

SWISS SOUND

NEWS AND VIEWS FROM STUDER

SWISS SOUND
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Editorial

After an extended pause «Swiss Sound» is back again with its latest edition which - so I hope - will be read with great interest. In the meantime much has happened at STUDER in Regensdorf. In the beginning of July 1994 we changed our former name, STUDER REVOX AG, to **STUDER Professional Audio AG**; the ownership of the company had changed already effective March 1, 1994. We are now part of **HARMAN INTERNATIONAL INDUSTRIES**, a group of companies, completely dedicated to audio electronics.

The HARMAN Group, headquartered in the USA, its companies in the US, Europe and the Far East achieve a combined annual turnover of nearly US \$ 1 billion. Familiar sounding brand names such as LEXICON, AKG, SOUNDCRAFT, JBL and now also STUDER - to name just a few - are part of this group and are able to offer a comprehensive range of professional technology and services for broadcasting services, the recording industry, PA systems, and related applications.

What does this new environment mean for STUDER? And what about the independence and the preservation of the traditions that have been developed over decades and have been much appreciated by our customers?

The corporate philosophy of HARMAN quickly dispels such anxieties. The established strategy is to preserve the specific identity of each company and to take advantage of possible synergies with other group companies. By being a member of the HARMAN Group our company will be strengthened, and this enhances our capabilities with respect to products and services.

The typically Swiss tradition, the concepts of product quality and business partnerships developed over many decades, will be preserved, which means that you - as our customer - will always be the focus of our attention.

Bruno Hochstrasser

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STUDER D731 QC

Quality Inspection of CDs and CD-Rs

by David Roth



David Roth

The compact disc is one of the most reliable media available today. The «red book» (CD) and the «orange book» (CD-R) specify not only the data formats but also the minimum quality criteria for CD players and CDs. The CD technology has become so refined that the manufacturing quality of CDs and CD players now exceeds the «red book» specifications. Playability problems have become rare. This is one of the many reasons why the CD has become so popular.

In order to achieve and maintain this high CD production quality, the minimum quality criteria must be monitored with special quality control CD players. Due to the wear caused by daily use, the quality of a CD can deteriorate to the point where these criteria are no longer met. This eventually results in playability problems which are unacceptable in professional applications, such as radio broadcasting. For this reason quality control CD players are recommended also for these users.

The new Studer D731 QC quality control CD player supersedes the A725 QC which in the meantime has become the de-facto standard QC player and can be found in virtually every CD pressing plant.

In this report I would like to discuss the principal quality measurements conducted with the D731 QC and their significance.

Digital error flags

The CD decoder works with a two-stage error correction system. The data recorded serially on the CD are taken to the input of the first error correction stage. The data are combined in error correction units, so-called frames. Each second 7,350 frames have to be processed for error correction (digital sampling frequency 44,100 Hz / 6 stereo samples).

In the **first error correction stage**, the so-called C1 decoder, these frames are checked for errors and corrected, if possible. The following digital error flags can be counted:

E11: Total of all frames that contain **one** faulty symbol (byte).

E21: Total of all frames that contain **two** faulty symbols.

E31: Total of all frames that contain **more than two** faulty symbols.

BLER: = $E11 + E21 + E31$. This is the total of all faulty frames. According to the «red book» no more than 3% of all frames may be faulty, that is, 220 of 7,350 frames.

BURST: This is the maximum number of consecutive frames with E21 and E31 error flags, that is, the number of consecutive frames with more than one faulty symbol. According to the «red book» no more than 7 consecutive frames may contain multiple errors. On the CD this corresponds to a length of approximately 1 mm.

The **first correction stage (C 1)** of a typical CD decoder is able to correct statistically distributed minor errors (E11 and E21). This is the principal function of the first error correction stage. But if the errors become larger (E31), the CD decoder can no longer correct these errors. These errors are principally caused by local faults such as air bubbles and scratches. In this case the C1 decoder does not perform any correction but flags the frames as C1 uncorrectable.

As can be seen from Fig. 1, the error-free frames, the corrected frames, and the C1 uncorrectable frames are input to the de-interleaving. This reverse interleaving process reas-

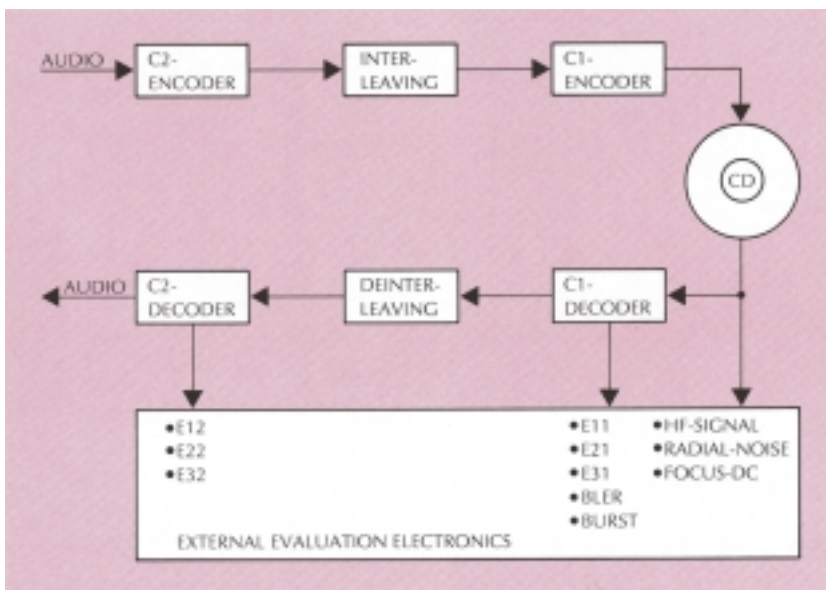


Fig. 1: Two-stage CD encoder and decoder

sembles the data in the proper sequence. Through this process errors flagged as C1 uncorrectable are «intermixed» with many error-free and corrected frames so that the C2 decoder can still correct them. As a result the decoder «sees» many faulty frames but these contain just a few errors which it can correct.

In the **second error correction stage (C2)** the frames are checked and corrected, if possible. In this process the following digital error flags can be counted:

E12: Total of all frames that contain **one** faulty symbol (byte).

E22: Total of all frames that contain **two** faulty symbols.

E32: Total of all frames that contain **more than two** faulty symbols.

For the second error correction stage there are no direct limits because the E12, E22 and E32 values depend on the performance of the first error correction stage in the CD decoder. Depending on the decoder IC, from one to four symbols per frame can be corrected in the C1 and the C2 decoder. The D731, for example, contains a decoder that can correct the theoretic maximum in both error correction stages. In the CD quality control technology, however, a medium error correction is used that corresponds to the one in most CD players. In the C1 as well as the C2 decoder up to 2 symbols/frame are corrected. One E22 flag consequently is the limit for this error correction. One E32 flag means that interpolations or even muting can occur. For this reason two decoders are used in the D731QC.

The decoder responsible for processing the digital and analog audio signals is capable of correcting the maximum number of errors, whereas the second decoder employs a medium error correction strategy and supplies the previously mentioned error flags, which are identical to those of the A725 QC player.

Fig. 2 shows the error signals of a test CD containing 3 different errors:

- Information layer interruptions of 0.4 to 1.0 mm
- Black dots on the read-out side: 0.3, 0.5, 0.6, 0.8mm
- Simulated fingerprint

The first 5 diagrams show the pattern of the error flags BLER, E22, E32, BURST and EFAB (system limit of CD error correction, see diagram 5).

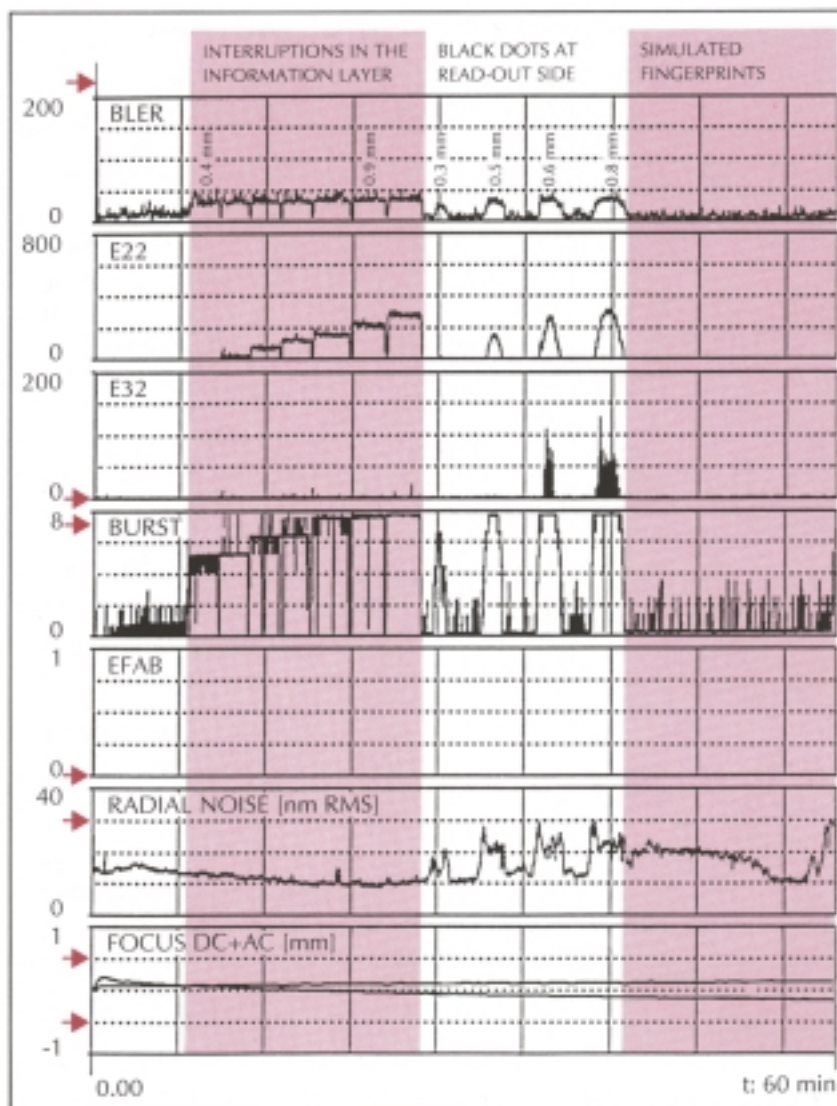


Fig. 2: Typical representation of a playability measurement

- **Diagram 1**

The **pattern of BLER** shows a base value of approx. 20, a typical value for good CDs. If an interruption in the information layer occurs or if black dots are present on the read-out side, the BLER increases only slightly to a maximum value of approximately 50, that is, it remains well below the limit of 220. Since a small local fault is very small in relation to a complete revolution, the BLER does not increase very much and does not exceed the BLER limit.

The BLER is consequently a measure of the **general pressing quality** of a CD. But it provides no information on local faults!

- **Diagram 2**

The **pattern of E22** shows that the first error correction stage was able to correct everything. For this reason E22 is 0. Only at the local faults where the information layer is interrupted and black dots occur does the E22 value increase, up to the limit of 300 for a fault with a length

of 1 mm. There is no official limit for E22, but an E22 should not exist, otherwise the faults will already become audible on a CD player that employs a very primitive error correction strategy.

The E22 measurement proves a good assessment of **local faults** on a CD.

- **Diagram 3**

The **pattern of E32** indicates how frequently the capability of a typical error correction strategy has been exceeded. In this example we see that practically all errors were correctable because the base value remains 0. Only for the larger 0.6 and 0.8 mm black dots can significant E32 values of approximately 100 be measured. Normally these produce audible noise in a CD player.

- **Diagram 4**

The **BURST pattern** is similar to E22. The main reason for measuring BURST is that it is specified in the «red book», which is not the case for E22. The base value of BURST typically less than 2 which is normal for a good CD. As in the case of E22, BURST increases as a function of the error magnitude and exceeds the value of 7 if the information layer interruption is longer than approx. 0.6 mm and black dots on the read-out side are greater than 0.5 mm.

- **Diagram 5**

The **pattern of EFAB** is normally not measured. EFAB specifies whether or not uncorrectable errors exist after the best error correction (as used in the D730, D731 and D732). Since EFAB in this case remains at 0, there are no interpolations or other audible faults when this test CD is reproduced by our CD players. The pattern of E32 and EFAB clearly shows why our CD players have a significantly better playability than other CD players.

Analog error signals

The best error correction is useless if the laser cannot follow the track. Particularly large local faults make heavy demands on the radial servo. A measure of faults in the radial control circuit is the **RADIAL NOISE**. The «red book» specifies a limit of 30 nm RMS ($1 \text{ nm} = 10^{-3} \mu\text{m}$).

- **Diagram 6**

The **RADIAL NOISE pattern** shows a base value of approx. 10 nm. This is a typical value for good CDs. At the black dots on the read-out side and the fingerprint on the test CD, the RADIAL NOISE increases up to the specified limit. Frequently the maximum value of RADIAL NOISE produced by a local fault does not oc-

cur in the middle (where E22 is the highest), but before and after. This behavior is characteristic for air bubbles. Because the laser beam deflection is strongest at the edge of a bubble, the reflected radial error signal is greater than in the middle where the laser beam deflection is not so strong. This example clearly shows that playability problems depend not only on the size but also on the type of the local fault.

- **Diagram 7**

The **pattern of FOCUS DC** shows the warp and wobble of the CD. The «red book» specifies a limit of $\pm 0.5 \text{ mm}$ peak and 0.4 mm RMS. This means that the test CD is neither concave nor convex. The wobble increases slightly toward the outer side of the disc where it reaches the uncritical value of $< 0.15 \text{ mm}$ peak.

RF signal from photodetector

Another very important signal is the RF signal produced by the D731QC. This RF signal is still analog and unconditioned as read by the player. This signal provides further indication as to why, for example, the digital error signals show poor measurement results.

The following measurements can, for example, be conducted with the RF signal:

- **JITTER:**

Jitter is a phenomenon that started to appear with the CD-R. The pits can have only a discrete length of 3, 4, 5...11T ($T = 231.38 \text{ ns}$). The «orange book» allows a maximum deviation of 35 ns RMS from the pit length. If this value is exceeded, digital error flags may possibly occur. These problems have particularly been noted with certain makes of CD-Rs that have a recording time of > 74 minutes.

- **ASYMMETRY:**

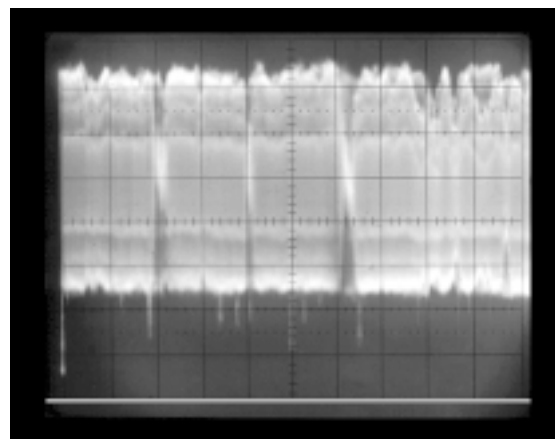


Fig. 3a): RF signal of a severely scratched CD. Time base = 4ms/div.

With a good RF signal the decision threshold should be in the middle of the large amplitude (I_{11}). If this is not the case the pit shape is probably not correct.

- I_{top} :

On the D731QC the maximum reflection (I_{top}) can be measured directly with the oscilloscope. A reading of 1 Volt corresponds to a reflection of 100%. The «red book» limit is 70% and normally does not present any problem.

- I_{11} / I_{top} :

The ratio of the large amplitude (I_{11}) to the total reflection corresponds to the modulation depth. According to the «red book» it must be > 0.6 . If this value is too small the pit depth is frequently too small. Since the pit depth is given by the stamper, all CDs of this batch have the same problem.

- I_3 / I_{top} :

The ratio of the small amplitude (I_3) to the total reflection should be > 0.3 . If this value is too small, the linear velocity is frequently too low. This also occurs with CDs having a playing time of > 74 minutes.

If only one of the foregoing parameters is outside the tolerance, this normally does not yet result in playability problems, but this will be the case with certain combinations of parameters. In the various CD players available on the market these problems manifest themselves differently, or not at all.

With the CD player STUDER D731QC and appropriate evaluation electronics additional parameters can be measured. The most important ones which are, for example, measured by pressing plants, have been described in this report.

The D731QC reads and tests all commercially

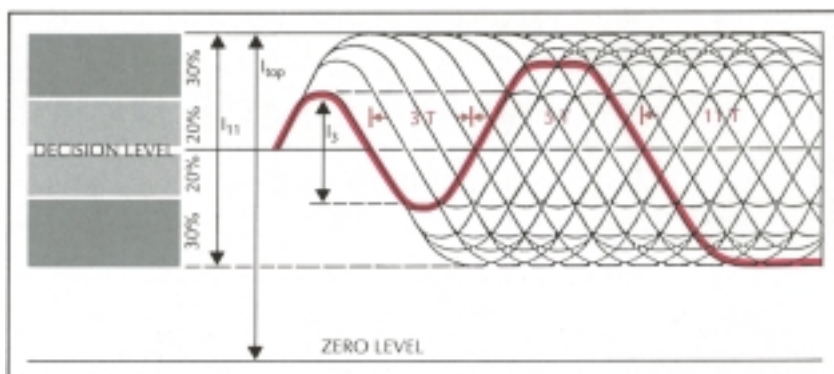


Fig. 4: Ideal RF signal

used CD formats

The D731QC can reproduce not only the CD-AUDIO but all commercially used CD formats (CD-ROM, CD-I, VIDEO CD, CD-R...). The digital error flags and the analog error signals are identical for all CD formats and can be measured with the D731QC.

Outstanding playability of STUDER CD player family

Allow me to conclude this report with an answer to question as to why STUDER has developed the D731QC. One of the principal considerations in the development of the new CD player generation D730, D731 and D732 was the playability of poor CDs. The excellent values were only achievable through an in-depth analysis of QC problems. For this purpose we have collected a CD horror set comprising over 100 CDs with different faults for accurate analysis. Numerous tests and experience in radio broadcasting have shown that the effort was worth while. The playability characteristics of the new CD players are without doubt the best that can be achieved today.

The standard broadcast version of the D731 was an excellent base, the step to the D731QC was logical rather than big.

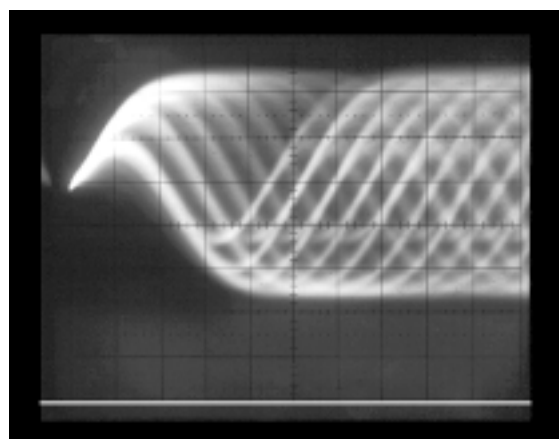


Fig. 3b): RF signal of a severely scratched CD. Time base = 0.4 μ s/div.

From Engineer to Engineer

The Algorithm in Digital Audio

by Silvio Gehri and Paul Zwicky



Silvio Gehri

Every function inside an audio device is performed by a «computer». We tend to forget that e.g. an analog filter circuit is simply an analog computer. Often its design is developed intuitively; the engineer glances at the circuit layout and immediately knows what it does. The selection of its components and their ratings is derived from certain limits (frequency response) of the transfer function which sometimes isn't even known.

On the other hand, the DSP engineer needs an accurate transfer function so that his digital filter will perform the desired filter function. The relatively diffuse specifications of customers or product managers don't suffice anymore.

We shall now demonstrate a possible approach by an engineer who wants to develop a simple shelving filter for the digital domain.



Paul Zwicky

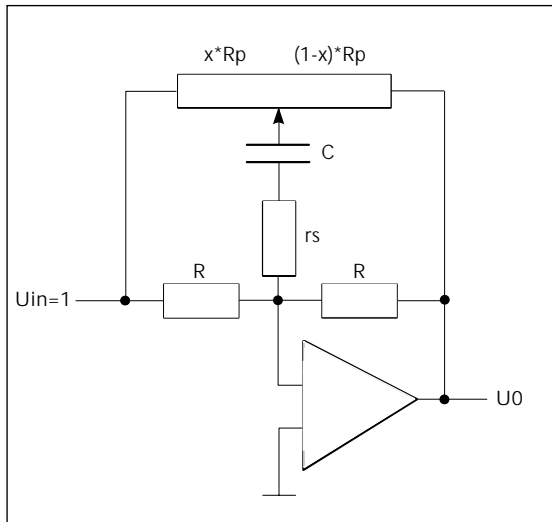


Fig. 1: Analog circuitry for a high-frequency shelf

An analog filter is designed

Given is the well-known circuitry for a high-frequency shelf in Fig. 1 where x denotes the position of potentiometer R_p . x can assume any value between 0 and 1. R_p is assumed to be small relative to r_s , so the potentiometer is simply a voltage divider with zero ohm source impedance.

Using the node equations for the inverting input of the opamp, we get Eq. (1). Solving for U_0 yields Eq. (2).

$$\frac{1}{R} + \frac{(1-x) \cdot R_p}{r_s + \frac{1}{s \cdot C}} = \frac{0 - U_0}{R} \quad (s = j \cdot 2\pi f) \quad (1)$$

$$-U_0 = \frac{(-C \cdot R \cdot x + C \cdot R + r_s \cdot C) \cdot s + 1}{(r_s \cdot C + C \cdot R \cdot x) \cdot s + 1} \quad (2)$$

Replacing the product $R \cdot C$ by the time constant τ yields

$$-U_0 = \frac{\left(1 - x + \frac{r_s}{R}\right) \cdot \tau \cdot s + 1}{\left(\frac{r_s}{R} + x\right) \cdot \tau \cdot s + 1} \quad (3)$$

The maximum boost is achieved with $x = 0$

$$A_{\max} = R \cdot \left(\frac{1}{R} + \frac{1}{r_s}\right) \quad (4)$$

which allows the determination of r_s/R , which, inserted in Eq. (3), leads to the desired transfer function $H(s)$:

$$H(s) = \frac{\left(\frac{1}{A_{\max} - 1} + (1-x)\right) \cdot \tau \cdot s + 1}{\left(\frac{1}{A_{\max} - 1} + x\right) \cdot \tau \cdot s + 1} \quad (5)$$

$$\text{Normalform: } H(s) = \frac{\alpha \tau s + 1}{\beta \tau s + 1} \quad (5)$$

We now have found a transfer function which in addition to the potentiometer setting x has only A_{\max} (maximum boost) and t (time constant). To check the result, we use math-software and simulate Eq. (5). The resulting graph in Fig. 2 reveals the correctness.

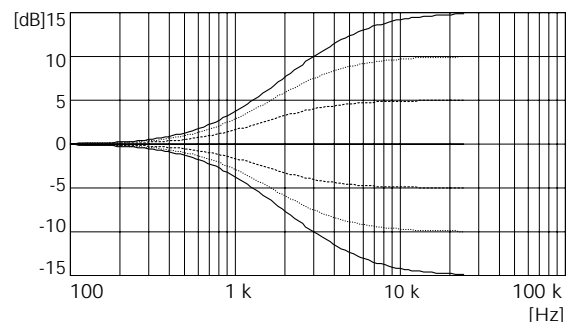


Fig. 2: Transfer function according to equation (5) for $A_{\max} = 15$ dB and $\tau = 1/(2 \pi \cdot 1000 \text{ Hz})$ for different potentiometer settings.

The Equivalent Digital Filter

Several steps are required to transform the analog filter into the digital domain. We first have to find a structure which is suited to audio applications. One possible structure is shown in Fig. 3, the so-called Direct-Form 1.

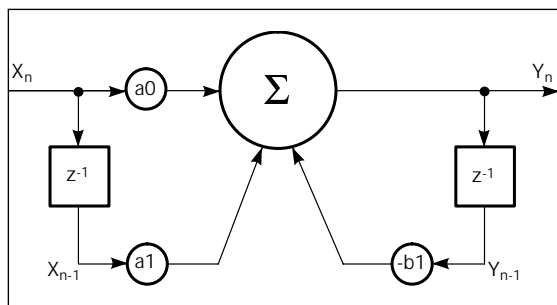


Fig. 3: Block diagram of the Direct-Form 1 for a first-order IIR

The input signal is denoted by X_n and the output by Y_n . The blocks z^{-1} can be treated as memories, i.e. X_{n-1} and Y_{n-1} are audio samples from the preceding sampling period. The output sample Y_n is computed as:

$$Y_n = X_n * a_0 + X_{n-1} * a_1 - Y_{n-1} * b_1 \quad (6)$$

In the digital domain, delaying an audio sample by k samples means multiplication by z^{-k} . So Eq. (6) can be rewritten as:

$$Y = X * a_0 + X * z^{-1} * a_1 - Y * z^{-1} * b_1 \quad (7)$$

Solving Eq. (7) for Y/X yields the transfer function for the digital filter of Fig. 3

$$H(z) = \frac{a_0 + a_1 * z^{-1}}{1 + b_1 * z^{-1}} \quad (8)$$

One possibility of getting the coefficients a_0 , a_1 and b_1 is the well-known bilinear transform which transforms the (digital) z -plane into the (analog) s -plane as follows:

$$s = \frac{2}{T_s} * \frac{1 - z^{-1}}{1 + z^{-1}} \quad (9)$$

T_s is the sampling period. We now insert Eq. (9) into the normal form in (5) and obtain an equation for $H(z)$. By comparing its coefficients with the ones in (8) we get

$$a_0 = \frac{T_s + 2 * \alpha_1 s}{T_s + 2 * \beta_1 s} \quad a_1 = \frac{T_s - 2 * \alpha_1 s}{T_s + 2 * \beta_1 s} \quad b_1 = \frac{T_s - 2 * \beta_1 s}{T_s + 2 * \beta_1 s} \quad (10)$$

At this point, the digital filter is completely defined. Of course, we now check whether it is the same filter as the analog original. This can be verified through simulation with the same math-software.

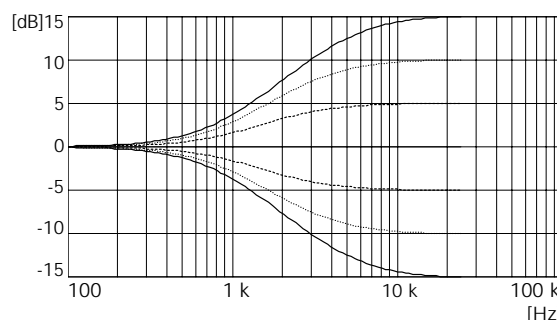


Fig. 4: Transfer function of Eq. (8) with the coefficients of (10).

Starting a design is easy

We have shown that a digital filter can be quickly computed, as long as the transfer function of an analog filter is known. Some pitfalls along our way have been neglected so far. For example: We suddenly find the bilinear transform to work incorrectly when high cut-off frequencies are involved - the frequencies become inaccurate. Even our simple example is not completely correct: Fig. 5 shows the difference between the analog and the digital transfer function.

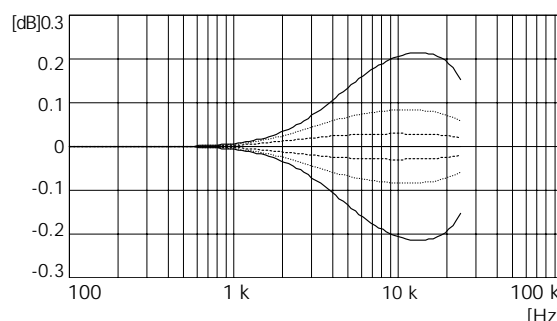


Fig. 5: Difference between the analog and the digital filter transfer function.

Conclusion

What seems to require a mystical touch isn't that sophisticated at all. On closer examination the design process is simply a logical consequence based on available knowledge. Digital filters - often called algorithms cannot be engineered without knowledge of their analog parents. Only the associated problems differ, of course, but their discussion is beyond the scope of his article. ■

STUDER D827 MCH

Extended Digital Resolution - or: 24 bits for the future

by Thomas Knäple



Thomas Knäple

The STUDER D827 MCH 24/48 channel DASH tape recorder made its début in October 1993. Its functionality and modularity are the result of the most advanced multitrack recording technology. Only one year later STUDER introduces a new world premiere: Extended Digital Resolution, EDR, retrofitable as an option on the D827 MCH. For the first time, linear 24 bit recording is now possible in digital multitrack technology.

Bigger, faster, wider - these often are attributes that describe modern technologies. In digital recording, the technological characteristics are often expressed as the number of processed or storable bits - the smallest information unit in digital signal processing. The aim is to have as many of them as possible, that is, to use the largest possible word length for recording the data.

Aside from earlier experiments, the 16 bit quantization of data soon became the standard in digital audio studio technology. A (theoretical)

signal-to-noise ratio of $S/N = 6.02 \times 16 \text{ (bit)} + 1.76 \text{ [dB]} = 98.08 \text{ dB}$ is remarkable. There are, of course, other advantages over analog recordings such as loss-free creation of copies, etc.

A word length of 16 bits has become established across the entire audio technology, on the CD, the DAT recorder, as well as in professional recording equipment such as the DASH tape machines.

Why now 24 bits?

Bigger, faster, wider - yes, even 16 bits have their limit which is easy to demonstrate. An analog signal basically has an infinitely fine resolution. But 16 bits allow only $2^{16} = 65,536$ steps for approximating the original, that is, not even close to infinity. By contrast, 24 bits give a resolution of $2^{24} = 16,777,216$ steps.

Admittedly, this is still far from the (infinite) analog truth, but at least the original analog signal can be recorded with a resolution that is 256 times more accurate than in 16 bit technology. (Using the same formula above this results in a theoretical signal-to-noise ratio of amazing 146.24 dB).

From theory to practice

The advantages described above do not sufficiently explain why a recording should be made in 24 bit technology when the origin is possibly limited to a 18 bit A/D conversion or when the ultimate playback device, as for example the CD, is limited to 16 bit.

A significant advantage of the signal-to-noise ratio is that a much larger headroom can now be used without sacrificing the resolution. But this is only one side of the coin.

But (digital) recording should not be regarded as an isolated process. In most cases the (digitally) recorded signal requires further processing, mixing, frequency response corrections, etc. Ideally, these tasks should also be performed at the digital level in order to prevent unavoidable losses in D/A - A/D conversion. This work can be performed most conveniently on a digital mixing console for which the



D827 MCH with the EDR option is the ideal (recording) partner.

Let us take another brief excursion into the naked theory and technology, that is, the principles of digital mixing consoles.

Modern signal processor architectures, like those found in the STUDER D940, use words with a width of up to 56 bits for all signal processing algorithms. Why? A rule applicable to calculations with binary numbers states that the multiplication of two data words with the width «n» (i.e. «n» bits) results in the word width «2n». For example, when two 16 bit words are multiplied in a digital mixing console, the result contains 32 bits.

The axiom is: keep the resolution as high as possible

It is desirable to maintain this high-resolution result as long as possible because losses occur inevitably in every conversion from a larger to a smaller word width.

Ideally, the word width should only be reduced to 16 bits in the final processing step, for example, the CD mastering. Usually, a special process is used for this purpose.

Large data volume

Recording with a larger word length also results in larger data volumes which means that greater storage capacity is needed. Up to now, magnetic tape still offers the best price/performance ratio for large storage capacities. With the enormous capacity of 25 Gigabytes of 1/2" tape on a 14" reel, 24 bit recording in DASH technology would be absolutely feasible, at least from an economical viewpoint. But the standardized DASH format does not foresee this application.

A change of the DASH format does not make sense. Compatibility must be maintained with the installed base of over 1000 DASH multitrack machines. But this is only achievable with a trick.

The solution is «Bitmapping»

Without changing the DASH format, the EDR option for the D827 MCH realizes the linear 24 bit recording of up to 24 channels. To accomplish this the (24 bit) data of a recording channel are distributed to two physical tracks of the tape.

The most significant 16 bits of the 24 bit word are recorded on track A, the least significant

8 bits on the associated track B. This approach is not new and is known from other solutions.

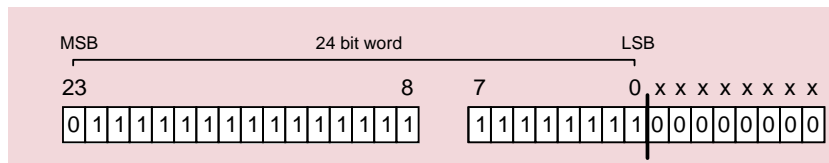


Fig. 1a: Original 24 bit word, allocated to two 16 bit tracks

New and unique, however, is that 24 bit processing is also possible on the recording device. This is essential if the 24 bit quality is to be continually preserved, for example, also during punch-in or punch-out. For these operations the recorder must calculate cross-fades - in 24 bit resolution of course - and this is where external solutions do not work.

The principle is illustrated in Fig. 1: A 24 bit word is allocated to two 16 bit tracks, for example, in the manner shown in Fig. 1a. If the tracks are processed independently (Fig. 1b), that is, if two independent 16 bit cross-fades are performed during a punch-in, this results in

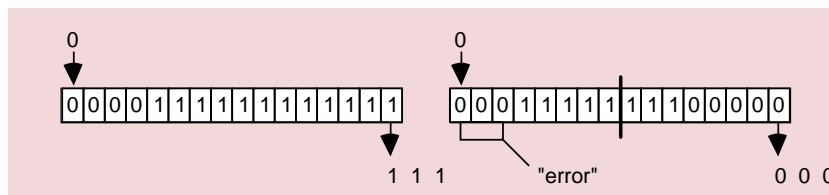


Fig. 1b: Independent 16 bit processing of two 16 bit words
Example: Division /8 (3 x shift right)

an error. However, if both tracks are processed jointly (Fig. 1c), unimpaired 24 bit quality is ensured also during cross-fades.

EDR for MADI and AES/EBU

In order to accomplish this, a special processing unit has been designed (Fig. 2 and 3) in which a 24 bit process is superposed on the standard 16 bit process. Access to this unit is implemented through the multichannel audio

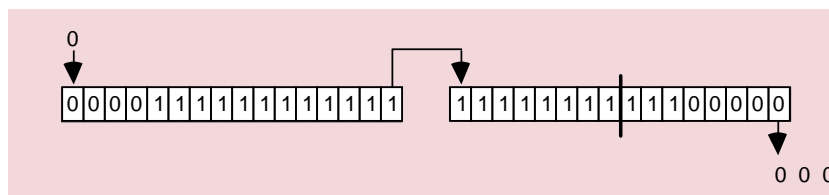


Fig. 1c: True 24 bit processing
Example: Division /8 (3x shift right)

digital interface, MADI (simultaneous for all tracks, coaxial or fiber-optic) or the AES/EBU

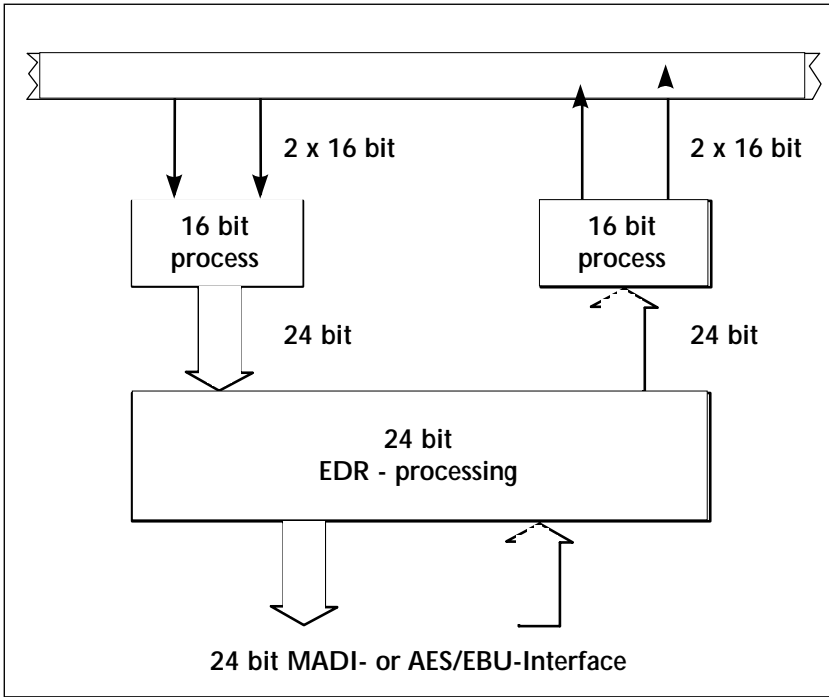


Fig. 2: Schematic representation of the 24 bit recording mode used by the D827 MCH

interface for up to 2 tracks simultaneously. Both formats are already specified for 24 bit transmission.

Compatibility

The data are distributed on the tape in such a way that high-order 16 of the 24 bit input data word are recorded in the DASH format on tracks 1 to 24, whereas the low-order bits are recorded on tracks 25-48. There is a fixed track relationship: track number B is always A+24.

The least significant 8 bits of the 24 bit words are recorded in bit positions 0..7 of the B

tracks, the most significant bits 8..15 of the B tracks are filled with zeros. (Fig. 4).

The advantage of this allocation is that a tape recorded in EDR mode can be played on a standard DASH machine without problems. The «A tracks» represent the recording in 16 bit quality. The «B tracks» are reproduced as «noise» with a level of approximately -50 dB below the maximum digital full scale (dFS).

16 or 24 bits - switchable

An important aspect of the EDR option is that the D827 MCH can also be operated in standard 16 bit mode. With the push of a button (through a simple menu setting) recording and playback of 48 tracks in 16 bit mode, or 24 tracks in 24 bit quality can be selected. Both formats are available without any hardware modifications.

Economy

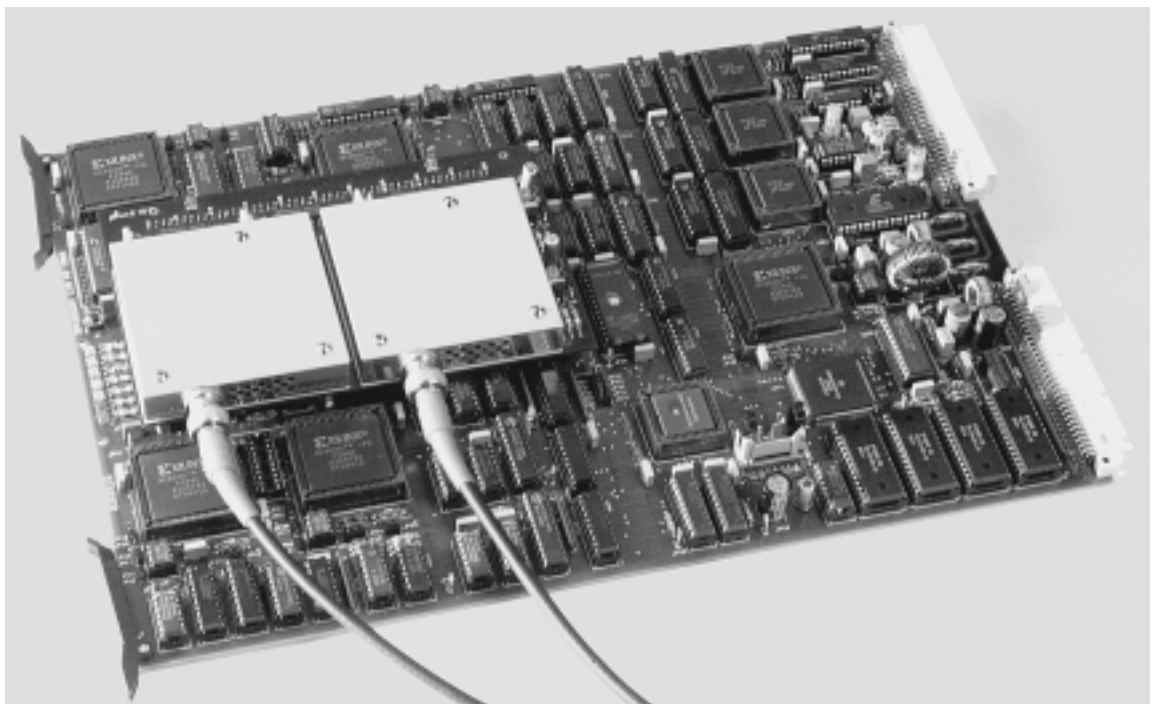
Due to the modular concept of the D827 MCH, the basic configuration can be enhanced with the EDR option. For interconnection with a digital mixing console, no A/D and D/A converters are needed in the tape recorder which results in a highly favorable price/performance ratio.

The EDR option can be retrofitted in a D827 MCH at any time. This work can be performed in the field and takes just a few minutes. An appropriate time may be when the investment in a digital mixing console is planned.

Future included

The D827 MCH has been enhanced with an-

Fig. 3: 24 bit MADI interface board with DSP



other world exclusive option. For the first time, linear 24 bit recording is now possible in multitrack technology. Compatible - and without using data reduction or compression.

Regardless of whether you produce in «conventional» analog technology and have come to appreciate the STUDER converters, or whether you want to record in 24 bit digital - or both, sooner or later - the D827 MCH is your ideal partner.

Bigger, faster, wider - as you see, is no problem with the future-oriented concept of the STUDER D827 MCH. ■

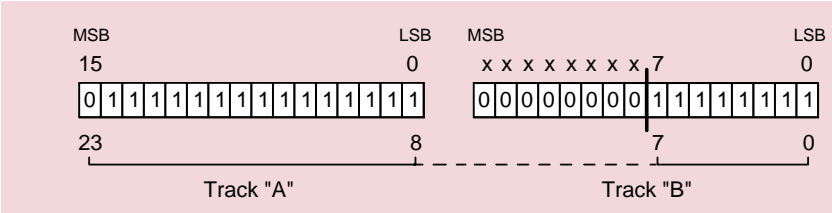


Fig. 4: Data allocation in EDR mode



STUDER D827 MCH- 24 bit EDR, with cover

Broadcast automation

Radio Rumantsch with Numisys II

by Jean-François Raoult and Marcel Bossart

Radio Rumantsch located in Chur, the capital of the Swiss Canton of Grishun, is the smallest unit of the Swiss Broadcasting Company, SRG. It covers the area in which the fourth national language, Romansh, a survival of the old Romance language, is spoken. From SRG Radio Rumantsch received the assignment to expand its daily in-house program from 6 hours to 10 hours. But one of the important constraints of this order was that this expansion had to be achieved with the existing personnel, that is, no new staff could be hired.

Rumantsch were taken into consideration. This process included the technology, the openness of the system, and the reputation of the supplier.

STUDER Numisys was short-listed among the preferred systems and in the evaluation it emerged as the winner.

After the TV symposium 1993 at Montreux the decision was finalized and we were able to book the order for a multi-user Numisys II system.



Chasper Stupan



Pius Paulin



Giacun Caduff



One thing was clear from the beginning: Such a major expansion was impossible to accomplish with a conventional technical infrastructure. For this reason Radio Rumantsch started to look for a broadcast automation system that would greatly help to alleviate the personnel bottleneck.

The short communication paths and the high professionalism of Messrs. *Stupan* (Director of Radio Rumantsch), *Paulin* (Engineering Manager), and *Caduff* (EDP and Networks Manager) resulted in a highly efficient evaluation process in which all factors that were critical for Radio

The automation system was installed and put into service toward the end of 1993. Much preparatory work had been performed by the radio staff who entered hundreds of music titles into the database of Numisys II. Today, 10 months after the Numisys II system was put into service, Director Stupan describes the result as follows: *«It is inconceivable to me how we could work without Numisys».*

Flexible technology

The heart of the Numisys system is installed in a remote equipment room and in a computer room. The servers with a hard disk capacity of

several gigabytes supply the individual workstations with data and audio via a star-shaped Ethernet network. In the machine room there are four CD changers which currently have a capacity of 1200 CDs. The hard disk players, broadcast control computer, controlled crossbars and VCA mixers are also located in this room.

In the main control room a 17" touch-screen monitor for operating the Numisys has been added to the existing Studer infrastructure. Here, the prepared play lists are called up and executed in semi-automatic or automatic mode. From the adjacent studio a news announcer can start the required contributions

and IDs by means of a simple keyboard.

The program contributions are assembled at two workplaces in the office of the music/program planning. The system automatically prepares a broadcasting schedule which can subsequently be modified by the editorial staff.

Four additional workplaces in the production studio are equipped with hard disk recording and editing facilities. Here, the jingles and other contributions are prepared. The material is recorded directly on one of the server hard disks from where it can be retrieved by the control room staff. ■

Right on success

Lexicon announces D827 MCH DASH Multitrack Sales

Closely following the recent appointment of **Lexicon** as the U.S. distributor of STUDER and STUDER EDITECH (Dyaxis) products, Lexicon has announced the sale of STUDER D827 MCH Digital Multitrack tape recorders to several leading U.S. recording studios.

Masterfonics Recording of Nashville, Tennessee, have taken delivery of their first D827, with plans to purchase two additional machines. Two recorders will be installed in their Music Row location. The third will be installed in their new Nashville facility, «The Tracking Room», which is set to open early 1996. Commenting on his purchase, Masterfonics studio owner *Glenn Meadows* said, «Studer may have created another legend like the A800, a machine against which all digital multitracks will be compared». Masterfonics clients include artists such as Little Texas, Willie Nelson, Michael McDonald, Reba McEntire, Vince Gill, Patty Loveless, Diamond Rio, Sting, Mark Knopfler, and Steely Dan.

Right Track Recording Studios of New York City have accepted delivery of two machines for their mid-town facility. The recorders are currently being installed in their studios. Right Track has recorded such legendary artists as The Rolling Stones, Carley Simon, Mariah Carey, Luther Vandross and a host of others. ■



Nashville's Masterfonics Recording Studios recently took delivery of their first Studer D827 MCH Digital Multi-Track Recorder which was immediately used on Willie Nelson's latest album «Healing Hands Of Time». Shown from Left to Right are: *John Guess*, Project Engineer, *Glenn Meadows*, Masterfonics Owner & Mastering Engineer, and *Jimmy Bowen*, President and CEO of Liberty and Patriot Records and overall Producer of the album.

ENVIRONMENT

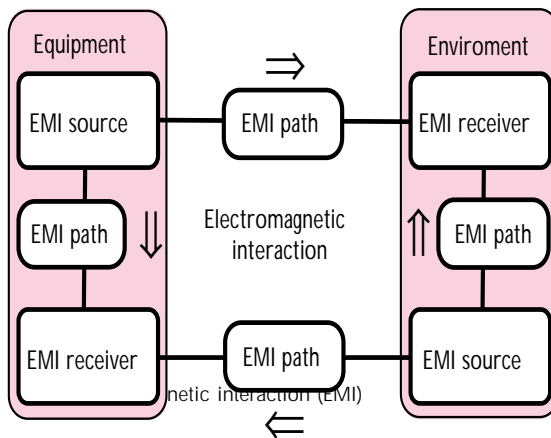
EMC in Professional Audio Systems

by Peter Fiala



Peter Fiala

Electromagnetic compatibility (EMC) is the ability of a system to work in its environment without being influenced by interference and without interfering with the environment. The EMC of a piece of equipment, be it integrated in a system or operating in a system environment, is an important prerequisite for reliable, trouble-free operation.



This problem was already addressed in the twenties (1928, radio interference protection law), but requires greater attention today. The continual miniaturization of technologically sophisticated circuitry considerably reduces the energy required for influencing or destroying an electronic element (1970: transistor 1µWs, 1990 IC 10 nWs), and consequently reduces the previously effective protection gaps due to the higher packing density of components and equipment. The number of electrical and electronic installations in a system is also growing which increases the possibility of mutual electromagnetic interference.

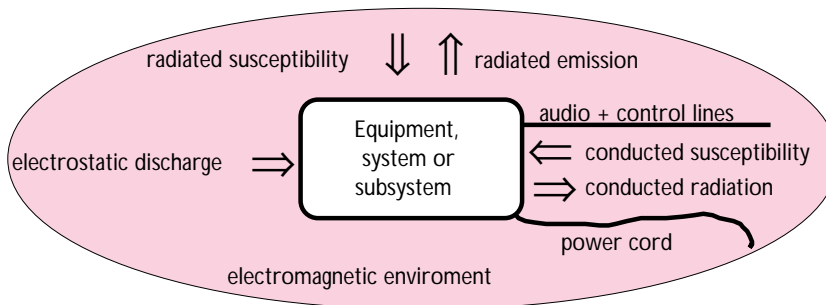


Fig. 2: Disturbance paths

Problem areas

Mutual interference of system, as mentioned in the EMC definition, takes place via so-called *disturbance paths* and allows the interaction of known, *physical phenomena*. Although the qualitative and quantitative assessment of such effects is often difficult, suitable precautions must be taken to ensure that potential sources of interference («offenders») do not exceed a certain maximum interference level and that the absorbing devices («victims») have a certain minimum interference or «noise» immunity. Only in this way can the desired EMC of a system be achieved (Fig. 1).

In this endeavor, also considerable *economic* and *legal aspects* must be taken into consideration. Without statutory regulations or standards concerning the noise immunity these problems could not be solved.

Disturbance paths, disturbance phenomena

The EMC is determined by the effect of physical phenomena whose disturbance paths are either conducted or radiated via fields, or take place in the form of electrical charges (Fig. 2).

Essentially it involves the following phenomena which act via electrical, capacitive or inductive coupling mechanisms or wave or radiation influence (Fig. 3).

Conducted phenomena in the AF range:

- Harmonic, intermediate harmonic components (voltage and current)
- Audio-frequency powerline carrier control
- Voltage fluctuations
- Voltage dips and interruptions
- Imbalances
- Frequency fluctuations
- Induced AF voltages
- DC components in AC supply systems

Radiated phenomena in the AF range:

- Permanent and transient magnetic fields
- Permanent and transient electrical fields

Conducted phenomena in the RF range:

- Surge transients, single or repetitive occurrence
- Transient oscillations
- Induced voltages and currents, permanently

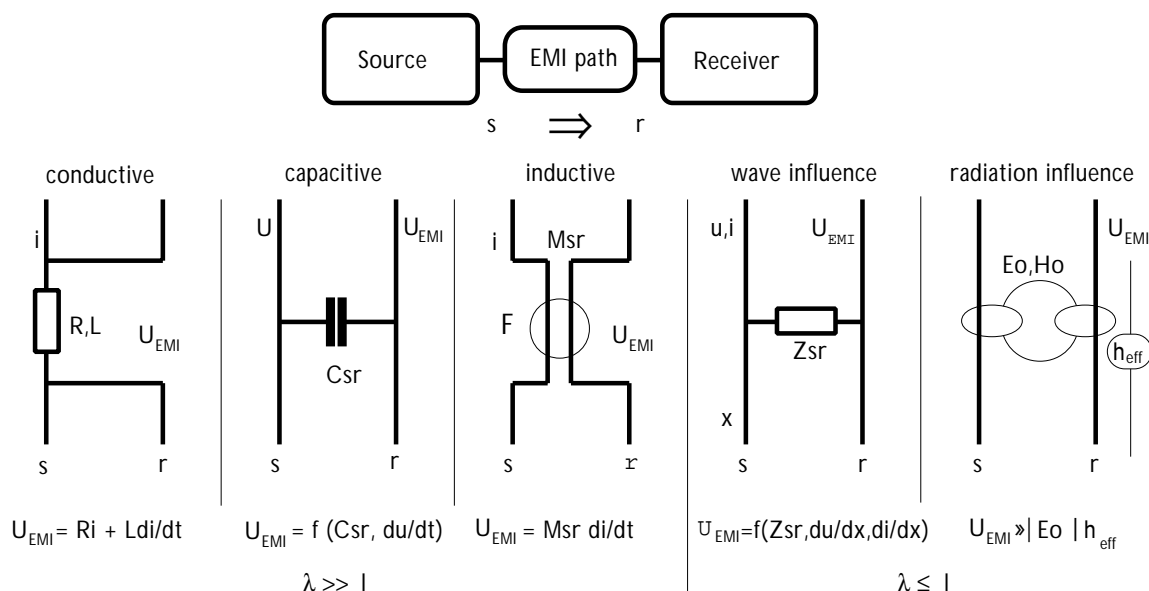


Fig. 3. EMI Coupling mechanism radiated phenomena in the RF range

Radiated phenomena in the RF range:

- Magnetic fields
- Electrical fields
- Permanent or transient electromagnetic fields, single or repetitive occurrence

Discharge phenomena:

- Electrostatic

Standardization

The objective of EMC standardization is to translate the intangible, abstract subject of phenomena and disturbance paths into practical and reproducible measurement and simulation methods, to evaluate today's (contaminated) electromagnetic world, and to define the limits of interference radiation and noise immunity. There is a worldwide trend in harmonizing the national methods (identical test equipment!) whereas the limits of the individual tests still differ considerably. As a result, the national technical standards are incorporated by the legislators into their national laws.

Europe

Although statutory requirements with respect to radio interference suppression have been in existence in some countries for a long time, the most important protection objectives related to EMC have been redefined for the EC market in the directive 89/336/EEC which ensures compatibility of the equipment and system within Europe. But this directive also implies the preparation and harmonization of corresponding technical standards that take into consideration a new assessment of the EMC phenomena. As requested by the EC commission (and the EFTA secretariat), these standards have been developed and promulgated as European

standards (EN) by CEN/CENELEC, ETSI. These standards are to be adopted without change by all CENELEC member states as the legal basis for EMC.

The rather complex set of EMC standards has been subdivided into four types of standards:

- Measuring methods and procedures (basic standards)
- Environment related standards (generic standards)
- Product family standards
- Product standards

With respect to audio, video and audiovisual products for professional applications, a product family standard is going through the consulting procedure. That means that after all the currently valid, environment related standards are the legal basis. These define the limits concerning the radiation of interference signals and the immunity to such signals in two areas:

- Residential, commercial and light industry
- Heavy industry

Proof that the product conforms to the requirements of this standard is mandatory, but it can be supplied in various ways. There is no certification requirement for professional audio systems.

Rest of the world

In the rest of the world the existing EMC radiation standards usually cover only the radiation of interference signals. Often the laws require official conformity certification of the product but for equipment developed in Europe this

hardly presents a technical obstacle.

Customer standards

In addition to national EMC standards there are also customer-specific EMC standards and guidelines (e.g. design specifications of the communications authorities) which may supplement or tighten the national standards. The conformity of the audio systems to such regulations is specified in the purchase agreement and consequently binding under civil law.

Studer EMC standard

The Studer equipment and systems that are currently in production or development satisfy the Studer EMC standard. This standard encompasses all essential statutory EMC requirements of the partner countries and has been supplemented based on the requirements in user standards or in-house experience. The limits

concerning the radiation of interference signals and the immunity to such signals conform to the electromagnetic environment typically found in residences, commerce and light industry where the vast majority of all electromagnetic equipment is installed.

The electromagnetic compatibility with Studer equipment and systems is only ensured if the installation site with its general electrical and electronic infrastructure satisfies the aforementioned EMC requirements. If equipment is installed in a more difficult EMC environments (OB vans, transmitter towers, etc.) than planned in the design, the corresponding modification or installations costs can be very high. ■

Patent News

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Method of applying negative impedance in multi-system speakers

The source impedance of the amplifier has a significant influence on the behavior of a dynamic speaker. The speaker is a mechanical high-pass filter with a cutoff frequency that corresponds to the resonance frequency of the built-in speaker. The quality (Q) of this high-pass filter is influenced by the source impedance. At least since 1953 circuit arrangements have been known that change the source impedance in such a way that the desired frequency response is achieved.

If the source impedance is changed to such an extent that a Q of zero is attained, the high-pass disappears completely and the resulting frequency response corresponds to the one of a differentiating element. To compensate the effect of this differentiating element, an integrating element is connected in series. The frequency response is now flat. Through this arrangement the delay of the high-pass disappears. This is essential for achieving phase-linear crossover networks according to Lipschitz/Vanderkooy. Our patent covers active multipath speaker systems that exploit the principle of negative impedance also outside the bass frequency range. This patent of Paul Zwicky and Roger Schultheiss has been registered with the US patent office on July 7, 1992, under the number 5,129,005.

Paul Zwicky

Method for utilizing the negative impedance in open speaker housings

All speakers have a bass cut-off frequency. In closed housing the air cushion acts as a restoring force on the diaphragm. This increases the natural resonance of the speaker. As a consequence also the bass cut-off frequency increases. By introducing a housing opening with an exactly calculated air volume, a system is created that is popularly called bass reflex. The bass cut-off frequency becomes lower and the acoustic efficiency higher. Unfortunately, bass reflex systems have inadmissibly high group delays. This means that the lower frequencies are radiated with a delay. The consequence is inaccurate bass reproduction. This fault can be corrected with the negative impedance. In the past this has been possible only in connection with closed housings. This new patent describes how group delay errors can be avoided without sacrificing the acoustical support of a Helmholtz resonator. In this way it is possible to build also small speaker housing capable of strong bass reproduction without adverse effect on the transient response time. This system is used in the speaker A623.

This patent by Roger Schultheiss and Paul Zwicky has been registered on January 19, 1993, with the US Patent Office under the number 5,181,251.

Paul Zwicky