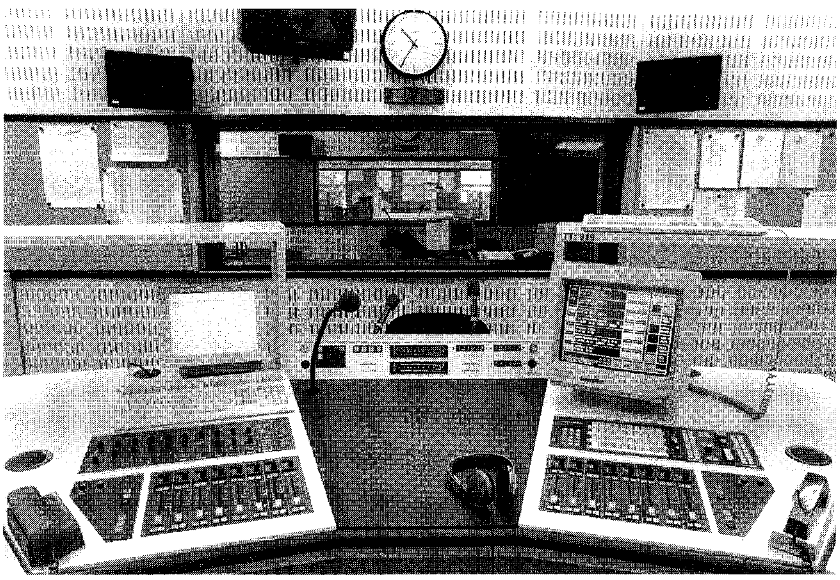
Hermann Stierli

After an intensive 2-year preparatory phase all programs of the radio studio in Zurich are broadcast since March 10, 1995, via the new, fully digital transmission complex of Swiss Radio and Television (SRG).

During the past year various European broadcasting companies have converted their production and broadcasting studios to digital technology. Unique for Europe is, however, that a completely new information and broadcasting complex (NISKO) was built at the same time. Already during the planning phase Swiss Radio DRS strongly emphasized a system solution that comprises the control rooms, studio rooms, as well as the master control room and the networking of the entire system. In the spring of 1993 tender invitations were issued according which four identical continuity and production control rooms with two announcer studios for each pair of control rooms, one MCR, and central engineering room, and corresponding installations had to be quoted.

We offered our MADI router, together with four digital mixing consoles, which are based on the existing D940 technology but are equipped with a new user interface.

After a comprehensive review of all offers by the technical and commercial committees of the customer, STUDER Regensdorf together with its subsidiary, STUDER Digitec, was awarded the order for the complete system in June '93.



Continuity and production control rooms

Swiss Radio DRS decided on so-called DJ mode. With few exceptions the entire radio program is controlled by the moderators themselves, that is, without assistance by technical personnel. Only in complex broadcasts involving many direct insertions from OB locations, for example during elections, sports events, and extraordinary events are the continuity consoles operated in the traditional manner, that is, with the assistance of audio engineers.

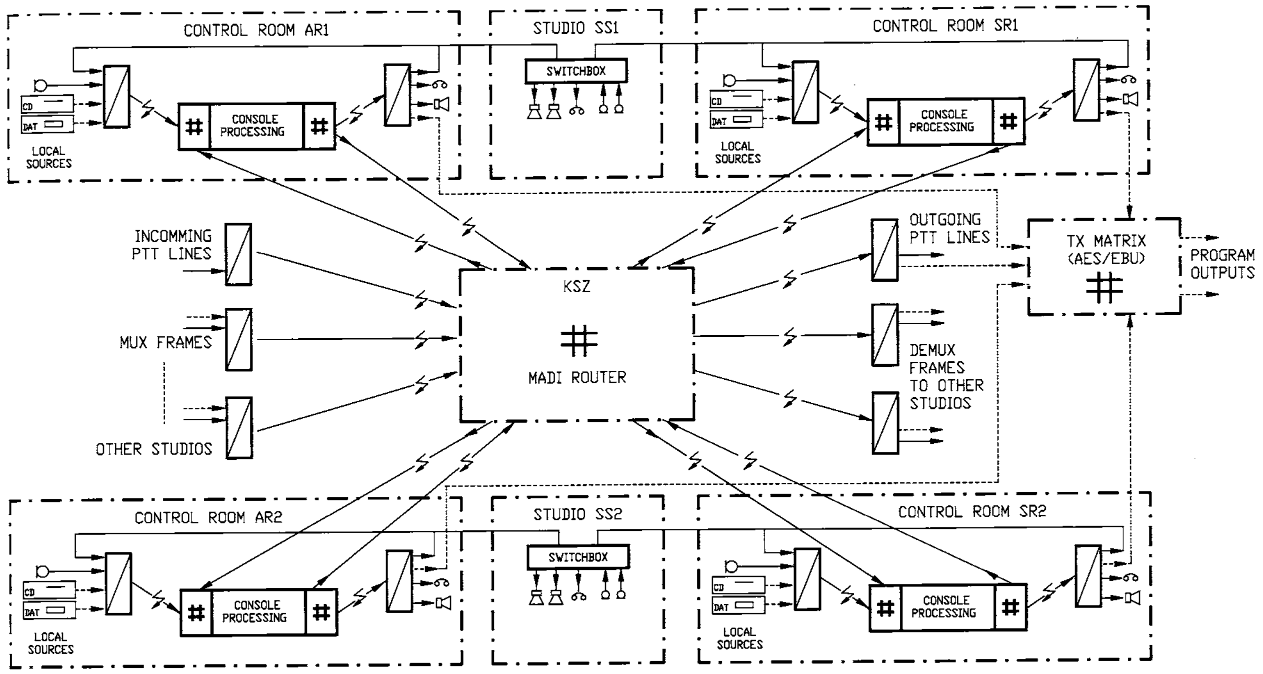
For DJ mode simple and convenient operation has top priority whereas in traditional sound engineer mode high flexibility that allows complex setups such as conference circuits to several OB locations and the utilization of filters, equalizers and limiters, must be feasible.

It was not a simple assignment and required many discussions with the users in order to get these two diametrically opposed requirements under one hat, or on the same user interface. The excellent cooperation paid off, the ergonomically designed userinterface has been fully accepted. Through excellent in-house training the switchover from audio engineer to DJ mode was mastered without major problems. The well-lit, pleasantly furnished rooms and the height adjustment of the continuity console for sitting or standing moderation are conducive to a positive work environment.

Master Control Room

The MCR is located in the base of the broadcasting complex. This is where all broadcast and transmission lines of the PTT are concentrated. The analog signals of the external connections are still transmitted via conventional cables, but also these will soon be replaced by optical connections for digital transmission, as is the case in the new continuity rooms.

The most important components are certainly the two matrices, that is, the main matrix in the



form of a MADI router, and the on air matrix (AES/EBU). Also the signal processing of the four control rooms, multiplexers and demultiplexers, modulation processing and monitoring, telephone hybrids as well as measurement and test equipment are located here.

Normally the master control room is not staffed because all switching operations are remote controlled. In the control rooms and other locations, PCs are available as input devices; these can program switching operation either for immediate or time-controlled execution. The special software runs under Windows and allows the assignment of specific menu screens to the different operating stations. In the control rooms only that part of the matrix is shown which is actually relevant to the corresponding workplace. This largely eliminates operating errors and greatly enhances the clarity.

Automatic connection of the forward and return path with only one switching command, automation insertion of summing during the transition from stereo to mono lines, and a complex monitoring and control system further enhance the ease of use.

The commissioning of the NISKO broadcasting complex at the Radio Studio Zurich is a decisive milestone in the digital age, both for STUDER and Swiss Radio DRS. Despite the much longer broadcasting hours at the Zurich studio, the greater flexibility allows the four new control rooms to replace four broadcasting and production studios built during the seventies.

The project manager and the maintenance manager of the Radio Studio Zurich commented: «The great advance of the new technology is on the user side. With conventional technology it would not have been possible for us to build four identical control rooms that cover the entire spectrum, from DJ mode to engineer assisted coverage of elaborate sports events.» ■

Technical data of the equipment

Four **Studer D941** mixing consoles, each with:

- **Input matrix**
with 122 input channels, of which 56 as MADI connections from the switching room,
20 analog stereo inputs, of which 9 with microphone amplifiers,
8 digital AES/EBU inputs

- **Output matrix**
with 122 output channels, of which 56 channels as MADI connections to the switching room,
8 analog stereo outputs
6 digital AES/EBU outputs

- **16 Stereo input units** with
high and low pass filter - three-band equalizer - limiter/compressor - linear fader - balance - call detector 1900 Hz - multiplex (N-1) circuit

3 Stereo master outputs - 2 Aux outputs - 16 multiplex outputs - metering - control room and studio monitoring - intercom

MADI Router	**AES/EBU Matrix**
16 MADI inputs (896 channels)	16 * 16 AES/EBU channels
10 MADI outputs (560 channels)	
(only partially used)	

STUDER 990 Software V4.1 and V2.4

Magic?

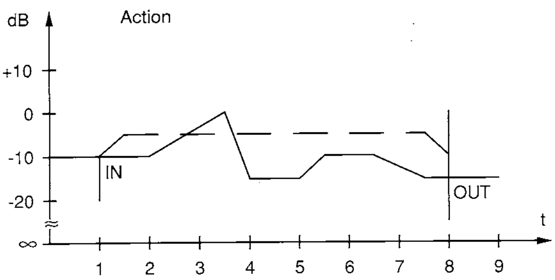
by Tibor Tamas

Tibor Tamas

As if activated by phantom hands, motor faders move up and down, keys switch on and off, the input gain adjusts itself automatically...

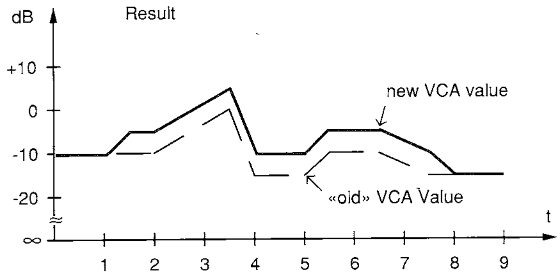
This, and much more, has little to do with magic but rather with dynamic automation as implemented in the Studer 990 mixing console. The latest software packages V4.1 and V 2.4 for the graphic user interface (graphical controller) offer new features and greatly simplified, fast and direct operation. This is important in radio and TV post production as well as recording studios where time is money and everything must be completed as quickly as possible but in perfect quality.

The basis of automation are the time code (TC). All fader and switch data are coupled to this TC. The user can choose between different automation modes. In WRITE mode, mix data are written and if data exist, they are overwritten. Since this is not always desirable, a relative change can be written in UPDATED mode.



The solid line represents the previously recorded fader movements, the dashed line symbolizes the new fader movements. The purpose is to make the mixdown only louder within the overall mix because in all other respects it is ideal. In the second diagram the solid red line represents the result, the dashed line the previous mix data. The fader movements have remained the same but the volume is higher by a certain level.

In addition to the WRITE and UPDATE mode there is the READ mode (mix data are only replayed, no intervention is feasible), the ISOLATE ENABLE mode (mix data are replayed and corrections can be made without being recorded), and finally the ISOLATE mode (automation disabled). All the automation modes are set with the ACU (Automation Control Unit).



The new software also supports the GLIDE mode. When GLIDE is activated the fader or the VCA (Voltage Controlled Amplifier) returns within the user-defined time to the previous mix data after a punch-out from WRITE or UPDATE. This eliminates the need for manual adjustment to the correct level.

In the step-by-step preparation of a mix, the ROLLBACK mode plays an important role. As soon as the time limit of the mix is passed, all fades that are currently in WRITE or UPDATE mode switch to active WRITE mode and produce mix data. This ensures that no holes in the mix occur.

The new software features also include the so-called OFFLINE functions: MOVE MIX, INSERT MIX, and FILL MIX. MOVE MIX which is decoupled from the time code, can be used for shifting a mix on the time axis. INSERT MIX allows a mix to be inserted into another. With FILL MIX a console setting can be written up to the specified time limit.

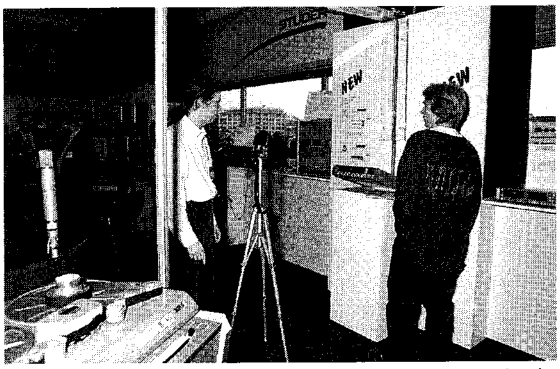
Conclusion: With the 990 mixing console the main fader, small faders, VCA master, all switches, and even the input gain can be automated. Together with the static automation (reset of all faders and switches via snapshot and recall of the potentiometers), the outstanding technical data and excellent sound characteristics as well as the almost legendary reliability the STUDER 990 mixing console offer everything that is required by professionals in demanding every-day studio applications. ■

AES Paris '95

D19 MicAD - a success at the first presentation

by Rudolf Kiseljak

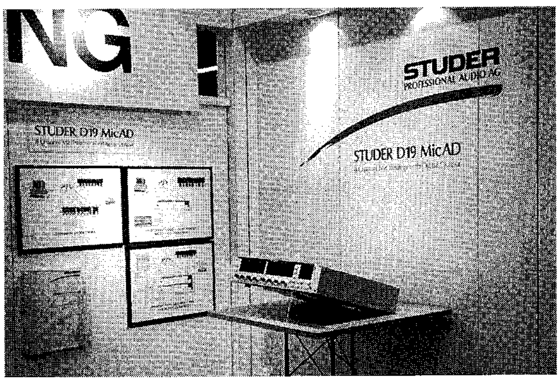
The new STUDER D19 MicAD 8-CH Mic/Line preamplifier with digital output had been an instant success at the AES Europe show in February. It's selection of features, the compact size and the mix of classical analogue circuitry with sophisticated digital technology has been received with enthusiasm by the salesmen and the customers alike. What a good way to start a product's life!



The D19 MicAD, one of the highlights of the show, mostly observed by journalists.

At this year's AES show in Paris, the MicAD was presented for the first time ever. It had been connected to 8 microphones suspended over the STUDER «bar» by an elaborate plexiglas cable duct. The output was connected via AES/EBU directly to the nearby D940 Digital Mixing Console. In such a way, the functionality of the product could be shown, only two months after start of development!

After the show we did some arithmetics and found out that only 4 days since the new product has been introduced, more than 50 units have been sold.



D19 MicAD at Frankfurt Musikmesse

A few weeks later the MicAD has been shown at the «Musikmesse» in Frankfurt, this time in the «STUDER Corner» of the AKG booth. This beeing a show, which is oriented for musicans and project studios, a second unit was shown at the ALESIS stand in the «Dream Studio».

Here the suitability of the MicAD for use with the 8-CH modular recorders could be shown. The Musikmesse has been a success as well, and the new product was extremely well received. ■

Successful products

My name is A807, I'm a jack of all trades!

by Martin Berner

At STUDER in beautiful Switzerland - were I was born - several A807 siblings leave the final inspection bay every day. After careful packing they are shipped to their final destination abroad where they prove themselves in demanding studio duty.

Martin Berner

Why am I so successful?, I have often asked myself. Is it only my stylish design? Or possibly my reliability and my application flexibility? Probably not. Many users praise me because I sing and yodel so well and fully master practically every musical instrument. If I'm honored with a high-quality tape I'm also very quiet and reward my master with a signal-to-noise ratio of over 70 dB, with a magnetic flux of 1000 nWb/m, measured with ASA-A filter. They say that I sound better than the original, but this is rather exaggerated, isn't it?

Are you in a hurry?
Read on nevertheless. If time is precious I am willing to spool a 1000m tape forward or backward in less than 120 seconds because my fast responding tape deck has strong AC motors. If due to the backcoating of the tape clean pancakes cannot be produced at high speed, I can also wind gently and leisurely in library wind mode.

Easy handling
I'm also appreciated for the easy handling and my ergonomical characteristics. To make points with my boss I even had my right (tape tension sensor) arm amputated so that he can edit better.

Easy handling - even my son is able to manage it!

Despite this mutilation I'm able to keep my tape tension stable across the full tape length. The secret: The left tape tension sensor arm controls the supply motor, whereas my microprocessor brain controls the take-up motor by calculating the pancake diameter from the speed ratio of the move roller and the corresponding spooling motor. Clever, isn't it? Using the same logic I can also calculate the current pancake diameter and accept reels of any size without switch settings. Two sapphire guides ensure that the tape remains exactly across the heads so that no information is lost.

Do you want to hitch me?
Due to the large variety of models and sophisticated, menu-controlled programming of individual keys I can be adapted to practically any application environment. If I'm equipped with time code electronics I can even be hitched with a partner. As a result I can march hand in hand, in lip synchronism, with a film. I also feel at home in a postproduction or editing studio or I speak to you directly from a radio station. Even at Olympic games I am a well liked guest who sets the proper tone.

Who has mislaid the screwdriver?
To ensure that a mislaid screwdriver does not get you into difficulties I can be aligned from the front without tools. A soft "prod" with a ball-point pen and I'm ready for alignment. I neatly store all audio data in my no-battery EEPROM memory that has twice the IQ which enables me to store all audio parameters for two different brand of tapes (with different tape flux).

For linearizing the frequency response at each of my three different tape speeds I offer separate controls for all parameters such as level, treble, and bias. The standard Dolby HX Pro® circuit ensures better treble response, particularly at low tape speeds.

Why do I have to hide?
Conscientiously I fulfill my daily duty; often under severe conditions, in a 12 hour day or even longer, during seven days a week. Many customers rely on me to such an extent that they lock me up in a dark room where my beauty can no longer be appreciated.
But important is that I'm fitted with the required interfaces. My generous connector panel features a parallel remote control port and a serial RS 232 interface as standard equipment .

Already tired?
If after a long period of heavy duty I should become old, tired and fail to perform there are well-trained service centers around the world that will rejuvenate me in no time at all. To the delight of the service technician my innards are simple and clean.
Incidentally, many are amazed that all important spare parts required for repair are still available after 10 years.

All this is responsible for my success?
No, not entirely. Not to be forgotten are the customers who have recommended me to their colleagues through mouth-to-mouth advertising. All of them deserve my sincere thanks. Without their assistance my success would not have been half of what it is. ■

SWISS SOUND

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