ENGINEERING GUIDELINES
THE EBU/AES DIGITAL AUDIO INTERFACE

EBU
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John Emmett 1995

european broadcasting union

C O N T E N T S

Editor's introduction

The EBU was formed some forty four years ago now, and most of us in Europe must have grown up, knowingly or not, to a background of television and radio programmes made and exchanged between members of the Union. Last year the EBU was joined by the countries of the old OIRT. Some have new names to us, and some have names as old as history itself. All of us, however, share a common love of looking and listening beyond our national boundaries, and improving the technical quality of those glimpses.

Fig. 1.1. European Network Map

One enormous contribution to this process has been the advent of digital audio, which allows sounds to traverse continents or the internal intricacies of our studios with equal and transparent ease. The essentials for programme exchange using digital audio reduce to one of only two things:

- a standard connection interface,
- or a common recording medium for physical interchange.

EBU working groups meet regularly to discuss the huge amount of work that surrounds these simple subjects, and a major part of this work involves liaison with industrial and academic bodies world-wide.

The close ties between the EBU and the AES and the IEC are two essential links to the outside world, but in the case of this Guide, I see the internal needs of broadcasters as differing in three distinct respects from those of the other organisations:

1. Recognition of the frequently changing "dynamic" nature of the digital audio installations used in broadcasting. This places special concern on interconnections between Members, analogue alignment levels, and common use of auxiliary data etc.

2. Recognition of the need inside broadcasting organisations for confidence testing of installations and audio quality control, in addition to acceptance tests of new equipment. Confidence tests must ideally use the simplest test equipment and shortest possible routines.

3. Recognition of the ability of, and need for, broadcasters to develop their own specific items of equipment incorporating the digital audio interface. Circuit design principles are therefore important, and so form a large part of this guide.

On this basis then, I have edited together contributions from EBU members, extracts from the draft AES guidelines, along with a little of my own linking material, which I hope you will excuse. Whilst on a personal note, I would like to thank all those who have contributed to this guide, especially those contributors who have not been involved as members of the EBU working groups; Bill Foster and Francis Rumsey of the AES, all of the contributors from the BBC (UK), IRT (Germany), DR (Denmark) and TDF (France), and finally, Thames Television and the ITV Association (UK) who made it possible for me to take on the whole task in the first place.

John Emmett

Chapter 1

A BRIEF HISTORY OF THE AES/EBU INTERFACE

In the late 1970s and early 80s, digital audio recording was at the experimental or prototype stage, and the hardware manufacturers began to develop digital interfaces to interconnect their various pieces of equipment. At this stage, this presented no major problem because the amount of digital audio equipment in use was small and almost all digital audio systems were installed in self contained studios and used in isolation.

1.1. The SDIF-2 interface

By far the most widely used interface was SDIF-2 from Sony. This interface used three coaxial cables, carrying the left channel, the right channel and a word clock. A number of other manufacturers, reacting to the increasing use of the Sony 1610 (and later 1630) processors for Compact Disc mastering, also adopted the SDIF-2 interface.

The SDIF-2 interface was very reliable over short distances but it became increasingly evident to broadcasters and other users of large audio facilities that an interface format was needed which would:

- work over a single cable,
- work over longer cable lengths,
- allow additional information to be carried.

1.2. The AES working group

In the early 1980's, the Audio Engineering Society formed a Working Group who were charged with the task of designing such an interface. The Group comprised development engineers from all the leading digital audio equipment manufacturers, and representatives from national broadcasting organisations and major recording facilities.

The criteria set for the new interface were:-

1. It should use a single cable, of type which was easy to obtain together with a readily available connector.

2. It should use serial transmission, to allow longer cable runs with low loss and minimal interference (RFI).

3. It should carry up to 24 bits of audio data.

4. It should be able to carry information about the audio signals, such as sampling frequency, emphasis, etc., as well as additional data, such as timecode.

5. The cost of transmission and receiving circuits should not add significantly to the cost of equipment.

The Working Group realised that unless a standard was endorsed by an independent body, a plethora of interface formats were likely to appear. They therefore put an enormous amount of effort into devising an interface that would satisfy all the above criteria, as well other requirements which came to light as the work progressed.

1.3. The first AES/EBU specifications

In October 1984, at the AES Convention in New York, the Working Group presented the Draft Standard, designated AES3. It was greeted with enthusiasm by both manufacturers and users, many of the latter stating that they would specify the interface on all future equipment orders.

This specification, AES3-1985, was put forward to ANSI, the American national standards authority, for ratification and also submitted to both the EBU in Europe and the EIAJ in Japan for their approval. Both bodies ratified the standard under their own nomenclature, although small modifications were made to both the text and the implementation. The most significant being the mandatory use of a transformer in the transmitter and receiver in the EBU specification. Despite these small discrepancies the interface is now commonly referred to as the "AES/EBU" interface.

Users of the AES/EBU interface have experienced relatively few problems and the interface is now widely adopted for professional audio equipment and installations. The only major teething problem was caused by the use of consumer integrated circuits, designed for the closely similar S/PDIF, in so-called professional equipment. This initially caused numerous interconnection problems, which are now well understood.

1.4. The second AES/EBU specifications

A small number of refinements, suggested by users of the interface, were recently addressed by the EBU and the AES (1992) when a second edition with a number of revisions and improvements was issued.

1.5. The IEC Publication

Meanwhile, the IEC followed quite a different line of development. At the 1980 meeting of IEC Technical Committee 29, a working group was formed to establish a consumer interface for the then new Compact Disc equipment. At the same time it was asked to ratify the AES and EBU work on the professional interface. The relationship between those interested in the consumer interface and those interested in the professional specifications was not always easy. Nevertheless, the IEC group has always seen the advantages of a basically similar interface structure for professional and domestic versions. The resulting IEC Publication 958 of 1986 contained closely similar consumer and professional interfaces. This ultimately produces greater economies throughout the whole audio industry. In fact, only the major difference between the two applications is in the areas of the ancillary data and the electrical structure. The two versions reflected the same division that existed in the analogue world: a professional version using balanced signals and a consumer version using unbalanced signals.

1.6. The future

As mentioned above, in 1990 a group was formed within the EBU to review the interface. As more and more EBU Members are installing digital audio equipment in production areas, this group has became an semi-permanent advisory body and the members maintain close contact with each other. It is this group which has pooled their experience in the present document.

At various points in this document we will mention areas where there have been proposals or agreement on developments. It is expected that these will be included in future editions of the specification but until this work is carried out, these developments will be recorded in these guidelines.

Chapter 2

STANDARDS AND RECOMMENDATIONS

As explained in Chapter 1, many different bodies have been involved in the development of the AES/EBU interface. A number of different documents now exist and these are listed below:

2.1. EBU documents on the digital audio interface

The EBU publications on and about the AES/EBU interface:

- EBU Tech Doc 3250: Specification of the Digital Audio Interface (2nd Edition 1992)
- EBU Tech Doc 3250, Supplement 1: "Format for User Data Channel".
- EBU Standard N9-1991: Digital Audio Interface for professional production equipment.
- EBU Standard N9, Supplement 1994: Modification to the Channel Status bits in the AES/EBU digital audio interface
- EBU Recommendation R64-1992: R-DAT tapes for programme interchange.
- EBU Recommendation R68-1992: Alignment level in digital audio production equipment and in digital audio recorders.
- These Engineering Guidelines.

2.2. AES documents on the digital audio interface

The Audio Engineering Society, AES, is an open professional association of people in the audio industry. Although based in America, it has many members world-wide.

The of following documents have been issued by the AES and adopted as American National Standards, ANSI:

- AES3-1992 (ANSI S4.40-1992): AES Recommended Practice for Digital Audio Engineering:- Serial Transmission Format for Two Channel Linearly Represented Digital Audio Data.
- AES5-1984 (ANSI S4.28-1984): AES Recommended Practice For Professional Digital Audio Applications Employing Pulse-Code Modulation - Preferred Sampling Frequency.
- AES10-1991 (ANSI S4.43-1991): AES Recommended Practice for Digital Audio Engineering - Serial Multichannel Audio Digital Interface (MADI).
- AES11-1991 (ANSI S4.44-1991): AES Recommended Practice for Digital Audio Engineering - Synchronisation of Digital Audio Equipment in Studio Operations.
- AES17-1991 (ANSI S4.51-1991): AES Standard Method for Digital Audio Engineering - Measurement of Digital Audio Equipment.
- AES18-1992 (ANSI S4.52-1992): AES Recommended Practice for Digital Audio Engineering - Format for the User Data Channel of the AES Digital Audio Interface.

The AES are also producing an Engineering Guideline document for AES 3. This is being assembled in parallel and in close association with this EBU text. The purpose of the AES document, and indeed this one, can be clarified by a quotation from the introduction by Steve Lyman CBC:

The information presented in the Guideline is not part of the AES3-1992 specification. It is intended to help interpret the specification, and as an aid in understanding and using the digital audio interface. The examples provided are not intended to be restrictive, but to further clarify various points. Hopefully, the Guideline will further encourage the design of mutually compatible interfaces, and consistent operational practices.

2.3. IEC Publications

The International Electrotechnical Commission, IEC, is the standards authority set up by international agreement which covers the digital audio interface. Its primary contributions come from the national standards authorities in its member countries. Its document which covers the interface is:

- IEC Publication 958: 1989 Digital Audio Interface

This text may also appear under different numbers when it is issued by national standards authorities in any particular country.

In recent years, the IEC has restructured itself to make it easier to accept input directly from any expert body, not just national standards bodies. Their aim is to reduce the costs and time scales involved in work in highly specialised fields. In practice the IEC has always accepted the inputs from the EBU and AES but the time scales of redrafting, etc., and the highly structured language of an inter-national standard has made it more difficult for the IEC to react quickly to developments. The IEC expect to redraft Publication 958 soon, to reflect the developments in the EBU and AES documents on the professional version. The Channel Status structure of the consumer version will also be revised and extended, ready for a new generation of digital audio home equipment to be launched onto the market.

2.4. ITU, CCIR and CCITT documents

The ITU, International Telegraph Union, is a United Nations body which is responsible for international broadcasting and telecommunications regulation. Until recently it worked through the CCIR (The International Radio Consultation Committee) and its associated body, the CCITT (The International Telegraph and Telephone Consultative Committee). In 1988 the CCIR adopted the AES/EBU interface specification as:

- Recommendation 647: A digital audio interface for broadcasting studios.

From 1993 the ITU has been restructured. The tasks of the CCIR have passing to the new Radiocommunications Sector. The specification was revised in line with the latest edition of the EBU and AES documents is now known as:

- ITU-R Recommendation BS 647: A digital audio interface for broadcasting studios.

2.5. The standardisation process

In March 1916, Henry D Hubbard, Secretary of the United States National Bureau of Standards, made the following comments in his Keynote Address to the first meeting of the then Society of Motion Picture Engineers:

"Where the best is not scientifically known and where inter-changeability or large scale production are not controlling factors, then performance standards serve".

"The user is the final dictator in standardisation. His satisfaction is a practical test of quality".

The same is still true today but among the major factors which have changed since 1916, we could included:

- the ever expanding size and effects of the world market,

- the changing economic and "managerial" scene within countries as well as between nations,
- the effects of modern communications with its advantages and values, as well as its problems.

Among the factors which remain unchanged are:

- the benefits of co-operation, negotiation and exchange of information,
- the need to make the best technology available on a timely and world-wide basis while not preventing the development of improved quality,
- the recognition that the user, or at least those who control the purchasing funds, are really the final judges.

It could be, therefore, that in the future, the criteria for standard-isation will remain as valid as they were in 1916, but the standardisation process may need to be radically adapted to change.

The AES/EBU Digital Audio Interface is perhaps an unusually wide-reaching standard in that it is accepted for applications in fields as varied as the computer industry, where standardisation has always been seen as a restriction on innovation, and the movie industry, where a unique set of audio standards has been adhered to since the advent of sound films.

Chapter 3

THE PHYSICAL (ELECTRICAL) LAYER

Broadcasting studios can be a very hostile environment for electrical signals. This is especially true for an interface signal with no error correction facilities. Successful use, therefore, depends on exploiting the limited error detection capability in the interface and on designing installations in which few errors occur. Parts 2 to 7 of this chapter cover the contribution of the equipment designer to this goal but this first section is almost totally about the work of the systems engineer and installer.

3.1. Transmission lines, cables and connectors

Whether or not a circuit is treated as a transmission line depends on the frequency of the signal and the length of the circuit. With modern equipment, the adverse effects of not matching impedances in cables are mainly caused by reflections which interfere with the wanted signal. Transfer of maximum power is not in itself very important. Thus although analogue audio distribution can suffer from transmission line effects over distances in excess of 1,500 metres, they are rarely met with in studio practice. As a consequence audio cable specifications pay little attention to parameters such as the characteristic impedance or the attenuation of signals above 1 MHz. For data signals like the AES/EBU digital audio interface, the situation is very different. Data signals start to suffer transmission line effects after only ten metres or so. This due to the higher frequencies and shorter periods involved. The system designer therefore has to modify his installation practice. This has to become in some respects quite unlike traditional analogue audio practice and much closer to traditional video practice. Fortunately, for a simple binary signal, there is no need to obey the transmission line rules anything like as strictly as for an analogue video signal.

In fact the original 1983 specification allowed up to a 2:1 mis-match of the line characteristics and this gave a certain flexibility to "loop through" receivers, or use multiple links radiating from transmitters. This concept was based on the theory that lossy PVC analogue audio cables would be used and it was predicted that:

- reflections in short cables were unlikely to interfere with the edges of the signal, due to the short delays involved,

- reflections in longer cables were likely to be attenuated so much that they would not significantly interfere with the amplitude and shape of the signal at a receiver.

In practice, however it was soon found that problems occurred with an open ended spur which happened to have an effective length of half a wavelength at the frequency of the "one" symbol. This length is also a quarter wavelength for the frequency of the "zero" symbol. This condition causes the maximum trouble for the signal characteristics on any connection in parallel with the spur.

It has been found that connectors are of little consequence since their electrical length is so short that any reflections due to mis-match are immediately cancelled out. Surprisingly, some "noisy" analogue connectors, such as brass ¼" jack plugs, work extremely well with digital interface signals. This is because all digital signals, even silence, are still represented by several volts of data signal and, by analogue standards, digital signals are very tolerant of crosstalk.

3.2. Guidelines for installation

The following practical guidelines have been produced for balanced circuits intended for the AES/EBU Digital Audio interface. They are based on experience gained from two installations by the BBC in London and from installations by CBC, Canada

Inter Area Cabling

- Cables should have a characteristic impedance of about 110 Ω (80-150 Ω is acceptable).

- Multicore twisted pair cables with a overall screen are best for installations where runs do not exceed 150 m. (Overall double screens give better EMC protection.)

- Multicore cables with individually screened pairs have higher capacity to the screens and hence greater loss but are satisfactory for shorter runs. Cables intended for data have some advantages even for short runs, and they also work quite well with analogue signals. This approach could therefore be considered if new cabling is needed.

- For cable runs greater than 150m, special cables together with re-clocking receiving devices may be needed.

- When using multicore cables it is good practice to keep signals travelling in one direction only within each cable. This will minimise crosstalk from the high level, fast rise-time signals at the transmitters to the possibly low level signals at the receivers.

- All circuits should be correctly terminated in 110 ohms.

- Avoid changes of cable type along a particular circuit. Changes in impedance cause reflections which can generate inter-symbol interference, (15-20 m. seems to be the critical length).

<u>Jumper Frames</u>

· IDC, insulation displacement connectors, blocks and normal twisted-pair jumpers can be used.
· Keep all wiring "one to one" with no spurs or multiple connections.
· Beware of open circuit lengths.

<u>Jackfields (see fig 3.1. below)</u>

· Use with caution.
· Do not "listen" across a digital line using a low impedance device,
· Do not connect a long transmission line.
· provide either:

  · monitoring jacks from a separate Distribution Amp output,
  · or pairs of jacks with "inners" connected so that inserting a jack breaks unterminated jack fields, as well.

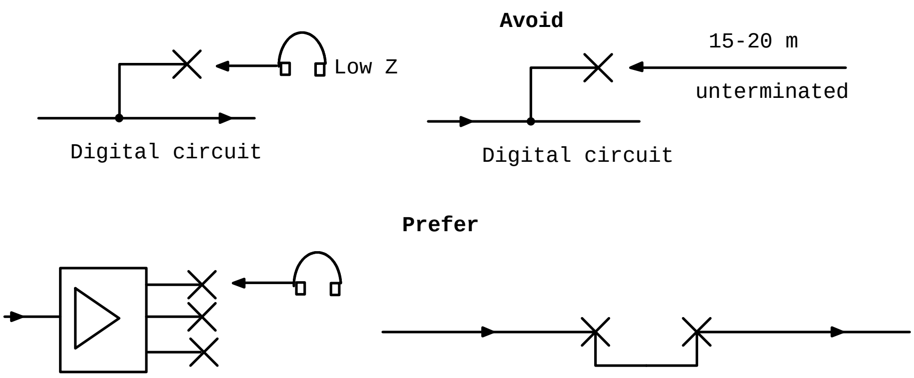


Figure 3.1 Good and bad practice on jack-fields carrying AES/EBU interface signals

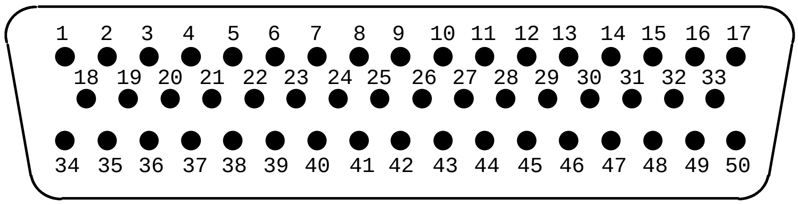
3.3. Coaxial cables and alternative connectors

The interface was originally designed to use existing audio cables but there has been much interest in some quarters in an alternative approach using coaxial cables and BNC connectors. This means that the electrical signals have to be converted from the normal AES/EBU signals to 75 Ω unbalanced signals video level signals, about 1 V. This can be done by using transformer adapters or, preferably, specially designed transmitters and receivers. The transformer adapters must be placed as close as possible to the AES/EBU sending or receiving devices, but even so the losses should be considered.

There is no theoretical or practical reason why this approach should not work satisfactorily. It does enables normal video circuits and components to be used but may not be an economic option for larger installations if new coaxial cable have to be provided. The advantage of using the transformers adapters is that it allows digital audio connections to made from existing equipment using existing equalised video tie-lines. Beware, however, that the interface signal is not a video signal. It may not pass through video equipment, such as vision mixers or switchers that clamp on black level, or equipment that expects to use the video sync pulse as a timing reference

3.4. Multiway connectors

For some equipment, such as routers, a large number of interface circuits need to be connected and there may be no space for an array of XLR connectors. The AES propose that a 50 way "D" connector can be used to carry up to 16 interface circuits. Ribbon cable may be used for short interconnections to nearby equipment. The AES propose the following connections should be used (See fig. 3.2.)



50 pin subminiture D connector: male plug seen from the front face

	Connector pin number			Ribbon cable wire number		
Interface Circuit	**Signal +**	**Signal -**	**Ground**	**Signal +**	**Signal -**	**Ground**
1	18	2	34	3	4	2
2	35	19	3	5	6	7
3	20	4	36	9	10	8
4	37	21	5	11	12	13
5	22	6	38	15	16	14
6	39	23	7	17	18	19
7	24	8	40	21	22	20
8	41	25	9	23	24	25
9	26	10	42	27	28	26
10	43	27	11	29	30	31
11	28	12	44	33	34	32
12	45	29	13	35	36	37
13	30	14	48	38	39	40
14	47	31	15	41	42	43
15	32	16	48	45	46	44
16	49	33	17	47	48	49
Chassis ground	1, 50			1, 50		

Fig. 3.2. Multiway connector used for up to 16 AES/EBU interface circuits

Notes:

- The connector housing shells and back shells should be metal and equipped with grounding indents.
- Male plugs should be used for input connections and female sockets should be used for output connections. In exceptional circumstances a connector may carry both input and output circuits.
- The signal polarity is defined so that the X, Y and Z preambles all start with positive going edges as seen on an oscilloscope whose non-inverting input is connected to “signal +” and inverting input to “signal -”. See section 2.4 and figure 3 and 4 of EBU Tech 3250 (AES3-1992)
- Signal polarity is only labelled for convenience because either relative polarity of the signal is allowed by the specification and can be accepted by receivers.

3.5. Equalisation and transformers

Two things happen to an interface signal as it passes along a cable:

- it is attenuated in a frequency selective fashion,
- the higher frequencies are "dispersed" or delayed relative to lower frequencies.

The attenuation can be predicted for uniform cable but in practice it may not have a smooth or predictable frequency response due to reflections at the cable and connector boundaries. Nevertheless, attenuation does not theoretically limit the range at which reception is possible because the losses can be corrected by equalisation. In practice accurate equalisation of each individual cable is expensive and complicated. Many receivers have a built-in fixed equaliser which partially corrects the cable response for long lengths of cable, at the expense of overcorrecting short lengths. A simple form of equalisation characteristic, given Tech. Doc. 3250, is shown below.

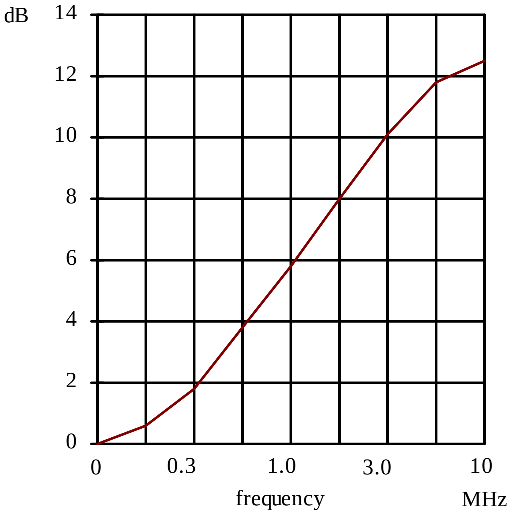


Figure 3.3 Suggested equalisation characteristic from EBU Tech 3250

This characteristic can be realised in a symmetrical constant impedance manner by the circuits suggested by Neville Thiele (IRT):-

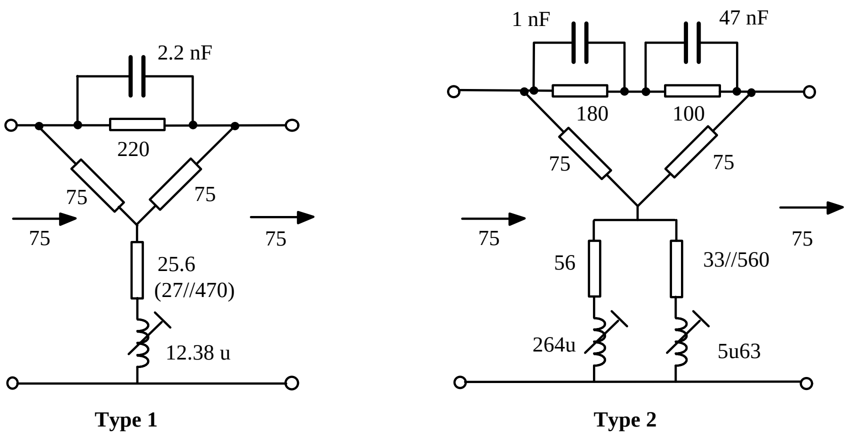


Figure 3.4. Suggested correction circuits for loss on longer cables

Note: these values are given for unbalanced 75 Ω systems but they can be easily adapted to a 110 Ω balanced configuration.

A different approach to equalisation would be to use a comb-filter made from an analogue delay line in the feedback loop of the input amplifier. The first half cycle of the comb response would be a near ideal equalisation characteristic.

In practice, however, trying to extend the range of a circuit too far by using higher levels of fixed equalisation is not recommended. The reason is simply that the available RS422 receiver circuits do not approach the theoretical performance in terms of slicing the incoming signal. High levels of equalisation will produce overshoots on shorter cable lengths which will cause malfunctions in these circuits. It is of course possible to use variable or switchable equalisation, set up for each individual circuit. However, in practice, this will greatly increase the cost and complexity of an installation.

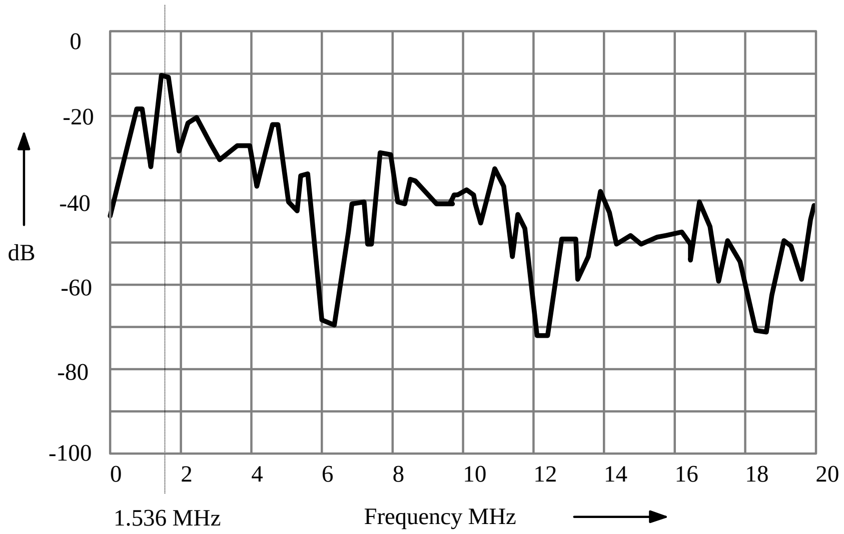
Passive equalisation will only improve the relative opening of the eye pattern at the receiver. Overall, there could be great gains in performance margins if more attention was given to the design of the receiver. This applies not just to improvements to the minimum eye performance and the acceptable dynamic range in the wanted band of frequencies, but also to rejection of out of band interference signals, both common mode and balanced. In Section 3.6 this is explored further. In section 6.2.1 there are some suggestions for testing circuits.

It is essential to use transformers on the inputs and outputs of broadcast equipment. The transformers should be tested for their current balance and common mode rejection at high frequencies. Some unsuitable types of transformer have been used in the past which have been found to badly affect the error performance when used in areas sensitive to EMC. These days, just about any area used by a broadcaster is sensitive to EMC!

Dispersion is a phenomena which is largely a factor of the cable dielectric. It provides a theoretical limit to the length of cable which can be used between repeaters. This limit cannot be improved by simple equalisation procedures. Nevertheless, tests have shown that for cables with polyethylene dielectric, dispersion is unlikely to be a limiting factor for lengths below 4,000 m. Therefore dispersion is unlikely to produce any practical limitations within studio premises.

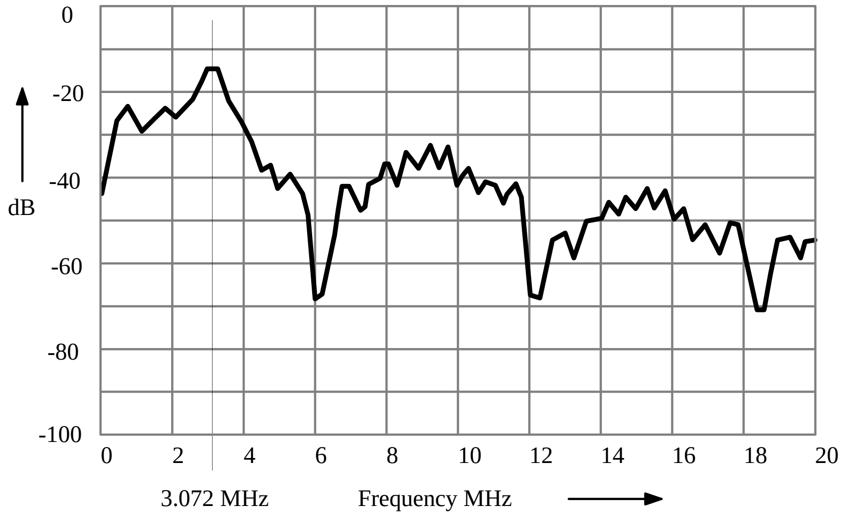
3.6. Clock recovery and jitter

Spectrum analyser traces for interface signals are shown in figures 3.5a & b below. As can be seen, interface signals contain high levels of clock signal at the "ones" and "zeros" frequencies, 1.536 and 3.072 MHz. These signals can be extracted by the receiver and used as a basis for decoding the bi-phase channel code.



Audio channel bits mostly = 0

Figure 3.5a: Interface signal in frequency domain



Audio channel bits mostly = 1

Figure 3.5b: Interface signal in frequency domain

Circuit techniques for clock recovery and decoding can vary greatly. The overall requirements are simply that a receiver should be able to decode correctly inputs at the widest possible range of sample frequencies with the shortest possible lock up time. Without this ability, any input disturbance which upsets the clock recovery will cause severe error extension. A block diagram of the electrical level of a receiver is given in figure 3.6. below.

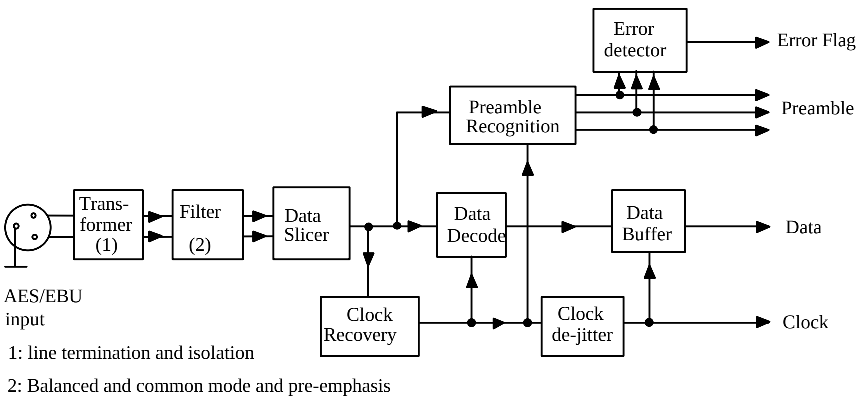


Figure 3.6: Block diagram of the electrical layer of a receiver

Ideally, the data slicer circuit should have no hysteresis. In practical circuits, used in the computer industry, hysteresis is often employed to reduce noise toggling. This technique, however, compromises other requirements of the receiver at low signal levels such as the need for low jitter gain. The same benefit could be obtained by filtering.

Jitter, which is uncertainty in the transition times of bit cells, can be passed on in a re-transmitted bit stream or can be introduced at the receiver due to shifts in the slicing levels. These shifts can be due to the performance of the receiver at both in-band and out of band frequencies. Shifts can be also be induced by the data content of the signal carried, as well as by external factors such as hum or noise on the line. These problems cannot be reduced by sending data with faster rise times. The energy of the fast rise times will either be rapidly attenuated in the interconnection cables, or worse still radiated as interference. Even if the fast rise time energy should reach the receiver, any well designed receiver will filter it out because all the wanted information is contained in a bandwidth below 6 MHz, as can be seen in the spectra in Figures 3.5.a & b.

Although the input stage of a receiver should be agile enough to decode signals in the presence of jitter, the agility should not be extend beyond the input stage. The clock jitter on the input should not be passed on. A "flywheel" clock buffer should be employed to reduce jitter on the output. An extreme case of this need occurs at A-D or D-A conversion stages where the clock used for the conversion should be extremely well isolated from any jitter present on any input, including the reference input. It would be quite acceptable for these stable sampling clock generators to take hundreds of frames to lock up correctly. in contrast, clock circuits used for decoding the channel code should, in general, lock up within a fraction of a sub-frame. Further work is in progress within the AES on an improved definition and specification of jitter, based on the CCITT specifications.

In summary, the design of the electrical layer of every professional interface receiver should aim at in the very minimum of error extension. If the signal is passed on by the equipment, there should be some ability to repair the bit stream so that downstream equipment can decode the electrical layer correctly. In terms of jitter performance, the widest possible window should be available at the input but, if the signal is passed on, the jitter should be attenuated at the output.

At the electrical layer, the analogue part of the receiver input is crucially important, so clean digital test signals are of little value as a guide to the practical performance of any piece of equipment (See Chapter 6). Tests of the electrical layer will need special test signals. One suggested technique is to add wideband analogue noise, in a balanced mode, to the interface signal at the input terminals of the receiver. This will be a confidence check on the margin of error of the installed link. It could also be a basis for quantitative measurements of the error extension and jitter attenuation when observed on a downstream output. Extensions of this approach would be to include out of band frequencies in the test signal, and to use common mode coupling. As well as testing the requirements of the specification, EBU Tech. 3250, such techniques could be extended to measurement of EMC susceptibility.

3.5. Preamble recognition

Each sub-frame of the interface signal contains a preamble. The preamble have at least one bit cell which is three clock periods long and thus does not obey the bi-phase mark rules (see Chapter 2,4 of EBU Tech 3250). As a result the preambles, and hence the sub-frames and the sample rate clock, are easy to detect at the electrical level once the data clock rate has been established. However, the preamble detection process has been known to fail before actual data clock is lost. Divided data clock is therefore a more reliable source for the sample rate clock than the preamble flags. Continuously incorrect preamble sequences, or inverted preamble patterns, are signs of incorrect design or faulty equipment, nevertheless an occasional corrupt of missing preamble pattern is a sure sign that the link is close to failure and should be flagged to the operators as a warning. Likewise, an incorrect count between Z preambles (Channel Status block start) is a direct indicator of an error in the channel status block. It is easy to give a warning based on this which is independent of the implementation of the CRCC in channel status. Beware, however, that some processes, such as editing or sample rate conversion, can produce this incorrect Channel Status block lengths without the audio data content being in error. Nevertheless, it would be useful to provide the operators with a warning whenever an incorrect block count is detected.

3.6. Regeneration, Delay and the Reference Signal

3.6.1. Frequency synchronisation

Synchronisation of interface signals has been thoroughly covered by the AES in their document AES11. The EBU fully supports this valuable work and will not publish its own duplicate documents on the subject. Broadcast equipment should obviously meet the professional tolerances given in AES11.

If any form of synchronous mixing or switching is required, all digital audio signals should be locked to the same fixed frequency reference. (See also 3.6.3. Framing.)

If the signals are not all locked, then "sample slips" will occur as one signal runs through a timing point with respect to another. This will happen if non-synchronous signals are being mixed or where one signal is being used as a reference for the processing device. These sample slips may be audible as clicks. The severity of the clicks from sample slipping will depend on the programme material. High frequency tone is a very demanding audible test. A sure sign that this slip process occurring is a wrong block count indication, mentioned in section 3.5. above,.

Any audible clicks in a programme should be investigated. They usually indicate that some piece of equipment is operating very close to failure in some parameter.

3.6.2. Timing (synchronisation) reference signals

Any normal AES/EBU signal which is locked to a stable reference can be used as a timing reference signal. In television, an audio reference signal should be locked to the video reference signal if one is present. However, for best performance, both audio and video reference signals should be locked to a common high frequency reference. If VTR timecode is implemented, there are advantages in arranging the video to audio timing so that the Z preamble of the audio interface is aligned with video frame sync and time code 00.00.00.00. at midnight.

Although video signals are usually derived from very stable master oscillators, it has long been realised by broadcasters using sound-in-sync on contribution circuits that normal video signals are not in fact very good as high quality reference sources. There are two reasons for this:

- The complex relationship between the digital audio frequency (48 kHz) and video frequencies (50 Hz, 15.625 kHz) result in the need to lock at low common denominator frequencies. This can compromise the jitter performance or lock-in range of the system.

- The rise time and general stability of a standard analogue video signal to CCIR specifications can result in a audio system stability that does not meet the requirements given in AES11.

For large installations it is important to maintain a very low jitter in the reference signal because of possible jitter amplification throughout the system. Good low jitter practice involves the minimum number of reference regeneration stages and using a low bit modulation of the reference signal.

3.6.3. Framing (phasing of the frames)

As well as the synchronisation of the sample clock, the framing of an interface signal is also important. Framing is a familiar concept to video engineers but its importance is not so obvious to someone used to analogue audio. The timing reference of a frame is the first edge of the X or Z preamble. AES11 specifies that receivers should maintain an acceptable performance with signals within a range of at least 25% of the frame period with respect to the timing point of the reference signals, (± 5 µs for a 48 kHz signal). Output signals from equipment should be within with 5% of the reference.(± 1 µs at 48 kHz)

Delays caused by the length of cables within a broadcast centre are unlikely to cause signals to fall outside this range, (it takes 1 km of cable to give a 4 µs delay) but poor equalisation will lead to the problems dealt with in previous sections which have an adverse effect on framing.

3.6.4. Reframers

A reframer is designed to give clean interface output signal whatever happens at its input. A reframer will:
repair a momentary discontinuity in the input data stream, caused, for example, by switching between digital audio signals, by inserting interpolated samples.
re-frame an incorrectly framed (timed) signal,
output a signal which represents silence if no signal is present at its input.

Any delay in a reframer will be in multiples of whole frames.

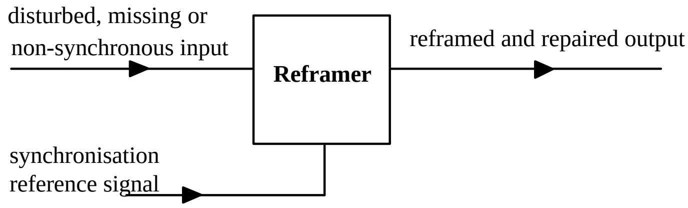


Figure 3.7: Reframer

3.6.5. Regenerators

A regenerator is a self-referenced receiver which is designed to attenuate jitter before re-transmitting the re-clocked signal. The delay in a regenerator should be as short as possible but there is no fixed delay because there is no external reference. The output may therefore have any framing relative to a station reference. Regeneration could compromise the framing demands within a studio area, and if it is used, a corresponding delay may be needed to the external reference signal to maintain the correct framing.

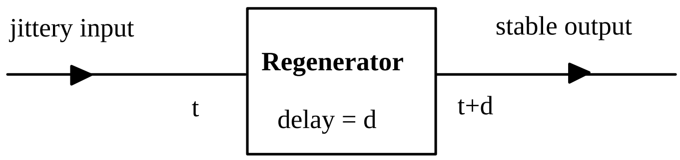


Figure 3.8: Regenerator

3.7. Electromagnetic compatibility, EMC

3.7.1. Background to EMC Regulations

The legal and political aspects of the recent EMC(Electromagnetic Compatibility) Directive for the European Union are outside the scope of these guidelines but some details are covered in Appendix 1. Equipment designers and installers need to know that almost all electrical or electronic products made or sold in the Europe Union must nowadays meet certain EMC requirements. Equipment must:

- not cause excessive electromagnetic interference
- not be unduly affected by electromagnetic interference;
- in some cases, such as RF transmitters, be subject to type-examination by an approved body;
- carry a "CE" mark.

All products should be testing against an approved EMC standard. Often a relevant European Standards is not available for a particular product. In this case the so called Generic Standards for equipment manufactured for domestic, commercial or light industrial environments should be applied. These cover:
- Radiated emissions from the enclosure and/or connecting cables,
- Conducted emissions from the connecting cables, in particular through the power supply.

The audio electronics industry has some difficulty in reconciling some aspects of the generic standards with the design of some equipment such as low level amplifiers. A group is trying to propose realistic EMC standards for the professional audio and video industries but progress is slow.

3.7.2. Product Development and Testing for EMC Compliance

Whilst the EMC standards themselves and general subject of EMC seem rather daunting, the design strategy to take into account the various factors for EMC compliance are fairly straightforward and easy to assimilate. It has always been the case that it is easier to design good EMC into equipment at the beginning rather than "add it on" afterwards. If anything, a good EMC design lends itself to having components removed rather than added at a later stage. There are a number of quite practical guides or checklists to assist the design engineer. One of these is reproduced below:-

3.7.3 Good design EMC check list

1 Design for EMC

· know what performance you require from the beginning;

2 Components and circuits:

- Use slow and/or high immunity logic circuits.
- Use good RF decoupling.
- Minimise signal bandwidths, maximise signal levels.
- Provide power supplies of adequate (noise free) quality.
- Incorporate a watchdog circuit on every microprocessor.

3 PCB layout:

- Ensure proper signal returns; if necessary include isolation to define preferred signal paths.
- Keep interference paths segregated from sensitive circuits.
- Minimise ground inductance with thick cladding or ground planes.
- Minimise loop areas in high current or sensitive circuits
- Minimise track and component lead-out lengths.

4 Cables:

- Avoid running signal and power cables in parallel.
- Use signal cables and connectors with adequate screening.
- Use twisted pairs if appropriate.
- Run cables away from apertures in the shielding.
- Avoid resonant cable lengths as far as possible.

5 Grounding:

- Make sure all screens, connectors, filters, cabinets, etc. are adequately bonded,
- Ensure that bonding methods will not deteriorate in adverse conditions.
- Mask or remove paint from any intended conductive areas.
- Keep earth leads short.
- Avoid common ground impedances.

6 Filters:

- Optimise the mains filter for the application.
- Use the correct components and filter configuration for input and output lines, [0.1 to 6 MHz is sufficient bandwidth for the AES/EBU Interface].
- Ensure a good earth return for each filter.
- Apply filtering to interfering sources such as switches and motors.

7 Shielding:

- Determine the type and extent of shielding required from the frequency range of interest.
- Enclose particularly noisy areas with extra internal shielding.

8 Testing:

· test and evaluate for EMC continuously as the design progresses.

Overall, this list of requirements might seem fairly daunting , yet practical experience has shown that they can be met by only using common sense.

A typical example of poor EMC design was found in an AES/EBU interface receiver which used line transformers designed for triggering triacs. When used for the AES/EBU application, the poor balance and cross-capacitance of these parts led to reception failures. Sometimes these failures extended over several hundred milliseconds. All this error extension resulted from just one spike induced at the input!

3.7.4. EMC and system design

It should be borne in mind that the overall system should be tested for EMC. If the product comprises a number of different elements e.g. A-D coder, link, supervisory computer, modems, etc. then it is the whole system which should be verified for EMC. Specific instructions for the installation of the system will have to be provided that ensure that the system complies as a whole.

3.8. Error detection and treatment at the electrical level

Strictly speaking, errors in the bit stream can only be detected after the validity bit is inspected. However, many other indications of problems have been mentioned above. These include:

- loss of lock of the clock,
- missing or corrupted preambles
- loss of framing,
- loss of the Z preamble sequence.

If an error is detected in a received bit stream or one of the above problems is encountered, some difficult decisions have to be made. This is true even if the problem is met at the electrical layer, as well as at the more complex channel status control layer.

Consider the simple case of a regenerator of an interface signal when the receiver data clock briefly looses lock. What should happen? Is it best to:

- reframe the output, with possibility of passing on un-repaired audio samples and set "non-valid" validity flags?
- interpolate the audio samples, resetting the "validity" flags and therefore leaving little or no evidence for downstream equipment that the error has occurred?

It is likely in practice that the economics of equipment design will determine the level of repair possible. The action or inaction is also influenced by the inherent lack of any way of showing the error history in the interface specification. In practice, a receiver may be best advised to ignore the validity bit. In any case the validity flags of the two audio channels in the interface should always be treated separately.

Fortunately it is simple to make recommendations on error handling in the case of a D-A convertor. Isolated audio samples which are fagged as non-valid or where an error is detected should be interpolated. A long sequence of errors should cause a slow mute to the audio signal, with a few milliseconds fade in and out. It would therefore be useful to have a few sample periods of delay between the decoding and D-A conversion stages. (Consumer digital audio equipment often incorporates this delay as part of an integrated interpolation and over-sampling filter.) A good test of how a D-A system handles errors is to "hot" switch several times between two non-synchronous sources of interface signals both carrying audio. The resulting switch between the sources should be free of loud clicks and long muting periods.

Chapter 4

THE DATA LAYER

4.1. Data structure of the interface

Once the data has been decoded from the serial bit stream in the Electrical Layer, it can be sorted into its various components. As shown in figure 4.1, these are:

- Auxiliary Data,
- Audio signal,
- Ancillary data i.e. V, U, and C,.

The audio data itself is normally passed on by the data layer unchanged. Nevertheless important information about the parameters of the audio signal can be carried in the ancillary data. It is also important not to ignore the sub-frame preambles, since they carry information which identifies the A and B audio channels, as well as the start flags for the Channel status data blocks.

0 3	4 27	28	29	30	31
Pre-amble	LSB MSB 24-bit audio sample word	V bit	U bit	C bit	P bit

(b) 24 bits

0 3	4 7	8 27	28	29	30	31
Pre-amble	Aux bits	LSB MSB 20-bit audio sample word	V bit	U bit	C bit	P bit

(a) 20 bits

Fig.4.1. Sub-frame format for audio sample words

(Fig. 1, EBU Tech 3250)

The first five bytes of the Channel status block carry information on how the other interface bits are used. The relevant bits are shown below in figure 4.2.

Since the second edition of the specification was published, a number of proposals have been made to use the same hardware for other interfaces, especially to carry bit rate reduced signals. The EBU has recognised a danger in this because these signals have a random data structure which is quite unlike that of a linear audio signal. If these non-linear signals are converted to analogue and they will cause high levels of high frequency energy which may lead to damage to equipment such as loudspeaker transducers. To try to prevent this danger, the EBU has issued a Supplement to EBU Standard N9 which modifies the meaning of the Channel Status information. As before byte 0 bit 0 signals consumer or professional use but bit 1 now signals "linear audio" or "not linear audio" (instead of "non-linear audio"). Thus a professional non-linear audio interface signal (such as a bit rate reduced interface) should be easy to tell apart from a normal linear interface signal. Obviously the remainder of the channel status information, defined in the specification EBU Tech 3250, does not apply to a bit rate reduced interface.

	Bit							
Byte	**0**	**1**	**2**	**3**	**4**	**5**	**6**	**7**
0	Professional/ Consumer	audio/ non-audio	Audio signal emphasis			SSFL flag	Sample frequency	
1	Channel mode				User bit management			
2	Use of auxiliary bits			Source word length			Reserved	
3	Multi-channel function description (future use)							
4	Digital audio reference signal		Reserved					

Fig. 4.2. Channel status data format (part)

4.2. Auxiliary data

The first four data bits can be either used as the LSB the audio coding, or they can be used for another purpose such as a separate low quality audio channel. Byte 2(0-3) in the channel status block is used to signal which option is used. In Appendix 1 of Tech. 3250 there is an example of a coding system for a low quality communication channel which has also been standardised by the ITU-R (CCIR). This system, however, is by no means the only proposal for this application. Other audio coding methods for lower quality signals may well be developed in future and accepted for general use.

4.3. Audio data

The interface is designed to carry two channels of "periodically sampled and linearly represented" audio. More precise information on the details of the audio signals that the interface is actually carrying can be found in the channel status. These details include: sample frequency, emphasis and word length.

4.3.1. Sampling frequency

CS Byte 0 signals the source sampling frequency. Most signals used by EBU Members in studios are expected to be sampled at 48 kHz. However 44.1 kHz may be used instead for some applications, such as recordings intended as masters for CDs. If signals are to be fed to transmission equipment, the frequency used may be 32 kHz.

4.3.2. Emphasis

CS Byte 0 also carries information on whether pre-emphasis is used and, if so, what sort. EBU Members do not normally use pre-emphasis for audio signals within studio areas, so any signal where any form of pre-emphasis is indicated should be treated with caution. The best procedure, if such signals are met, would be to immediately de-emphasise them, preferably digitally, and reassembled them into a new interface signal carrying the "no emphasis" flag. This should be done before they enter any system, otherwise it might be too late to prevent mixed operation. Remember that a lot of equipment does not transmit the Channel Status; for instance a digital recorder may only record the audio data. Therefore, on replay the original channel status information will be lost, and the pre-emphasis flag will be lost along with it. There is now no way of knowing how to treat the signal.

4.3.3. Word length and dither

The word length of the audio signals can be useful information for processing equipment. CS Byte 2 (bits 3-5) is used to show the state of the Least Significant Bits of the audio data by indicating the number of bits used in the original coding. Any further LSB are assumed to be unused. Correct "rounding up" and dithering of the LSBs can greatly enhance the apparent audible dynamic range of a PCM signal. Theoretically this can be by up to the equivalent of three extra bits. So if for any reason the audio bit stream has to be truncated, for instance from 20 bits to 16 for recording or before D-A conversion, the LSB of the output 16 bit signal should be re-dithered. The dithering will take into account the 4 LSBs of the 20 bit signal that are to be discarded. In this way the full potential of the 16 bit system will be realised and the minimum of audio quality loss will occur. It is significant that the noise levels given in the "codes of practice" for general audio performance of many organisations, written with analogue practice in mind, can only be met by 16 bit digital coding if intelligent dithering is implemented.

Note that extra LSBs may be present on the interface between two items of signal processing equipment. These can be generated as overflow bits in the processing of the earlier stage. In theory this should be signalled in the Channel Status but it is quite possible that Channel status will not be changed. Therefore the maximum word length of the audio samples actually present may be longer than the "encoded sample word" indicated by Byte 2 of channel status. For the sake of the overall signal quality it is important that these extra LSBs are not truncated. As a general rule, it is worth considering CS byte 2 bits 3 to 5 as indicating the inactive audio bits present.

Generally the default condition of 20 audio bits per sample will be sufficient for practically all broadcasting purposes. 24 bit distribution will be needed in only a few cases.

4.3.4. Alignment level - EBU Recommendation R68

Everyone agrees that it would be very useful if all digital audio equipment, particularly recorders, used the same alignment levels, so that signals could be processed more easily. EBU Recommendation R68 defines an alignment level for digital audio production and recording in terms of the digital codes used for the signal levels described in ITU-R (CCIR) Rec. 645. These CCIR levels are basically:

- the maximum permitted signal level which is allowed in a system,
- an alignment level, which is a convenient standard reference related to the maximum level.

These levels are illustrated in figure 4.3., below.

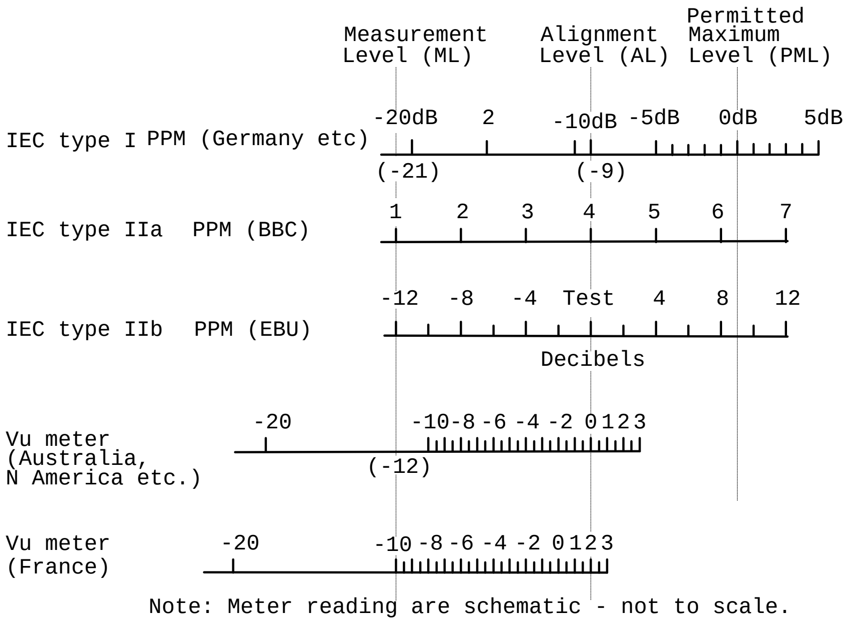


Figure 4.3: Indications produced by various types of programme meter with the recommended test signals
(after ITU-R (CCIR) Rec. 645)

In practice this means that it is the coding levels on the input and output AES/EBU interfaces of a recorder that are defined, rather than any parameter of the recording process. The alignment level recommended by EBU is digitally defined as peaking 3 bits below full scale, which is approximately -18 dBfs. This level is recommended to be used both for 625 line television and sound broadcasting applications in Europe. The digital definition enables both simple bit shifting and simple alignment metering in the digital domain. It also matches well the dynamic range of the IEC 268-10 analogue Peak Programme Meter, at the same time as providing the maximum dynamic signal range within the CCIR Recommendations.

Unfortunately one of the obstacles to a universal agreement is that many individual EBU Members use different analogue levels for the ITU-R (CCIR) signal levels. This means that it is not possible to define a single relation between analogue voltages and digital coding levels. Most broadcasters use a "line up" or "identification" or "reference" tone before each recording which is used to define the level used on the recording. However, national practices vary here too. The EBU has prepared a demonstration R-DAT cassette containing examples of the ITU-R (CCIR) and EBU line up levels coded to EBU Recommendation R68. The tape also contains typical material taken from the SQAM disc (EBU Tech 3253). The modulation levels of these extracts have been audibly selected for equal loudness. It is hoped that this tape will result in a better understanding of the relationship between the digital code full scale and the maximum permitted level.

Because, in the studio, a number of analogue voltage equivalents to these levels are used, A-D and D-A converters will have a fairly wide range of gain adjustment.

In EBU Recommendations R64, the EBU has specifies the R-DAT format for programme exchange and this format is now used extensively for this. The alignment tone on these recordings should correspond to the level defined in EBU R68.

However, things are not quite so straightforward n the field of television operations and programme exchange. In the all digital sound editing, some trouble has been experienced because some, but not all, recordings made in the D-2 or D-3 digital television recording formats have used an alignment level specified by the SMPTE , nominally -20 dBfs. It is hoped that this will fall out of use in Europe as all digital recorders are aligned to the EBU Recommendation. The manufacturers have been asked to do this in EBU Statement D77.

4.3.5. Single or multiple audio channels

The digital audio interface described in EBU Tech 3250 permits four modes of transmission which are signalled in channel status (Byte 1, bits 0-3). The two channel, stereophonic and monophonic modes are easily understood, however the fourth mode, primary/secondary, is not well defined.

The EBU has identified three possible uses for the primary/secondary mode, namely:-

- A mono programme with reverse talkback.
- A stereo programme in the M and S format.
- Commentary channel and international sound.

Ideally, these three uses should be signalled in the Channel status by separate codes. The EBU is seeking support from the AES to allocate further codes for this purpose. In the mean time the EBU has published Recommendation R73, which is based on the existing practice for the allocation of audio channels in D1, D2 and D3 Digital Television Tape Recorder formats.

EBU Recommendation gives the following details of the channel use:-

				Primary/Secondary		
	Mono programme	Stereo Programme	Two Channel	Mono and Talkback	Stereo M and S	International sound and Commentary
Ch 1	Complete mono mix	Complete mix L	Channel A	Complete mono mix	Mono signal	Mono Commentary
Ch 2	Complete mono mix	Complete mix, R	Channel B	Talkback	Stereo difference signal	International sound

Byte 3 of the Channel Status has been reserved for descriptions of multi channel functions. This is intended for the situation where a number of associated interface signals emerge from sources such as a multi-track recorder. It certainly was not intended to cover the use of a single interface with a number of bit rate reduced signals. This latter use would be outside the scope of the specification, since it is implicit in the specification that the audio signals are linearly coded. In Section 3.1 there are details of the use (recommended by the AES) of a multiple interface equipment connector for the electrical level. It is possible that, in the future, byte 3 may be developed to provide identification codes for use on such a multiple interface.

4.3.6. Sample frequency and synchronisation

Two signals about the sample frequency are carried in the Channel Status:

- byte 0, bits 6 and 7, is the coded sample frequency used for the original encoding of the audio ,
- byte 0, bit 5 is a flag (Source Sampling Frequency Locked ,SSFL) which shows if the sample frequency of the current interface is locked to a reference.

The SSFL flag indicates that the clock oscillator of the sending device is synchronised with (or locked to) a reference signal. The reference signal serves to keep all the local clock oscillators involved in processing the programme signal on the same frequency, and in phase with each other. (The "colour black" signal serves the same purpose in a television system.) The SSFL flag is set to "0" when the local clock oscillator is locked and to "1" when it loses lock with the reference signal.

At this point, it is worth considering a few practical situations, defined by the AES group, and the flag signals associated with these situations. These are based on a typical programme chain made up of digital audio devices (DAD) and digital audio reference sources (DARS) as shown below.

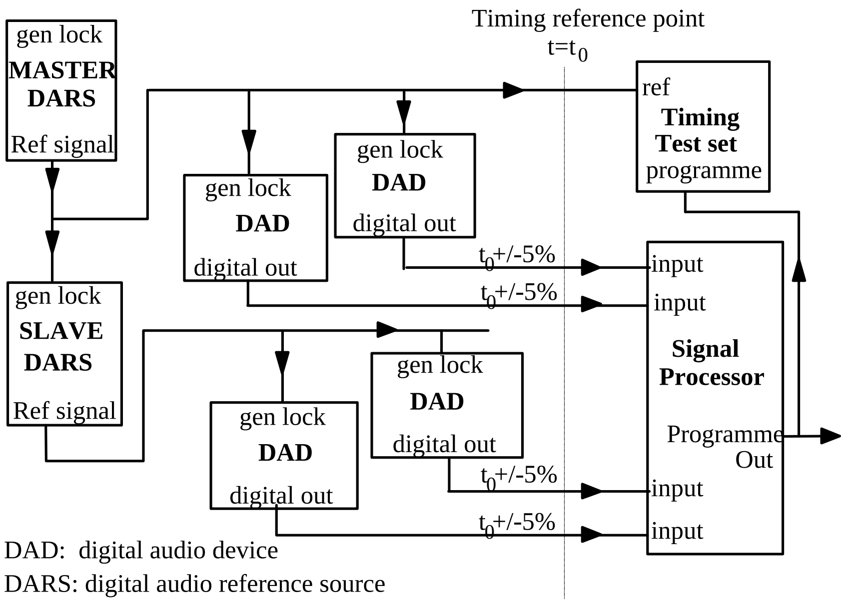


Figure 4.4 System timing and synchronisation

Most equipment will have two options for the source of the synchronisation signal. The first is the "house sync" signal, provided by a Digital Audio Reference Source (DARS) generator, The second reference signal is the programme input signal itself. Depending on the type of operation, the internal clock oscillator should lock to either one or other of these sources.

The DARS generator may be itself be locked to an external reference, which is usually the same high frequency reference used for any associated video and station time clocks. For reasons of jitter accuracy, the edges of a video signal can be unsuitable as a high quality reference for DARS generation. Therefore it is better to lock a DARS generator directly to the common high frequency reference rather than via a video signal.

A DARS generator is accurate and stable enough that it does not need to be locked to a reference signal. Nevertheless, a "slave" DARS generator will normally be locked to a master DARS somewhere in the plant, simply to ensure phase integrity of the whole plant. A DARS should give some type of indication on its front panel when it is locked to an external source.

A number of other points can be made about figure 4.4.

If there is any danger of the system timing at any point going outside the limit specified in AES-11 (25% of a frame) there should be timing adjustments provided to adjust the phase of output signals from equipment.

A slave DARS should have quite a large range of timing adjustment (including "advance"), so that the reference output can be brought into phase with of the master DARS at the timing reference point, t_0, on the right of the diagram. This will eliminate any difference between t_0 and t'_0.

Each DADs should also have internal compensation for any delays in its internal signal paths, thus bringing all the outputs timings to within at least the 5% phase margin to the t_0 point of the system.

The Timing Test Set should indicates the phase (timing) of the output signal relative to the master DARS reference signal. It can be used to detect non-locked signals appearing at the programme output of the system (see below).

Operationally, it is important to know if there is a free running clock somewhere in the system. This condition can be signalled using the SSFL flag (CS Byte 0 bit 5) in two different ways.

1. A device whose clock oscillator is unlocked should set the SSFL flag to "1". This flag should be propagated through all downstream equipment to warn of the non-synchronous condition. This effectively makes the last device in the signal chain the system monitoring point. In terms of individual devices, this means that:-

- All equipment which cannot change the SSFL flag should pass the incoming flag to its output.

- All equipment which can change the SSFL flag should:
  (i) set the output flag to "1" if its own clock oscillator is unlocked,
  (ii) pass the incoming unlocked flag to the output.

- Equipment with more than one input should set its output SSFL flag(s) to "unlocked" if any one of the inputs contributing to the output signal shows an unlocked SSFL flag.

2. A device whose clock oscillator is unlocked should set the SSFL flag to "1". This flag bit should not be propagated throughout the downstream equipment. A front panel (or equivalent) indicator should warn the operator of the non-sync condition. In this situation, the Timing Test Set shown in the diagram becomes the system monitoring point. This is similar to the way that a vectorscope is used to monitor the output of a TV studio.

Note that a DARS generator is a special case. It should always set the SSFL flag to "0" (locked), as mentioned above.

4.4. Validity bit

The Validity, V, bit in the interface gives a warning that an audio sample word is not "suitable for conversion to an analogue audio signal". However the judgement of what is "suitable" can be different for different applications. For instance, interpolations or concealed errors may be perfectly acceptable in a signal in a news programme, but completely unacceptable for a signal intended for a CD master.

In the signals from interactive CDs (CD-I), for example, the Validity bit is used to indicate that the audio information has been replaced by a stream of non-audio data which it would be clearly unsuitable to convert to analogue audio. Using the V bit in this manner avoids a delay before the audio/data flag bit can be recovered from the channel status data. The delay could be up to one CS block (192 samples), and during the interval there would be a burst of noise on the audio channel.

There are a number of different reasons which may make a sender may want to declare an audio sample word "unsuitable for conversion" and a number of different ways a receiver might want to react to the reception of such a message. With only a single bit to convey the messages, the use of the V bit clearly depends greatly on the application. The equipment supplier must clearly define and document how and why his equipment transmits the V bit and how it reacts to it on reception. Users, on their part, must be aware that different equipment may use different criteria for setting the V bit. The choice of these criteria may or may not be under the control of the user. There is, thus, a joint responsibility on users and equipment designers for correct system implementation. All this is reflected in the broad definition of the V bit, given in the specification, as indicating that the audio data is "unsuitable for conversion to an analogue audio signal".

It is important not to confuse the Validity bit, which is present after every audio sample, with the audio/non-audio flag contained in channel status.

It therefore seems to be difficult to formulate even simple rules for the use of the V bit, even for the equipment classifications that have been established for audio operations. However in section 5.2 below there are some guidelines which should be studied if this form of operation is likely to be even remotely possible within any installation.

4.5. User bit

On bit per audio sub-frame has been allocated as a User bit. In Tech Doc 3250, there is no restriction on the use of this User bit, U. However four bits have been allocated in Channel status (CS byte 1, bits 4-7) which can be used to indicate the form of the data in the user channel. One of these is the packet HDLC protocol, defined in supplement 1 to Tech Doc 3250 and this is directly equivalent to AES18-1992. This protocol defines an asynchronous format which is particularly useful if a computer HDLC protocol is already available in associated equipment. The audio interface will then act as a virtually transparent data link, even when synchronisation or sample rate changing takes place along the audio chain.

More information on this system is expected to be available in a separate Guideline document which will be published later.

The other defined format for User data uses the same 192 sample block as the channel status data. The use of this format is not defined in any further detail.

4.6. Channel Status bit

A number of the uses of the channel status, C, data have been mentioned above. After the audio data, the channel status is probably the most valuable data to a broadcaster of all the signals carried in the interface. It consisting of a single bit per sub-frame, and is assembled in blocks of 192 bits in 24 8 bit bytes. The start of a CS block is indicated by the unique "Z" preamble in sub-frame 1 every 192 frames. The CS data in a block applies to the audio bit stream in its own sub-frame. So only for stereo use would there necessarily be identical C bits in both sub-frames. In normal broadcast use, at 48 kHz sampling, C data is updated every 4 milliseconds. Fig. 5 in Tech Doc 3250 gives a global map of this data format. the present section is only concerned with the recovery of the data from the serial bit stream. Chapter 5 covers how the information is used in more depth.

4.7. The Parity bit and error detection

On transmission. the parity bit, P, for each sub-frame is set to make an even number of ones and even number of zeros in bits 4-31 of the sub-frame. This has two consequences, an odd number of errors resulting from malfunctions in the interface can be detected and also the sub-frames are linked so that a constant polarity of the bi-phase mark channel coding is maintained during the whole running time of the equipment.

Error detection based on the parity bit can therefore be performed in two ways:

- at the data layer, by testing the parity bit in each sub-frame
- at the electrical layer, by verifying the properties of the bi-phase mark signal for time slots 4 to 31 and the specific patterns of the preambles for time slots 0-3.

Section 3.8 above covers what to do if errors are detected, as well as the case of multiple errors. This section is principly concerned with the use of the Parity bit in error detection. If an error is detected, any consequential treatment will usually be limited to correction of the audio data. There may be an advantage for downstream equipment in inserting a localised mute for a single audio sample. This can be done by setting all the audio bits to "0" and the validity bit to "1". Muting in this way does rely on downstream equipment being able to detect and correctly interpret the validity flag.

Error concealment can be carried out very simply by interpolation between neighbouring samples. This method generally gives good audible results at the output of D-A converters.

Chapter 5

THE CONTROL LAYER (CHANNEL STATUS)

The channel Status information contains information which can be used to control the equipment receiving the interface. The ways in which this control can be achieved are outlined below.

5.1. Classes of implementation

Because the interface can be used with so many different types of equipment and in so many different circumstances, it is virtually impossible to lay down hard and fast rules for procedures that can apply to all equipment. The EBU and AES have tried to simplify matters by proposing class of equipment with more clearly defined rules of operation. The following proposals are based on the AES Engineering Guidelines. The aim is to introduce a universal "implementation data sheet" together with agreed receiver classifications that are easy to understand and which will relate to real products. The current proposal is to divide equipment into classes: A, B1, B2, B3 and C. These classification is based on how the device deals with the data it receives and to what extent it stores the data during audio processing and how it passes the data on to its output. Currently the AES classification is made on the assumption that the equipment will handle the V, U and C bits in the same way, whereas it is quite possible that real devices will need to handle them differently. Furthermore, the V bit is simpler to deal with than the C and U bits which can carry much more information.

In order to arrive at the right classification for equipment, the following questions must be asked:

- How does the receiver behave on receiving certain data?
- How much of the received non-audio data does it store and/or pass through to the output?
- How can the receiver **modify** the non-audio data before re-transmission?

A correctly completed implementation data sheet, together with an equipment classification, will indicate how a receiver will behave. The implementation data sheet will also indicate the transmitter implementation, and from it could be deduced to what degree the device conforms to the 'minimum', 'standard' or 'enhanced' transmitter categories given in the specification EBU Tech 3250.

Receiver implementation can be divided according to how the V, U and C data are handled. These are discussed in more detail below.

5.1.1. V Bit handling

It is straight-forward to indicate on the implementation chart (see example below) the way in which the V bit is handled. It does not then need to influence the equipment classification. The only problem is whether the state of the V bit is stored and passed through. This could possibly be done with an extra symbol on the implementation chart (e.g. 'S'), but see below on modification of stored Channel status bits before re-transmission.

5.1.2. U Bit handling

This need to indicate how the U bit is handled depends on the growth in the use of user bits. At present the use is small and should not influence the receiver classification of the equipment, which should be based entirely on how the Channel status information is handled. The implementation chart can show which modes of user bit management are recognised, and a separate section of the equipment manual could cover the practical application of user bits. A section of the implementation chart, similar to that of the V bit, would show the basic transmission/recognition of user bits. The 'S' symbol could be used to indicate the U data is stored and passed through.

5.1.3. C Bit handling

It is important to note that whatever the receiver classification, the non-audio data **transmitted** by a device should represent the actual status of the audio and data signal. Thus received channel status data which has been stored or will be passed through may have to be modified by the device to represent the true status of the output.

Categories of equipment at present defined are:-

Group A Acts like a wire, and simply passes the input signal to the output with no intermediate decoding or processing (e.g. a simple crosspoint switcher).

Group B1 Decodes only audio data. Does not decode channel status data. Does not pass or store any input C data. (N.B.: clearly this would be highly unlikely in practice and rather dangerous, since such things as emphasis in the received signal would go unnoticed downstream.

Group B2 Decodes the input channel status and recognises or acts on the data to the extent shown in the implementation sheet. It does not store or pass on received channel status data.

Group B3 Decodes the input channel status and recognises or acts on the data to the extent shown in the implementation sheet. It also stores and/or passes on the received channel status information denoted with the 'S' symbol in the implementation sheet, modified if necessary to reflect the true status of the output signal (e.g. emphasis, sample rate, source ID, timecode, etc.).

Group C1 Is a terminal-end equipment with no digital audio output (e.g. a D/A convertor). It only decodes the audio data. It does not decode channel status data. (N.B.: the same proviso applies as

Group C2 Is terminal-end equipment with no digital audio output. It decodes channel status data and recognises and acts on the data to the extent shown in the implementation sheet.

5.1.4. Implementation data sheet

An example of a standard format of chart which could accompany any item of equipment implementing the AES/EBU interface is shown below. It has been designed for maximum commonality with the MIDI implementation chart. It shows channel status implementation in detail, indicating both what is transmitted (TX) and what is recognised on reception (RX). On the RX side three symbols can be used:

- 'X' for data which is not recognised,
- 'O' for data which is recognised,
- 'S' for data which is recognised and stored and/or passed through to output where appropriate. ('Where appropriate' may need qualification in the Remarks column. The 'S' indicates that the received data is available for inclusion in the output data and will be used unless the equipment takes action to replace it with something else, such as a indication of a change of emphasis). The 'Remarks' column allows the manufacturer to comment on particular details of a response or implementation.

The chart also has entries to show how the V bit is handled and whether user, U, bits are recognised, stored or transmitted. A separate chart should be designed in the future to give the details of U bit handling.

AES/EBU implementation chart (draft example)

Model	
Receiver classification	B2
Transmitter classification	Standard

Key: TX: X: not transmitted 0: transmitted
RX: X: not recognised 0: recognised S: recognised and stored

Channel Status (C) data

Byte	Bit	Function	RX	TX	Remarks
0	0	[0] Consumer use [1] Professional use	X 0	X 0	
0	1	[0] Audio [1] Non audio	0 0	0 X	 Analogue o/p mutes (RX)
0	2-4 Emphasis	[000] Not indicated [100] No emphasis [110] 50/15 us [111] CCITT J17	0 0 0 X	X 0 0 X	RX defaults to no emphasis RX defaults to no emphasis
0	5 Fs locked	[0] locked [1] unlocked	X X	0 X	
0	6-7 Sample Freq.	[00] Not indicated [01] 48 kHz [10] 44.1 kHz [11] 32 kHz	0 0 0 X	X 0 0 X	default to 48 kHz
1	0-3 Channel mode	[0000] Not indicated [0001] Two channel [0010] Mono [0011] Prim/sec [0100] Stereo [0101-1111] undefined	0 0 0 0 0 X	X 0 X X X X	RX defaults to 2 channel mode Normal condition Ch A on both O/Ps RX defaults to 2 channel mode Same as 2 channel Don't care
1	4-7 User bit mode	[0000] Not indicated [0001] 192 bit block [0010] AES18 (HDLC) [0011] User defined [0100-1111] undefined	0 X X X X	0 X X X X	 Don't care
2	0-2 Aux. bit use	[000] Not indicated [001] Audio data [010] Co-ordn [011-111] undefined	0 0 X X	0 X X X	 RX redithered to 16 bits
2	3-5 Sample length	[000] Not indicated [100] All other states	0 0 0	X 0 X	Default and re-dither to 16 bits 16 bits Re-dither to 16 bits
3	0-7	Multichannel modes	X	X	
4	0-1	AES11 sync ref. signal	X	X	
5	0-7	Unused	X	X	
6-9		ASCII Source ID	X	X	
10-13		ASCII Destination ID	X	X	
14-17		Local sample add code	X	X	
18-21		Time of day add code	X	X	
22	0-7	C reliability flags	X	X	
23	0-7	CRCC	0	0	

	RX	TX	Remarks
Validity (V) bit	S	0	True for uncorected samples (TX)
User (U) bit	X	X	
Audio sampling frequency (kHz)	44.1,48(RX):44.1,48(TX)		
Audio sample word length (bits)	16-24, re-dithered to 16(RX).16(TX)		

5.2. Examples Of Classification of Real Equipment

5.2.1. Digital tape recorder or workstation

Most digital recorders would be in either receiver classifications B2 or B3. If a recorder is in classed B3, then only that channel status data denoted by the 'S' symbol in the RX column on the implementation sheet would be stored on tape/disk and made available on replay for use in the TX data.

5.2.2. Digital studio mixer

These would normally be in either receiver classifications B2 or B3. The implementation sheet will show whether any input Channel status data is carried through to output. The likelihood is that most mixers will be B2, since when more than one channels are combined, it will rarely be clear to the internal logic which C data from which input should be carried through to which output. The C data transmitted at the output(s) will be have to be newly generated by the mixer and reflect the true state of the output.

5.2.3. Routing switcher

A simple crosspoint matrix would most likely be Group A, since it does not even decode the bi-phase mark signal; merely regenerating it. A TDM router would probably be B3, since it would pass on any received channel status data and might modify it, although not necessarily.

5.3. Reliability and errors in the Channel Status data

Since the channel status data, almost alone of the data carried in the interface, can be used to control equipment, any errors in the CS data could prove very damaging in operations. For instance an error in the destination data may cause a router to switch the signal at the wrong moment. The general philosophy that should be adopted for dealing with errors in the Channel status data should depend on the application and the type of data actually carried.

Although there are two mechanisms for detecting errors built into the Channel status data channel, both require care in use for reasons explained below. In practice equal of better reliability can be obtained by checking the integrity of the data as explained below.

Even this does not always guarantee accuracy. As an example, in a typical audio device there are a number of ways to determine the sample frequency of the input signal:

- by decoding Byte 0 of the channel status data,
- from the repetition rate of the frames of the interface signal
- from the frequency of the received word clock,
- from the setting of a hardware switch.

Recently, when several DAT recorders from different manufacturers were examined, it turned out that the different designers had made different choices of which of these to act on.

It is also important to remember that some codes used in CS bytes may not interpreted correctly by all equipment. As an example, it has been found that some equipment can misinterpret the sample frequency code (CS byte 0, bits 6-7). If this is set to "00" it should mean just that the sample frequency is not indicated but it has been interpreted to mean that the sampling is indeterminate.

5.3.1. Static Channel Status information

Bytes 0 to 5 carry static information which does not normally change, so any action can usually afford to wait until after the reception of several blocks have confirmed that a change has actually taken place. Similarly any change in the source data (bytes 6-9) or, particularly, the destination data (bytes 10-13) ought to be confirmed several times before any action, particularly switching, takes place.

5.3.2. Regularly changing Channel Status information

Bytes 14-27 and 18-21 carry sample address codes which should normally be progressively increasing. So after an single error they can be interpolated or extrapolated if necessary. But beware that the sample address does nor always increment regularly. It will change suddenly if the signal is switched. It may also behave irregularly if the signal comes, for instance, from a recorder which is spooling.

5.3.2. Dynamic Channel Status information, validity flags and the CRC

Sometimes rapid action is required on a change to the data in the Channel status. In this case the Reliability flags (byte 22) and the Cyclic Redundancy Check sum (byte 23) should be consulted to confirm that a valid change has taken place. But both require considerable care in how they are interpreted.

The reliability flags are effectively "validity" flags for sections of channel status data. One problem with this is that the first flag (byte 22 bit 0) refers to the data in all of bytes 0 to 5. This covers so many parameters that in practice the uncertainty will be so great that only solution to an invalid indication would be to wait for the next block and check again. This, of course, is the recommended practice given above.

In Section 5.1 there is a list of the classes equipment based on how they handle channel status data. From this list a very complex set of possibilities can be constructed for the action to be taken downstream when there is a difference between the data and the CRCC. In this situation, two sound pieces of advice are to:

- use majority logic on CS block information wherever possible,
- if the block length is wrong, treat this as a sign of an error whatever the CRCC indicates!

5.4. Source and destination ID

In the flexible studio installations which are common today, information on the source of a signal, and its intended destination are extremely valuable to the users. For the AES/EBU interface signals, the source identity data is probably one of the most used pieces of channel status information. It is very often shown under monitors in television studios, or on the front panels of recording or mixing equipment. It readily takes the place of the hand-written pieces of tape which have always served to identify channels (with less reliability!) in analogue equipment. Although limited to four alphanumeric characters, the amount of information which these four bytes can show is surprising. They can also be used to indicate more than the source address. For instance, a distributed digital audio reference source (DARS) which carrying digital silence might have the source identification "MUTE". Such an indication on the displays of a mixer or recorder gives the operator confidence that an actual locked input is present, rather than that the silence is due to no input.

The value and use of the destination ID data is less certain. Potentially it could be used as part of the control of an automated routing system. However, there a number of problems with this idea, such as the lack of an return channel for "tally" signals. As far as is known, the destination ID data has not been used on a large scale in broadcasting.

5.5. Sample address and timecodes

The sample address codes in the Channel status are intended to label the signals to make future processing more straightforward. The sample addresses simply increase throughout the day and thus the information has to be processed to extract the time in hours, minutes, seconds, etc. Ideally the first audio sample will occur at midnight. A channel status block will also start at the same time. Bytes 18-21 of this first block of channel status will thus carry the sample address code of zero. The subsequent blocks will carry the binary equivalent counts of 192, 384, nx192, and so on, since each channel status block is 192 audio samples long. If the first audio sample of the first block actually occurred X audio samples after midnight, the "time of day" sample address counts would still increment throughout the Day in steps of 192 but each would have an offset of plus X.

Where the audio interface signals are associated with pictures, there may be some practical difficulties in using the sample address code. The VTR time-code traditionally used for video (IEC 461) gives time in hours, minutes, seconds and television frames. Transcoding between VTR time-code and the large 32 bit binary sample address count (SAC) has not be found to be simple, although there is some simplification if the block start of the audio reference occurs at midnight. The need for frequent transcoding, in both directions, between the SAC and VTR time-code will still remain. Partly to avoid this, some installations have used the "time of day" sample address bytes (18-21) to carry VTR timecode in ASCII form. This has been found to work very well for 625/50 television systems but may not work as efficiently for 525/59.94 NTSC television systems, due to the complexities of the relation of the television signal to the audio sample rate. If the VTR timecode is carried in this way it should be remembered that it is an "illegal" or "heretical" use of the interface and should not be used outside an enclosed plant without prior agreement. Another advantage in carrying VTR time-code is that the audio reference signal can be used as a distribution medium for the time-code, saving the cost of a separate system.

Chapter 6

EQUIPMENT TESTING

6.1. Principles of acceptance testing

All broadcasters test their technical equipment to a more or less rigorous extent on acceptance. It is worth considering what this process involves before considering its effect on the testing of AES/EBU interface equipment.

Acceptance testing can be thought as just a commitment by the end user to technical quality control of his equipment. However, in practice, it often represents the end of a long period of discussions with the supplier. This process involves discussions, evaluations, measurements and more discussions. It frequently means far more than just filling in a results sheet.

Before a broadcaster shows serious interest in a buying a particular piece of equipment he has usually satisfied himself that it meets some basic criteria:

- it does what it is supposed to
- it will meet the performance specification set by the end user.

However care should be taken with which specification is used. Often there is a specification to be met on a day-to-day basis by equipment selected at random. Many manufacturers quote adherence to Technical Performance Codes (Code of Practice) which although they may be nationally or organisationally based, are usually this form of specification. Equipment should, however, be capable of achieving a better performance when properly set up, and it is this tighter specification that should be used for acceptance testing of individual items.

The current mixture of analogue and digital equipment may cause unexpected problems with specifications. A traditional specification for a particular item may be based on the use of analogue distribution circuits with an allowance for progressive degradation of technical parameters of the signal downstream. Use of equipment with digital interfaces should mean that the parameters of the original signal from the A-D coder are preserved at virtually the same level throughout the chain. Logically, therefore, for the same overall performance, their could be some relaxation of the source coding specification in a full digital interface implementation. There is need to strike a balance between getting better overall performance and avoiding unnecessary costs from over specification.

Acceptance testing begins long before purchase with an evaluation of the products of various manufacturers. Differences in performance would be noted and discussed. Control panel layouts and operational procedures may need to be examined. New systems must interface with current equipment without too many problems. For example the control surface layout and switching capabilities of a large matrix would need to be discussed with the operational staff to find out if any changes are possible and what would be the cost penalties.

Armed with test results and reports from all interested parties, the various alternative products will be discussed with the suppliers. Eventually a decision will be taken and the order placed.

For some products the final acceptance test on delivery could be fairly straightforward: checking the operational parameters and making measurements. However, with some equipment, which may have been extensively modified to make it suitable for the operational requirements, "type" approval tests on the initial items may be required to ensure that the final product will perform as expected.

Some equipment will have to be matched to make sure all the different units track together and possibly that they track with existing equipment. Confirmation of the control ranges for different conditions may also be necessary. All of this adjustment forms part of acceptance testing and, where possible, it should be performed on the premises of the manufacturer or his agent. This enables queries and problems to be identified and solved far quicker than when the testing is done at the end user's own premises after delivery.

The overall complexity of the installation will determine to what extent special procedures apply to digital audio interface equipment. However, the testing can usually be divided into three separate areas which can be individually assessed. These are based on the following layers:

- Electrical layer,
- Audio layer,
- Control layer.

These are covered in more detail below.

6.2. Testing the electrical layer

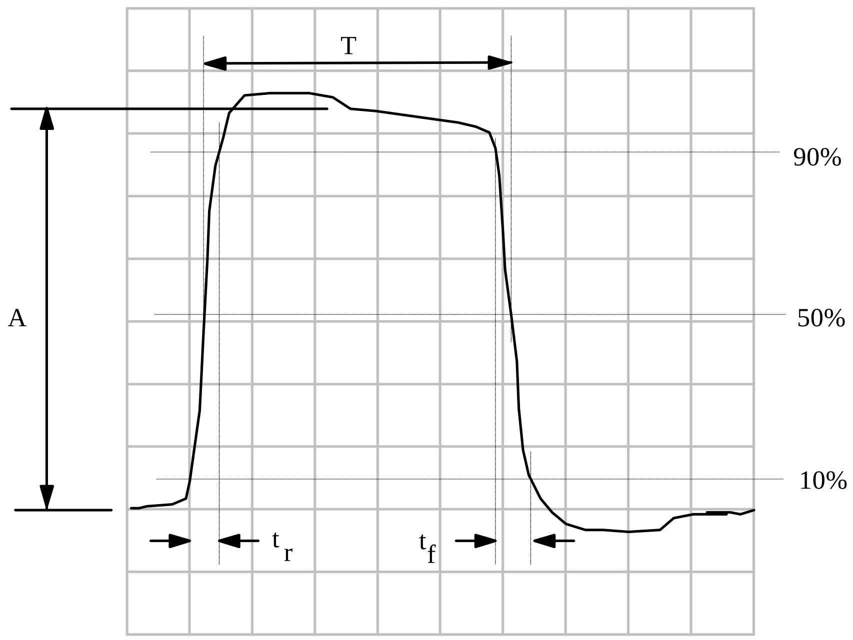
However important it might seem to check the audio performance or the channel status response of a system, there is little point in proceeding with these checks until the equipment can be relied on to send or receive the serial data. This is the job of the electrical layer and this should be tested first.

The electrical layer is also the aspect of a system which is by far the most susceptible to good or bad installation techniques (cables connectors etc.) which are covered in Chapter 3. In spite of all the hazards, carefully designed, installed and tested interface equipment has proved remarkably reliable in service. This has been found even for very large systems, such as those described in the Chapter 6, where the reliability of analogue equipment would have been a constant problem.

The following section gives a review of some of the basic tests that can be easily performed during an installation. It is based on the work Klaus Altmann of the IRT (Germany).

6.2.1. Time and frequency characteristics

The Digital Audio Interface according to the EBU Tech Doc 3250 is a serial interface, which is primarily designed to carry monophonic or stereophonic programmes in a studio environment at 48 kHz sampling frequency. This means that a bit stream of 3.072 Mbit/s has to be transmitted. The signal is bi-phase mark coded; Figure 6.1. below shows part of the signal in time domain as it can be observed on an oscilloscope.

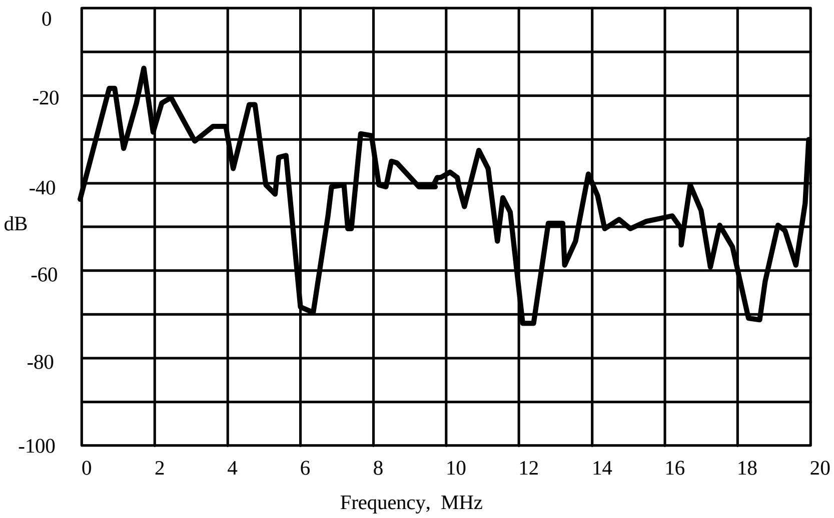


A: pulse amplitude   T: pulse width

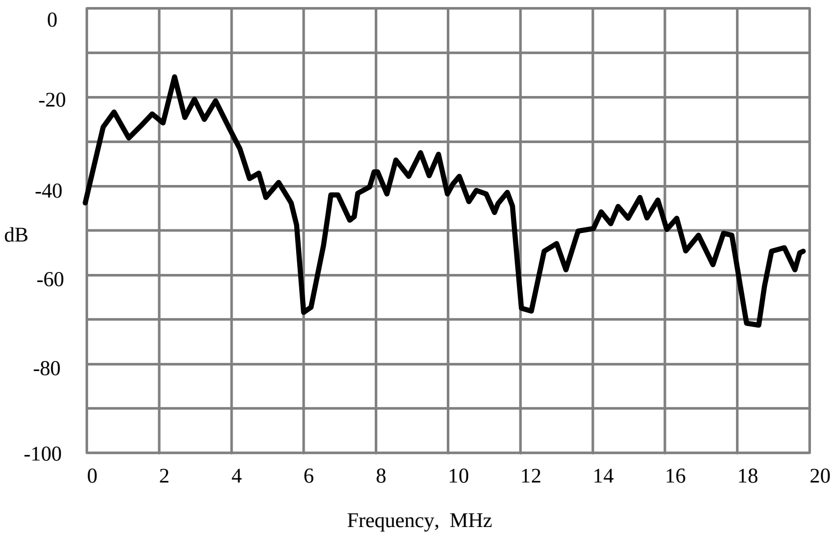
t_r: rise time t_f; fall time

Figure 6.1: Pulse parameters

In the frequency domain, the signal has a spectrum with a maximum at 1.536 MHz, if the audio channels carry only logic "0" and a maximum of 3.072 MHz with logic "1". This is shown in two spectral response diagrams below, Figure 6.2.a & b.



a) Audio channel bits mostly = 0



b) Audio channel bits mostly = 0

Figure 6.2: Interface signal - frequency domain

6.2.2. Impedance matching

Signals which cover a frequency range up to 3 MHz and higher are very well known in television engineering. In order to transmit these signals over cables, the impedance of interface driver, cables and interface receiver must be well matched. Otherwise the shape of pulses will be altered by reflections with the consequence that an interpretation of the received signal may be difficult or impossible.

For video signal distribution television engineers have developed a strategy which allows them to supply several units from one source by means of a "loop through" technique. For this, all receivers are equipped with high impedance inputs and two connectors in parallel. The signal passes through the first receiver and is fed to the next. At the end of the chain the last input has to be terminated. This technique is not applicable to the Digital Audio Interface because the transformers have low impedance. Transmissions of digital audio signals are point-to-point connections. For distribution to more than one receiver, special distribution amplifiers are necessary.

6.2.3. Use of transformers

The electrical characteristics of the Digital Audio Interface were based on the data interface, RS 422. The interface signal is balanced, and, if transformers are used at the input and output of the interface, it allows one of the two poles to be grounded without damage. However, many manufacturers replace the transformers by electronic balance circuits, which do not always tolerate such grounding.

6.2.4. Effect of jitter at an input

Jitter is a source of problems in all digital transmission links. It can be seen as phase modulation of the pulse slopes. Its effect depends on different factors, whether the jitter is of periodical or statistical nature or influenced by the data stream itself. The presence of jitter makes the recovery of clock signal more difficult and leads as a consequence to misinterpretations of the signal status. If passed through to the D-A stage, jitter will also adversely affect the audio performance because it is converted to noise and distortion.

Jitter can be measured by means of an oscilloscope in form of so-called eye patterns, which can be observed by repeatedly displaying of many cycles of the signal. The time base of the oscilloscope is triggered by a stable clock signal without any jitter. The edges of the signal form broad stripes, whose width corresponds to the peak-to-peak amplitude of jitter. This is illustrated in figure 6.3. below.

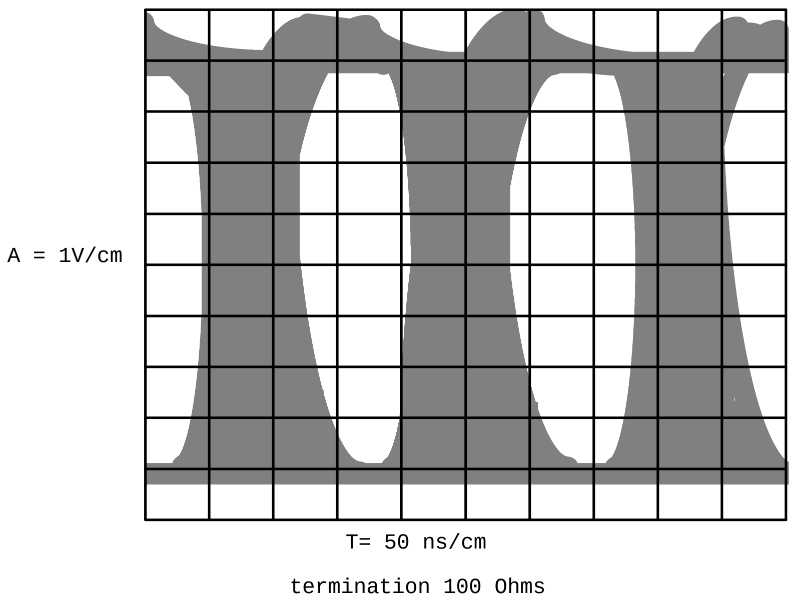


Figure 6.3: Eye pattern of interface signal with 100 Hz jitter modulation

Jitter, at least at higher frequencies, can be reduced by phase lock loop circuits. In order to investigate the jitter characteristic of an interface receiver it can be helpful to generate an adjustable amount of jitter at various frequencies. A circuit to do this is shown in figure 6.4., below.

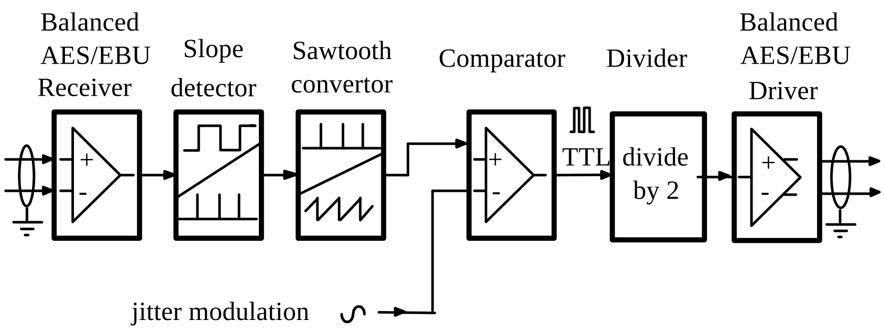


Figure 6.4: Jitter test generator

6.2.5. Output impedance

An accurate measurement of the output impedance at frequencies between 0.1 and 6 MHz is possible only by means of a vector voltmeter or an impedance bridge. When using one of these methods a current has to be fed into the output circuit; therefore the internal signal must be switched off.

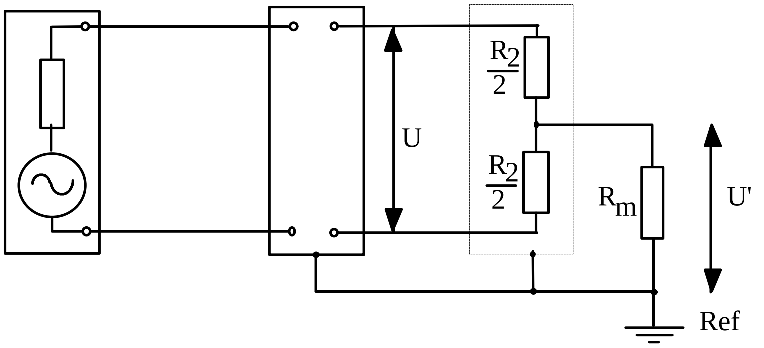
A practical test method is to measure the output voltage of the transmitter using an oscilloscope with and without a termination resistor The impedance can be calculated from the ratio of the two voltages.

6.2.6. Signal amplitude

Figure 6.1 above shows an oscillogram of a pulse form the interface waveform. It is shown how the pulse amplitude, A, should be measured. Namely between two (imagined) lines, which take into consideration any undershoot and overshoot of the pulse.

6.2.7. Balance

There are several possible causes of unbalance of an output circuit, for instance inequality of the internal impedances or different voltages from the two halves of the driver. In principle, the balanced-to-common-mode signal ratio can be measured by a method standardised in IEC 268-2 (1987) which is shown in figure 6.5, below.



$$\text{Balanced – to – common – mode signal ratio } = \frac{U}{U'} \text{ dB}$$

Note: If screened resistors, matched to the required degree of precision (and of suitable value and power rating) are not available, use may be made of a suitable balanced centre tapped winding of an inductor or transformer (repeating coil). In this case the ends of the windings are connected in parallel with a resistor R_2 and the output terminals.

Figure 6.5: Test method of balance measurement, after IEC 268-2

Beware, however, that this IEC publication deals with sound system equipment at audio frequencies and in practice it may be very difficult to guarantee uniformity of the test resistors at higher frequencies due to stray capacitance.

6.2.8. Rise and fall times

Both rise and fall times can be measured directly by means of an oscilloscope, provided, of course, that the risetime of the instrument is better than that of the signal being measured! The method of measurement, between the 10% and 90% levels, is illustrated in the diagram of the pulse wave form in figure 6.1., above.

6.2.9. Data jitter at an output

A simple method to determine the amount of jitter at an output is to measure the eye-pattern of the interface signal at the output terminals using an oscilloscope (see figure 6.3. above). The eigen-jitter of the oscilloscope, caused by inaccuracies of the trigger circuit, has to be considered . More sophisticated techniques may be needed to detect jitter at the sample frequency rate.

6.2.10. Terminating impedance of line receivers

The vector voltmeter and impedance bridge methods mentioned above for output impedance can also be used for the measurement of the input impedance of a receiver. Also, in a similar way as that described above, a rapid measurement can be made using an oscilloscope and a generator with the appropriate source impedance (or padded to the correct value by a series resistor). The output voltage of the generator is measured before and after it is connected to the input terminals of the line receiver. The input impedance can then be simply calculated from the voltages and the known output impedance of the generator.

6.2.11. Maximum input signal levels

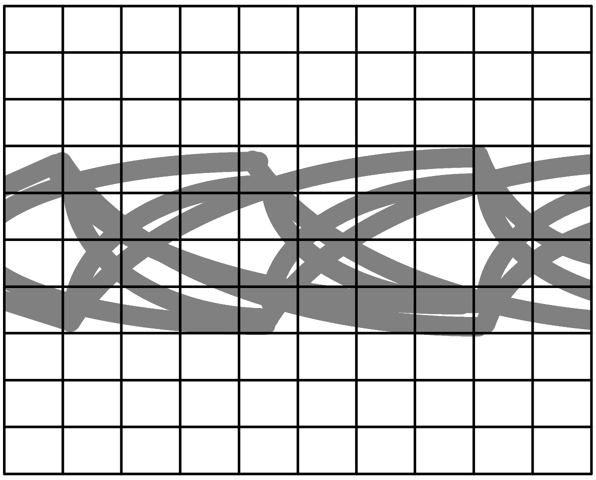
A receiver should function correctly when connected to a generator with an output up to the maximum level of 7 V. If an interface driver which can provide this level (with a termination resistor!) is available, this gives the possibility to confirm that the receiver functions correctly with an input of this voltage. This test may bring additional peace of mind. Note that in the earlier edition of the specification, EBU Tech 3250, the maximum voltage was specified as 10 V. It is not thought that any equipment actually gave an output at this level.

6.2.12. Minimum input signal levels

In order to check the limit of input sensitivity, the jitter generator shown in figure 6.4. above with the addition of a variable attenuator on the output can be used for this test.

Figure 6.6. below illustrates the appearance of an interface signal after passing through 100 m of audio frequency cable (2 x 0.5 mm screened).

A = 1V/cm



T= 50 ns/cm
termination 100 Ohms

Figure 6.6: Eye pattern of an interface signal after 100 m audio cable

This can be compared with the typical signal at the output of a transmitter shown in figure 6.7. below.

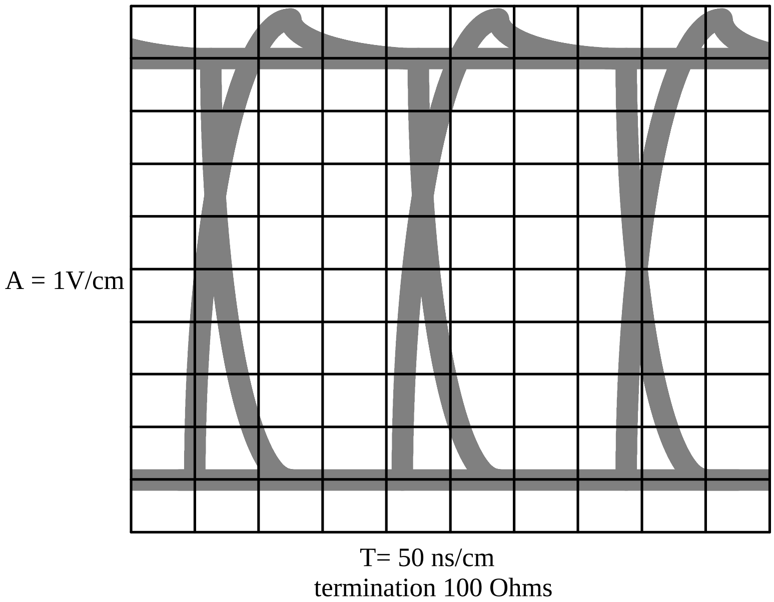


Figure 6.7: Eye pattern of an interface signal at the output of a line driver

6.3. Testing the audio layer

The quality of the audio signal in the interface signal can be tested provided that the audio data can be accessed transparently. Tests can be made on three areas that affect the quality of the coding of the audio:-

- A-D Conversion,
- D-A Conversion,
- Audio processing stages.

The techniques of A-D and D-A conversion are outside the scope of this guide, but in general, faults to be expected from converters vary according to the technology used. They come into the area of laboratory rather than installation testing.

Surprisingly, faults can be consistently detected from simple listening tests using familiar high quality excerpts of sound material. The excerpts in the SQAM disc (EBU Tech. 3253) are particularly suitable. Bit shifting is a useful technique when used to examine the performance with small signals.

Audio processing can produces defects which are can be examined in a similar manner to the defects found in codecs. In addition, commercial test equipment is available which can be used for tests purely in the digital domain. Sometimes it is possible to determine the processing history from clues in the channel status, provided the correct implementations are used. This can sometimes show that "bottlenecks" have been encountered which have restricted the sample size, or that sample rate conversion has taken place. Sample slips are very difficult to confirm and will almost certainly only be revealed by a CRCC error in channel status if at all. The cause of an apparently transparent audio bit stream which in fact contains concealments can be a particular problem in a large systems. Examination of the Channel status or validity flag may help. A high frequency continuous tone audio signal, even if it is audibly masked under programme material, will show up concealments or muting in a quite audible manner. This technique is especially useful if an "audio only" bottleneck has removed all the other information from the interface signal.

A high level pseudo-random pattern in the audio data is often used in automated error detection systems. However this technique is dangerous in audio systems because it is rarely compatible with loudspeaker monitoring, and difficult to use with audio processing of any form. It can only be used on isolated links. Similarly, repetitive single bit transitions or multi tone tests are powerful tools for use for time or spectrum analysis, but the effective audio levels must be controlled to avoid damage to loudspeaker transducers.

6.4. Testing the control layer

A fairly quick test of the control layer, but of necessity an automated one, is to present the system inputs with a cycle of Validity, User bit and Channel status conditions and automatically log the system response both in terms of audio and control layer responses. In operational use all that is needed is to flag any deviation from "normal" channel status or non-valid flags. This can be done, for instance, by displaying error codes visibly on source identification displays.

6.5. Operational testing

Operational testing of interfaces should ideally consist of a rapid confidence check that all the parameters given in Section 6.2, established by acceptance and installation tests, are still in place. In reality, only a small sub-set of essential tests can be performed, and restricted access to various parts of a large installation may mean that only a few of the possible electrical routes can be explored at one time. For example, electrical noise, in balanced or unbalanced form, can be inserted onto accessible input ports in order to test the error margin of that input, but any audio or control faults seem at the output of a system may be due to error extension processes, occurring in items of equipment further down the chain. It is particularly important, therefore, that error amplification factors in the audio and control domains have been explored fully at the installation test stage.

In installations in the future, it may be found that sufficient self-analysis software is available in the equipment to carry out confirmation tests during non-operational time. Equipment which analyses the Channel status or User bits at the system output may be sufficient to flag many failing parameters within individual items of equipment before these manifest themselves as audible faults. Apart from proposals for concealed audio test signals, it is just this efficient handling and display of the control and error data which users would most like to see incorporated into systems employing the interface. Automation of the operational confidence testing aspects of large installations would not only increase the popularity of the interface for such systems, but, before long, may become an essential backbone for future automated broadcasting networks.

Chapter 7

INSTALLATION EXPERIENCES

7.1. Thames Television (UK) London Playout Centre

7.1.1 System description

Thames Television acquired experience in the use of the Digital Audio Interface which derived from their leading role in laboratory developments on the interface dating from 1980 and latterly from the practical installation of a large and integrated distribution system at the Euston Studio Transmission Centre in London.

The system at Euston consists of some 70 quasi 18 bit ADCs which feed 96 central digital distribution amplifiers, DDAs. A number of sources are also already digital and it is expected that this number would grow, with a corresponding reduction in ADC requirements. All the inputs to the DDAs are accessed via standard ¼" jackfields located in the same bays as the DDAs. Outputs from the DDAs are taken directly to the inputs of a number of large TDM matrices which provide simple mixing and switching facilities for feeding the local Thames TV transmitters as well as the ITV network in the UK and other output lines from the building. In addition, for safety, there is a small emergency relay switcher which can be selected by means of a downstream switcher in the event of major system failure. The final outputs from the various matrices are fed, via DDAs, to 20 DACs which are required for audio monitoring and, in the short term, feeding the analogue inputs of the DSIS (dual sound in syncs) link equipment to the local television transmitter and the rest of the UK-ITV network. Direct digital conversion to the NICAM is now a practical possibility, giving a fully digital route from studio direct to viewer. As with the inputs to the DDAs all the principle digital output signals in the system are accessible on ¼" jackfields for maintenance and monitoring purposes.

The whole system is synchronised to a Thames designed reference generator which is in turn locked to the same 10 MHz rubidium source as the video SPGs. EBU Timecode is inserted into the channel status bits, and the reference generator also carries stereo line up tones to EBU R49. All the digital sources are uniquely coded with an appropriate source ID in the channel status bits. Particular care has been taken in the distribution and routing of this signal to the various elements that require it for synchronisation. Elements such as ADCs and TDM matrices have dual control and reference systems.

No particular changes to normal wiring and construction methods were made other than the selection of an 8 twisted pair Belden type of cable for interconnecting the blocks of DDAs into the TDM matrices. The cable specification was confirmed as being nominally 110 ohms at data frequencies. For other cable runs within the Central Apparatus Room individual twisted pair cables with foil screens and drain wires were chosen, rather than the heavier DEF10 type cables.

The main problems encountered in the early stages of the commissioning the system were found to be due to excessive jitter in the reference signal and, unexpectedly, with reflections on the links between the DDAs and the TDM inputs. These cables were found to be of a critical length (approximately 12m) and this, coupled with unnecessary receiver equalisation components, was responsible for signal corruption of these links.

7.2. Eye height measurements

Eye height measurements were made on a number of representative signals within the Central Apparatus Room of the installation. Values of between 4.0 and 5.8 volts were measured across the input terminals of the DAC inputs at the end of up to 16 m of cable. An eye height of 2.9 volts was measured at the end of a 50 m link between an ADC and a DDA input carrying a VTR transmission signal.

As a result of doing 20 or so of these measurements it was discovered that eye measurements were a tedious, subjective and error prone process. The experience suggests that two different operators, not conversant with the measurement technique and using different oscilloscopes could produce significantly different results.

7.3. Error counting

One other valuable approach to the testing of digital audio systems, including those using the AES/EBU interface, is the Error Counting technique. The Audio Precision Dual Domain test set is capable of generating a number of Pseudo Random Number Sequence test signals in the digital domain. These can be fed through the device under test and back to the test set. The AP test set is able, subsequently, to lock to the input signal and compare it to the output of its generator and provide if desired, a total count of errors against time This technique is particularly valuable in determining the transparency of digital equipment such as DAT recorders and the PCM tacks of MII VTRs. Clearly this technique can only be used if the device under test is transparent. This requirement precludes using this method to test any device which alters the signal in any way, e.g. digital mixing desks and effects devices unless these can be set for a direct-through mode without any processing.

7.4. Further comments by Brian Croney, Thames TV

Certainly the tightening of the specification to reduce the termination impedance to 110 ohms will help to ensure a more reliable point to point transmission system. Only when the characteristics of all interactive elements of the link are tightly controlled with it be possible to use margining devices and enumerated the results. It may well be that improvements in the specification of the circuit elements that comprise the transmitter part of the interface will also enable a much greater communication distance to be reliably achieved.

It may also be worth considering whether it might be possible to implement a digital equivalent of the Insertion Test Signals applied to video signals in order to quantify the error rate present in any given interface or system. This may take the form of a low data rate PRNS encoded in the user bits or at higher rates if the auxiliary bits are not required. The receiver/analyser, suitably programmed to anticipate the encoded PRNS, would be set to ignore Validity and CRCC error masked words and thus quantify the absolute error rates to which any link or part thereof was subject.

A means of exploring the range of satisfactory operation of receivers and systems when subject to a varying duty cycle of sample frequency would be useful. This would equate with the concept of digital "wow" whilst the existing specification for data jitter (for transmitters) equates with "flutter". Some of the very early (pre-commissioning) problems with the Euston system might possibly be categorised in this way.

Appendix

Electromagnetic Compatibility

1. Background to EMC regulations

Due to various initiatives from the European Union, EU, and its predecessor the European Commission, the whole subject of EMC (Electromagnetic Compatibility) has come to the fore of importance when the design of new equipment is contemplated.

A free movement of goods lies at heart of the drive to create a single market in Europe. All European Union countries have laws on product safety and so on. Differences in these laws can cause technical barriers to trade. In May 1985 the then EC ministers agreed on a "New Approach to Technical Harmonisation and Standards" to tackle this long standing problem to business. "New Approach" Directives set out "Essential requirements" (for safety for example), written in general terms, which must be met before products can be sold anywhere within the EC. European "Standards" fill in the detail and are the main way for businesses to meet the "essential requirements". Products meeting these requirements carry the "CE" mark.

Under the EC EMC Directive, almost all electrical or electronic products made or sold in Europe must:
- be so constructed that they do not cause excessive electromagnetic interference and are not unduly affected by electromagnetic interference;
- in some cases, such as RF transmitters, be subject to type-examination by an approved body;
- carry a "CE" mark.

All products should be subject to satisfactory testing against approved EMC standards. Where no approved standard is available for the product, or the manufacturer applies a standard only in part, compliance with the essential requirements of the Directive may be demonstrated by the preparation of a technical construction file. This file would be drawn up by the manufacturer or his representative in the Community, and must include a report or certificate obtained from a competent body.

CENELEC (European committee for Electrotechnical standardisation) is developing about 30 new standards, and revising about 120 existing generic and product related standards on a phased timetable. Until all the relevant standards are in place, transitional arrangements will apply. In the absence of the relevant European Standard, a national standards approved for the purpose by the European Union may be used instead.

On a world-wide scale, all developed countries insist that imports meet certain minimum performance criteria. The original manufacturer must ensure that his products meets the standards in force in the target market. Europe has taken on the challenge of developing and setting a workable range of harmonised EMC standards. These are designed not only to limit the emissions from products but also cover their immunity performance.

2. Generic EMC standards

The new European standards are all prefixed with the letters EN (Euro-Norm). Items not covered by specific or product related standards are subject to the essential requirements set out in the so called Generic Standards. There are two of these:-

- EN 50 081-1 which covers Emissions from equipment manufactured for domestic, commercial or light industrial environments.

- EN 50 082-1 which covers the Immunity of equipment manufactured for domestic, commercial or light industrial environments.

There two parallel standards which are suffixed "-2" which cover equipment used in a heavy industrial environment. The limits set for these latter standards are more relaxed or less stringent than those set for the domestic and commercial environments. It is the more stringent set that apply to broadcasting premises.

The Generic Emission standard cover disturbances in the frequency range 0 Hz to 400 GHz although no limit figures appear to be specified above 1 GHz at present. The standard contains sections that discuss the scope, objective, definitions, description of locations, conditions during measurement, documentation and applicability, before presenting the mandatory tests and limits of acceptability. Finally there is an "informative annex" which includes additional tests for possible inclusion in the standard at a later date. These standards all refer across to other standards either in part or whole. These original standards were produced to cover:

- specific types of equipment, (e.g. EN 55 022 Information Technology Equipment)
- specific phenomena, (e.g. EN 60 555 (parts 1 to 3) Low frequency disturbances to the electric power system).

The Generic Emission standard covers two main areas:

- Radiated emissions from the enclosure and/or connecting cables,

- Conducted emissions from the connecting cables, in particular through the power supply.

Radiated emissions and limits are covered by EN 55 022 Class B and conducted emissions are covered in sections of three different standards:-

- EN 60 555-2 and EN 60 555-3
- EN 55 022 Class B
- EN 55 014

The standards contain graphs which show these limits. The generic immunity standard is presented in a similar style but contains a section on performance criteria. This provides guidelines to the classification of acceptable performance degradations which might be expected to take place when the equipment is adversely affected by its operating environment. The following mandatory immunity tests are required:-

- RF field 27 to 500 MHz at 3 V/m (IEC 801-3),
- Electrostatic discharge of 8 kV into case of equipment (IEC 801-2),
- Fast transients induced into cables (IEC 801-4).

The Informative Annex covers 9 other types of tests which include magnetic field and mains disturbances.

3. Product related EMC standards

Where there is a relevant dedicated EMC standard for a product or product-family, then it should take precedence over the generic standard. Indeed, much effort is presently being expended in preparing product related standards, especially in areas where the generic standards are not very satisfactory. A case in point is the audio electronics, where the industry has great difficulty in reconciling some aspects of the generic standards, for instance in the design of suitable low level microphone pre-amplifiers. In the UK a group drawn from the Professional Audio and Video Industries has been formed to propose realistic EMC standards applicable to its products. Progress in preparing these standards is fairly slow and even then it will take several years before the standards are likely to be accepted by the Commission of the EU. It does seem likely that as soon as the standard is agreed at industry level then it will become adopted in advance of its final ratification.