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Digital Audio Broadcasting (DAB); Conformance Testing for DAB Audio

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Foreword

This Technical Specification (TS) has been produced by the Joint Technical Committee (JTC) Broadcast of the European Broadcasting Union (EBU), Comité Européen de Normalisation ELECtrotechnique (CENELEC) and the European Telecommunications Standards Institute (ETSI).

NOTE 1: The EBU/ETSI JTC Broadcast was established in 1990 to co-ordinate the drafting of standards in the specific field of broadcasting and related fields. Since 1995 the JTC Broadcast became a tripartite body by including in the Memorandum of Understanding also CENELEC, which is responsible for the standardization of radio and television receivers. The EBU is a professional association of broadcasting organizations whose work includes the co-ordination of its members' activities in the technical, legal, programme-making and programme-exchange domains. The EBU has active members in about 60 countries in the European broadcasting area; its headquarters is in Geneva.

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EUREKA Project 147

EUREKA Project 147 was established in 1987, with funding from the EC, to develop a system for the broadcasting of audio and data to fixed, portable or mobile receivers. Their work resulted in the publication of a European Standard, ETS 300 401 [2], for DAB (note 2) which now has worldwide acceptance. The members of the Eureka 147 Project are drawn from broadcasting organizations and telecommunication providers together with companies from the professional and consumer electronics industry.

NOTE 2: DAB is a registered trademark owned by one of the EUREKA 147 partners.

Introduction

ETS 300 401 [2] specifies the coded representations of audio information to be transmitted over the DAB system. This allows for large flexibility, in terms of bitrate efficiency and protection against errors for transmission in an error prone environment. The flexibility is obtained by including parameters in the bitstream that define the characteristics of coded bitstreams. Examples are the audio sampling frequency and bitrate parameters.

The present document specifies a test procedure and defines test bitstreams, which can be used to verify whether bitstreams and decoders meet the requirements as specified in ETS 300 401 [2]. These tests can be used for various purposes such as:

- manufacturers of encoders, and their customers, can use the tests to verify whether the encoder produces valid bit streams;

- manufacturers of decoders and their customers can use the tests to verify whether the decoder meets the requirements specified in ETS 300 401 [2] for the claimed decoder capabilities.

1 Scope

The present document specifies how tests can be designed to verify whether bitstreams and decoders meet the requirements specified in ETS 300 401 [2]. In the present document, encoders are not addressed specifically. An encoder may be considered to be a DAB audio encoder if it generates bitstreams compliant with the syntactic and semantic bitstream requirements specified in ETS 300 401 [2].

The characteristics of a bitstream such as the sampling frequency, mode and bit rate define the subset of the standard that is exploited in the bitstream. Decoder characteristics define the properties and capabilities of the applied decoding process. An example of a property is the accuracy with which the PCM audio signal is reconstructed. The capabilities of a decoder specify which coded bitstreams the decoder can decode and reconstruct, by defining the subset of the standard that may be exploited in decodable bitstreams. A bitstream can be decoded by a decoder if the characteristics of the coded bitstream are within the subset of the standard specified by the decoder capabilities.

Procedures are described for testing the conformance of decoders to the requirements defined in ETS 300 401 [2]. Bitstreams for conformance testing, which are available from Eureka 147, are described briefly, and there is an annex describing a test for the operation of dynamic range control (DRC) in receivers. The provision of DRC is optional.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

[1] ISO/IEC 11172-4 (1993): "Information technology - Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbit/s - Part 4: Compliance testing".

[2] ETSI ETS 300 401: "Radio broadcasting systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers".

[3] Gilchrist, N.H.C., 1994: "DRACULA: Dynamic range control for broadcasting and other applications". BBC Research & Development Report No. 1994/13.

[4] Gilchrist, N.H.C., (1994): "DRACULA: Eine Dynamiksteuerung für Rundfunk- und andere Anwendungen". 18 Tonmeistertagung (Karlsruhe, 1994). Main text in English.

[5] Hoeg, W., Gilchrist, N.H.C., Twietmeyer, H. and Jünger, H., (1994): "Dynamic range control (DRC) and music/speech control (MSC) programme-associated data services for DAB". EBU Technical Review No. 261 (Autumn 1994) pp. 56-70.

3 Symbols

The mathematical operators used to describe the present document are similar to those used in the C programming language. However, integer divisions with truncation and rounding are specifically defined. Numbering and counting loops generally begin from zero.

For the purposes of the present document, the following symbols apply:

3.1 Relational operators

=	Equal to

3.2 Arithmetic operators

+	Addition
-	Subtraction (as a binary operator) or negation (as a unary operator)
*	Multiplication
^	Power
/	Division
√	Square root
sqrt	Square root

4 Audio conformance testing

Test signals applied to audio decoders has to be free of errors. The purpose of each test is described in subclause 4.5.

4.1 Audio decoder tests

To test audio decoders, Eureka 147 supplies a number of coded test bitstreams. Testing is performed by comparing sample-by-sample the digital output of a decoder under test with a reference PCM bitstream, also supplied by Eureka 147.

The Eureka 147 coded test bitstreams carrying noise signals (test02, test03 and test08) may be used both for testing the computational accuracy of decoders and for determining the RMS difference between the decoded PCM bitstream and the reference PCM bitstream. These bitstreams contain noise signals with an amplitude of -10 dB relative to full scale.

The calculations, which follow to determine the RMS difference, are normalized to full scale so that the output of the decoder lies in the range -1,0 to +1,0. The reference PCM bitstreams associated with the coded test bitstreams have a precision (P) of 24 bit, i.e. the MSB represents the value of -1, the MSB-1 bit represents the value of +1/2, etc.

MSB	-1/2^0	=	-1
MSB-1	1/2^1	=	1/2
MSB-2	1/2^2	=	1/4
MSB-23	1/2^23	=	1/8 388 608

The output signal of the decoder under test requires to be in the same format. In the case that the output of the decoder has a precision of P' bits and if P' is smaller than 24, then the values for the bits between the positions P'-1 and 24 shall be set to zero.

MSB	-1/2^0	=	-1
MSB-1	1/2^1	=	1/2
MSB-(P'-1)	1/2^P'	=	1/2^P'
MSB-P'	0	=	0
MSB-23	0	=	0

In the next step the difference (diff) of the samples of these signals has to be calculated. If two channels are present (in case of stereo and joint-stereo) both channels shall be tested independently. The total number of samples for each channel is N.

diff(n) = output sample from decoder under test(n) - corresponding sample from reference signal (n)'

for n = 1 to N

The values of all difference samples shall be squared, summed, divided by N and then the square-root shall be calculated. This calculation finally gives the RMS difference.

RMS = sqrt(1/N*sum(diff^2))

To be called a **full accuracy DAB audio decoder**, the decoder digital output should be such that the RMS difference between the output of the decoder under test and the reference PCM bitstream is less than $2^{-15}/\sqrt{12}$.

In addition, the difference between the individual decoded samples and the corresponding reference samples (i.e. diff(n), as defined above) shall have a maximum absolute value of at most 2^{-14} relative to full-scale (ISO/IEC 11172-4 [1]).

To be called a **limited accuracy DAB audio decoder**, the decoder digital output should be such that the RMS difference between the output of the decoder under test and the reference PCM bitstream is less than $2^{-11}/\sqrt{12}$.

The tests described above verify only the computational accuracy of an implementation. The criteria for decoder accuracy apply only for tests with the noise signals (test02, test03 and test08).

4.2 Decoder testing arrangement

For conformance testing it is necessary that the decoder under test provides a digital output of the reconstructed audio signal. The test arrangement is shown in figure 1.

An audio decoder within a DAB receiver cannot be accessed directly; the audio test bitstreams have to be transported using the DAB signal. In practice, this means that the DAB receiver has to be connected to a DAB chain. Error free bitstreams at the input of the audio decoder shall be ensured, and verified by monitoring the BER.

A linear PCM (i.e. decoded digital) output shall be available from the DAB receiver, and the bitstream at the output of the decoder and the reference output shall be accurately aligned. If the receiver does not have a linear PCM output, a test point should be provided within the receiver, from which a decoded digital audio signal may be taken and converted into a form suitable for connection to a workstation. The receiver manufacturer should ensure that such a test point is available and clearly identified in receivers submitted for testing, and (ideally) provide a conversion module suitable for interfacing to the workstation.

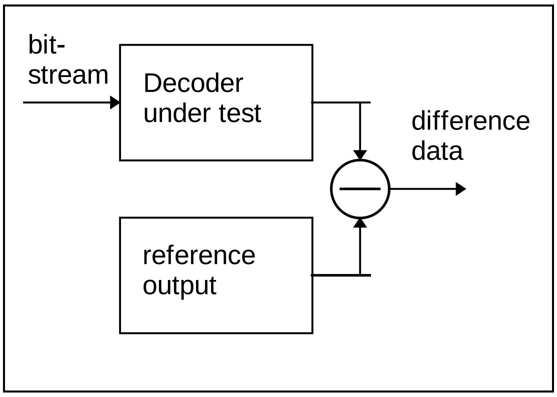

Figure 1: Test configuration for a decoder with a digital output

In receivers where D/A conversion is performed with a clock which is locked to the received bit clock, any buffering in the decoder should convey the decoded PCM signal to the D/A converter unchanged except for the timing. In receivers using a free-running clock for D/A conversion, the management of buffering in the decoder may cause sample slipping to occur. It may be possible to prevent sample slipping by taking the D/A clock and locking the master clock for the DAB chain to this. To facilitate this, the receiver manufacturer should ensure that a suitable test point from which to obtain this clock is available in receivers submitted for testing.

When the DAB audio is sampled at 24 kHz (LSF) some receivers may convert the sampling frequency to 48 kHz in the decoder prior to D/A conversion. Depending upon the nature of the sampling frequency up-conversion, it may, or may not, be possible to run the conformance tests with 24 kHz sampling by sub-sampling the decoded signal.

4.3 Audio decoder characteristics

A DAB Audio decoder may support only specific values, or a specific range, or a specific combination of values or ranges of the following parameters in audio bitstreams. These parameters are encoded directly or indirectly in the bitstream.

a) bitrate_index (see subclause 7.2.1.3 of ETS 300 401 [2]);

b) id (see subclause 7.2.1.3 of ETS 300 401 [2]);

c) mode (see subclause 7.2.1.3 of ETS 300 401 [2]);

d) mode_extension (see subclause 7.2.1.3 of ETS 300 401 [2]);

e) dynamic range control (DRC) (see subclause 7.4.1 of ETS 300 401 [2]).

A DAB Audio decoder shall be capable of handling error protection (crc_check and scf_crc_check) in order to conceal transmission errors. The method to be applied in the case of transmission errors is not standardized and depends on the specific implementation of a DAB audio decoder.

The provision of dynamic range control (DRC) is not mandatory and the implementation of DRC in the decoder may differ between decoders. A method for testing dynamic range control in receiver decoders is described in annex A.

Conformance of an audio decoder to ETS 300 401 [2] requires that the output signal of the decoder is reconstructed accurately.

Decoders that support all combinations are designated as Full DAB audio decoders. Other decoders may be able to support at least one but not all of the options defined above.

4.4 Requirements for audio bitstreams

Bitstreams for DAB, including test bitstreams, shall conform to the following requirements:

layer: the Layer field shall not be encoded with the binary values '01', '11'.

bit_rate_index: the bitrate field shall not be encoded with the binary values '0000' and '1111'.

sampling_frequency: the sampling frequency field is fixed to the binary value '01'.

padding: since no padding is necessary the value is fixed to '0'.

emphasis: since no emphasis is allowed this field is fixed to the binary value '00'.

protection_bit: this bit is fixed to '0' since error protection is mandatory, hence the correct CRC16 value shall be in the crc_check field.

scalefactor: the scalefactor[sb][p] or scalefactor[ch][sb][p] field shall not refer to index 63.

samples: for un-grouped samples the coded representation of subband samples the valid range is from zero up to (nlevels -2), where nlevels equals the number of levels used for quantization of that sample, that is the coded representation of a sample shall not consist of a bitstring with only '1's. For grouped samples the range shall be from zero up to 26 if nlevels equals 3, from zero up to 124 if nlevels equals 5, and from zero up to 728 if nlevels equals 9.

frame length (1): the bit allocation and the scalefactor select information shall be such that the total number of bits for a frame does not exceed the frame length.

frame length (2): the frame length shall equal the number of slots times the slot size.

The audio test bitstreams available from Eureka 147 and described in subclause 4.5 conform to these requirements.

4.5 Descriptions of the audio test bitstreams

All bitstreams used to test the decoder capabilities are error free, fully compliant DAB Audio bitstreams as defined in ETS 300 401 [2]. All bitstreams carry a correct F-PAD, ISO-CRC and ScF-CRC field some bitstream are prepared for carrying X-PAD data field with variable length. This data is set to zero. Testing of the X-PAD field and its content is beyond the scope of the present document.

The bitstreams can be obtained from ftp://ftp.worlddab.org/bitstreams.

Compressed bitstreams are provided, according to ETS 300 401 [2]. Detailed descriptions of the bitstreams are furnished below. The following file name extensions are used to identify different parts:

testXX.mpg: DAB audio bitstream;

testXX.dec: reference output;

testXX.txt: description file.

Where XX stands for the bitstream number as indicated in table 1.

Table 1: Characteristics of the test sequences

feature	Bitstream number									
	01	02	03	04	05	06	07	08	09	10
sampling frequency	48	48	48	48	48	48	24	24	24	48
id	1	1	1	1	1	1	0	0	0	1
bitrate (in kbit/s)	192	384	256	256	48	96	80	144	160	192
mode	st/js	st	st	st	si	js	st/js	st	si	st
number of frames	50	177	177	65	50	50	50	105	65	749
F-PAD: speech	-	-	-	-	-	-	-	-	-	-
F-PAD: music	-	-	-	-	x	x	-	-	-	-
F-PAD:DRC	x	-	-	-	-	-	x	-	-	x
X-PAD	-	x	x	x	-	-	-	x	x	-
NOTE: st: stereo; js: joint stereo; si: single channel.										

4.5.1 Test bitstream #1

Specification: Channel #1: sine 0,4; 2,0; 3,2; 5,3; 6,8; 8,3; 9,8; 11,4; 13,0 and 15,0 kHz; -22,3 dB.

- Channel #2: sine 0,3; 2,5; 3,5; 4,0; 7,0; 8,0; 10,5; 11,5; 14,0 and 20,0 kHz; -22,3 dB.
- The mode is switched consecutively through the following set: stereo, joint stereo bound #1 to joint stereo bound #4.
- The bitstream carries DRC information which covers the complete range of possible gain factors. The DRC data for the frames #5 to #47 is: 0, 3, 6, ..., 60, 63, 60, ..6, 3, 0.

Purpose: Check that the decoder can handle the worst case of switching the mode between stereo and joint stereo. Furthermore the DRC implementation can be tested.

4.5.2 Test bitstream #2

Specification: Noise signal at a level of -20 dB. All possible bit allocations are used. The bitstream carries a variable length X-PAD field.

Purpose: To check audio decoder conformance (as described in subclause 4.1 of the present document) at 48 kHz sampling frequency and to check that the decoder can handle the worst case of switching the bit allocation. Furthermore the capability of handling X-PAD fields of variable length is tested.

4.5.3 Test bitstream #3

Specification: Noise signal at a level of -20 dB. All possible bit allocations are used. The bitstream carries a variable length X-PAD field.

Purpose: To check audio decoder conformance (as described in subclause 4.1 of the present document) at 48 kHz sampling frequency and to check that the decoder can handle the worst case of switching the bit allocation. Furthermore the capability of handling X-PAD fields of variable length is tested. (This test is similar to test #2, but at 256 kbit/s. It may be used, for example, to test receivers which are incapable of operating at the maximum specified bit rate of 384 kbit/s).

4.5.4 Test bitstream #4

Specification: A sine tone at 500 Hz, at a level between -0,08 dB and $-\infty$. All possible scalefactors occur switched each frame. The bitstream carries a variable length X-PAD field.

Purpose: Check that the decoder can handle all possible scalefactor values. Furthermore the capability of handling X-PAD fields of variable length is tested.

4.5.5 Test bitstream #5

Specification: Three sine tones at 250 Hz, 800 Hz and 2 kHz, at a level of -10 dB. The F-PAD field indicates the content as 'music'.

Purpose: Check that the decoder can handle lower bitrate modes which requires a different table set in the decoder.

4.5.6 Test bitstream #6

Specification: Three sine tones at 250 Hz, 800 Hz and 2 kHz, at a level of -10 dB. The F-PAD field indicates the content as 'music'.

Purpose: Check that the decoder can handle lower bitrate modes which requires a different table set in the decoder.

4.5.7 Test bitstream #7

Specification: Channel #1: sine 0,4; 2,0; 3,2; 5,3; 6,8; 8,3; 9,8 and 11,0 kHz; -22,3 dB.

- Channel #2: sine 0,3; 2,5; 3,5; 4,0; 7,0; 8,0 and 10,5 kHz; -22,3 dB.
- The mode is switched consecutively through the following set: stereo, joint stereo bound #1 to joint stereo bound #4.
- The bitstream carries DRC information which covers the complete range of possible gain factors. The DRC data for the frames #5 to #47 is: 0, 3, 6, ..., 60, 63, 60, ..6, 3, 0.

Purpose: Check that the decoder can handle the worst case of switching the mode. Furthermore the DRC implementation can be tested.

4.5.8 Test bitstream #8

Specification: Noise signal at a level of -20 dB. All possible bit allocations are used. The bitstream carries a variable length X-PAD field.

Purpose: To check audio decoder conformance (as described in subclause 4.1 of the present document) at 24 kHz sampling frequency and to check that the decoder can handle the worst case of switching the bit allocation. Furthermore the capability of handling X-PAD fields of variable length is tested.

4.5.9 Test bitstream #9

Specification: A sine tone at 250 Hz, at a level between -0,08 dB and -∞. All possible scalefactors occur switched each frame. The bitstream carries a variable length X-PAD field.

Purpose: Check that the decoder can handle all possible scalefactor values. Furthermore the capability of handling X-PAD fields of variable length is tested.

4.5.10 Test bitstream #10

Specification: A sine tone at 1 kHz, at 3 different levels (-26 dB, -20 dB, -14 dB). The bitstream carries DRC information that is equivalent to a gain adjustment of 0 dB, 12 dB and 6 dB respectively (see annex A). The gain adjustments in the receiver should occur at exactly the same times as the tone changes in level.

Purpose: Check DRC according to the method described in annex A.

Annex A (informative): A Test Method for Dynamic Range Control in Digital Audio Broadcasting Receivers

DAB potentially allows the listener to adjust the dynamic range of the programme material to suit the particular listening environment. Information in the Programme Associated Data (PAD) fields of the coded audio allow the receiver to adjust the audio gain for each coded frame. This feature is called Dynamic Range Control (DRC). Programmes can therefore be broadcast with a dynamic range suitable for a good listening environment, and the listener can choose to reduce the dynamic range if the listening environment is not ideal (see [3], [4] and [5]).

For a correct implementation of DRC, the receiver has to apply gain adjustment to the audio output at the correct time. Timing errors produce an incorrectly adjusted audio output level, which are particularly evident with large changes in audio level over a short period of time.

This annex describes a test method developed by the BBC to check receivers with DRC for timing errors. The test arrangement does not require an audio coder or PAD generator, and the test bit stream conforms to the DAB standard ETS 300 401 [2]. Consequently, there is no risk that the source of the signal can introduce any mistiming of the DRC data relative to the audio. Examination of the analogue audio output of the receiver with an oscilloscope is normally sufficient to determine whether or not the timing of the DRC process is correct with respect to the audio.

DRC data is included in the coded audio bitstream of the DAB signal, which uses a 24 ms frame structure. Therefore, once per 24 ms a new DRC value is available to the decoder as the PAD from a frame is decoded. This value shall be applied to the decoded audio derived from the following audio frame. Figure A.1 shows this timing relationship.

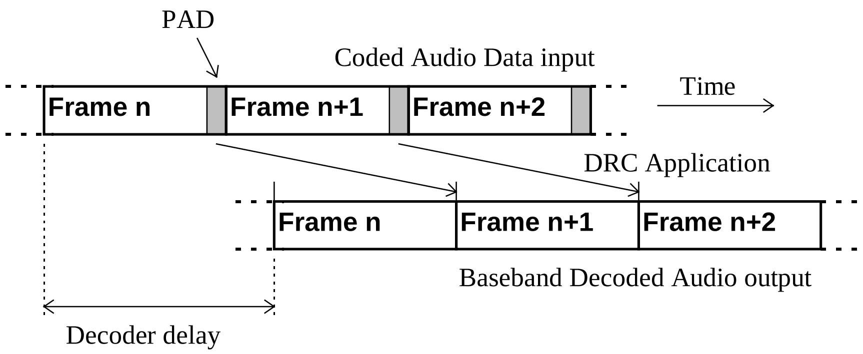

Figure A.1: Timing Relationship Between Coded Audio Data, Baseband Audio and DRC Application

Test Method

To determine the timing relationship between the decoded audio and DRC gain adjustment, a reference-coded bitstream is used. Figure A.2 shows the test waveform envelope, test DRC values and decoded output with correct DRC timing.

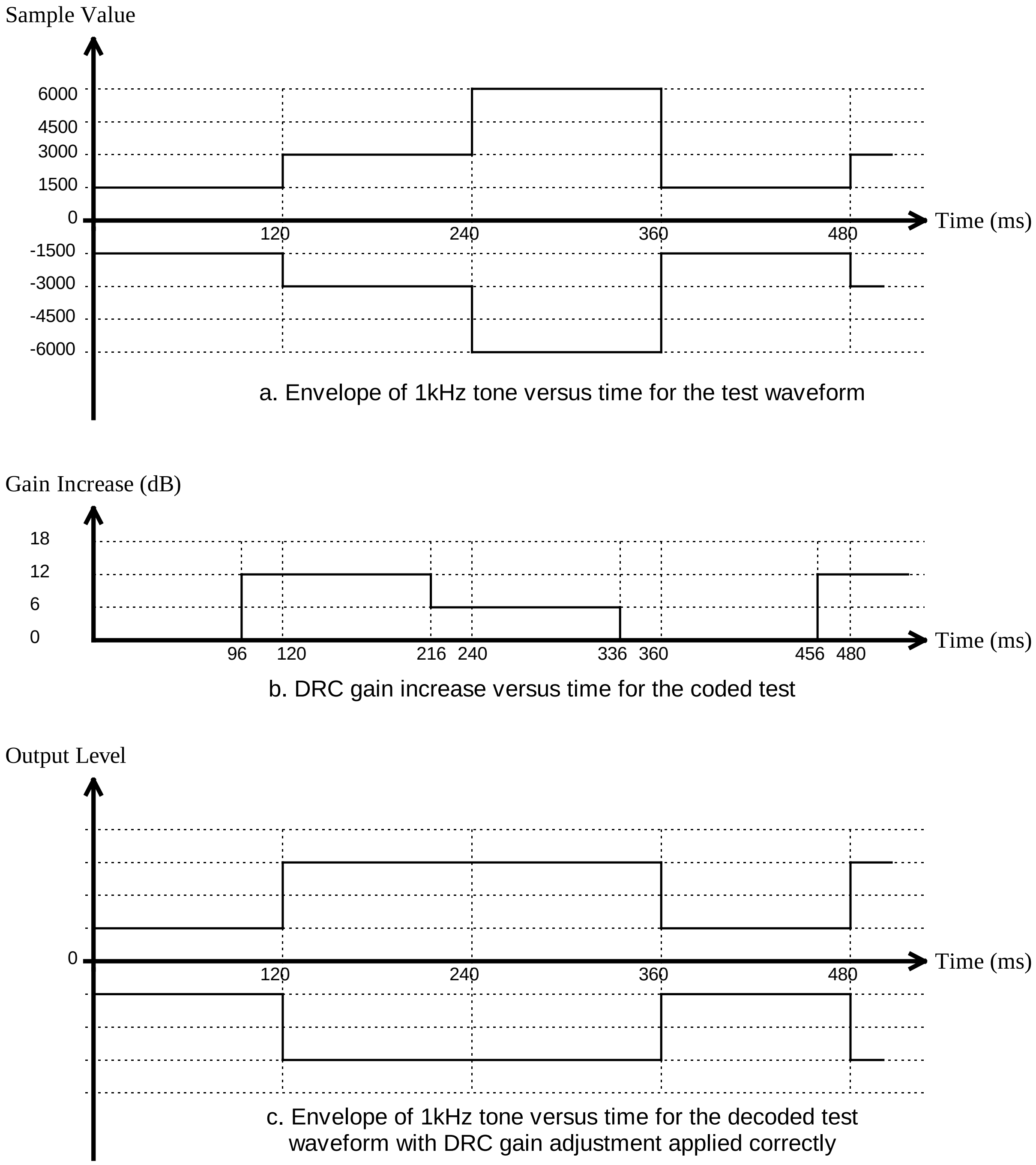

Figure A.2: DRC Test Waveforms

Figure A.2a shows the envelope of the 1 kHz tone used to create the reference bitstream. Between 0 and 120 ms (five 24 ms frames) the tone is of a low level, which then doubles in level at 120 ms and doubles again in level at 240 ms. This pattern is repeated 50 times in the reference bitstream.

Figure A.2b shows the variation in DRC value in the PAD of the reference bitstream. Note the offset of 1 frame between changes in DRC data and changes in audio level, as the DRC value in the PAD of each frame should be applied to the audio from the next frame after decoding.

Figure A.2c shows the output from an audio decoder which has the correct timing between decoded audio and DRC gain changes. Between 0 and 120 ms, 0 dB of gain is applied to the output signal. Between 120 ms and 240 ms 12 dB is gain is applied, and 6 dB of gain is applied between 240 and 360 ms. The effect of the gain change is to make the level equal between 120 and 360 ms.

The vertical scale on figure A.2c cannot be predicted, as this will vary with the particular implementation of DRC gain adjustment. For example, some audio decoders may only be able to apply a variable attenuation to the output level.

If a timing error occurs between the DRC gain adjustment and audio decoding, deviations from the square shape envelope of the 1 kHz tone will be visible, appearing at the 120, 240 and 360 ms points.

Figure A.3 shows the test arrangement required for DRC timing measurements.

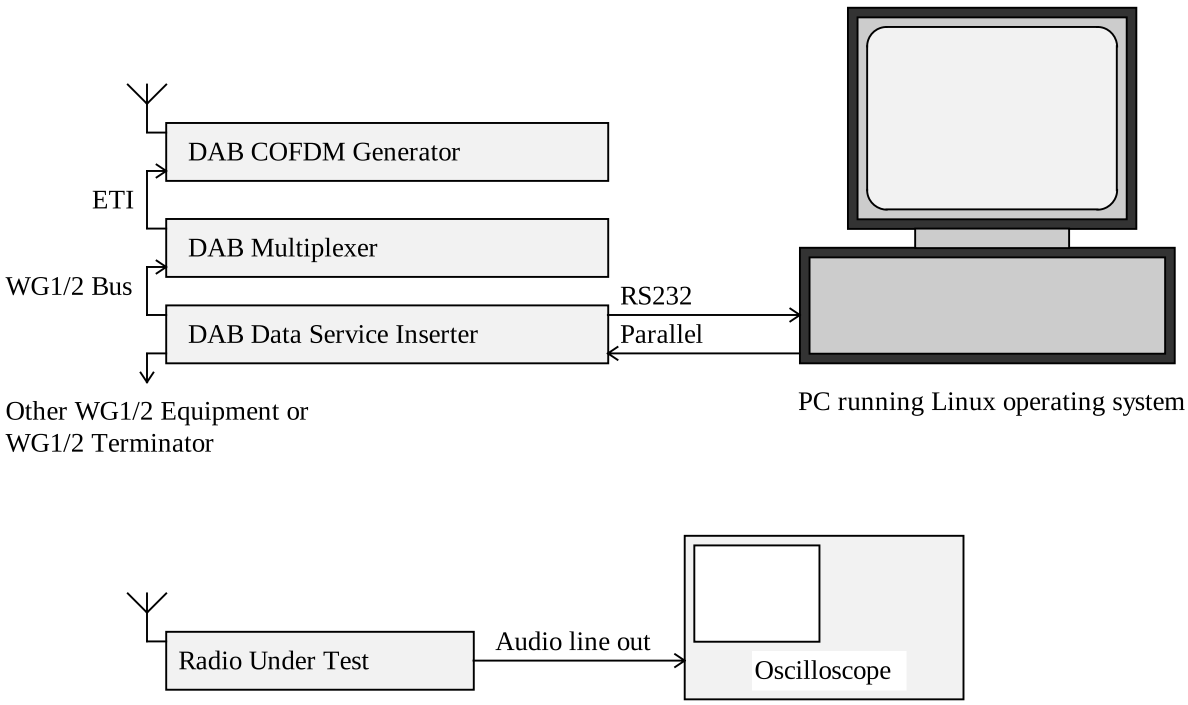

Figure A.3: Test arrangement for DRC timing measurements

To play a test sequence through the receiver under test, the following steps are required:

1) Set up a configuration on the DAB multiplexer containing a suitable audio subchannel, i.e. a stereo 192 kBits/s channel.

2) Tune the receiver to the ensemble from the COFDM generator and check that tuning and synchronization are successful.

3) Play the coded audio file into the multiplexer.

4) Observe the audio output from the receiver on the oscilloscope with DRC turned on and with DRC turned off. The oscilloscope can be triggered on the rising edge of the louder tone after 120 ms, which is present if DRC is turned on or off. Use of a storage oscilloscope in 'single trigger' mode may ease the measurement.

5) Timing errors can be observed as deviations of the tone envelope from that shown in figure A.2c. Examples of oscillograms obtained during testing at the BBC are shown in figures A.4 to A.6.

Note that for the 24 kHz sampling frequency test sequence, the sample rate is half that described up to this point, therefore the frame length becomes 48 ms and the decoded tone has a frequency of 500 Hz. The envelope of the decoded audio remains the same amplitude, but lasts twice as long.

Sample Test Results

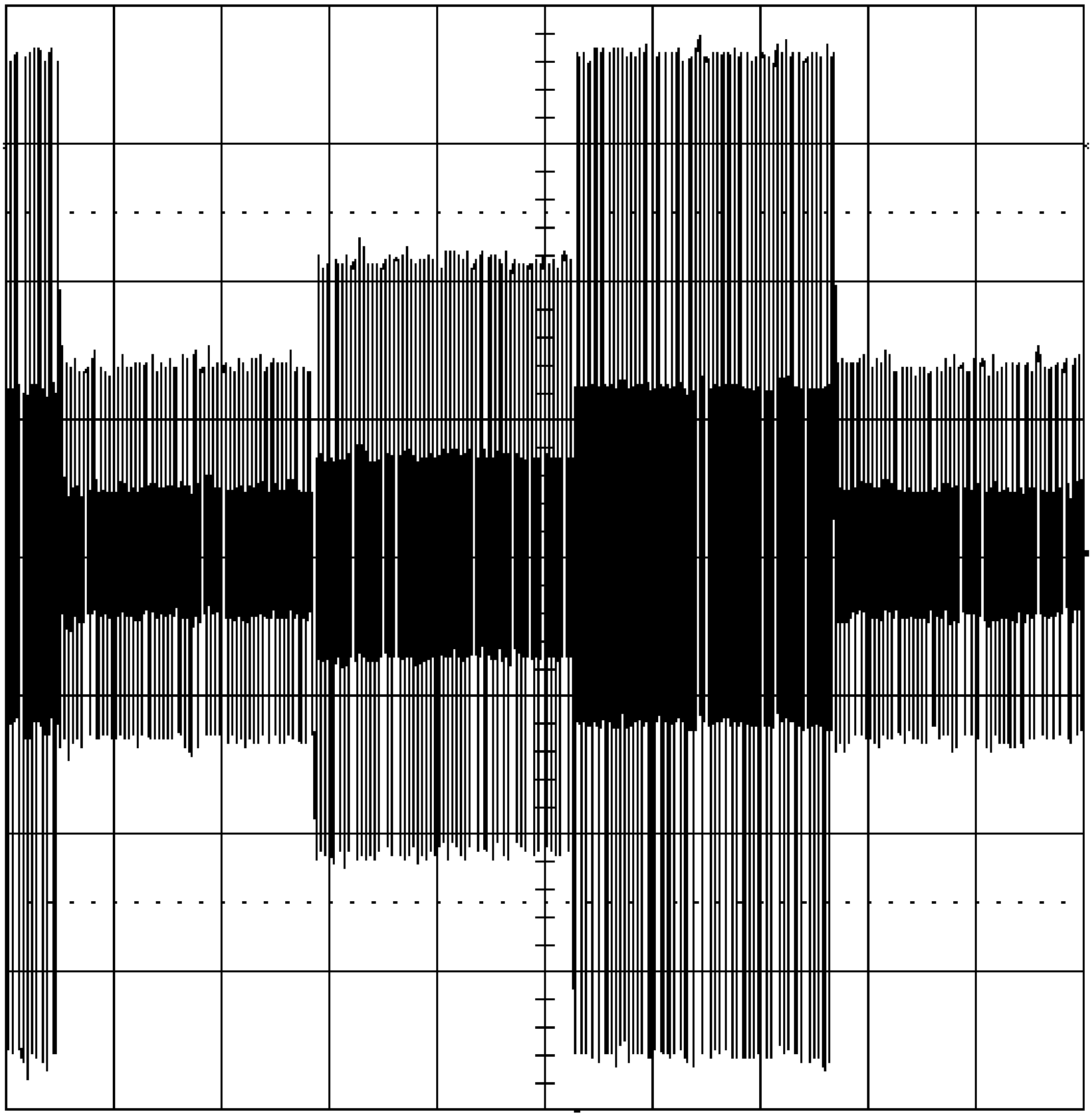

**Figure A.4: Decoded test signal with DRC turned off.
Horizontal scale is 50 ms/div. (Audio sampling frequency is 48 kHz)**

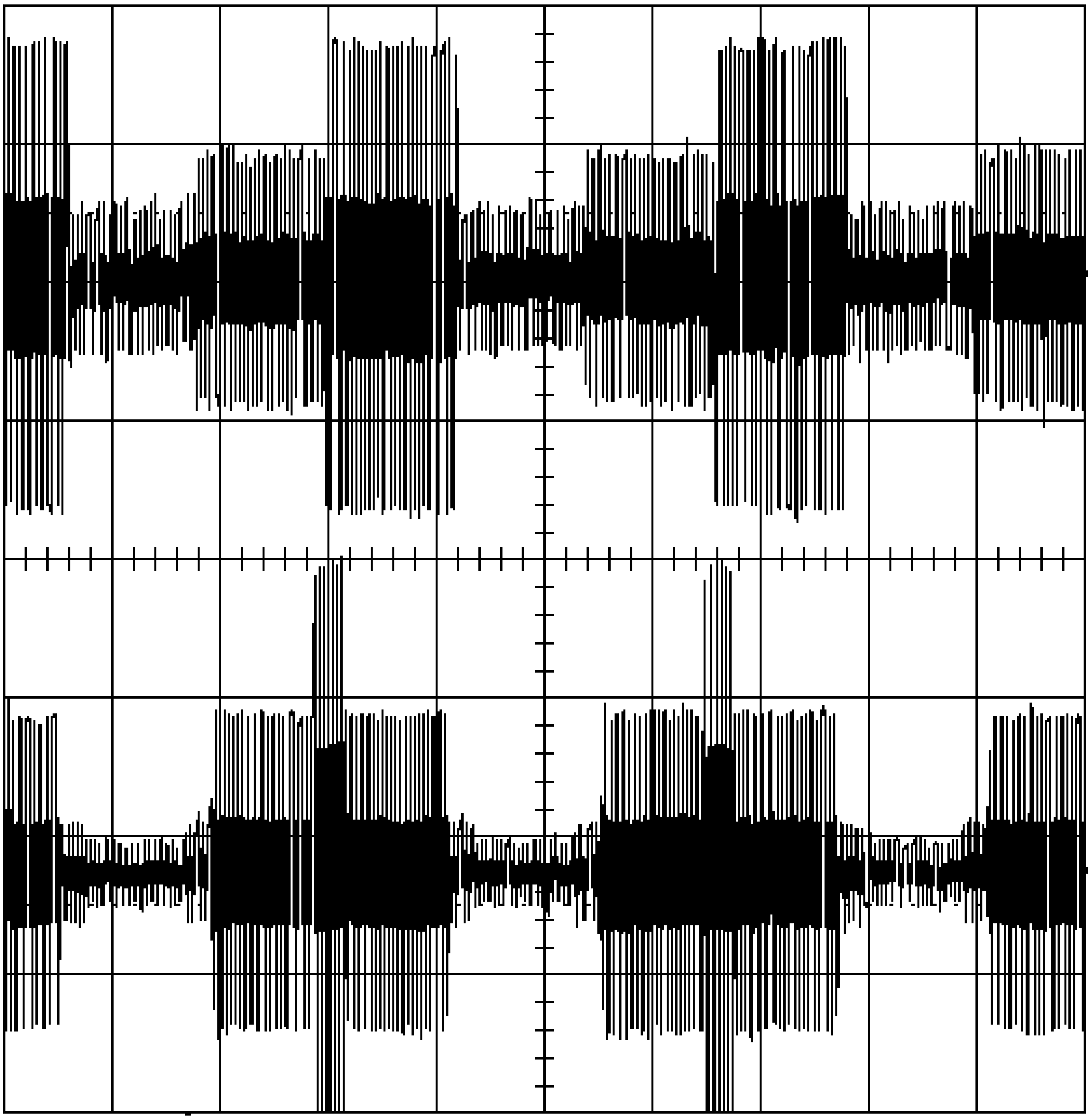

**Figure A.5: Upper trace - Decoded test signal with DRC turned off;
Lower trace - decoded test signal with DRC value applied 24 ms too early.
Horizontal scale is 100 ms/div. (Audio sampling frequency is 48 kHz)**

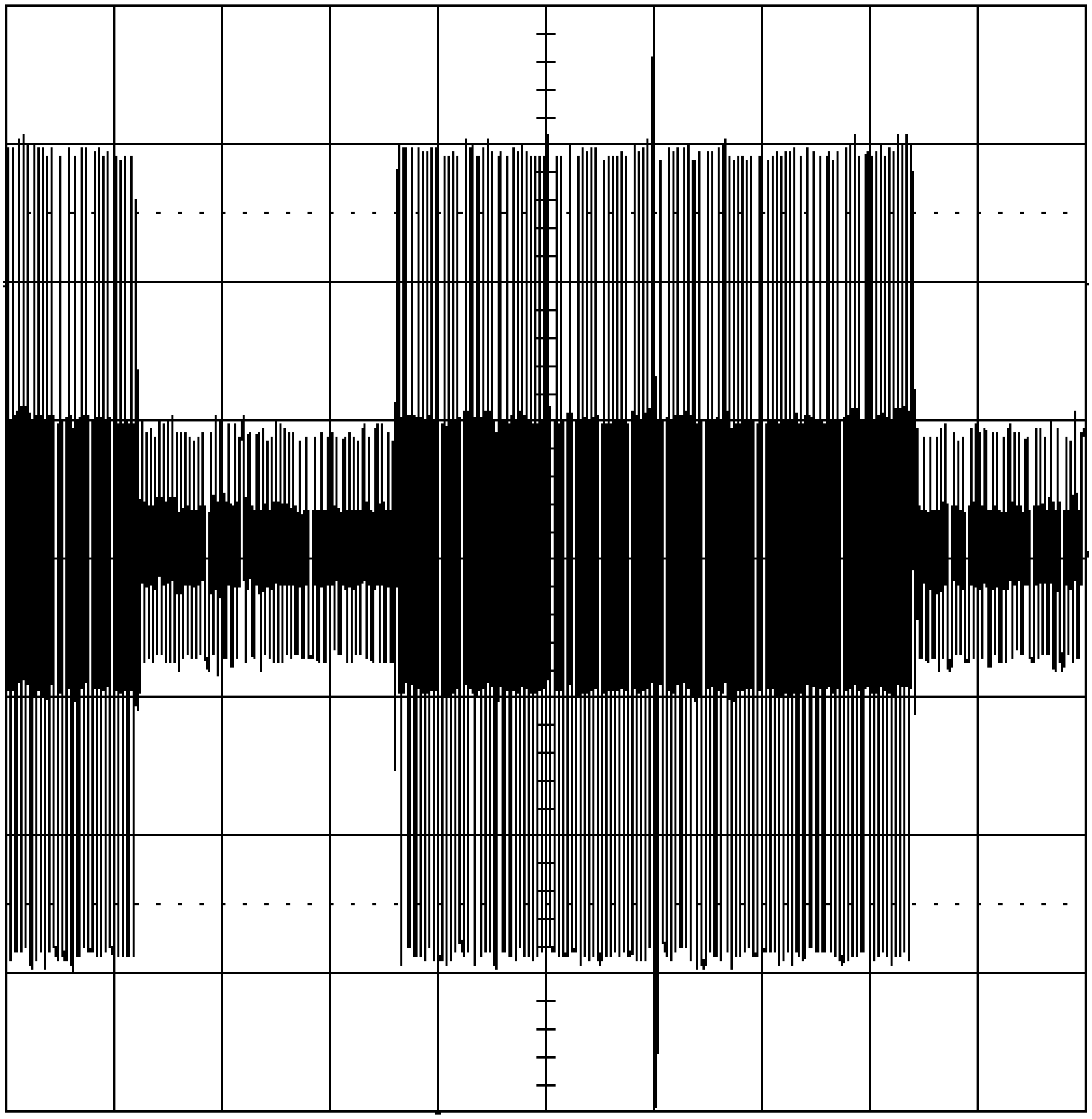

Figure A.6: Decoded test signal with DRC applied with less than 1 ms timing error. Horizontal scale is 50 ms/div (Audio sampling frequency is 48 kHz)

Bibliography

The following material, though not specifically referenced in the body of the present document (or not publicly available), gives supporting information.

- ISO/IEC 11172-3 (1993): "Information technology - Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbit/s - Part 3: Audio".

- ISO/IEC 13818-3 (1996): "Information technology - Generic coding of moving pictures and associated audio information - Part 3: Audio".

- ISO/IEC 13818-4 (1996): "Information technology - Generic coding of moving pictures and associated audio information-Part 4: Conformance testing ".

History

Document history		
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