SWISS SOUND

NEWS AND VIEWS FROM STUDER

SWISS SOUND
A PUBLICATION OF
STUDER
PROFESSIONAL
AUDIO AG

NOVEMBER 1996
No 38

In this issue:

Dear SWISS SOUND reader

Bruno Hochstrasser

For a long time the buzzword "Multimedia" was not clearly characterized with respect to its technical parameters. With today's DVD standard proposal a format has been defined that fulfills all preconditions for spreading this technology.

DVD is a consumer format; in production, however, the aspects of professional studio technology, particularly in the field of pre-mastering, predominate (similarly to the CD). This is where our company sees an interesting field of work which we shall pursue based on an exclusive agreement with Philips. The reports on the following pages provide more information on this field of immediate interest.

The possibility of using the MADI format also for routers has already been described some time ago. Today we can present you the complete family of digital system components which serve not only as input and output interfaces for a MADI node or Series 940/941 digital mixing console, but can also be used for a variety of other functions. In conjunction with digital multitrack machines (SONY or STUDER) they can be used as source multiplexer and as an interface to eight-channel VHS machines, as well as a small router in time-division multiplexing format. The characteristics of this amazing new family of boards are described in the report beginning on page 9.

Some time ago our STUDER 990 console was put into service at the PRO 7 broadcasting complex. The cover picture is reprinted with the kind permission of the studio.

I hope that you will enjoy reading this latest SWISS SOUND edition.

Sincerely yours
Bruno Hochstrasser

STUDER at international trade shows

The jubilee conference of the Audio Engineering Society took place in Copenhagen in May 1996. Again the major companies of the professional audio industry participated with an impressive performance show. STUDER presented for the first time its new digital On-Air 2000 mixing console, the new 928 analog mixing console, and as a European première the D424 MO recorder, as well as a number of units, principally from the digital product range (D940 and D941 digital mixing console, MADI router, D19 line). A continuous stream of visitors to our stand demonstrated that our new developments are of high interest to the market.

Upcoming trade shows

STUDER will participate in the following trade shows either directly or via its national marketing organization:

SATIS PARIS	22.-25.10.1996
AES LOS ANGELES	8.-11.11.1996
INTERBEE TOKYO	13.-15.11.1996
MEDIA BROADCAST LEBANON	13.-17.11.1996
INTERNATIONALE TONMEISTERTAGUNG KARLSRUHE	15.-18.11.1996
PRO AV JOHANNESBURG	18.-21.2.1997
AES MÜNCHEN	22.-25.3.1997
NAB LAS VEGAS	7.-10.4.1997
ITVS MONTREUX	12.-17.6.1997

SWISS SOUND

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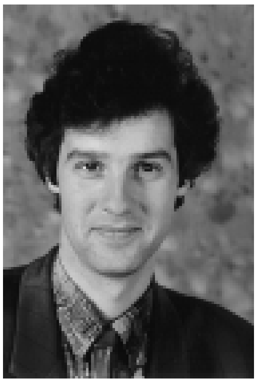
A Harman International Company

Printed in Switzerland
10.26.3110 (Ed. 1196)

Pre-Mastering Equipment for DVD

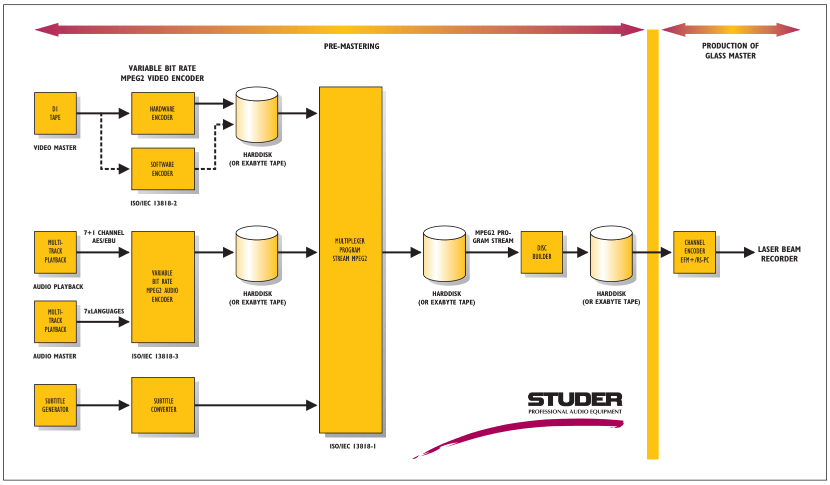
In June 96, PHILIPS and STUDER Professional Audio AG have announced an agreement for DVD-Video Pre-Mastering equipment. The Pre-Mastering equipment for DVD-Video is developed and manufactured at PHILIPS. STUDER is responsible for the world wide and exclusive marketing, sales and service activities.



David Roth

1. The Production of a DVD Master

The production of a DVD Master is devided in two parts. The Pre-Mastering of a DVD, which is covered by Studer and the production of the glass master. The channel encoder for the production of the glass master with the laser beam recorder is also developed and manufactured by Philips, but it will be made available through companies offering equipment to pressing plants. Since both parts, the Pre-Mastering and the production of the glass master is developed by Philips, the complete production process for a DVD stamper is guaranteed through compatible equipment.



2. Pre-Mastering Equipment Video Encoding

2.1. Video Encoding

Video encoding for DVD is standardised as MPEG2. This is identical for all countries. Just the number of lines and the picture rate are different for NTSC and PAL/SECAM countries.

2.2. STUDER DVD320 MPEG2 Video encoder

As a video source, a professional video recorder such as a D1 or a digital Betacam with a serial digital video interface according to CCIR 601/656 is used (digital component video 4:2:2). The STUDER DVD320 encodes the digital video in two steps realtime. In the first pass, each picture is analysed for its complexity and the appropriate data is calculated for an optimum picture quality. If required, the calculated data rate can be adjusted manually before the encoding takes place. The so defined bit rate is used in the second pass for the encoding. During the encoding pass, the encoded MPEG2 video stream is directly decoded, so that with two video monitors an A/B-comparison between the original and the encoded/decoded MPEG2 can be done. The MPEG2 video file will then be stored on the harddisk of the PC which is used with the DVD320. If required, the MPEG2 video file can be downloaded to an Exabyte tape.

Features
- Outstanding picture quality near D1
- Variable bit rate (1..11 Mbaud), depending on the picture complexity
- Encoding in two single passes realtime
- Motion compensated prediction for efficient encoding
- Graphic display of the bit rate (complexity) for each picture
- Possibility to manually allocate more data for pictures in very critical or important scenes
- Definable average bit rate depending on:
  - bit rate required for audio
  - bit rate required for subtitles
  - length of the movie
  - capacity of the disc
- Definable picture rate for 25 Hz (PAL/SECAM) and 29.97 Hz (NTSC)
- Definable aspect ratio 4:3 or 16:9
- Realtime operation with fixed bit rate for other applications

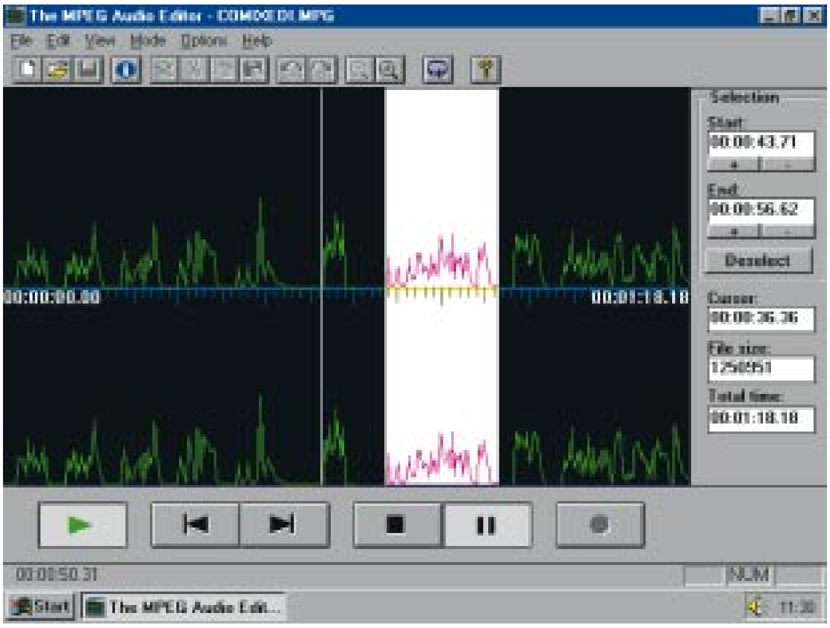
2.3. Audio Encoding

Audio encoding for DVD is standardised as MPEG2 or linear PCM for PAL/SECAM countries. For NTSC countries, Dolby AC-3 or linear PCM is the standard. Up to 8 audio streams might be recorded on a DVD, each containing MPEG2 stereo, MPEG2 5+1, MPEG2 7+1 surround sound, linear PCM or AC-3. To start with, STUDER will support one audio stream with either MPEG2 Stereo, MPEG2 5+1 or MPEG2 7+1 surround sound. Additional audio streams as well as linear PCM and AC-3 will be supported later, if required.

2.4. MPEG2 stereo audio codec for DVD

The MPEG2 stereo audio codec for DVD is a PC plug-in card, which allows a cost effective MPEG encoding and decoding. The software is running under Windows 3.1 or Windows 95. The MPEG2 stereo audio codec can also be used for a lot of other applications than just for DVD. For example the MPEG files created with this card and software can also be used by the broadcast automation system STUDER Digimedia 95.

Features
- Superb audio quality due to optimized algorithms
- 20 bit resolution
- Encoding and decoding in one single pass realtime
- Supports MPEG layer I and II
- Definable bit rate up to 384 kbaud (layer II)
- Definable operating mode (stereo, joint stereo, dual channel and mono)
- SMPTE timecode triggered recording
- Editor function
- Cost effective solution, easy operation
- Suitable for other applications
- Easy operation



Editor zum MPEG2 Stereo Audio Codec

2.5. STUDER DVD310 MPEG2 Surround Codec
The STUDER DVD310 MPEG2 surround codec allows surround recordings with 5+1 channels or even 7+1 channels with variable bit rate. (See also article MPEG2 SURROUND SOUND - THE COMPATIBLE SYSTEM WITH VARIABLE BIT RATE - in this issue on page 5). As an audio source, linear PCM from an 8-channel MDM-Recorder is used. In the first pass audio is analysed for its complexity and the appropriate data is calculated for an optimum sound quality. In the second pass the encoding takes place, using the bit rate calculated in the first pass. The encoding of the audio can be synchronised with the video encoding of the DVD320 MPEG2 video encoder. This allows a direct control of the MPEG2 audio- and video quality in one step if required. The MPEG2 audio file will then be stored on the harddisk of the PC which is used with the DVD310 MPEG2 surround codec. If required, the MPEG2 audio file can be downloaded to an Exabyte tape or to a CD-R with the CD-recorder Studer D741 through the SCSI interface.

Features
- 5.1 and 7.1 recordings
- Compatibility with MPEG2 stereo decoders
- Superb audio quality due to optimized algorithms
- 24 bit resolution
- Variable bit rate, depending on the surround sound complexity for efficient encoding (up to 912 kbaud)
- Encoding in two single passes realtime
- Possibility to synchronize the encoding processes for simultaneous control of the audio- and video quality
- SMPTE timecode triggered recording

2.6. Subtitle Generator
Up to 32 subtitle channels can be recorded on the DVD. A subtitle generator from a third party will be integrated in our Pre-Mastering equipment for DVD.

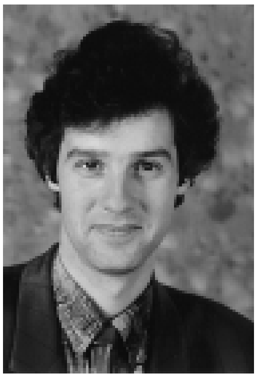
2.7. Multiplexer
The multiplexer is a PC with software, which merges the video MPEG2 stream, the audio stream(s) and the subtitle stream(s) into one DVD program stream. Two versions of this multiplexer will be available. One with just the basic functionality and one with additional features such as parental coding possibilities. The parental code allows an additional data stream, which defines the content of sex and violence for each part of the movie. If during play back on a DVD-player the sexual or violence content should surpass a certain level which is defined in the DVD-player, the corresponding scene will be cut out during play back without visual and audible effects.

2.8. Disc builder
The disc builder is a software process, which translates the multiplexed video, audio and subtitle data into a disc image ready for channel encoding. Therefore, a table of contents is added and the DVD program stream is devided into data sectors.

2.9. System controller
The system controller, which is also part of the Pre-Mastering equipment is not shown in the figure on page 3. It is used for checking, if all parameters such as bit rates are within the DVD-specification. ■

The compatible system with variable bit rate

ISO/MPEG2 SURROUND SOUND



David Roth

With DVD a new era of multi channel sound will begin. The concept of multi channel sound is an old one. Early demonstrations date back to the thirties. It is generally believed, that DVD will mean the breakthrough into the consumer market, as the next step after the analog surround systems such as DOLBY™ Prologic.

Part of the DVD standard is the MPEG2 surround sound system. In the design phase of this world-wide international standard special consideration has been given to compatibility issues. MPEG2 is a compatible surround sound system.

Multi channel sound set-up

Everybody will be familiar with the first system using multiple audio channels: the stereo system, in which a representation of a sound stage is reproduced using two audio channels, Left and Right (Fig. 1). This system has been employed for many years now, and will remain for many years to come.

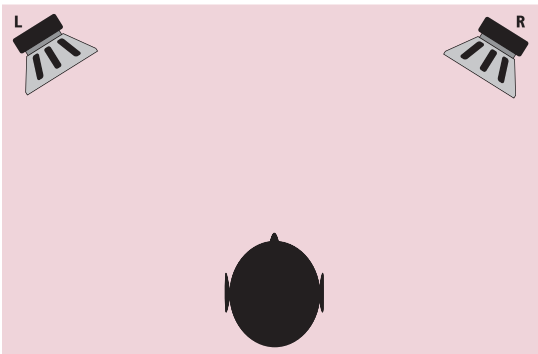


Fig. 1: Stereo reproduction

It was soon discovered that the addition of a third Centre channel provides a wider zone of good stereo localisation and a better center localisation, if the Left and the Right channel are located at a too far distance to each other (Fig. 2).

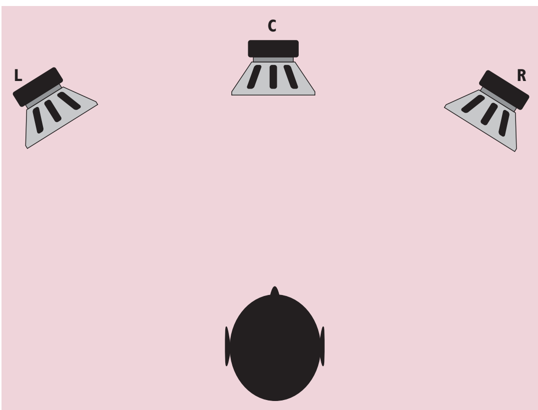


Fig. 2: Stereo with center channel

DOLBY™ Surround, the analog system that is currently in use to bring surround sound effects into the home theatre has a fourth Surround channel, which is reproduced with two speakers behind the listener (Fig. 3). This system is in use in cinemas since the mid of the seventies . Thousands of movies are encoded with this surround system from Dolby, which shows some stunning ambient effects.

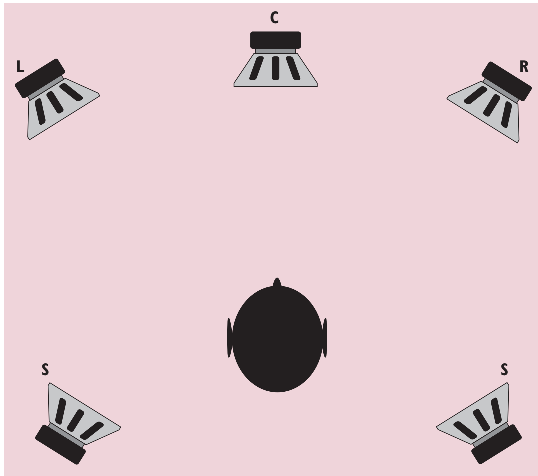


Fig. 3: Dolby™ surround

However, since the four channels are encoded in two channels with phase coding, this system lacks of limited dynamic range and limited bandwidth in the Surround channel. For music reproduction, the system is very sensitive to phase shifts in the transmission channel, which can produce an unstable sound image. In addition the system lacks a natural "all around" characteristic, because both Surround speakers have the same signal. This can be overcome using two separate Surround channels (Fig. 4), Left surround (Ls) and Right surround (Rs).

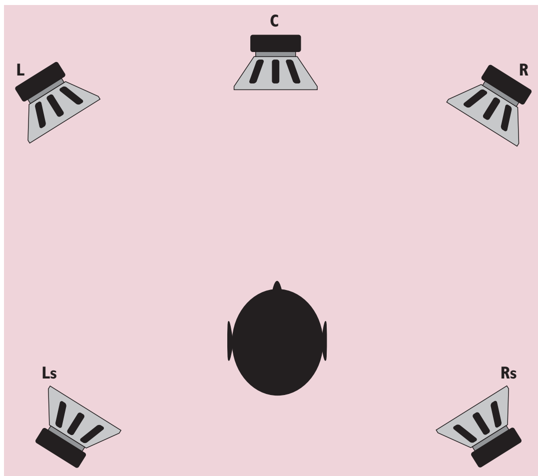


Fig. 4: 5 Channel surround

MPEG2 surround sound is based on this five channel set-up. However, to provide the listener with the option to extend the low frequency content of the reproduced audio in both frequency and level, a sixth Low Frequency Enhancement (LFE) channel has been added (Fig. 5). This system is normally referred to as 5+1 or 5.1. The subwoofer used to reproduce the LFE channel does not need to be located in any particular location, as the human ear is very limited in detecting the direction of these low frequencies from 20 Hz to 120 Hz.

Please note, that all other channels (L, C, R, Ls and Rs) contain the complete frequency spectrum from 20 Hz to 20 kHz. Specially, if small speakers are used at home for the above mentioned channels, a subwoofer is recommended

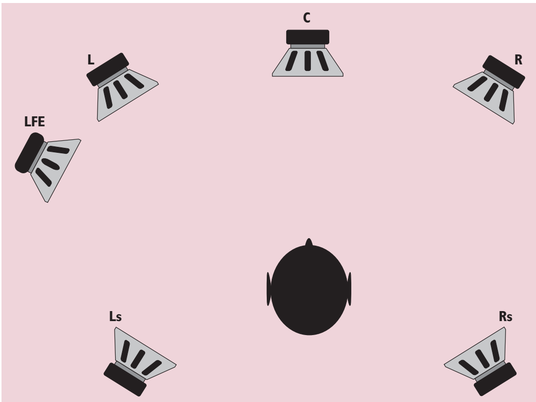


Fig. 5: 5 channel surround with subwoofer

to enhance the low frequency content which can not be reproduced by the small speakers. For large screen movie theatres sometimes two additional channels are provided which compensate for the wider viewing angle (Fig. 6).

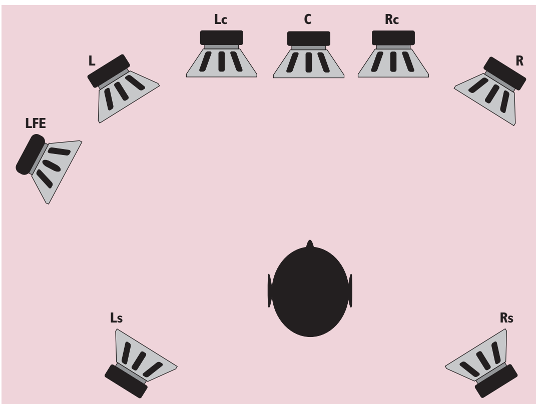


Fig. 6: 7 + 1 Channel surround

The ISO/MPEG2 bit stream: built in compatibility

The MPEG2 bit stream has been designed with compatibility in mind. The minimum number of channels to be reproduced is assumed to be two, for stereo (although it is of course possible to even make a monaural signal). Therefore, the backbone of the MPEG2 bit stream is a stereo encoded MPEG1 signal (Fig. 7). A first extension, identified in the figure as MC 5 (+1), provides additional information with which it is possible to retrieve the 5 (+1) channel information.

Fig. 7: MPEG2 data stream

With more additional information in a second extension MC 7, it is possible to reproduce the full 7 (+1) channels.

This compatibility is achieved by employing matrix techniques during encoding and decoding.

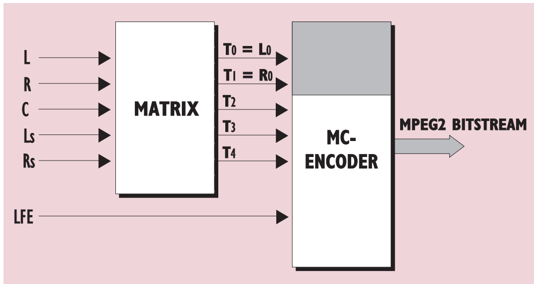


Fig. 8: Multichannel MPEG2 encodierung

During encoding, as shown in Figure 8, the input information is used in an encoding matrix to generate new signals that will be psychoacoustically encoded . Two of these (T0 and T1) are combinations of the input signals that will be encoded as an MPEG1 stereo signal. These signals will be the outputs of a simple decoder that only provides a stereo signal (see Fig. 9).

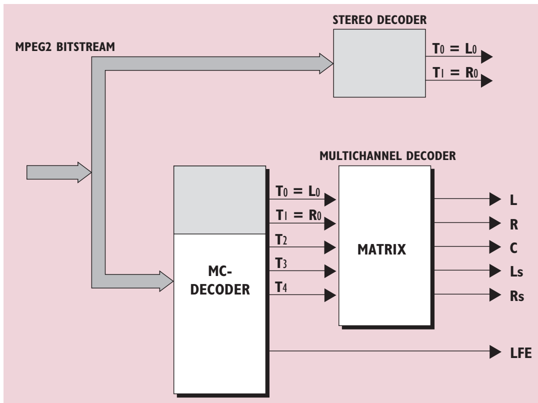


Fig. 9: MPEG2 decodierung options

Therefore they are generated in such a way that the resulting stereo signal is a combination of all input signals.

Unlike other systems a stereo decoder does not need to calculate the multi channel information before a stereo downmix can be made. With MPEG2 the stereo downmix is already in the bit stream. As a consequence, the basic MPEG stereo decoder is less complex. The stereo portion of the MPEG2 signal is also used in multi channel decoders. Together with additional information in one or both multi channel extensions, the signal is fed to a decoding matrix which regenerates exactly the original five or seven channels. One of the most important features of MPEG2 audio is the forward and backward compatibility with MPEG1. An MPEG2 decoder will properly decode an MPEG1 bit stream (either mono, stereo or dual channel), and an MPEG1 decoder will decode the basic stereo information from a multi channel MPEG2 signal.

Fig. 10: Compatibility of multi-channel MPEG2 sound

		Decoder:		
		2 ch DVD	5 + 1	7 + 1
Source:	Stereo	Stereo	Stereo	Stereo
	Prologic	Prologic	Prologic	Prologic
	5 + 1	Prologic	5 + 1	5 + 1
	7 + 1	Prologic	5 + 1	7 + 1

DOLBY™ Prologic compatibility

As surround sound brings the viewer in touch with the cinema experience, and as such is a vital part of the present day home movie entertainment, it has been decided to equip even the most basic DVD player with the possibility to provide a Prologic signal on it's stereo outputs. Therefore the MPEG2 system has been made compatible with Dolby Prologic. This has been achieved by applying a special coding matrix (Fig. 8), which generates the T0 and T1 signals identical to a two channel Prologic signal.

There are two possible multi channel decoding methods (Fig. 11):

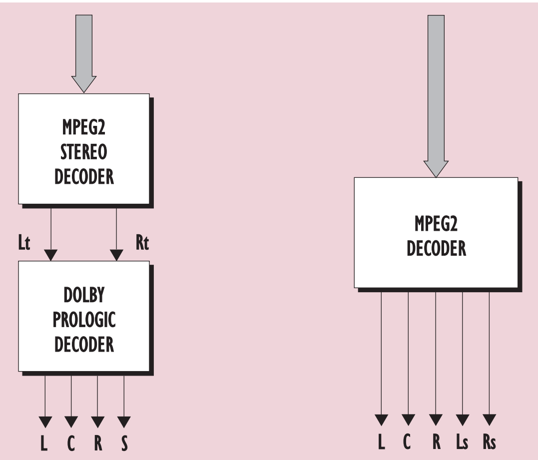


Fig. 11: Decoding options

Either a normal Prologic decoder applied which operates on the stereo signal as decoded by a two channel MPEG1 or MPEG2 decoder. In this case, only one Surround channel is created which will be reproduced by the two speakers behind the listener. This is ideal as an intermediate solution, because the first DVD-Players will probably have just a stereo output, which could be connected to already existing Dolby Prologic equipment. The second solution is a multi channel MPEG2 decoder, which will decode the bit stream into five channels. This should be the preferred long term solution because the quality is far better due to digital encoding and decoding with 5 complete descrete channels.

Variable bit rate for efficient coding

Variable Bit Rate (VBR) is a method to significantly increase the efficiency of MPEG encoding of audio and video. MPEG2 audio offers the possibility to adapt the bit rate used by the encoder every 25 ms to the complexity of the sound to be encoded. Simple pieces of music demand a low instantaneous bit rate, while complex sound demands much higher instantaneous bit rates. For most sound material, the average bit rate required is much lower than the bit rate required to code the most complex sounds without artifacts. This is especially true for movie sound tracks. For some scenes there is a lot of activities in all channels requiring high bit rates. Other scenes contain practically nothing in the Surround and Centre channels allowing for very low bit rates. If VBR is used, the capacity required on the disc for an audio stream will correspond to the average bit rate required for that sound track. In contrast, a Fixed Bit Rate (FBR) encoder requires a bit rate necessary to encode the most complex piece of sound. This means that the encoding efficiency is low for simple passages of sound e.g. silence. Please note, that variable bit rate means no variable bit rate around a fixed average bit rate using a buffer. Variable bit rate allows for example to encode the first ten minutes of a movie with 256 kbits/sec only, while the last 5 minutes are encoded using the maximum bit rate of 912 kbits/sec, which is allowed by the DVD system.

Fixed bit rate encoding is inherently a subset of variable bit rate encoding, therefore the use of variable bit rate is always optional at the encoder side. A decoder for DVD has to support VBR and FBR. Therefore the content provider is free to exploit the advantages of VBR or to use FBR. Encoding with variable bit rates is usually performed in two steps. In the first pass, the audio is analysed for its complexity. In the second pass the encoding takes place, using the bit rate calculated in the first pass. In case of FBR, only one pass, the encoding is required.

The advantages of VBR should become clear in the following figures. Figure 12 shows the distribution of bits rates in the VBR case, for a relatively long part in reel 6 with 20 minutes of the soundtrack of the movie "Immortal Beloved". This is a digital soundtrack that contains different types of classical music, dialogue and some effects. The average bit rate to encode this soundtrack without any audible artifacts is about 384 kbits/sec. It can however be seen from this histogram that the highest instantaneous bit rates required are around 600 kbits/sec. This means that when using FBR, a bit rate of around 600 kbits/sec would have to be used to encode the soundtrack without any audible artifacts. Using VBR, a bit rate of 384 kbits/sec is sufficient, which is a gain of 36%!

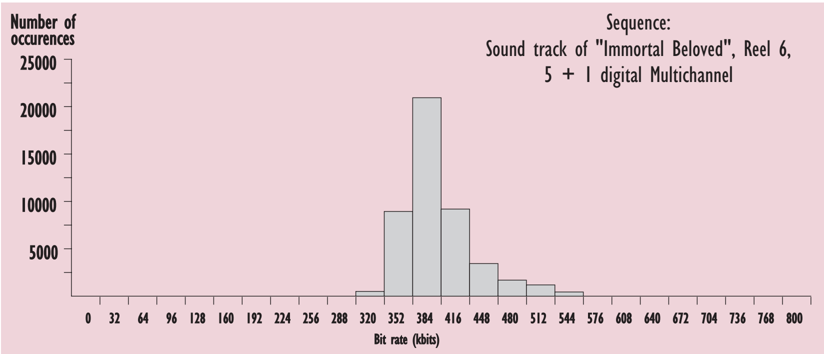


Fig. 12: Bit rate statistics

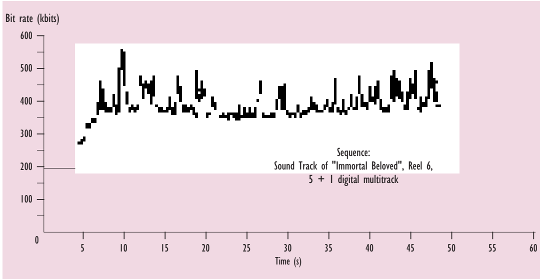


Fig. 13: Changes of the bit rate

Fig. 13 shows how the requested bit rate varies over the time when VBR is used. Here, a shorter part with 48 seconds of the same soundtrack is used. The bit rate can, and it does change every 25 ms according to the actual contents of the sound. It can also be seen that a bit rate of around 384 kbits/sec is sufficient for most time, but that for short periods of time a significantly higher bit rate is necessary to obtain the highest quality. ■

STUDER D424 magneto-optical recorder supports SADiE 3 Interchange format

Alex Rüegg

During this years AES convention in Copenhagen STUDER has announced the cooperation with SADiE in order to create a common audio interchange format.

The D424 offers 16, 20 and 24 bit resolution and can be equipped with either a Pioneer 2.2 GB or standard ISO 2.6/1.3 GB 5¼ inch magneto optical drive. In 16 bit/48 kHz mode and 2.6 GB media, recording time is stated at 100 minutes per side.

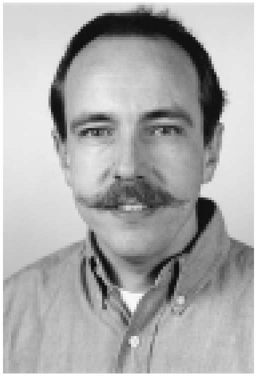
During recording audio files are created and written in the common Microsoft WAVE format. Because of the 100 % edit decision list (EDL) compatibility, the media can instantaneously be interchanged between the two systems without any loss of information or timeconsuming file conversion. Even PQ data is completly transparent for both systems.

Futhermore the functionality of the D424 has been enhanced in such a fashion that the internal MO drive can be mounted to any PC via SCSI. This allows not only to perform data exchange via removable media but also by being able to use the D424 MO drive as an additional storage device to the SADiE workstation.

D424 recorders already in use will be upgraded with Software V 1.1, which is planned to be released at the end of the year. Beside of many new features it will support the SADiE 3 Interchange format and automatically convert the existing EDL. ■

D19m – a new system of digital components

Karl Otto Bäder



Rainer Kunzi

For building a complete system, additional components are often needed that handle system functions which the individual units cannot fulfill.

In the analog field corresponding STUDER components have been available for a number of years. For digital systems STUDER has now begun to close the corresponding gap by introducing the D19m series (m stands for modular).

With respect to the circuit technology there are clear similarities to the D19 series. The same high-resolution converters are used that can be found in MicAD and MultiDAC. Also the interface design corresponds to the D19 series.

The mechanical dimensions, however, are completely different. Whereas the individual units of the D19 series are designed for the 19″ format occupying 1 or 2 units of vertical space, the D19m series consists of boards with a width of 100 mm (Euro card width) that can be inserted into a chassis occupying 1 or 3 units of vertical space (Fig. 7). Each configuration can be individually tailored. Of course, the card chassis has a built-in power supply and can optionally be fitted with a redundant power supply.

Components

The following boards are currently available (see also block diagram Fig. 1):

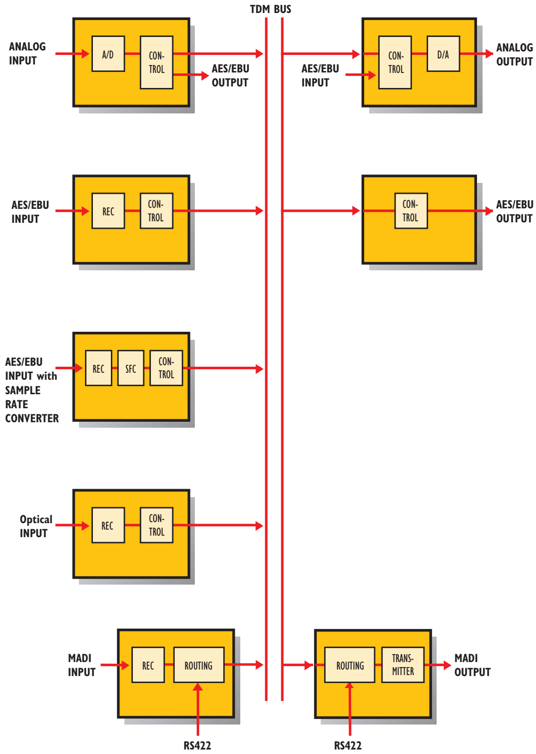


Fig. 1: Blockschaltbild

- D19m C4AD
  Quad 20-bit A/D converter with four balance output inputs, interface on the TDM bus plus additional AES/EBU outputs (Fig. 2). This board converts two stereo signals or four mono signals; two mono channels each are packed into an AES/EBU frame. Each channel is allocated to a subframe on the TDM bus and subsequently also in the MADI frame.
- D19m C4DA
  Quad 23-bit D/A converter with four balanced analog outputs, interface from TDM bus plus additional AES/EBU inputs (Fig. 3). This board recovers up to two stereo or four mono signals.
- D19m AESI
  Dual AES/EBU input board with interface to the TDM bus (Fig. 4). The two AES/EBU signals occupy four subframes on the TDM bus and subsequently also in the MADI frame.

Fig. 2:
D19m C4AD

Fig. 3:
D19m C4DA

- D19m AESI - SFC
  Dual AES/EBU input board with unbalanced sampling frequency converter per input; outputs to the TDM bus. This board corresponds to the D19m AESI except that the inputs are equipped with asynchronous sampling frequency converters. These are suitable for connecting not only sources with a different sampling frequency but also 48 kHz sources that are not synchronized. In addition, this board features two AES outputs which means that it can be used also as an autonomous sampling frequency converter.
- D19m DA
  AES/EBU dual distribution amplifier 1 to 8. The inputs can be connected in parallel so that a 1 to 16 distribution amplifier is created.
- D19m ARG
  Synchronization and AES/EBU blank frame generator. Creates a "DARS" signal conforming to AES 11; can be controlled by an AES/EBU signal or by a word clock signal.
- D19m VS video synchronization unit
  This board is additionally required if a system is to be controlled by a video clock. It

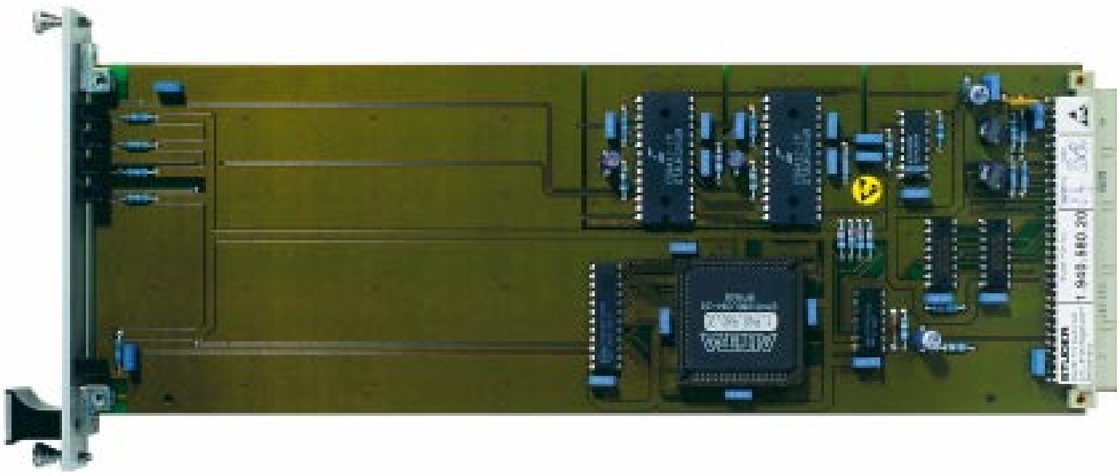


Fig. 4:
D19m AESI

first converts it to a word clock signal that can control the D19m ARG board.

- D19m AESO
  Dual AES/EBU output board with interface from the TDM bus.
- D19m MADO
  MADI output (optical or coax) with TDM input. Conversion of the TDM signal to the MADI format. The assignment of the TDM subframes to the MADI subframes can be defined via an RS 422 interface (Fig. 6).
- D19m MADI
  MADI input (optical or coax) with TDM output (Fig. 5).
- D19m Dual ADAT™ interface
  Interface between the Alesis ADAT™ optical 8-channel format and the TDM bus. For connecting ADAT™ units or STUDER D19 MicAD preamplifiers.
- D19m M81
  AES/EBU 8 to 1 router. These boards can be cascaded for building a small router.

Fig. 5:
D19m MADI

Also included are two card racks:

- D19m frame/1U
  Rack chassis 19″, 1U, incl. power supply, accommodates up to 4 boards.
- D19m frame/3U
  Rack chassis 19″, 3U, incl. power supply, accommodates up to 15 boards.

- Universal unit with A/D converter, D/A converter, sampling frequency converter and digital signal distribution.
- Analog input rack and analog output rack with more stringent specifications for MADI oriented digital recorders (for example, from STUDER or SONY).

- Individual boards can also be integrated into other systems.

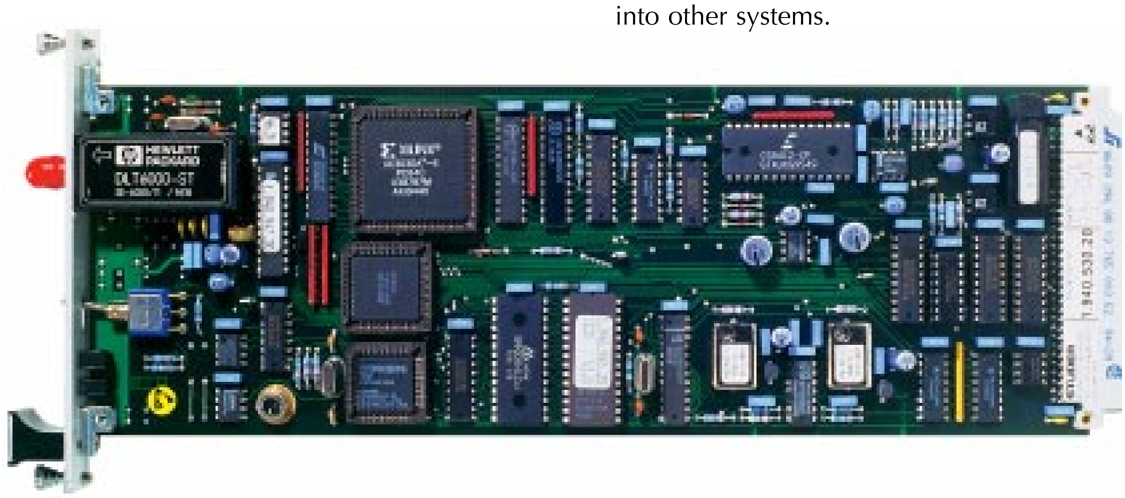


Fig. 6:
D19m MADO

Applications:

With the D19m components a variety of applications can be solved, such as:

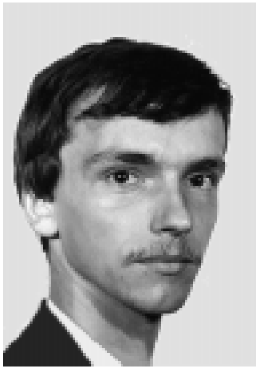
- Custom wired card racks for:
  - Sync generator incl. AES/EBU distribution system
  - Small TDM routers with variable interface configuration (digital and analog). For example a 28 x 28 or a 56 x 56 router can be installed in one or two 3U racks respectively.
  - Multiplexing and demultiplexing arrangements
  - Stage boxes with optical link to the audio production (for example, in conjunction with STUDER D19 MicAD modules).

The D19m series will be continually expanded in the future.

Fig. 7: D19m chassis with power supply

Sound quality can be measured

Karl Otto Bäder

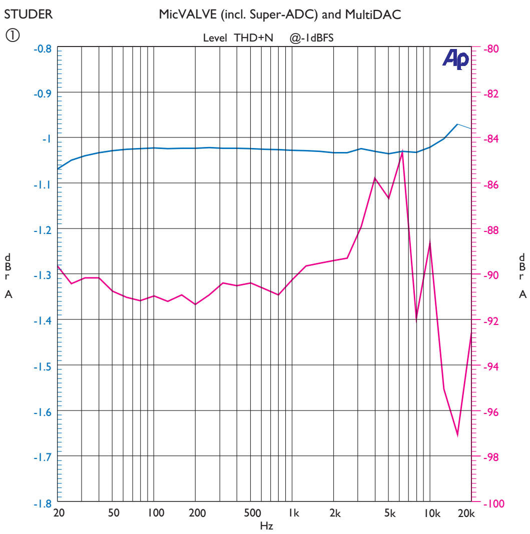


Silvio Gehri

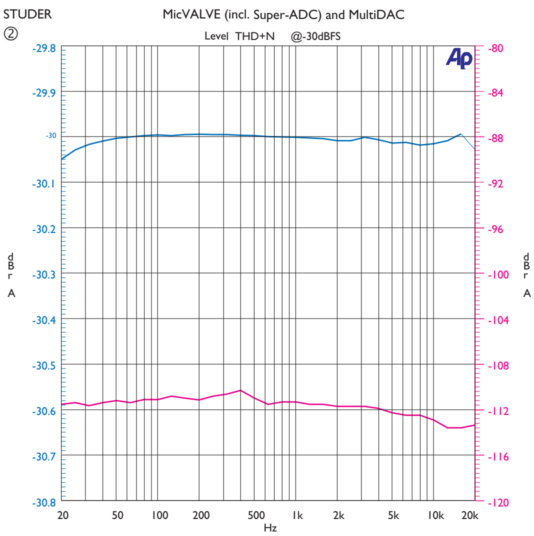
Behind the “excellent sound” of the STUDER equipment there are, of course, technical reasons. The corresponding parameters are measurable and can be weighted. The following report gives some examples of the quality control procedures to which our equipment is subjected.

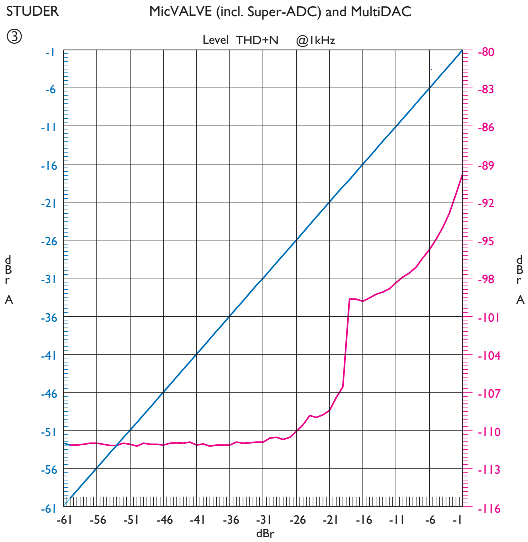
Converter

Although the introduction of the sigma-delta converters resulted in a significant improvement over previous designs, the conversion of analog signals to the digital level and vice versa is still a potential weak spot in the observed system.

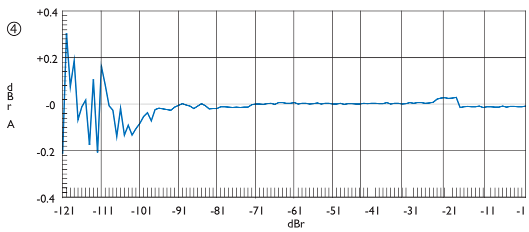


The measurements show the results of a chain of STUDER equipment of the D19 family connected in series, that is, the MicVALVE with 22-bit A/D conversion and the MultiDAC with 23-bit D/A conversion. This chain covers not only the complete conversion and reconversion in the measurement path, but also all other components of the equipment (including input and output transformers) and therefore corresponds to the situation encountered in practice.



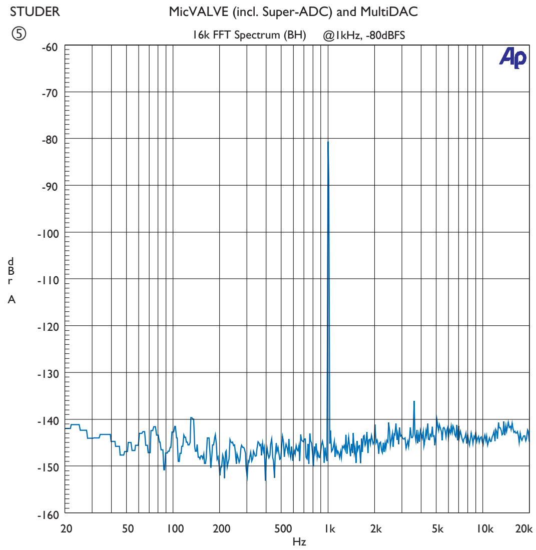


Measurement 1 shows the frequency response (left-hand ordinate) as well as the THD+N (right-hand ordinate) at a level 1 dB below full scale. In the THD+N measurement the stimulus is filtered out. Striking is that despite the transformers the harmonic distortion does not increase at lower frequencies. Even at approx. 6 kHz the harmonic distortion is still less than 0.01 %. As shown in measurement 2 this distortion is caused by an analog circuit (same measurement with a stimulus of -30 dB). Here the distortion components are no longer recognizable, the THD+N value consists purely of noise.

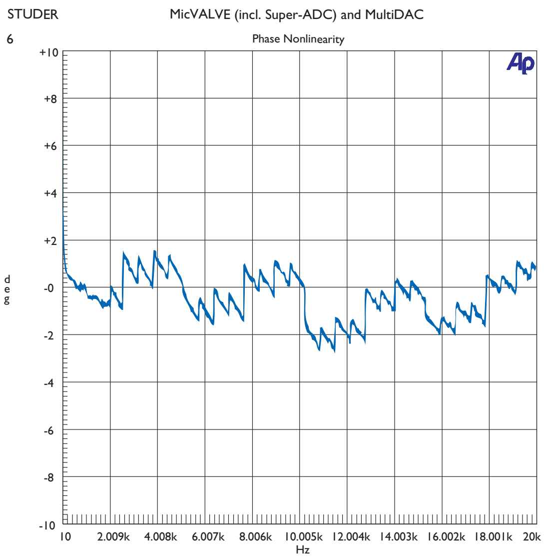


Whereas the two first measurements were conducted with a sliding frequency but fixed level, measurement 3 shows a recording with a fixed frequency (1 kHz) but rising level. The left-hand ordinate again relates to the level, the right-hand ordinate to the THD+N value. At lower levels there is a constant background noise of approx. 111 - 112 dB below full scale which increases only at higher levels.

Of major interest is measurement 4 that investigates the non-linearity errors of the converters. The measurement was conducted with a bandpass. Throughout the investigated spectrum the maximum deviation remained below 0.2 dB where the values below approx. 105 dB are caused by inaccuracies of the measurement system.

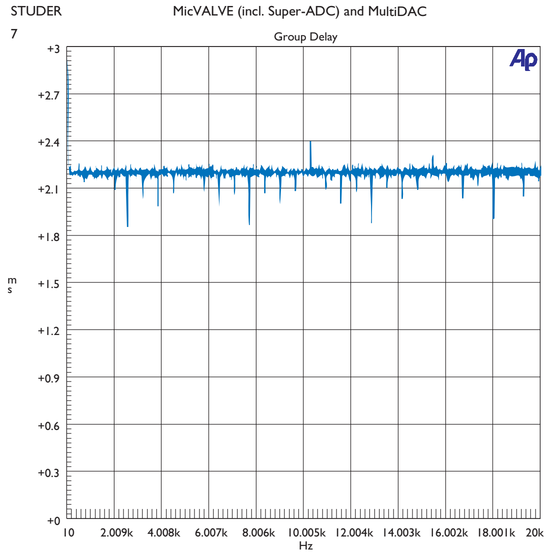


Measurement 5 illustrates an FFT analysis of the noise and distortion spectrum with a measurement tone of 1 kHz at -80 dB. At 16 k FFT points and using a Blackmann/Harris window a constant bin of 2.7 Hz over the measurement range is observed. As this value remains constant the noise does no longer increase with the root of f as it would be typical for white noise.

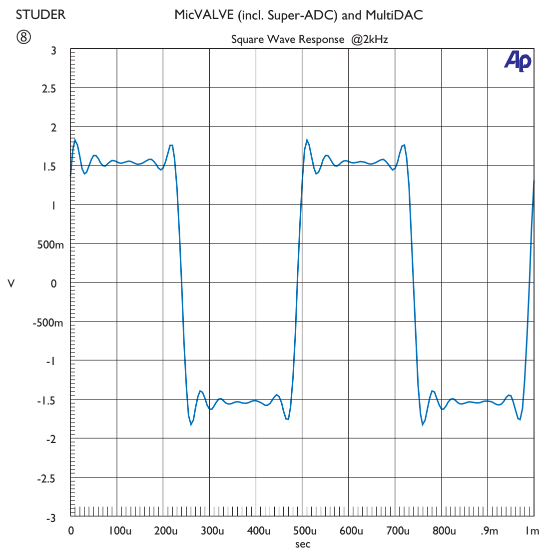


The measurement shows that hardly any harmonic components occur; only at 2 kHz can a very small value be detected. Also discrete low frequency components in the noise cannot be detected.

Measurement 6 is a very tricky measurement because for measuring the non-linearity of the phase the basic delay of the measurement path must be compensated. In the measurement system (SYSTEM ONE from Audio Precision) very high phase values occur before the compensation. Due to these high values the system must change over to floating point mode. The continuous mode switching results in the uniform staircase of the measurement. Accurate qualitative statements on the characteristics of the test object cannot be made; certain is only that they are significantly better than those of the measuring equipment.



The group delay error was computed from measurement 6 and depicted in fig. 7. It also shows the superposition of the effects from the test object and the measuring equipment. It is apparent, however, that the threshold of 0.4 ms referred to in the literature (Blauert/Schlichthärle) is never exceeded. In the tested STUDER equipment no group delay effects are noticeable; only at extremely low frequencies (below 50 Hz) can a rise be observed.



In view of these excellent results in the phase error measurement it comes as no surprise that also the square-wave behavior is very good. Measurement 8 shows a highly symmetric pattern of the overshoots with a 2 kHz signal.

News from the STUDER World

- With an order from the renowned Studios *Roberto Carlos* in Rio de Janeiro, 200 digital STUDER D827 DASH multitrack machines are now in worldwide operation. *Robert Carlos* became acquainted with this machine during recordings in the USA; decisive for the purchase decision was finally the excellent reputation enjoyed by this machine worldwide. This D827 is the first digital STUDER machine in Brazil.

- Analog tape is still extensively used in radio studios because the inventory in the archives is enormous. Based on its positive experience with STUDER products in the past the Italian RAI has ordered 80 type A807 and A812 tape recorders.

- RADIO VARNA located directly at the Black Sea in Bulgaria is building a new radio complex. The order for the technical equipment and system concept with a total volume of almost CHF 2 million was awarded to STUDER.

- West-German Broadcasting (WDR) in Cologne continues to upgrade its radio studios to digital technology. After nearly two years of experience with a STUDER D940 digital mixing console two additional 32-channel consoles were ordered. For the on-air operation five digital studios will be set up, each equipped with a STUDER D941 mixing console.

- The advantages of the MADI format for router applications are increasingly recognized. Today's users include not only Swiss organizations such as DRS Zurich, RSR Lausanne and RSI Lugano, but also ARTE in France, Billboard in the USA, ORB in Potsdam/Germany, Radio VARNA in Bulgaria, and NRK Oslo/Norway.

- After extensive tests the renowned artist and producer, Alan Parsons, has decided on the STUDER D19 MicVALVE. Not only did Parsons confirm the excellent characteristics of the adjustable tube sound parameters but he also described the built-in converter as the best on the market.

- NOB in Hilversum/Netherlands was the first TV broadcaster to use a STUDER 980 mixing console with surround monitoring in an OB van. Based on the excellent experience two more OB vans will be equipped in the same manner.

- Radio Bukarest has started a large project for upgrading its broadcasting center. STUDER will supply seven mixing consoles of various types and forty A807 tape recorders.

- Audio tasks in theaters are becoming increasingly complex because modern authors call for special audio effects or because the standards established by television and multimedia technology also have to be met on stage. As a result also the theaters must increasingly be equipped with modern control rooms. For the Staatstheater Kassel/Germany STUDER was able to supply a 990 mixing console with 56 channels, automation, and in-audience remote control.

- Radio TV Zagreb (Croatia) expands its regional stations as well as the TV continuity. For the regional studios three STUDER 904 mixing consoles and for the TV continuity four STUDER 962 as on-air mixing consoles were purchased.

- Despite the growing use of hard disc recording devices for multichannel recording, tape is still in demand due to the enormous storage capacity and the easy exchangeability of the medium. STUDER D827 digital tape recorders with 24 or 48 channels have recently been supplied to RAI in Italy, to Taiwan, Korea, Thailand, Poland, Ukraine, KSS and Nihon University in Japan, and to several studios in the USA such as Enterprise Burbank, Elysian Fields Boca Raton, and Criteria Miami.

- In its department of music the University of Bombay offers not only courses on vocal and instrumental Hindustan music but since 1993 also a study program on music recording and reproduction. For this purpose a complete music recording studio, essentially furnished with STUDER equipment, was installed. This study program under the supervision of Prof. Manchar Kunte is the only one of its type in India and has been included by AES in its "Educational Register".

News from the STUDER World

On Tour

In our age of rapidly changing technologies the information exchange between users and manufacturers is more important than ever. In July 1996 we were able to welcome a delegation from Spain which after the visit of a few installations (DIGIMEDIA '95 at Radio Pilatus in Lucerne and NISKO at DRS Zurich) engaged in detailed technical discussions at our factory.

The picture shows from left to right:
J. Rueda, Catalunya Radio, Barcelona, *L. Marti*, Televisio de Catalunya, Barcelona, *G. Nuez*, Cadena Cope, Madrid, *F. Fuchs,* STUDER, Regensdorf, *R. Herraez,* Antena 3 Television, Madrid, *F. de Corral*, Telco Electronics, Madrid

We congratulate our distributor in Saudi Arabia, *Said bin Saad al - Ghamdi*, to his 20th anniversary of his company. This period saw the renovation of Radio Jeddah, the installation of numerous REVOX language training systems as well as equipment in Median, Mecca, Taif, Dammam and the FM station of Aramco in Dahran.

Through his correct and reliable nature *Said bin Saad* has made many friends, in his own country as well as here in Switzerland.

Already since 1992 a STUDER 990 with 28+8 channels has been in operation in the television studio of Televisione Svizzera Italiana (TSI). It allows the concurrent recording of two productions in the large production studio 1 and often the sound has to be recorded and assembled in two languages. The snapshot automation has greatly changed the operating procedures of the audio engineers; the preparation times have become much shorter and the flexibility has greatly improved.
(From an experience report by TSI Lugano).