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TEST SIGNALS AND MEASUREMENT TECHNIQUES FOR TRANSMISSION CIRCUITS CARRYING MAC/PACKET SIGNALS OR HD-MAC SIGNALS

ITU-T Recommendation J.67

(Formerly "Recommendation ITU-R CMTT")

FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Conference (WTSC), which meets every four years, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

ITU-T Recommendation J.67 was revised by ITU-T Study Group 9 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 22th of August 1994.

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NOTE

In this Recommendation, the expression “Administration” is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

CONTENTS

SUMMARY

The purpose of Recommendation J.67 is to give the tools of the transmission methodology of the MAC packet family signals. Section 1 is devoted to the traditional MAC packet signals (D/D2), while Section 2 deals with the high-definition television systems HD-MAC.

Each part of this Recommendation begins with the definition of the test signals and the test lines that are the basis of the transmission methodology. To have the best compatibility with the MAC signals, test lines for HD-MAC are as close as possible to the test signals previously defined in Section 1, but nevertheless slightly different, due to the different nature of HD-MAC. Furthermore, the main quality measurement parameters are defined, as well as the corresponding application methods.

Section 2 of this new version of the Recommendation replaces the previous part B, the contents of which were only subclause titles with text “to be defined”.

INTRODUCTION

A clear description of what is meant by the transmission circuit is essential for defining the measurement problem. Figure Intro. 1 shows a studio encoder driving the transmission circuit, and a studio decoder driven by the transmission circuit. MAC is a multiplex of luminance, chrominance, and sound/data signals. The video inputs to the encoder are the luminance component and the two colour difference components. These are also present at the output of the decoder. The measurement methods described in this Recommendation are for automatic measurements of the transmission circuit between the MAC/HD-MAC encoder and the MAC/HD-MAC decoder.

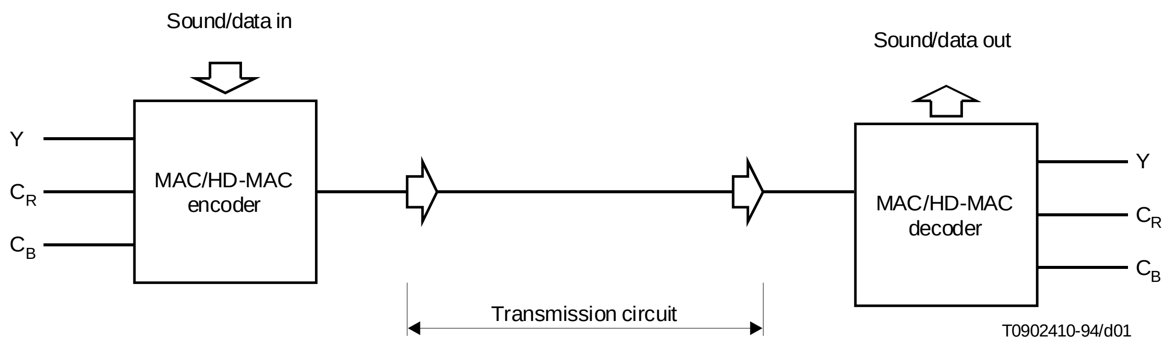

FIGURE Intro. 1/J.67

Transmission of MAC signals

The Recommendation contains two sections listed below:

- Section 1: Definition of test signals and measurement methods for transmission circuits carrying MAC/packet signals.
- Section 2: Definition of test signals and measurement methods for transmission circuits carrying HD-MAC signals.

Recommendation J.67[1)]

TEST SIGNALS AND MEASUREMENT TECHNIQUES FOR TRANSMISSION CIRCUITS CARRYING MAC/PACKET SIGNALS OR HD-MAC SIGNALS

(1992; revised 1994)

The ITU-T,

considering

the need of a Recommendation concerning the transmission methodology of television signals using multiplexed analogue components,

recommends

(1) that for MAC/packet signals, quality parameters defined in 1.2 should be measured using the test signals defined in 1.3 and measurement methods defined in 1.4;

(2) that for HD-MAC signals, quality parameters defined in 1.2 and 2.3 should be measured using the test signals defined in 2.2 and measurement methods defined in 2.3.

SECTION 1 – DEFINITION OF TEST SIGNALS AND MEASUREMENT METHODS FOR TRANSMISSION CIRCUITS CARRYING MAC/PACKET SIGNALS

1.1 Introduction

Quality parameters to be measured are defined in 1.2, test signals are defined in 1.3 and in Annex A. Measurement methods are given in 1.4.

1.2 Definition of the quality parameters of a MAC/packet signal

1.2.1 MAC signal

1.2.1.1 Waveforms and line allocations

The MAC analogue waveform is directly derived from the standard 4:2:2 sampling ratio used for digital television (ITU-R Recommendation BT.601). MAC coding produces a sequential transmission of a chrominance signal, compressed in a 3:1 ratio, and the luminance signal, compressed in a 3:2 ratio.

Given the sampling frequencies defined for the digital television standard (13.5 MHz for luminance and 6.75 MHz for chrominance), the consequent MAC sampling frequency is 20.25 MHz. The resulting nominal bandwidth required for the coded MAC signal is 8.4 MHz. After decompression the luminance bandwidth is 5.6 MHz.

It is important to note that, even though the MAC signal is derived through a sampling process, the resulting signal has an analogue form for transmission. A remarkable feature of the MAC coding system is that there is no absolute limit for the bandwidth. This characteristic can be used to broadcast the MAC signal in a narrow-band channel.

[1)] Formerly Recommendation ITU-R CMTT.772.

1.2.1.2 Quality parameters

1.2.1.2.1 Nominal signal amplitude

The nominal amplitude of a MAC signal is 1 V. It is defined as the difference between the white level and the black level of the reference signal of line 624.

1.2.1.2.2 Distortions

1.2.1.2.2.1 Gain/frequency response

The gain/frequency characteristic of the circuit is defined as the variation in gain between the input and the output of the circuit over the frequency band extending from the field repetition frequency to the nominal cut-off frequency of the MAC signal, relative to the gain at a suitable reference frequency.

1.2.1.2.2.2 Phase distortion

The phase-frequency distortion is defined as the difference in degrees relative to a linear phase characteristic over a frequency band extending from, ideally, 0 Hz to a defined upper frequency.

1.2.1.2.2.3 Group-delay distortion

The group-delay distortion, expressed in ns, is defined by the difference between the group delay for each measured frequency and the group delay for a given reference frequency.

1.2.1.2.2.4 Long-time waveform distortion

If a test signal, simulating a sudden change of the luminance from a black level to a white level or vice versa, is applied to the input of a circuit, a long-time waveform distortion is present if the variations of the clamp level (medium grey) of the output signal do not precisely follow those of the clamp level of the input signal. This failure may be either in exponential form, or more frequently in the form of damped very low frequency oscillations.

1.2.1.2.2.5 Field-time waveform distortion

If a square-wave signal with a period of the same order as one field and of nominal luminance amplitude is applied to the input of the circuit, the field-time waveform distortion is defined as the change in shape of the square wave at the output. A period at the beginning and end of the square wave, equivalent to the duration of a few lines, is excluded from the measurement.

1.2.1.2.2.6 Line-time waveform distortion

If a square-wave signal with a period of the same order as one line and of nominal luminance amplitude is applied to the input of the circuit, the line-time waveform distortion is defined as the change in shape of the square wave at the output. A period at the beginning and end of the square wave, equivalent to a few picture elements, is excluded from the measurement.

1.2.1.2.2.7 Short-time waveform distortion

If a short pulse (or a rapid step-function) of nominal luminance amplitude and defined shape is applied to the input of the circuit, the short-time waveform distortion is defined as the departure of the output pulse (or step) from its original shape.

1.2.1.2.2.8 Distortions due to echoes

This distortion is that caused by the superposition of the direct signal in the RF paths and an attenuated version of that signal delayed in time and shifted in phase relative to the direct signal.

1.2.1.2.2.9 Low frequency non-linear distortion

For a particular value of average picture level, the low frequency non-linear distortion is defined as the departure from proportionality between the amplitude of the input signal and the output signal, when the input signal is shifted from the black level to the white level within the duration of a line period.

1.2.1.2.3 Noise

1.2.1.2.3.1 Continuous random noise

The signal-to-noise ratio for continuous random noise is defined as the ratio, expressed in decibels, of the nominal amplitude of the luminance signal (1 V) to the r.m.s. amplitude of the noise measured after band limiting. A signal-to-weighted-noise ratio is defined as a ratio, expressed in decibels, of the nominal amplitude of the luminance signal, to the r.m.s. amplitude of the noise measured after band limiting and weighting with a specified network.

One possibility is that wideband random noise should be measured in a bandwidth of 8.4 MHz using a constant impedance noise-weighting network with a time constant of 90 ns. Such a network is based partly on the assumption that with the trend towards larger picture displays and with the improved picture quality available from the MAC/packet television standard, future subjective tests will more commonly employ a viewing distance of four times the picture height, rather than six times, as at present.

The second possibility uses the existing unified weighting network, scaled according to the 3:2 compression ratio, as a common weighting network for all MAC systems. This filter gives the same results as would be obtained from a signal in decompressed form with the unified weighting filter described in Recommendation J.61[2]. It also takes account of the noise carried in the more-compressed colour-difference signals. The possibly greater noise sensitivity due to the higher bandwidth HD-MAC signals when these use the same networks that are designed for present-day MAC signals is also considered. The definition of this network and its amplitude/frequency response are given in Figure 1.

1.2.1.2.3.2 Low frequency noise

The signal-to-noise ratio for low frequency noise is defined as the ratio, expressed in decibels, of the nominal amplitude of the luminance signal (1 V) to the mean square value of the noise.

1.2.1.2.3.3 Interference

The signal-to-interference ratio is defined as the ratio, expressed in decibels, of the nominal amplitude of the luminance signal (1 V) to the peak-to-peak amplitude of the interfering signal.

1.2.2 Data signals

1.2.2.1 Data signal waveform

The data signals have very different characteristics within the family of MAC systems. They are defined in the ex-CCIR special publication “Specifications of Transmission Systems for the Broadcasting-Satellite Service”.

1.2.2.2 Quality parameters for digital signals

1.2.2.2.1 Bit-error ratio

The bit-error ratio (BER) is defined as the ratio of the number of detected bit errors to the number of transmitted bits over a given period of time.

1.2.2.2.2 Eye diagram

The eye diagram is defined as the superposition of all the configurations of the data signals.

1.2.2.2.3 Equivalent impairment

The data signal quality is evaluated by adding a Gaussian noise signal to the received signal and plotting the bit-error ratio versus the noise level. For a given bit-error ratio, the difference in dB between the measured noise level and the theoretical level produces, by definition, the “equivalent impairment”.

1.2.2.2.4 Decoding margin

Another method to evaluate the data signal quality by adding a Gaussian noise, is to measure the level of added noise to obtain a given bit-error ratio. This is, by definition, the “decoding margin”.

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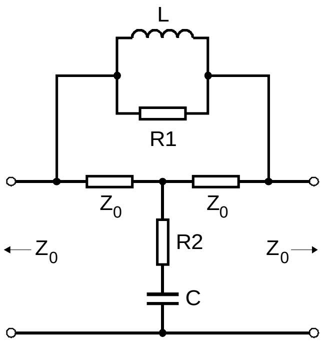
[2] Formerly Recommendation ITU-R CMTT.567.



$$L = Z_0 \cdot \tau$$

$$C = \frac{\tau}{Z_0}$$

$$R1 = a \cdot Z_0$$

$$R2 = \frac{Z_0}{a}$$

$$\text{Insertion loss } A = 10 \log \frac{1 + \left[\left(1 + \frac{1}{a}\right)\omega\tau\right]^2}{1 + \left[\frac{1}{a}\,\omega\tau\right]^2} \text{ dB}$$

$$\text{where } \tau = \frac{245 \text{ ns}}{1.5}\,; \; a = 4.5$$

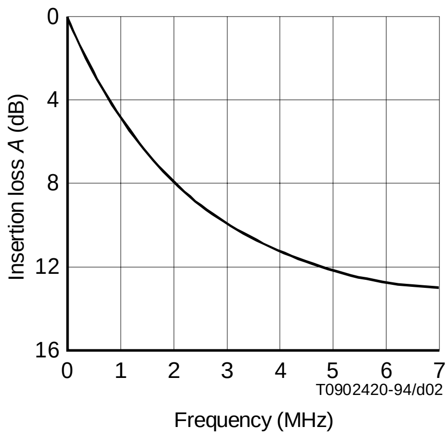


FIGURE 1/J.67

Unified random noise weighting filter for MAC circuits using a compression ratio of 3:2

1.3 Description of the test signals for MAC/packet systems

1.3.1 General remarks

Three insertion test signals are defined, primarily for automatic measurement. In addition, three optional waveforms may be used either in test line or as full field signals for monitoring purposes (see Annex A).

As far as possible, essential signal elements have been allocated to the luminance period of the line (samples 590 to 1286). This will enable these signals to be used also for testing MAC coders and decoders.

Only in a restricted sample range between 245 and 1277 do the proposed signals vary from the 0 mV level. This allows blanking and reinsertion of the test signals at appropriate points of the transmission chain (e.g. between terrestrial and satellite section).

The spectral content of all waveforms is restricted to 8.5 MHz (–6 dB).

High frequency signal amplitudes are restricted to ± 250 mV to avoid non-linear distortion and to allow conversion into an AM-VSB-MAC-system with Nyquist filtering at the transmitting end.

1.3.2 Definition of elementary waveforms

1.3.2.1 Basic definitions

T is the MAC sampling period ≈ 49.38 ns,

k is the MAC sample number (see ITU-R Recommendation BO.650 and the ex-CCIR special publication "Specifications of Transmission Systems for the Broadcasting-Satellite Service").

1.3.2.2 Transition

A transition is a signal defined for a duration 4 T according to the shaping:

$$0.000 - 0.114 - 0.500 - 0.886 - 1.000$$

(Hamming window integral on 4 T)[3].

1.3.2.3 Pulse

A pulse is a signal defined for a duration 6 T according to the shaping:

$$0.000 - 0.130 - 0.630 - 1.000 - 0.630 - 0.130 - 0.000$$

(Blackman window on 6 T)[4].

1.3.2.4 Ramp

A ramp is a signal defined for a duration $n\,T$ in the equation:

$k = 0$ to n: $y_k = k / n$ for a rising ramp

$y_k = 1 - k / n$ for a falling ramp

1.3.2.5 Complex wobbulation

A complex wobbulation is made up of two signals defined for a duration 512 T according to equations:

Real part:

$$k = 0 \text{ to } 512: \qquad y_k = \left(\cos \frac{\pi (k - 256)^2}{512}\right) W(k)$$

Imaginary part:

$$k = 0 \text{ to } 512: \qquad y_k = \left(\sin \frac{\pi (k - 256)^2}{512}\right) W(k)$$

where

$W(k)$ is a window defined as:

$k = 0$ to 28: $W(k) = 0$

$k = 28$ to 53: $W(k) = \sin^2 \frac{\pi (k - 28)}{50}$

$k = 53$ to 459: $W(k) = 1$

$k = 459$ to 484: $W(k) = \sin^2 \frac{\pi (484 - k)}{50}$

$k = 484$ to 512: $W(k) = 0$

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[3] Hamming: $y(t) = 0.54 + 0.46 \cos \pi t / 2 T$.

[4] Blackman: $y(t) = 0.42 + 0.50 \cos \pi t / 3 T + 0.08 \cos 2 \pi t / 3 T$.

The complex wobbulation signals are transmitted in a 4-frame sequence in positive and negative (inverted) polarity as follows:

- Even frame: real part not inverted
- Odd frame: imaginary part not inverted
- Even frame: real part inverted
- Odd frame: imaginary part inverted
- Even frame: real part not inverted,

  etc.

1.3.2.6 Modulated pulse

A pulse modulated at a frequency *f* (MHz) is a signal defined for a duration 81 *T* according to the equation:

$$k = 0 \text{ to } 81: \qquad y_k = \cos^2 \frac{4\pi f k}{81} \sin^2 \frac{\pi k}{81}$$

1.3.2.7 Burst

A burst, modulated at a frequency *f* (MHz) is a signal defined for a duration 81 *T* in the equation:

$$k = 0 \text{ to } k_1: \qquad y_k = \sin \frac{8\pi f k}{81} \cdot \left[\frac{1}{2\pi} \left(\frac{2\pi k}{k_1} - \sin \frac{2\pi k}{k_1}\right)\right]$$

$$k = k_1 \text{ to } 81 - k_1: \qquad y_k = \sin \frac{8\pi f k}{81}$$

$$k = 81 - k_1 \text{ to } 81: \qquad y_k = \sin \frac{8\pi f k}{81} \cdot \left[\frac{1}{2\pi} \left(\frac{2\pi (81 - k)}{k_1} - \sin \frac{2\pi (81 - k)}{k_1}\right)\right]$$

$$f = 1 \text{ to } 6: \qquad k_1 = 15$$

$$f = 7: \qquad k_1 = 25$$

$$f = 8: \quad y_k = \sin \frac{8\pi f k}{81} \cdot \sin^2 \frac{\pi k}{81}; \quad k = 0 \text{ to } 81$$

1.3.3 Test signal description

1.3.3.1 **Test signal No. 1** (see Figure A.1 and Table A.1)

Test signal No. 1 is a mandatory signal and allocated to line 312. It is intended for automatic measurement and composed of a bipolar bar signal with inverse polarity in even and odd frames. Positive and negative Blackman pulses are contained in the even frame signal only.

The first part of the signal ($k = 225$ to 612) is provisionally set to 0 mV. It may be used in the future for insertion of other test signals for HD-MAC.

1.3.3.2 **Test signal No. 2** (see Figure A.2 and Table A.2)

Test signal No. 2 is a mandatory signal and allocated to line 623. It is intended for automatic measurement of noise and non-linear distortion. It comprises a rising ramp (even frames) and a falling ramp (odd frames). This allows separation of linear distortions (e.g. tilt) from non-linear ones.

1.3.3.3 **Test signal No. 3** (see Figure A.3 and Table A.3)

Test signal No. 3 is a mandatory signal and allocated to line 624. The first part of this line is already defined in the MAC/packet standards. The second part of this line contains a complex wobbulation.

1.3.3.4 **Test signal No. 4 (national option)** (see Figure A.4 and Table A.4)

This optional signal is intended for evaluation of linear distortions on a waveform monitor. This signal consists of a bipolar pulse and bar signal and eight modulated pulses (1 to 8 MHz) of 500 mV amplitude. It may be used also with full amplitude (1000 mV) if non-linear distortions are unlikely to occur.

If used as a test line signal it should be inserted on line 311.

1.3.3.5 **Test signal No. 5 (national option)** (see Figure A.5 and Table A.5)

This optional signal comprises an 8-riser staircase waveform and is intended for evaluation of non-linear distortions using a waveform monitor.

If used as a test line signal it should be inserted on line 1.

1.3.3.6 **Test signal No. 6 (national option)** (see Figure A.6 and Table A.6)

This national option is intended for display of the amplitude frequency response on a waveform monitor and is composed of eight multiburst signals (1 to 8 MHz) of 500 mV amplitude preceded by a reference bar. It may be used also with full amplitude (1000 mV) if non-linear distortions are unlikely to occur.

If used as a test line signal it should be inserted on line 313.

1.4 Measurement methods

1.4.1 General remarks

The measurement techniques described below use the test signals in 1.3 above and are to be used for automatic measurement. These methods are based largely on digital signal processing techniques.

1.4.2 Measurements related to the vision signal

1.4.2.1 Low-frequency noise

Low-frequency noise (below line frequency) is measured on a 50% white picture signal. High-frequency noise is reduced by averaging the signal value within each line. The spectral noise density is estimated by Fourier transform of the 625 signal values derived from one frame. The obtained analysis pitch is 25 Hz, allowing separation between the different sources of noise in the frequency band 25 Hz to 7.8 kHz.

1.4.2.2 High-frequency noise

High-frequency noise measurement is carried out using test signals No. 2a and 2b (ramp signals).

One method consists of estimating the noise level by averaging a large number of observations of the same complete test line and subtracting this average waveform from each observation. The result is the true noise content. From this, the true spectral noise density can be calculated.

The other method for weighted and unweighted random noise measurement uses the 200 kHz high-pass filter as specified in Recommendation J.64[5)] to remove the ramp signal. The filter also reduces the influence of static non-linearities to the noise measurement.

1.4.2.3 Dynamic non-linearity

When comparing the shape of the two polarities of the Blackman pulse and the Hamming slopes in test signal No. 1, information on high-frequency non-linearity can be obtained.

1.4.2.4 Static non-linearity

Static non-linearity is measured using test signals No. 2a and 2b. The two polarities are used to separate linear from non-linear effects.

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5) Formerly Recommendation ITU-R CMTT.569.

One method is to sample the ramp signals and to remove the noise by averaging. After removal of linear distortion, the processed waveform is approximated by a polynomial of degree *k*. Analysis of the coefficients of the polynomial provides information on the overall non-linear characteristic. It should be noted that any difference between the above-mentioned processed waveform and the polynomial indicates the presence of quantization distortion.

The second method also uses the ramp signals but is analogous to the luminance non-linearity measurement on the 5-riser staircase described in Recommendation J.64 and provides comparable results. The ramp level is measured at timing instants which are equally separated from one another by 9 μs, and centred with respect to the active period of 50.5 μs. The amplitude value at each of the six timing instants is calculated as the arithmetic mean of the ramp signal from 0.5 μs before to 0.5 μs after the timing instant. This reduces the influence of quantization errors and superimposed high-frequency noise. The six samples are processed to a single figure as described in 2.9/J.64.

1.4.2.5 Amplitude and phase/group delay frequency response

The complex wobbulation signal (test signal No. 3) is processed through a Fourier transform into amplitude and phase frequency response. The latter can be processed further into group delay. Prior to applying the sampled complex wobbulation signal to the FFT, non-linearities can be minimized using the two opposite polarities and noise can be reduced by averaging.

1.4.3 Specific data signal measurements

1.4.3.1 Bit-error ratio measurement

The bit-error ratio can be measured during programme transmission on synchronization words, the Golay code on packet headers, and on dummy packets. In the case of duobinary coding, the violation rate of this code gives an excellent estimate of the bit-error ratio.

1.4.3.2 Decoding margin measurement and equivalent impairment

Bit-error ratio measurements give no indication on the actual safety margin for possible additional distortions.

One method of deriving such information is to measure the error ratio using variable deviations from the threshold levels until a predefined error ratio is reached. A second method adds Gaussian noise to the received signal until the predefined error ratio is reached.

1.4.3.3 Analysis of eye diagrams

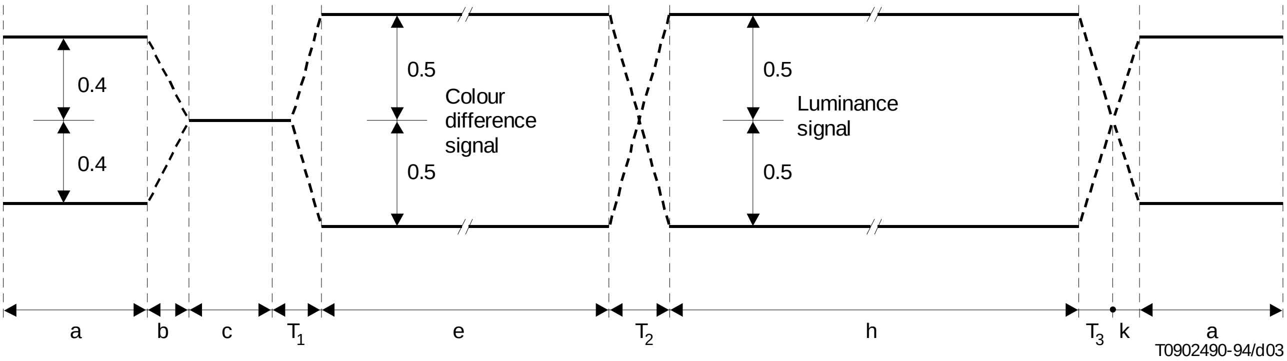
The analysis of the eye diagrams of a two-level or a duobinary data signal – either made by means of an oscilloscope or microprocessor – provides eye-height and eye-width indications and additional information about the location of the optimum sampling phase and threshold levels with respect to their nominal values.

SECTION 2 – DEFINITION OF TEST SIGNALS AND MEASUREMENT METHODS FOR TRANSMISSION CIRCUITS CARRYING HD-MAC SIGNALS

2.1 Introduction

Because of the specific features of HD-MAC signals, test signals, quality parameters and measurement methods are slightly different from those of D- and D2-MAC signals.

The baseband multiplex structure for the (D2) HD-MAC/packet signal is shown in Figure 2. It should be noted that the video signal is non-linearly pre-emphasized. For high-definition television transmissions the baseband signal includes two different data multiplexes, one in the LBI referred to as the sound/data multiplex (10.125 Mbit/s) and one in the FBI referred to as the DATV/data multiplex (20.25 Mbit/s). In addition, optional teletext data lines have been defined. Figure 3 gives the general HD-MAC/packet TDM structure.



a = 209 clock periods for 105 bits of data and sync.
b = 4 clock periods for transition from end of data
c = 15 clock periods – clamp period (0.5 V)
T_1 = 10 clock periods for weighted transition to colour-difference signal
e = 349 clock periods for colour-difference component
T_2 = 5 clock periods for weighted transition between colour-difference signal and luminance signal
h = 697 clock periods for luminance component
T_3 = 6 clock periods for weighted transition from luminance signal
k = 1 clock period for data transition

Clock frequency: 20.25 MHz

FIGURE 2/J.67

HD-MAC multiplex signal

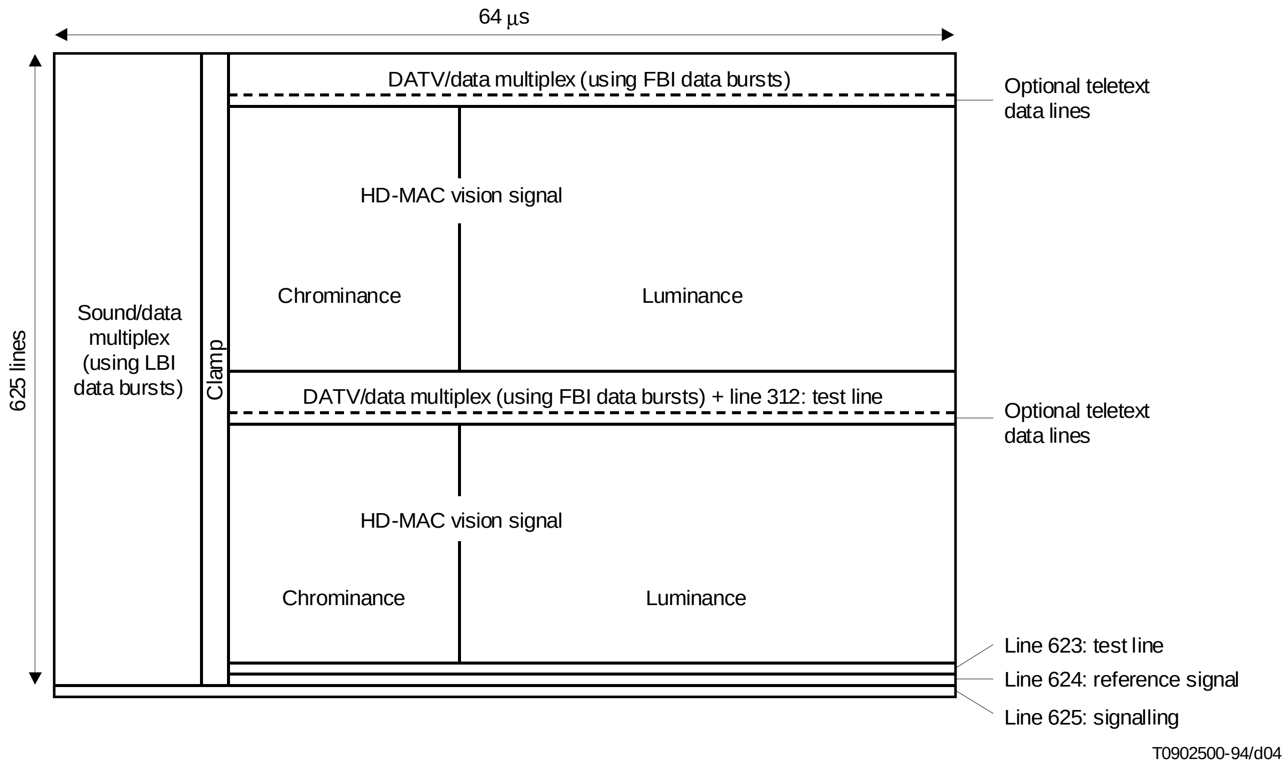


FIGURE 3/J.67

General HD-MAC/packet TDM structure

The transmission chain from the transmitter multiplex to the receiver demultiplex is given in Figure 4. The transmission circuit may typically include satellite section and/or cable section. Point M is the baseband measuring point.

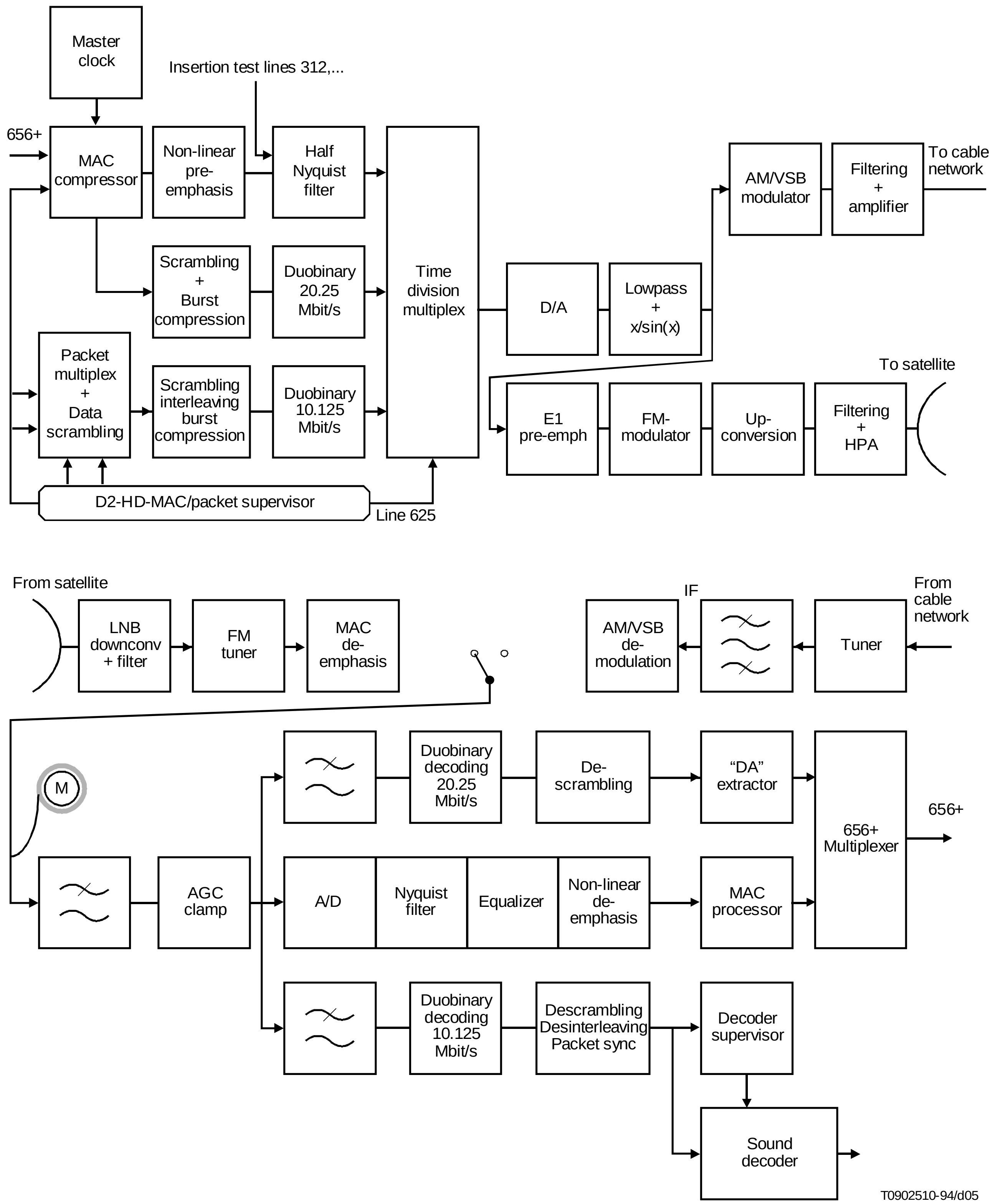


FIGURE 4/J.67

HD-MAC multiplexing and demultiplexing

In the sequence, it should be understood that the basic measurements are defined and made without an equalizer in the receiver. However, many receivers will have equalizer or adaptive data decoding. Taking this into account, some measuring equipment will integrate these functions. This has to be considered when defining templates in the future.

2.2 Definition of used test lines and signals

2.2.1 General remarks

Basically there are two types of measurements. Set-up measurements are made when starting the service or during off-service hours without programme contents and possibly using special test signals (like full-field signals). In-service measurements can be made during the normal programme hours, using the test lines (312, 623, 624) existing in the multiplex signal.

It should be noted that these test lines are not processed by the non-linear pre-emphasis but they are processed by the half-Nyquist filter (roll-off = 0.1) in the transmitter in order to fit the used transmission channel.

Two modes can be used to broadcast test signals. In the basic mode the test signals are sent as described hereafter. In a cyclical mode, test signals can cyclically be sent in the MAC part of the line 623 according to a determined sequence. This sequence is not yet defined, but it will include the test signals subsequently described for line 623.

The mode used is signalled in line 625.

2.2.2 Description of test lines

2.2.2.1 Test signal No. 1 (see Figure B.1)

Test signal No. 1 is a mandatory signal allocated to line 312 and which has been revised to include parts needed in HD-MAC receivers for equalization purposes. This line is also intended for automatic measurements.

The first part of line 312 is filled over two frames by a pseudo-random sequence of 511 + 1 = 512 bits with the levels –250 mV and +250 mV corresponding to the bit values "0" and "1" respectively, the first function of which is to help the equalizer process.

The polynomial generator is: $X^9 + X^4 + 1$ and gives a pseudo-random sequence of 511 bits clocked at 20.25 MHz. The first half of the sequence (256 bits) is transmitted in the even frame and the second part is transmitted in the odd frame. The last bit (256th bit) of the odd frame is identical to the first bit of the next sequence.

The pseudo-random sequence (PRBS) generator is initialized at the beginning of each even frame with the binary word 111111111.

The first bit of the sequence generated by the PRBS generator on the even frame is the value present at the output after it has been loaded and before any shift operations have taken place.

In addition, two inverse half-amplitude pulses and a half-amplitude transition are inserted in even frames, the first function of which is to distinguish between linear and non-linear perturbations. The full-amplitude pulses and transitions in the D2-HD-MAC test line 312 are not weighted by Blackman and Hamming windows.

The pseudo-random sequence is defined in Figure B.2.

The allocation of samples levels are given in Annex B.

2.2.2.2 Test signal No. 2 (see Figure B.3)

This is identical to the conventional MAC test signal (1.3.3.2), line 623.

The test signal attributed to line 623 consists in a ramp defined over a period of 1000 T (1/T = 20.25 MHz) from –500 mV to +500 mV in the even frame and +500 mV to –500 mV in the odd frame.

The allocation of samples levels are given in Annex B.

2.2.2.3 **Test signal No. 3** (see Figure B.4)

This test line is directly derived from the D2-MAC test signal (see 1.3.2.5). The only difference is that the wobbulation is not filtered. This signal is then defined up to 10.125 MHz.

The first part of line 624 contains grey, white and black references which are specified as follows:

- Grey level: from sample 210 to sample 370;
- White level: from sample 374 to sample 532;
- Black level: from sample 536 to sample 694.

The second part contains a test signal that consists in a complex wobbulation constituted by two signals defined on a period of 512 T according to the following relations:

Real part

$$k = 0 \text{ to } 512; \qquad y_k = \cos \frac{\pi (k - 256)^2}{512}$$

Imaginary part

$$k = 0 \text{ to } 512; \qquad y_k = \sin \frac{\pi (k - 256)^2}{512}$$

These signals are transmitted in sequence on four digital frames as follows:

- Even frame: non-inverted real part;
- Odd frame: non-inverted imaginary part;
- Even frame: inverted real part;
- Odd frame: inverted imaginary part.

The allocation of line 624 samples is given in Annex B.

2.3 Definition of quality parameters and measurement methods

2.3.1 Video signal

2.3.1.1 Video level

Video level can be measured as the black and white level difference on line 624. This serves as a normalization parameter for other measurements.

2.3.1.2 Wideband random noise

2.3.1.2.1 Unweighted signal-to-noise ratio

The signal-to-noise ratio for continuous random noise is defined as the ratio, expressed in decibels, of the nominal amplitude of the luminance signal (1 V) to the r.m.s. amplitude of the noise measured after band limiting.

The measurement is made using line 623 ramp signal, having a 10.125 MHz low-pass filter and a 200 kHz first order high-pass filter. The purpose of the high-pass filter is to eliminate hum, pulses of line frequency, etc.

This measurement can conveniently be made using FFT analysis.

2.3.1.2.2 Weighted signal-to-noise ratio

The weighted S/N ratio is measured in line 623. The upper noise spectrum is limited by a 10 MHz low-pass filter as described in Annex C.

The lower noise spectrum is limited by a 200 kHz first order high-pass filter. The weighting filter is described in Annex C.

2.3.1.2.3 Noise-spectrum analysis

Noise-spectrum analysis is the study into the frequency domain, of the amplitude of the noise superimposed on the wanted signal. The analysis is very important for different purposes, i.e. to study the shape of the noise (and eventually to calculate its level), to detect interfering signals, etc.

The spectrum can be calculated using an FFT on any test signal once the content of the test signal has been suppressed. The most convenient signal for this is the ramp of line 623 since it has no high-frequency component.

The noise signal can be evaluated by subtracting the mean value of a large number of acquisitions from each acquisition.

The use of this test signal, defined on 1024 points at a sampling frequency of 20.25 MHz allows a resolution of approximately 19.78 kHz.

2.3.1.3 Low-frequency noise

2.3.1.3.1 Unweighted low-frequency noise

This requires a full-field test signal (medium grey level, no teletext, no DATV) and is a set-up measurement. The measured band is from 25 Hz to 7.8 kHz. In order to have results comparable to the traditional method (which excludes the clamping noise), the bandwidth 25 Hz to 1 kHz may also be used.

2.3.1.3.2 Spectrum analysis

Spectrum analysis is possibly more important for low-frequency noise than for wideband noise. It enables the different sources of noise to be distinguished: residual mains voltage, energy dispersal (line spectrum) or clamping noise (concentration of random noise at a very low frequency).

As for all the low-frequency measurements, a full-field grey signal is required, with no teletext, nor DATV.

The spectrum analysis can be carried out by means of a 625 points FFT (radix 5). Each point, needed for the calculation, is given by the mean value of the MAC part of the line.

2.3.1.4 Clamping performance evaluation

2.3.1.4.1 Clamping noise

In the MAC/packet family signals, the clamping period is very short. Since clamping noise is a very annoying phenomenon, it is important to measure it.

The measurement requires the two following signals:

- a full-frame grey level D2-MAC/packet signal;
- a wideband additive white noise signal, resulting in a given input S/N (e.g. 50 weighted dB, measured in 7.5 MHz).

Three measurements are carried out in the frequency band of 0 to 7.8 kHz:

- noise measurement at the output of the equipment with the additive white noise signal (B_{o1} expressed in dBm);
- noise measurement at the input of the equipment with the additive white noise signal (B_i in dBm);
- noise measurement at the output of the equipment, without the additive white noise signal at the input (B_{o2} in dBm).

Clamping noise, expressed in dBm, for the given S/N ratio is then calculated as:

$$B_c = -10 \log \left(10^{-B_{o1}/10} - 10^{-B_i/10} - 10^{-B_{o2}/10}\right)$$

For the validity of the measurement, some precautions need to be taken:

- when measuring B_{o1}, one needs to check that the signal is not affected by any interference;
- for a better precision, the level of the added noise has to be adjusted so that the induced clamping noise has a significant value compared to B_{o1}.

2.3.1.4.2 Clamping impairment

The method consists of adding a sine signal to the input signal and measuring the attenuation ratio of the added signal at the output in dB.

Appropriate sine signal characteristics are:

- frequency: 50 Hz;
- peak-to-peak level: 150 mV.

2.3.1.5 Linear response of the channel

In-service measurement uses line 624 complex wobbulation. This complex wobbulation is processed through a Fourier transform into amplitude response giving precise information of the transfer function of the transmission. The calculation method for this signal is similar to the one used for conventional MAC, and provides the transfer function- H(ω) of the channel.

The complex wobbulation signal is analysed as follows:

$$W(\omega) = WOB(\omega) \cdot H(\omega) \cdot Nyq(\omega)$$

where

W(ω)	is the Fourier transform of the received complex signal;
WOB(ω)	is the Fourier transform of the wobbulation test signal;
Nyq(ω)	is the transfer function of the Nyquist filter;
H(ω)	is the transfer function of the channel.

Because the wobbulation signal stops at 10.125 MHz, this method does not directly provide information about the channel response around Nyquist shape.

Following the description of test signal number 3 previously defined, the measurement method is based on acquisition of 512 samples of line 624 in a four-frame sequence. A difference of both real and imaginary positive and corresponding opposite parts can be applied to avoid non-linearities before FFT process and noise can be reduced by averaging.

2.3.1.5.1 Amplitude-frequency response

The amplitude-frequency response is based on FFT analysis where the ratio between the module of complex wobbulation measured and the theoretical value of this module is calculated.

2.3.1.5.2 Phase-frequency response (group delay)

Based on line 624, as amplitude frequency response, this measurement is also made using FFT calculation.

Within the MAC specification, a template for phase/frequency response instead of group delay/frequency has been chosen. As for amplitude frequency response, acquisition will be made in a four-frame sequence and will follow the same process.

The phase-frequency response corresponds to the difference between the linear phase and the phase error. This phase error is the difference between the arguments of the received and the theoretical signals (argument of the complex wobbulation of FFT analysis). The linear phase is obtained by calculating the regression line of the phase error for low frequencies.

The group-delay measurements will be obtained by calculating the derivative of the phase frequency response.

2.3.1.6 Distortions

2.3.1.6.1 Short-time distortion: pulse and bar ratio

The pulse and bar ratio measurement (Pbr) is carried out on line 312 and consists in measuring the ratio expressed in percentage, of the amplitude of the full-amplitude pulse (P) and the difference of the black level and the white level (B).

The result, expressed in percentage, is then:

$$Pbr = \frac{P}{B} \times 100$$

2.3.1.6.2 Static non-linear distortion

The measurement is carried out on the ramp of line 623. This signal is approximated by a polynomial of degree 3. In order to minimize noise, several acquisitions of ramps of same polarity are averaged. The coefficients of the polynomial are calculated by minimization of the quadratic error. They give information about the type of non-linearity, and several parameters can be defined.

Let $f(x) = \sum_{j=0}^{3} a_j x^j$ be the polynomial.

$NL2 = \frac{a_2}{a_1} \times 100$ gives the quadratic non-linearity as a percentage.

$NL3 = \frac{a_3}{a_1} \times 100$ gives the cubic non-linearity as a percentage.

$$NL1 = \frac{\max|f'(x)| - \min|f'(x)|}{\max|f'(x)|} \times 100 \text{ where } f'(x) = \frac{df}{dx}$$

$NL1$ is a parameter close to the one defined for the traditional TV systems.

2.3.1.7 Impulse response

Based on line 312, this is an HD-MAC specific new item. This impulse response is very important since it gives a lot of information about the channel around 10 MHz. Some are redundant with the measurement that can be achieved with the wobbulation of line 624, but it enables an easy oscilloscopic check. They are:

- echoes (very visible on TV screen);
- group delay (asymmetric response).

It is important to establish a weighting function (and appropriated template) of the impulse response that would give the maximum tolerance level of an interfering impulse in the vicinity of a reference impulse. If the amplitude of the pulse is not very critical, its spread can have some very disturbing effects. These can be seen either as staircase effects on diagonal lines or echoes. For a scrambled picture, some coloured spots can appear on the screen (comet noise).

Half-amplitude impulses are used to define linear distortions and full-amplitude impulses are used to define the sum of linear and non-linear distortions.

For automatic measurement, the peak distortion of a HD-MAC signal is given by the relationship:

$$D = \frac{A - h_0}{A} \times \rho_0 + \sum_{i \neq 0} \frac{|h_i|}{A} \times \rho_i$$

where

A is the nominal impulse level;

h_i are the impulse response samples;

ρ_i is the weighting coefficient.

2.3.1.8 Step response

Step response is measured using a full-amplitude transition of line 312 and fitting a template. This measurement is very important especially for scrambled pictures.

2.3.2 Data signals

2.3.2.1 Code violation

Duobinary code violations provide an excellent estimate of Bit-Error Rate (BER). Redundancies in the duobinary code are used to make a measurement. This method can be operated in transmission whatever the sound or the data carried. The other methods using Golay code redundancy on packet headers and synchronization word can be used to get the real BER, but requires a longer monitoring time.

The 20.25 Mbit/s stream is more sensitive to distortion than the 10.125 Mbit/s. But, as the 20.25 Mbit/s is only carried by a few lines in the picture part of the frame (*according to what you want to qualify DATV or data and sound*) all the measurements specified for digital data can also be made on 10.125 Mbit/s stream.

The code violation measurements on the 10.125 Mbit/s stream remain the same as the D2-MAC code violation qualification, which will guarantee the compatibility with existing test sets.

2.3.2.2 Equivalent noise degradation

In most cases, signals to be qualified in a transmission chain are virtually error free. The BER measurement or code violation can confirm this situation, but give no information on the available margin.

This margin can be evaluated by adding Gaussian Random Noise to the baseband signal and plotting a curve showing the BER as a function of the noise level. For a given BER (10^{-4} for automatic measurements), by definition, the difference between the measured S/N ratio for a predefined bandwidth (5 MHz for LBI data, 10 MHz for DATV) and that given by theory will represent the equivalent degradation.

The equivalent degradation must be measured even in the case in which the residual error reaches 10^{-4} with no added noise. Errors depend on the nature of the noise or distortions and the equivalent degradation may remain low even under these conditions:

a) *Manual measurement*

This measurement requires the availability of an instrument for measuring the BER (or code violation) and a white Gaussian noise generator applied to the baseband.

b) *Automatic measurement*

Two methods can be used and will produce similar results:

– *Addition of noise to the signal:*

Gaussian noise is filtered over a predefined bandwidth (see 2.3.2.2) and added to the HD-MAC baseband signal.

Its level is then automatically adjusted such that the measurement bit-error rate is of the order of 10^{-4}.

– *Variation of decoding thresholds:*

It may be demonstrated that there is a relation between the bit-error rate due to thresholds shifting and the error rate due to added noise. The error rate as a function of the noise level is obtained by taking the convolution product of the bit-error rate as a function of the thresholds and the noise probability density.

2.3.2.3 Eye diagram

Eye diagram gives a great deal of information on the quality of the digital signal (10 and 20 Mbits/s streams). This information is complementary to the code violation and error rate measurements. It gives an estimation of timing jitter and noise degradation.

However, it is difficult to quantify the eye diagram opening because there is no parameter that really represents the degradation of the signal. For this reason, no template nor measurement figures will be suggested.

An eye diagram should therefore be considered as a subjective check parameter.

2.3.2.4 Jitter on data

The following jitter measurements are derived from digital data transmission experience. The low level of knowledge and experience on HD-MAC jitter measurements does not allow conclusions to be drawn at this stage.

Jitter definition

With a theoretical digital transmission, the clock which is driving the information always has the same period T. However, considering a real network, the period is not perfect and a phase modulation appears. Pulses arrive from the reference clock at times which are either too early or too late. This phenomenon is called jitter, and can generate errors, as the data sent is not really synchronized with the reference clock.

The jitter is measured using two main parameters:

– The *jitter amplitude* – It corresponds to the peak to peak deviation, of the phase function versus time. The unit of this characteristic is the Unit Interval (UI), which is equivalent to the deviation of one clock period.

– The *jitter frequency* – It corresponds to the frequency of the deviation. However, in practice, the jitter variation is not a pure sinusoid. It is more likely to be a combination of different frequencies.

Annex A

(This annex forms an integral part of this Recommendation)

A.1 Introduction

Detailed definitions of the test signal elements contained in the following tables and figures (e.g. transition, pulse, etc.) are given in 1.3.

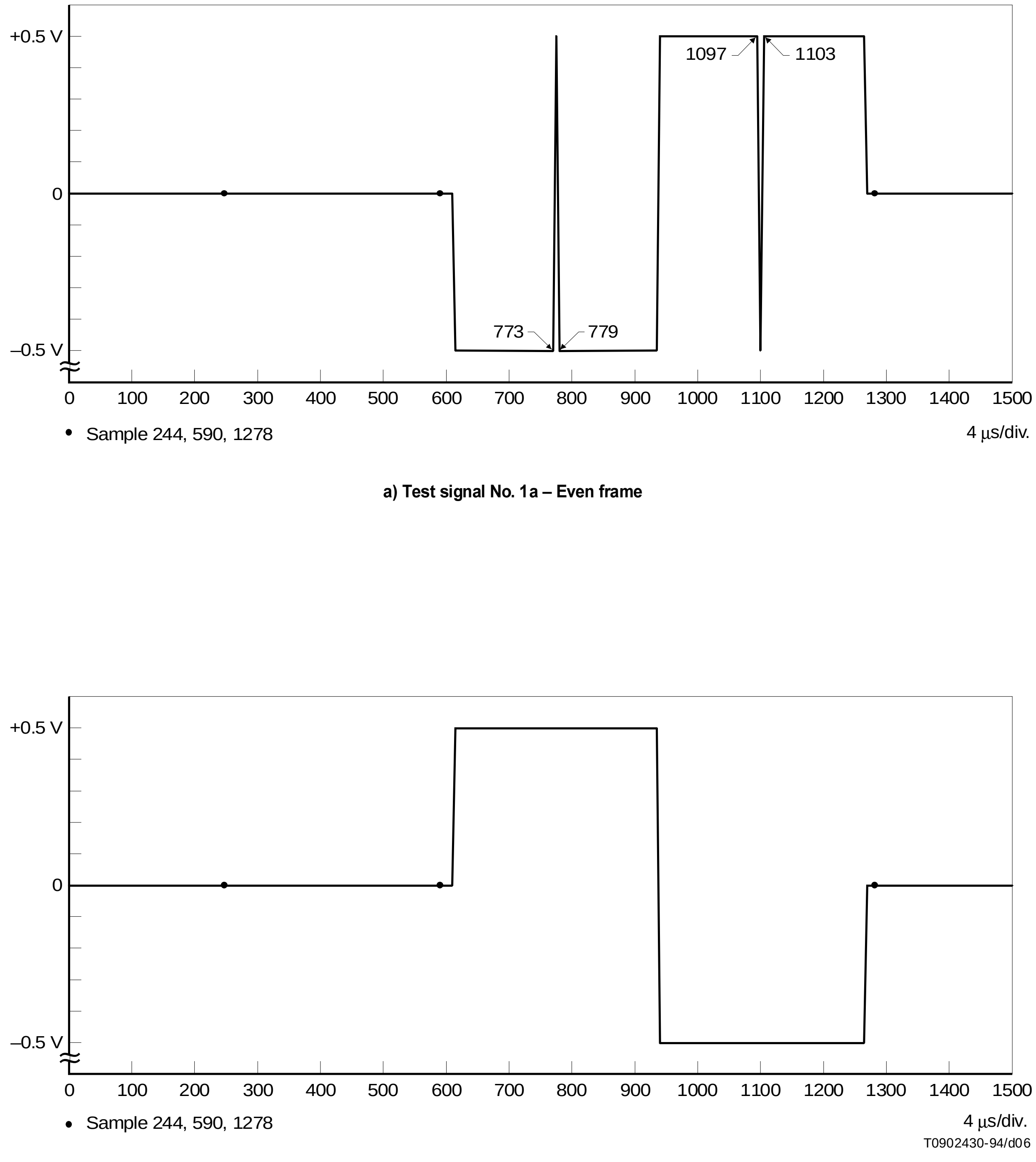
TABLE A.1/J.67

Definition of signal 1
a) Even frames

k =	255	to	612:	level 0 mV
k =	612	to	616:	transition from 0 mV to –500 mV
k =	616	to	773:	level –500 mV
k =	773	to	779:	pulse (base –500 mV; peak +500 mV)
k =	779	to	936:	level –500 mV
k =	936	to	940:	transition from –500 mV to +500 mV
k =	940	to	1097:	level +500 mV
k =	1097	to	1103:	pulse (base +500 mV; peak –500 mV)
k =	1103	to	1260:	level +500 mV
k =	1260	to	1264:	transition from +500 mV to 0 mV
k =	1264	to	1292:	level 0 mV

b) Odd frames

k =	255	to	612:	level 0 mV
k =	612	to	616:	transition from 0 mV to +500 mV
k =	616	to	936:	level +500 mV
k =	936	to	940:	transition from +500 mV to –500 mV
k =	940	to	1260:	level –500 mV
k =	1260	to	1264:	transition from –500 mV to 0 mV
k =	1264	to	1292:	level 0 mV



a) Test signal No. 1a – Even frame

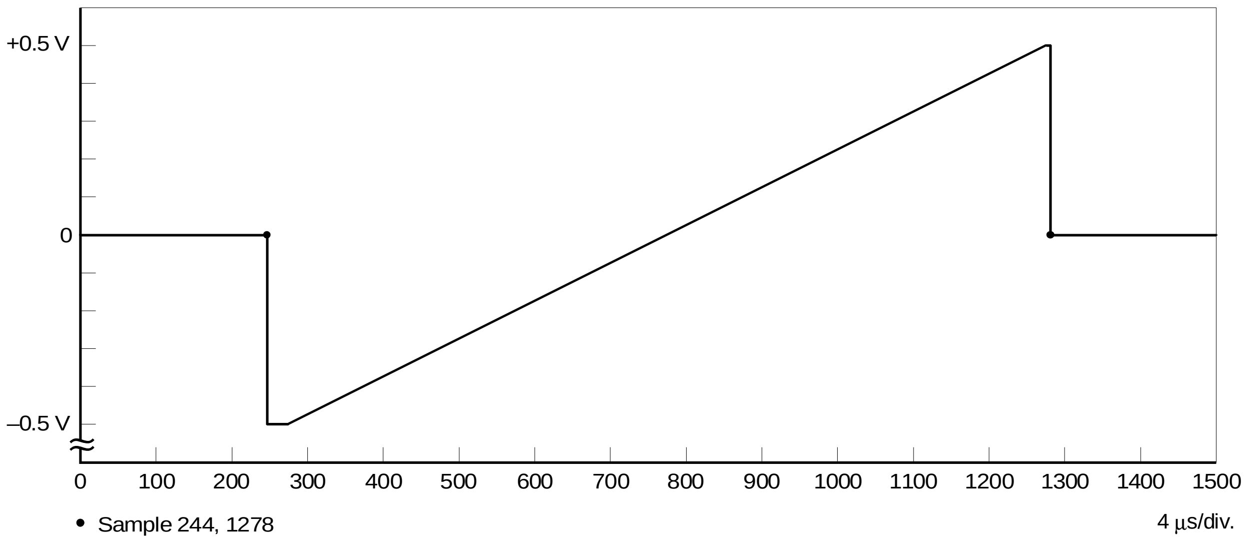
b) Test signal No. 1b – Odd frame

FIGURE A.1/J.67

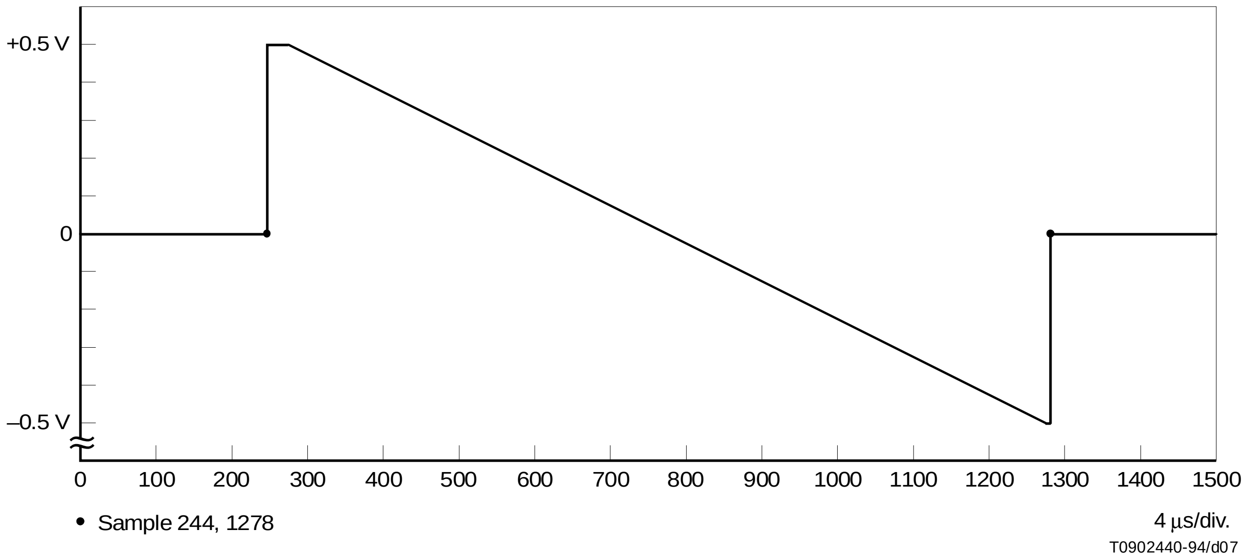
TABLE A.2/J.67

Definition of signals 2a and 2b
Signal 2a: even frames

k = 225 to 244: level 0 mV
k = 244 to 248: transition from 0 mV to −500 mV
k = 248 to 268: level −500 mV
k = 268 to 1268: −500 mV to +500 mV ramp
k = 1268 to 1274: level +500 mV
k = 1274 to 1278: transition from +500 mV to 0 mV
k = 1278 to 1292: level 0 mV
NOTE – Signal 2b (odd frames) has timing as above, levels inverted.



a) Test signal No. 2a – Even frame



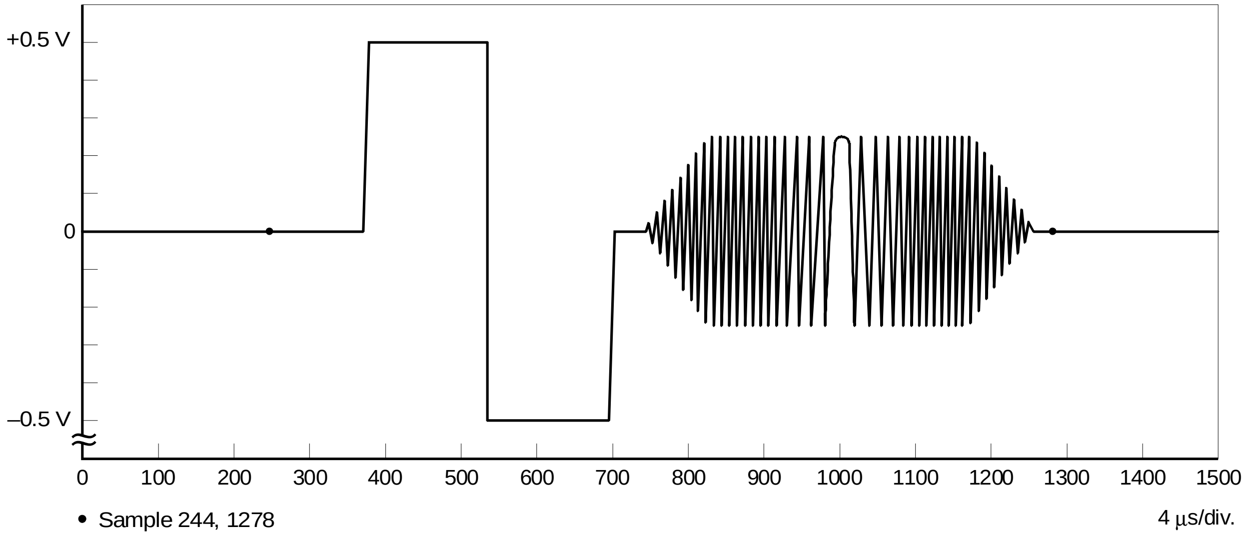
b) Test signal No. 2b – Odd frame

FIGURE A.2/J.67

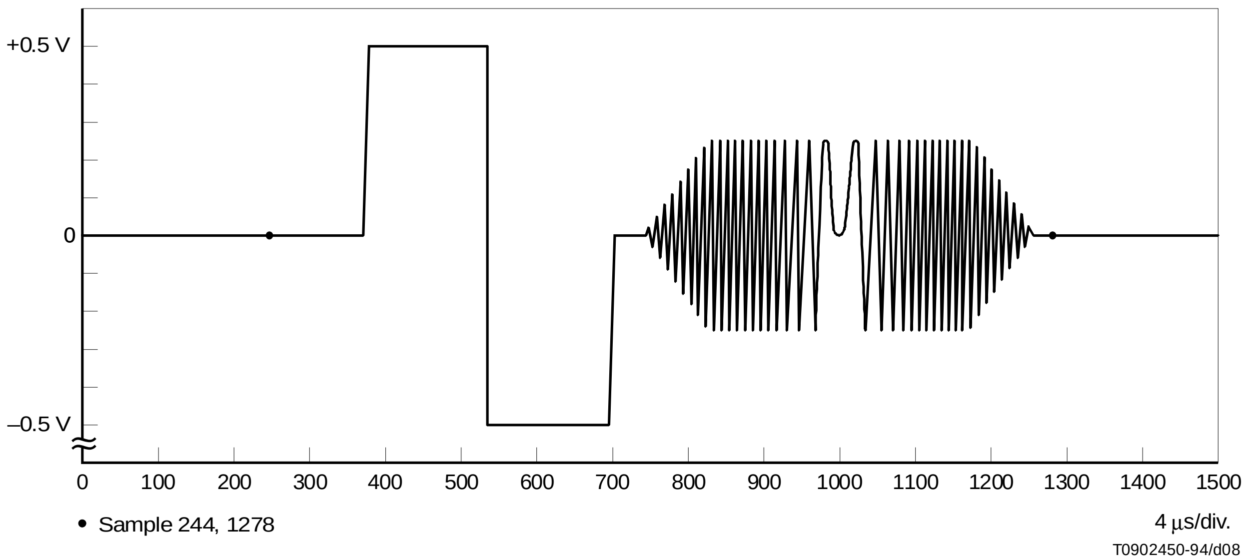
TABLE A.3/J.67

Definition of signals 3a and 3b

k = 255 to 370:	level 0 mV
k = 370 to 374:	transition from 0 mV to +500 mV
k = 374 to 532:	level +500 mV
k = 532 to 536:	transition from +500 mV to –500 mV
k = 536 to 694:	level –500 mV
k = 694 to 698:	transition from –500 mV to 0 mV
k = 698 to 739:	level 0 mV
k = 739 to 1251:	complex wobbulation of amplitud ± 250 mV transmitted in a 4-frame sequence: real part positive, imaginary part positive, real part negative, imaginary part negative.
k = 1251 to 1292:	level 0 mV



a) Test signal No. 3a – Even frame – Positive polarity of complex wobbulation signal



b) Test signal No. 3b – Odd frame – Positive polarity of complex wobbulation signal

FIGURE A.3/J.67

TABLE A.4/J.67

Definition of signal 4

k =	225	to	244:	level 0 mV
k =	244	to	248:	transition from 0 mV to +250 mV
k =	248	to	324:	level +250 mV
k =	324	to	330:	pulse (base +250 mV; peak –250 mV)
k =	330	to	406:	level +250 mV
k =	406	to	410:	transition from +250 mV to –250 mV
k =	410	to	486:	level –250 mV
k =	486	to	492:	pulse (base –250 mV; peak +250 mV)
k =	492	to	607:	level –250 mV
k =	607	to	688:	pulse modulated at 1 MHz
k =	688	to	690:	level –250 mV
k =	690	to	771:	pulse modulated at 2 MHz
k =	771	to	773:	level –250 mV
k =	773	to	854:	pulse modulated at 3 MHz
k =	854	to	856:	level –250 mV
k =	856	to	937:	pulse modulated at 4 MHz
k =	937	to	939:	level –250 mV
k =	939	to	1020:	pulse modulated at 5 MHz
k =	1020	to	1022:	level –250 mV
k =	1022	to	1103:	pulse modulated at 6 MHz
k =	1103	to	1105:	level –250 mV
k =	1105	to	1186:	pulse modulated at 7 MHz
k =	1186	to	1188:	level –250 mV
k =	1188	to	1269:	pulse modulated at 8 MHz
k =	1269	to	1274:	level –250 mV
k =	1274	to	1278:	transition from –250 mV to 0 mV
k =	1278	to	1292:	level 0 mV

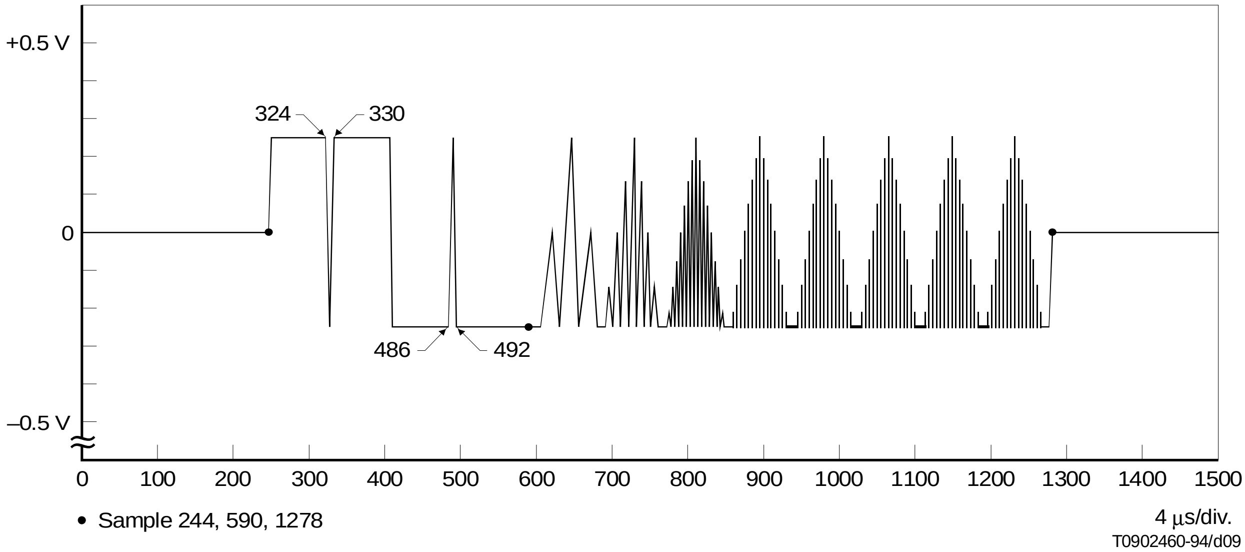


FIGURE A.4/J.67

Test signal No. 4

TABLE A.5/J.67

Definition of signal 5

k =	225	to	598:	level 0 mV
k =	598	to	602:	transition from 0 mV to –500 mV
k =	602	to	674:	level –500 mV
k =	674	to	678:	transition from –500 mV to –375 mV
k =	678	to	749:	level –375 mV
k =	749	to	753:	transition from –375 mV to –250 mV
k =	753	to	824:	level –250 mV
k =	824	to	828:	transition from –250 mV to –125 mV
k =	828	to	899:	level –125 mV
k =	899	to	903:	transition from –125 mV to 0 mV
k =	903	to	974:	level 0 mV
k =	974	to	978:	transition from 0 mV to +125 mV
k =	978	to	1049:	level +125 mV
k =	1049	to	1053:	transition from +125 mV to +250 mV
k =	1053	to	1124:	level +250 mV
k =	1124	to	1128:	transition from +250 mV to +375 mV
k =	1128	to	1199:	level +375 mV
k =	1199	to	1203:	transition from +375 mV to +500 mV
k =	1203	to	1274:	level +500 mV
k =	1274	to	1278:	transition from +500 mV to 0 mV
k =	1278	to	1292:	level 0 mV

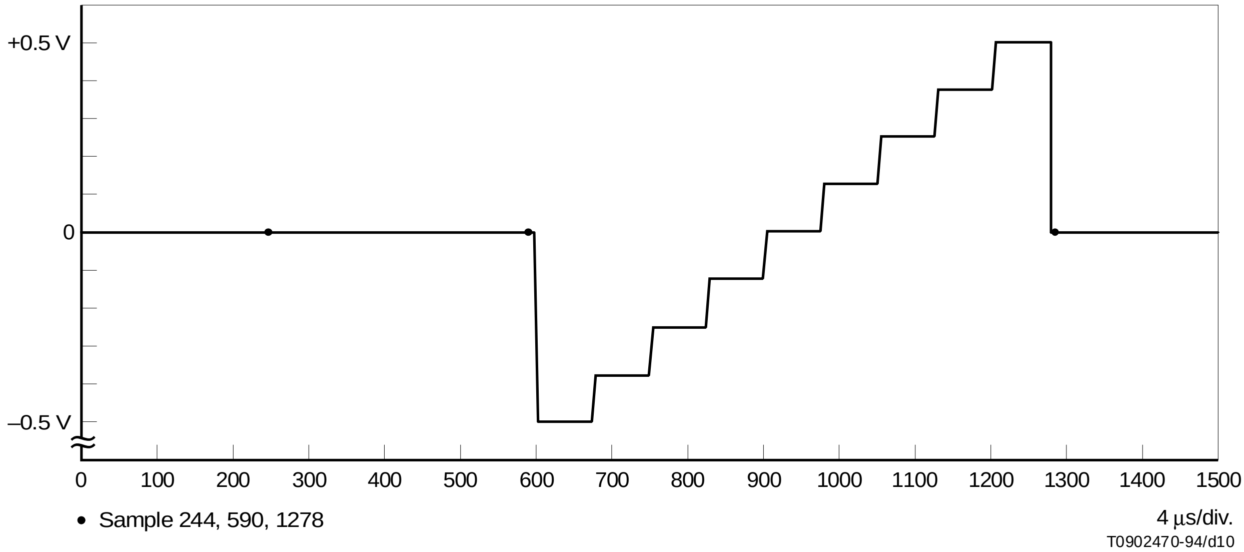


FIGURE A.5/J.67

Test signal No. 5

TABLE A.6/J.67

Definition of signal 6

k =	225	to	244:	level 0 mV
k =	244	to	248:	transition from 0 mV to –250 mV
k =	248	to	365:	level –250 mV
k =	365	to	369:	transition from –250 mV to +250 mV
k =	369	to	486:	level +250 mV
k =	486	to	490:	transition from +250 mV to 0 mV
k =	490	to	607:	level 0 mV
k =	607	to	688:	burst 1 MHz
k =	688	to	690:	level 0 mV
k =	690	to	771:	burst 2 MHz
k =	771	to	773:	level 0 mV
k =	773	to	854:	burst 3 MHz
k =	854	to	856:	level 0 mV
k =	856	to	937:	burst 4 MHz
k =	937	to	939:	level 0 mV
k =	939	to	1020:	burst 5 MHz
k =	1020	to	1022:	level 0 mV
k =	1022	to	1103:	burst 6 MHz
k =	1103	to	1105:	level 0 mV
k =	1105	to	1186:	burst 7 MHz
k =	1186	to	1188:	level 0 mV
k =	1188	to	1269:	burst 8 MHz
k =	1269	to	1292:	level 0 mV

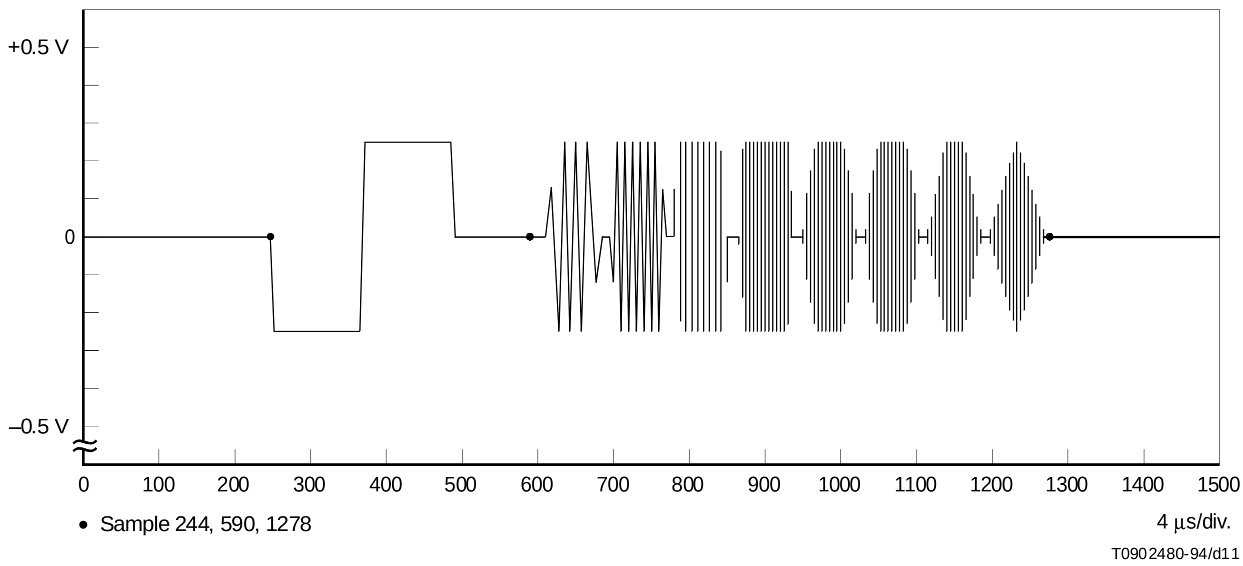


FIGURE A.6/J.67

Test signal No. 6

Annex B

(This annex forms an integral part of this Recommendation)

B.1 Introduction

Detailed definitions of the test signal elements contained in Tables B.1, B.2 and B.3 are given in 2.2.

In Tables B.1, B.2 and B.3 the transitions quoted are four *T*-transitions described in 1.3.2.2, and *k* is the sample number.

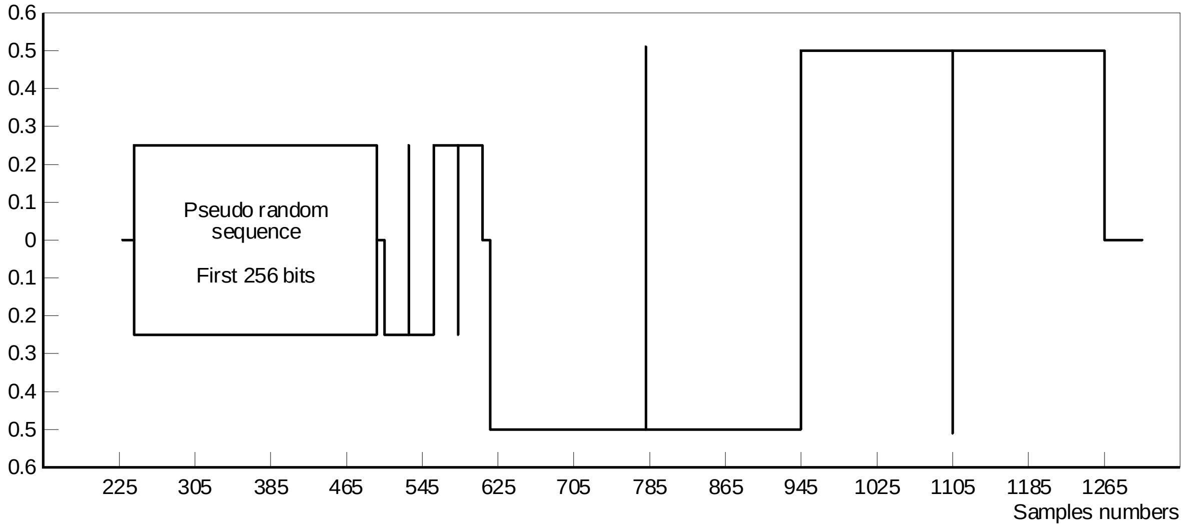
TABLE B.1/J.67

Test signal No. 1: Line 312
a) Even frame [see Figure B.1 a)]

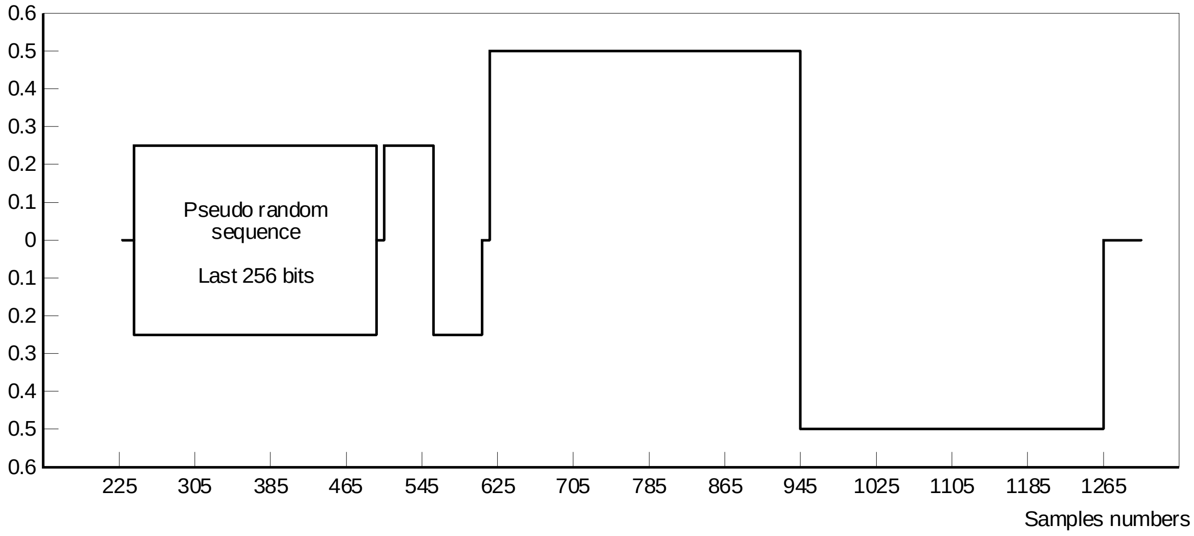
k =	225	to	233:	level 0 mV
k =	234	to	489:	level –250 mV or 250 mV (pseudo-random sequence)
k =	490	to	499:	level 0 mV
k =	500	to	524:	level –250 mV
k =	525:			level 250 mV
k =	526	to	550:	level –250 mV
k =	551	to	575:	level 250 mV
k =	576:			level –250 mV
k =	577	to	601:	level 250 mV
k =	602	to	614:	level 0 mV
k =	615	to	775:	level –500 mV
k =	776:			level 500 mV
k =	777	to	938:	level –500 mV
k =	939	to	1099:	level 500 mV
k =	1100:			level –500 mV
k =	1101	to	1262:	level 500 mV
k =	1263	to	1292:	level 0 mV

b) Odd frame [see Figure B.1 b)]

k =	225	to	233:	level 0 mV
k =	234	to	489:	level –250 mV or 250 mV (pseudo-random sequence)
k =	490	to	499:	level 0 mV
k =	500	to	550:	level 250 mV
k =	551	to	601:	level –250 mV
k =	602	to	614:	level 0 mV
k =	615	to	938:	level +500 mV
k =	939	to	1262:	level –500 mV
k =	1263	to	1292:	level 0 mV



a) Even frame

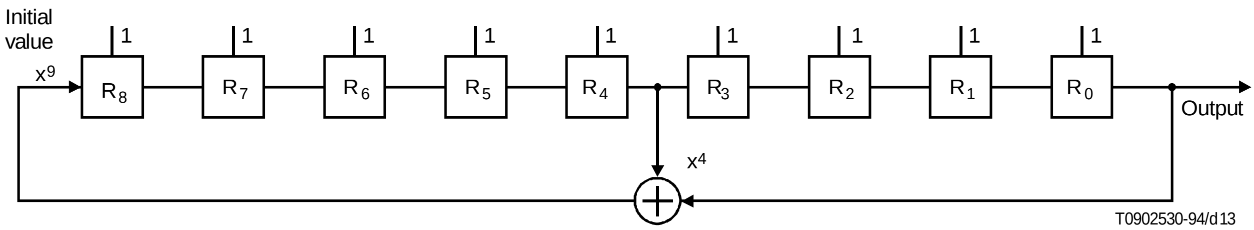


T0902520-94/d12

b) Odd frame

FIGURE B.1/J.67

Test signal No. 1



R 0 à R 8: Registers

FIGURE B.2/J.67

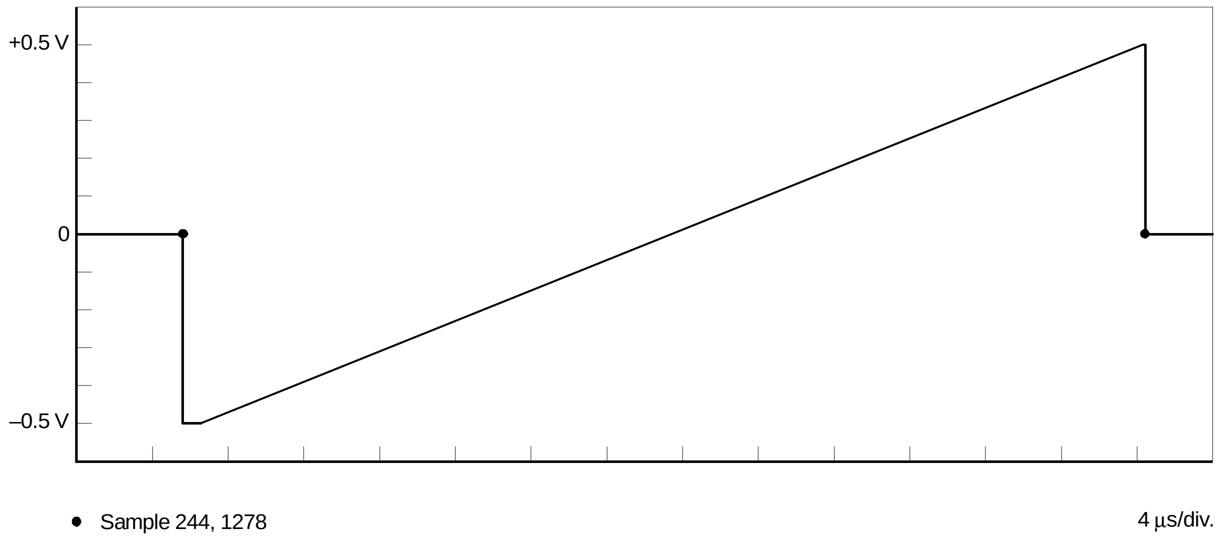
Pseudo-random generator for equalization

TABLE B.2/J.67

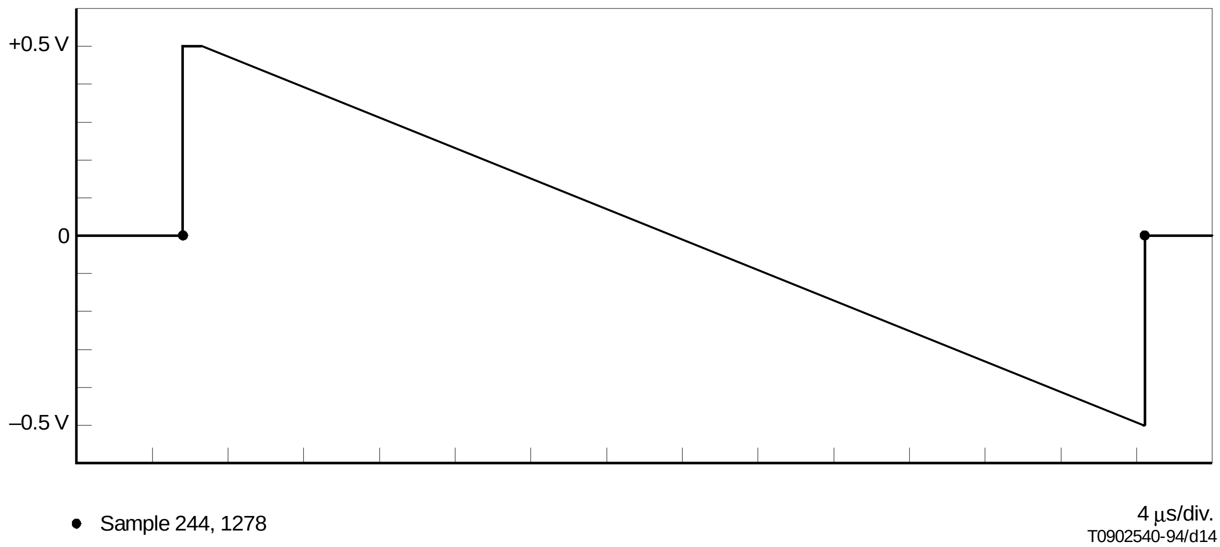
Test signal No. 2: Line 623
a) Even frame [see Figure B.3 a)]

k =	225	to	244:	level 0 mV
k =	244	to	248:	transition from 0 mV to –500 mV
k =	248	to	268:	level –500 mV
k =	268	to	1268:	ramp from –500 mV to +500 mV
k =	1268	to	1274:	level +500 mV
k =	1274	to	1278:	transition from +500 mV to 0 mV
k =	1278	to	1292:	level 0 mV

In the odd frames, the levels are inverted [see Figure B.3 b)].



a) Even frame



b) Odd frame

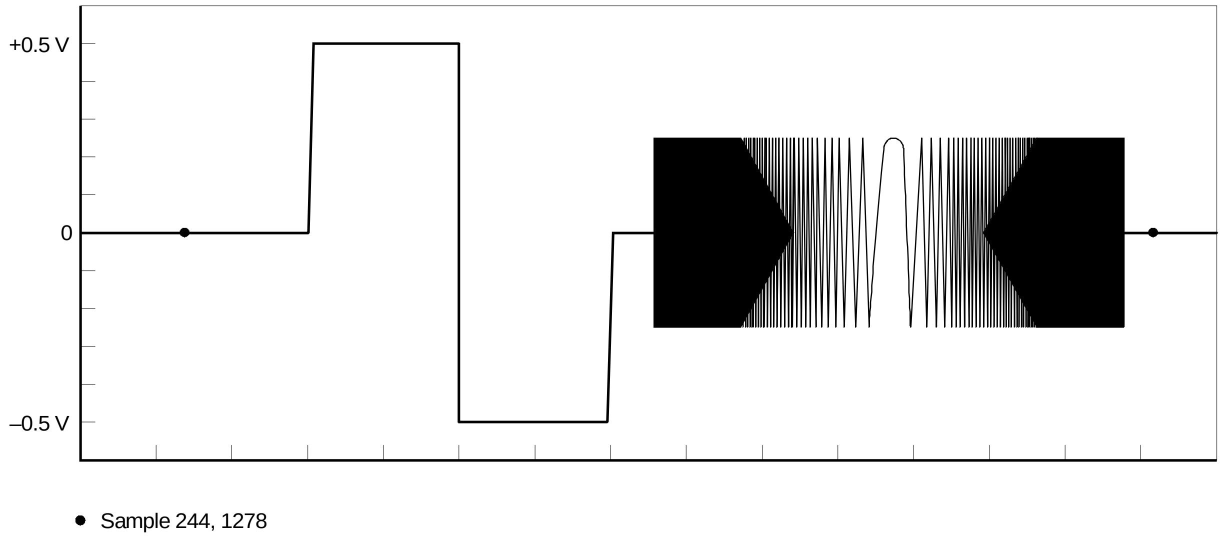
FIGURE B.3/J.67

Test signal No. 2

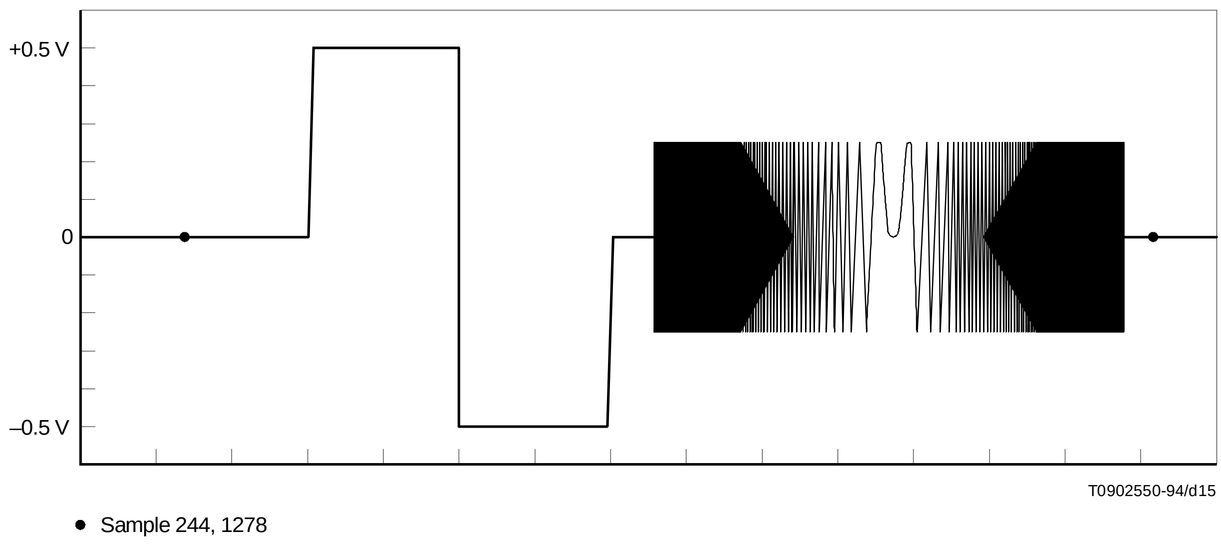
TABLE B.3/J.67

Test signal No. 3: Line 624
(see Figure B.4)

k = 225 to 370:	level 0 mV
k = 370 to 374:	transition from 0 mV to +500 mV
k = 374 to 532:	level +500 mV
k = 532 to 536:	transition from +500 mV to −500 mV
k = 536 to 694:	level −500 mV
k = 694 to 698:	transition from −500 mV to 0 mV
k = 698 to 739:	level 0 mV
k = 739 to 1251:	complex wobbulation with the levels ± 250 mV transmitted in a 4-frame sequence: real part positive, imaginary part positive, real part negative, imaginary part negative.
k = 1251 to 1292:	level 0 mV



a) Even frame



b) Odd frame

FIGURE B.4/J.67

Test signal No. 3

Annex C

(This annex forms an integral part of this Recommendation)

C.1 Introduction

Two filters are used for the measurement of a weighted signal-to-noise ratio. Both are derived from the former CCIR Recommendation 421-3, Annexes 2 and 3, with new suitable values.

C.2 Low-pass filter for use in measurements of continuous random noise

This filter is defined according to the former CCIR Recommendation 421-3, with $f_c = 10$ MHz, without group-delay compensation (see Figures C.1 and C.2, and Table C.1).

NOTE – This filter is the same as the one defined in Recommendation J.61 (formerly Recommendation ITU-R CMTT.567-3), Annex II to Part C, with $f_c = 5$ MHz.

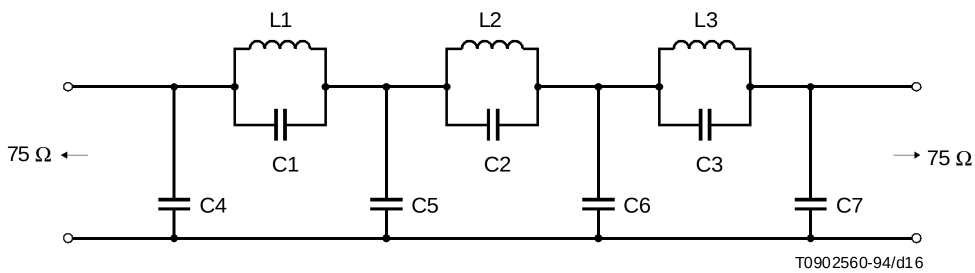


FIGURE C.1/J.67

TABLE C.1/J.67

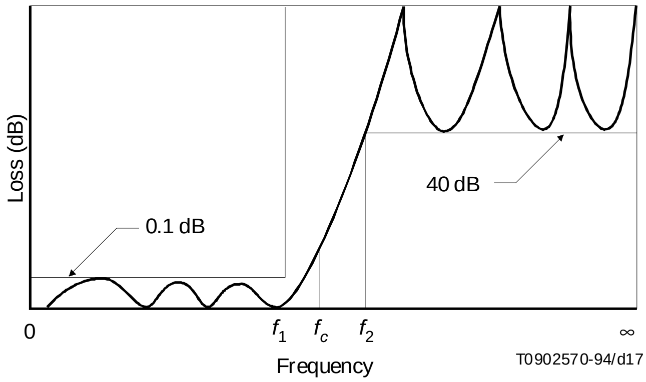
	Nominal upper video-frequency limit: f_c (MHz)		
	L (µH)	C (pF)	f (MHz)
1	$14.38/f_c$	$497.6/f_c$	$1.8816\ f_c$
2	$7.673/f_c$	$2723/f_c$	$1.1011\ f_c$
3	$8.600/f_c$	$1950/f_c$	$1.2290\ f_c$
4		$2139/f_c$	
5		$2815/f_c$	
6		$2315/f_c$	
7		$1297/f_c$	

NOTES

1 Each capacitance quoted is the total value, including all relevant stray capacitances, and should be correct to ± 2%.

2 Each inductor should be adjusted to make the insertion loss a maximum at the appropriate indicated frecuency, f (MHz).

f/f_c	dB	f/f_c	dB
0.98	0.1	1.04	14.8
0.99	0.5	1.05	18.8
1.00	1.8	1.06	23.0
1.01	4.2	1.07	27.7
1.02	7.3	1.08	33.3
1.03	10.9	1.09	41.0



Theoretical insertion loss
$f_1 = 0.9 f_2$ by design
Ringing frequency = f_c by design
$f_1 = 0.9807 f_c$
$f_2 = 1.0897 f_c$

NOTES

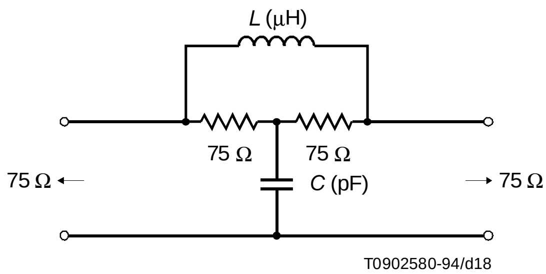
1 The theoretical insertion loss curve above corresponds to an infinite Q-factor. In practice, Q should be at least of the order of 100 at frequency f_c.

2 Limits for the insertion-loss/frequency characteristics are specified indirectly by the indicated tolerances on the component values.

FIGURE C.2/J.67

C.3 Continuous random-noise weighting network

This filter is defined according to the former CCIR Recommendation 421-3, with $\tau = 50$ ns (see Figure C.3).



$$L\,(\mu H) = 75\,\tau\,(\mu s);\ C\,(pF) = \frac{\tau\,(\mu s)}{75} \cdot 10^4$$

$$\text{Insertion loss (dB)} = 10 \log_{10} [1 + (2\pi\tau f)^2]$$

FIGURE C.3/J.67