INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

P.56

(03/93)

TELEPHONE TRANSMISSION QUALITY
OBJECTIVE MEASURING APPARATUS

OBJECTIVE MEASUREMENT OF ACTIVE SPEECH LEVEL

ITU-T Recommendation P.56

(Previously “CCITT Recommendation”)

FOREWORD

The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of the International Telecommunication Union. 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, established the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

ITU-T Recommendation P.56 was revised by the ITU-T Study Group XII (1988-1993) and was approved by the WTSC (Helsinki, March 1-12, 1993).

NOTES

1 As a consequence of a reform process within the International Telecommunication Union (ITU), the CCITT ceased to exist as of 28 February 1993. In its place, the ITU Telecommunication Standardization Sector (ITU-T) was created as of 1 March 1993. Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.

In order not to delay publication of this Recommendation, no change has been made in the text to references containing the acronyms "CCITT, CCIR or IFRB" or their associated entities such as Plenary Assembly, Secretariat, etc. Future editions of this Recommendation will contain the proper terminology related to the new ITU structure.

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

CONTENTS

OBJECTIVE MEASUREMENT OF ACTIVE SPEECH LEVEL

(Melbourne, 1988; amended at Helsinki, 1993)

1 Introduction

The CCITT considers it important that there should be a standardized method of objectively measuring speech level, so that measurements made by different Administrations may be directly comparable. Requirements of such a meter are that it should measure active speech level and should be independent of operator interpretation.

In this Recommendation, a meter is a complete unit that includes the input circuitry, filter (if necessary), processor and display. The processor includes the algorithm of the detection method.

In its present form, this meter can safely be used for laboratory experiments or can be used with care on operational circuits. Further study is continuing on:

a) how the meter can be used on 2-wire and 4-wire circuits to determine who is talking and whether it is an echo; and

b) how such an instrument can discriminate between speech and signalling, for example.

The method described herein maintains maximum comparability and continuity with past work, provided suitable monitoring is used, e.g. an operator performing the monitoring function. In particular, the new method yields data and conclusions compatible with those that have established the conventional value (22 microwatts) of speech power at the input to the 4-wire point of the international circuit according to Recommendation G.223. A method using operator monitoring can be found in Annex A.

This Recommendation describes a method that can be easily implemented using current technology. It also acts as a reference against which other methods can be compared. The purpose of this Recommendation is not to exclude any other method but to ensure that results from different methods give the same result.

Active speech level shall be measured and reported in decibels relative to a stated reference according to the methods described below, namely:

– *Method A* – Measuring a quantity called speech volume, used for the purpose of real-time control of speech level (see clause 4);

– *Method B* – Measuring a quantity called active speech level, used for other purposes (see clause 5).

Comparison of readings given by meters of methods A and B can be found in the *Handbook on Telephonometry*.

NOTE – This meter cannot be used to determine peak levels but sufficient information exists [1] giving the instantaneous peak/rms ratio, provided the signal has not been restricted or modified in any way, e.g. peak clipping.

2 Terminology

The recommended terminology is as follows:

speech volume	Until now used interchangeably with *speech level*, should in future be used exclusively to denote a value obtained by method A;
active speech level	Should be used exclusively to denote a value obtained by method B;
speech level	Should be used as a general term to denote a value obtained by any method yielding a value expressed in decibels relative to a stated reference.

The definitions of these terms [2], and other related terms such as those for the meters themselves [3], should be adjusted accordingly.

3 General

3.1 Electrical, acoustic and other levels

This Recommendation deals primarily with electrical measurements yielding results expressed in terms of electrical units, generally decibels relative to an appropriate reference value such as one volt. However, if the calibration and linearity of the transmission system in which the measurement takes place are assured, it is possible to refer the result backwards or forwards from the measurement point to any other point in the system, where the signal may exist in some non-electrical form (e.g. acoustical). Power is proportional to squared voltage in the electrical domain, squared sound pressure in the acoustical domain, or the digital equivalent of either of these in the numerical domain, and the reference value must be of the appropriate kind (1 volt, 1 pascal, reference acoustic pressure equal to 20 micropascals, or any other stated unit, as the case may be).

3.2 Universal requirements

For speech-level measurements of all types, the information reported should include: the designation of the measuring system, the method used (A, B, or B-equivalent as explained in clause 4, or other specified method), the quantity observed, the units, and other relevant information such as the margin value (explained below) where applicable.

All the relevant conditions of measurement should also be stated, such as bandwidth, position of the measuring instrument in the communication circuit, and presence or absence of a terminating impedance. Apart from the stated band limitation intended to exclude spurious signals, no frequency weighting should be introduced in the measurement path (as distinct from the transmission path).

3.3 Averaging

Where an average of several readings is reported, the method of averaging should be stated. The *mean level* (mean speech volume or mean active speech level), formed by taking the mean of a number of decibel values, should be distinguished from the *mean power*, formed by converting a number of decibel values to units of power, taking the mean of these, and then optionally restoring the result to decibels.

Any correction that has been applied should be mentioned, together with the facts or assumptions on which any such correction is based. For example, in loading calculations, when the active levels or durations of the individually measured portions of speech differ widely, $0.115\ \sigma^2$ is commonly added to the median or mean level in order to estimate the mean power, on the grounds that the distribution of mean active speech levels (dB values) is approximately Gaussian.

4 Method A – Immediate indication of speech volume for real-time applications

Measurement of speech volume for rapid real-time control or adjustment of level by a human observer should be accomplished in the traditional manner by means of one of the devices listed in Recommendation P.52.

The choice of meter and the method of interpreting the pointer deflexions should be appropriate to the application, as in Table 1.

Values obtained by method A should be reported as *speech volume*; the meter employed, the quantity observed, and the units in which the result is expressed, should be stated.

TABLE 1/P.56

Application	Meter	Quantity observed
Control of vocal level in live-speech loudness balances	ARAEN volume meter (SV3)	Level exceeded in 3 s
Avoidance of peak limiting	Peak programme meter	Highest reading
Maintenance of optimum level in making magnetic tape recording	VU meter	Average of peaks (excluding most extreme)

5 Method B – Active speech level for other applications than those mentioned in method A

5.1 Principle of measurement

Active speech level is measured by integrating a quantity proportional to instantaneous power over the aggregate of time during which the speech in question is present (called the active time), and then expressing the quotient, proportional to total energy divided by active time, in decibels relative to the appropriate reference.

The mean power of a speech signal when known to be present can be estimated with high precision from samples taken at a rate far below the Nyquist rate. However, the all-important question is what criterion should be used to determine when speech is present.

Ideally, the criterion should indicate the presence of speech for the same proportion of time as it appears to be present to a human listener, excluding noise that is not part of the speech (such as impulses, echoes, and steady noise during periods of silence), but including those brief periods of low or zero power that are not perceived as interruptions in the flow of speech [4]. It is not essential that the detector should operate exactly in synchronism with the beginnings and ends of utterances as perceived: there may be a delay in both operating and releasing, provided that the total active time is measured correctly. For this reason, complex real-time voice-activity detectors depending on sampling at the Nyquist rate, such as those that have been successfully used in digital speech interpolation, are not necessarily the most suitable for this application. Their function is to indicate when a channel is available for transmission of information: this state does not always coincide with the absence of speech; on the one hand, it may occur during short intervals that ought to be considered part of the speech, and on the other hand, it may be delayed long after the end of an utterance (for reasons of convenience in the allocation of channels, for example).

This Recommendation describes the detection method that meets the requirements. The method involves applying a signal-dependent threshold which cannot be specified in advance, so that accurate results cannot be guaranteed while the measurement is actually in progress; despite that, by accumulating sufficient information during the process, it is possible to apply the correct threshold retrospectively, and hence to output a correct result almost as soon as the measurement finishes. Continuous adaptation of the threshold level in real time appears to yield similar results in simple cases, but further study is needed to find out how far this conclusion can be generalized.

5.2 Details of realization

The algorithm for method B is as follows.

Let the speech signal be sampled at a rate not less than f samples per second, and quantized uniformly into a range of at least 2^{12} quantizing intervals (i.e. using 12 bits per sample including the sign).

NOTE – This requirement ensures that the dynamic range for instantaneous voltage is at least 66 dB, but two factors combine to make the range of measurable active speech levels about 30 dB less than this:

1) Allowance must be made for the ratio of peak power to mean power in speech, namely about 18 dB where the probability of exceeding that value is 0.001.

2) Envelope values down to at least 16 dB below the mean active level must be calculated: these values may be fractional, but will not be accurate enough if computed from a quantizing interval much exceeding twice the sample value; that is to say, it should not be expected that an active speech level less than about 10 dB above the quantizing interval would be measurable.

Let the successive sample values be denoted by x_i where i = 1, 2, 3, …. Let the time interval between consecutive samples be $t = 1/f$ seconds.

Other constants required are:

v	(Volts/unit) scale factor of the analogue-digital converter;
T	Time constant of smoothing in seconds;
$g = \exp(-t/T)$	Coefficient of smoothing;
H	Hangover time in seconds;
$I = H/t$	Rounded up to next integer;
M	Margin in dB, difference between threshold and active speech level.

Let the input samples be subjected to two distinct processes, 1 and 2.

Process 1

Accumulate the number of samples n, the sum s, and the sum of squares, sq:

$$\begin{aligned} n_i &= n_{i-1} + 1 \\ s_i &= s_{i-1} + x_i \\ sq_i &= sq_{i-1} + x_i^2 \end{aligned}$$

where s_0, sq_0 and n_0 (initial values) are zero.

Process 2

Perform two-stage exponential averaging on the rectified signal values:

$$\begin{aligned} p_i &= g \cdot p_{i-1} + (1-g) \cdot |x_i| \\ q_i &= g \cdot q_{i-1} + (1-g) \cdot |p_i| \end{aligned}$$

where p_0 and q_0 (initial values) are zero.

The sequence q_i is called the envelope, p_i denotes intermediate quantities.

Let a series of fixed threshold voltages c_i be applied to the envelope. These should be spaced in geometric progression, at intervals of not more than 2:1 (6.02 dB), from a value equal to about half the maximum code down to a value equal to one quantizing interval or lower. Let a corresponding series of activity counts a_j, and a corresponding series of hangover counts, h_j, be maintained:

for each value of j in turn,

if $q_i > c_j$ or $q_i = c_j$, then add 1 to a_j and set h_j to 0;

if $q_i < c_j$ and $h_j < I$, then add 1 to a_j and add 1 to h_j;

if $q_i < c_j$ and $h_j = I$, *then* do nothing.

In the first case, the envelope is at or above the jth threshold, so that the speech is active as judged by that threshold level. In the second case, the envelope is below the threshold, but the speech is still considered active because the corresponding hangover has not yet expired. In the third case, the speech is inactive as judged by the threshold level in question.

Initially, all the a_j values are set equal to zero, and the h_j values set equal to I.

It should be noted that the suffix i in all the above cases is needed only to distinguish current values from previous values of accumulated quantities; for example, there is no need to hold more than one value of sq, but this value is continually updated. At the end of the measurement, therefore, the suffixes can be omitted from s, sq, n, p, and q.

Let all these processes continue until the end of the measurement is signalled. Then evaluate the following quantities:

$$\text{Total time} = n \cdot t$$

$$\text{Long-term power} = sq \cdot v^2/n$$

NOTE – If it is suspected that there may be a significant d.c. offset, this may be estimated as $s \cdot v/n$, and used to evaluate a more accurate value of long-term power (a.c.) as $v^2\,[sq/n - (s/n)^2]$. However, in this case, the effect of the offset on the envelope must also be taken into account and appropriate corrections made.

For each value of j, the active-power estimate is equal to $sq \cdot v^2/a_j$.

At this stage, the powers are in volts squared per unit time. Now express the long-term power and the active-power estimates in decibels relative to the chosen reference voltage r:

Long-term level $L = 10 \log (sq \cdot v^2/n) - 20 \log r;$

Active-level estimate $A_j = 10 \log (sq \cdot v^2/a_j) - 20 \log r;$

Threshold $C_j = 20 \log (c_j \cdot v) - 20 \log r.$

For each value of j, compare the difference $A_j - C_j$ with the margin M, and determine (if necessary, by interpolation on a decibel scale between two consecutive values of A_j and of C_j) the true active level A and corresponding threshold C for which $A - C = M$. If one of the pairs of values A_j and C_j fulfils this condition exactly, then the true activity factor is a_j/n, but in all cases it can be evaluated from the expression $10^{(L-A)/10}$.

For simplicity, the algorithm has been defined in terms of a digital process, but any equivalent process (one implemented on a programmable analogue computer, for example) should also be considered as fulfilling the definition.

5.3 Values of the parameters

The values of the parameters given in Table 2 should be used. They have been found suitable for the purpose and have stood the test of many years of application by various organizations [4].

TABLE 2/P.56

Parameter	Value	Tolerance
f	694 samples/second	Not less than 600
T	0.03 seconds	± 5%
H	0.2 seconds	± 5%
M	15.9 dB	± 0.5
NOTE – The value M = 15 dB might appear to be implied in [4], but the threshold level there described equals the *mean absolute voltage* of a sine wave whose *mean power* is 15 dB below the reference. The difference of 0.9 dB is 20 log (voltage/mean absolute voltage) for a sine wave.		

The result of a measurement made by means of the above algorithm with parameter values conforming to the above restrictions should be reported as *active speech level*, and the system should be described as *using method B* of this Recommendation.

NOTE – Where noise levels are very high, as they are for example in certain vehicles or in certain radio systems, it is often desirable to set the threshold higher (i.e. use a smaller margin) in order to exclude the noise. This may be done provided the margin is also reported. The result of such a measurement should be reported as *active speech level with margin M*, and the measurement system described as *using method B with margin M*.

The activity factor should preferably be reported as a percentage, with a specification of the margin value if this is outside the standard range.

6 Approximate equivalents of method B

Other methods under development use a broadly similar principle of measurement but depart in detail from the algorithm given above.

It is not the intention to exclude any such method, provided it is convincingly shown by experimental evidence to yield results consistent with those obtained by method B in a sufficiently wide range of conditions. For this reason, a class of methods called *B-equivalent methods* is recognized.

A B-equivalent method of speech-level measurement is defined as any method that satisfies the following test in all respects.

Measurements shall be carried out simultaneously by the method in question and by method B on two or more samples of speech in every combination of the following variables:

Voices	One male and one female voice.
Speech material	A list of independent sentences, a passage of continuous speech, and one channel of a conversation, each lasting at least 20 s (active time).
Bandwidth	300 to 3400 Hz and 100 to 8000 Hz.
Added noise	Flat within the measurement band at levels (M + 5) dB and (M + 25) dB below the active speech level, where M (the margin) is normally 15.9 dB, but smaller in high-noise applications.
Levels	At intervals of 10 dB over the range claimed for the system in question.

From the results, 95% confidence limits for the difference between the level given by the method in question and the active speech level given by method B shall be calculated for each of the above 24 combinations.

If, for every combination, the upper confidence limit of this difference is not higher than +1 dB and the lower confidence limit is not lower than –1 dB, then the method shall be deemed to be a B-equivalent method.

This verification procedure is valid until a suitable speech-like signal has been recommended and found suitable to perform this function.

Further, a method qualifies as B-equivalent if it gives results that fall within the specified limits when corrected by the addition of a fixed constant, known in advance of the measurement and not dependent on any feature of the speech signal (except possibly the bandwidth if this is known independently).

The results of measurements by such a method should be reported as *B-equivalent active speech level*, and the activity factor as *B-equivalent activity factor*.

Certain measurement systems with fixed thresholds (instead of the retrospectively selected threshold as described in 5.3), may still give an active speech level according to the definition in cases where the margin turns out to be within the specified limits.

7 Specification

A speech voltmeter normally consists of three parts, namely:

i) input circuitry;

ii) filter; and

iii) processor and display.

Figure 1 shows a typical layout of such a meter.

Whether all or part of the components that make up i) and ii) are used will depend on where the meter is to be used. However, it is recommended that a meter for general usage should conform to this specification.

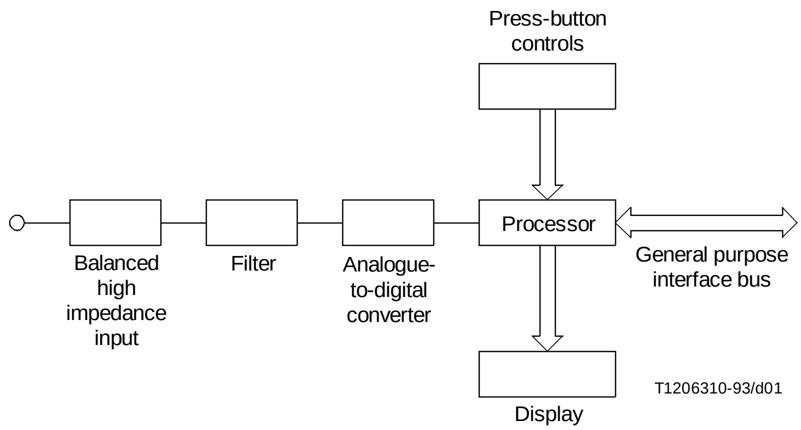

FIGURE 1/P.56

Block diagram of a typical meter

7.1 Signal input

7.1.1 Input impedance

The meter is normally used as a bridging instrument and, if so, its impedance must be high so as not to influence the results. An impedance of 100 kohm is recommended.

7.1.2 Circuit protection

It is recommended that the meter should withstand voltages far in excess of those in the measurement range as accidental usage may occur and the circuit under test may have higher voltages than anticipated. Examples of this are mains 110/240 V or 50 V exchange voltages.

7.1.3 Connection

It is recommended that the connection should be independent of polarity. The meter should have the facility of connection in both balanced and unbalanced modes.

7.2 Filter

When measuring the speech levels of circuits in the conventional telephony speech bandwidth (300-3400 Hz), it is often practical to use a filter that will reject unwanted hum, tape noise, etc. yet pass the frequencies of greatest interest without affecting the speech level measurement. The set of coordinates in Table 3 meet these requirements. Figure 2 gives an example of such a filter.

The following noise requirements should also be met:

– Output noise level:

  wideband (20-20 000 Hz) < –75 dBm;

  telephone weighted < –90 dBmp.

TABLE 3/P.56

Frequency (Hz)	(dB)
	Upper limit response relative to 1 kHz
16 160 7 000 70 000	– 49.75 + 0.25 + 0.25 – 49.75
	Lower limit response relative to 1 kHz
Under 200 200 5 500 Over 5 500	– ∞ – 0.25 – 0.25 – ∞

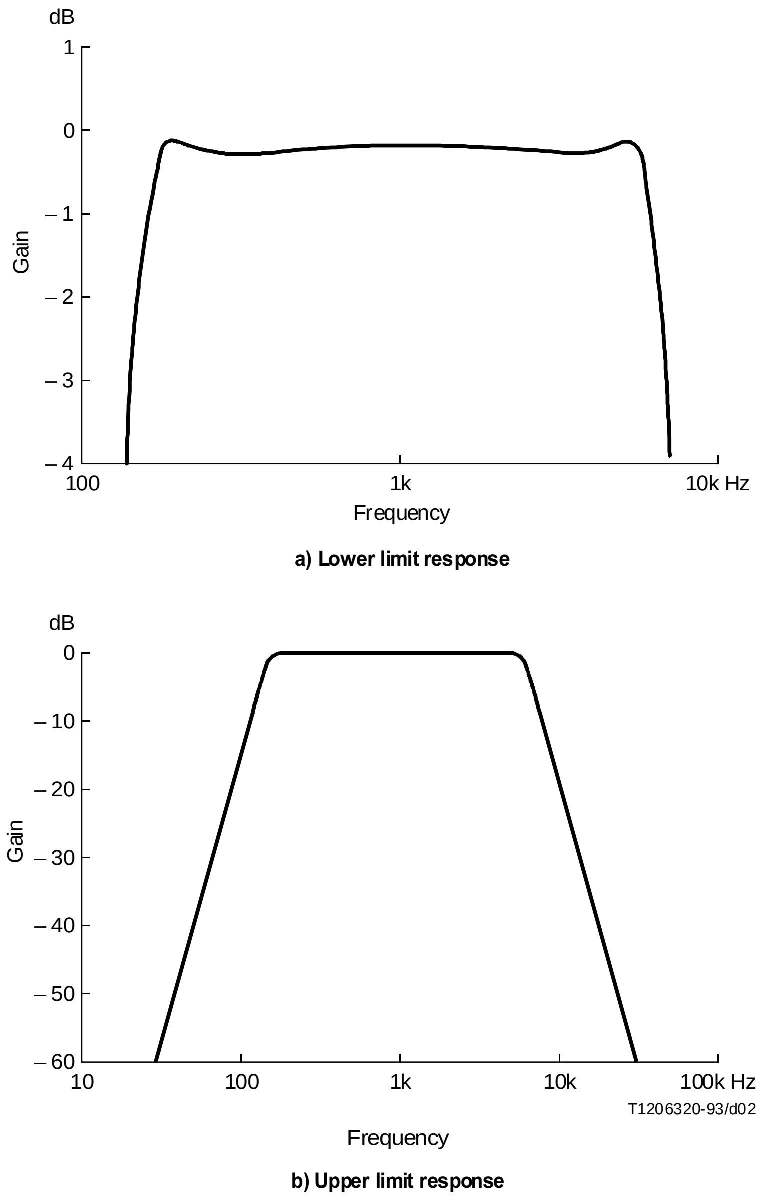

FIGURE 2/P.56

Filter passband response

7.3 Speech level measurements

7.3.1 Working range for speech

The recommended working range for speech refers to the active level and should be at least 0 to −30 dBV.

NOTES

1 The dynamic range of the instrument will depend on the analogue-to-digital converter (ADC). If the ADC is set to a 10 volt maximum input level (i.e. the all 1 code) and 12-bit arithmetic is used, based on the most significant bits from the ADC, then 1 sign bit +11 bits magnitude provides a 66 dB range. The measurable range will be some 35 dB less when allowance is made for the peak/mean ratio of 18 dB (peaks of speech will only exceed the maximum input level for less than 0.1% of the time [1]) and margin M of 15.9 dB; the largest speech signal is therefore around +2 dBV with a smallest speech signal of –30 dBV. However, the practical working range has been found to be +5 dBV to –35 dBV.

2 To cater for a wider range of speech levels, an attenuator or low noise amplifier may be inserted in the input circuitry. Care must be exercised to maintain the input requirements of 7.1.1.

7.3.2 Linearity

The linearity of the meter is specified for rms sine wave measurements since for speech the algorithm is correct by definition, and only the precision or repeatability of measurements need to be considered; this is specified in 7.3.4.

Assuming that

a) the measurement is for a minimum period of 5 s,

b) the sine wave is present for the whole of the measurement period, the linearity specified is:

Frequency (Hz)	Input range (dBV)	Accuracy (dB)
100 to 4000	+ 16 to – 45	± 0.1
4000 to 8000	+ 13 to – 45	± 0.3

NOTE – The maximum input for the frequency range 4000 Hz to 8000 Hz should ideally be the same as for 100 Hz to 4000 Hz, but practical limitations in commercially available ADCs (due to the limited "slewing rate" of the input circuitry) means that this cannot be obtained. However, as the power in the 8000 Hz band for speech is 30 dB down on the level at 500 Hz it is likely that any error will be extremely small.

7.3.3 Frequency response

The frequency response of the meter without filter when measured in the frequency range 100 Hz to 8000 Hz should be flat within the specified tolerances:

Frequency (Hz)	Input range (dBV)	Accuracy (dB)
100 to 4000	+ 16 to – 45	± 0.2
4000 to 8000	+ 13 to – 45	± 0.4

NOTES

1 Tolerances are referred to 1000 Hz.

2 The Note of 7.3.2 applies.

7.3.4 Repeatability

When a given speech signal, having its active level within the recommended working range and its duration not less than 5 s active time, is repeatedly measured on the same meter, the active-level readings shall have a standard deviation of less than 0.1 dB.

8 Routine calibration of method-B meter

The following routine calibration procedures, using non-speech-like signals, will ensure that the meter is performing satisfactorily. The calibration can only be made using speech.

A suitable circuit arrangement is shown in Figure 3. Wherever suitable, measurements should be made with two settings of the attenuator, 0 and 20 dB. All source signals are from a 600 ohm source and the meter is terminated in 600 ohm.

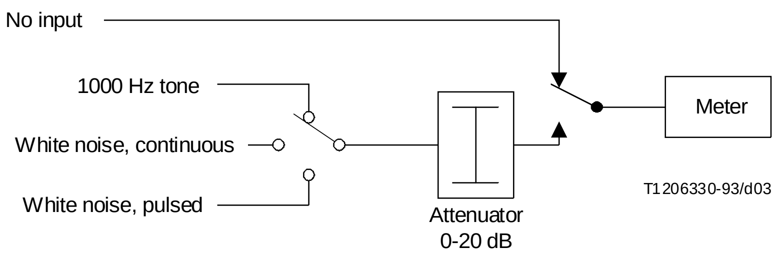

FIGURE 3/P.56

Switching arrangement

8.1 No input signal

With no input applied, the meter should display the following results:

Activity factor	0 + 0.5%
Active-level	< – 60 dBV
Long-term level	< – 60 dBV

8.2 Continuous tone

With a 1000 Hz sine wave calibrated to be 0 dBV, the meter should display the following results for the two settings of the attenuator when applied for 12 + 0.2 s:

	Attenuator = 0 dB	Attenuator = 20 dB
Activity factor	100 to 0.5%	100 to 0.5%
Active-level	0 ± 0.1 dBV	–20 ± 0.1 dBV
Long-term level	0 ± 0.1 dBV	–20 ± 0.1 dBV

8.3 White noise

8.3.1 Without filter

With the meter having no filter in circuit and the white noise source calibrated to be 0 dBV, the meter should display the following results for the two settings of the attenuator when applied for 12 + 0.2 s:

	Attenuator = 0 dB	Attenuator = 20 dB
Activity factor	100 to 0.5%	100 to 0.5%
Active-level	0 ± 0.5 dBV	–20 ± 0.5 dBV
Long-term level	0 ± 0.5 dBV	–20 ± 0.5 dBV

8.3.2 With filter

With the meter having the filter in circuit and the white noise source calibrated to be 0 dBV, the meter should display the following results for the two settings of the attenuator when applied for 12 + 0.2 s:

	Attenuator = 0 dB	Attenuator = 20 dB
Activity factor	100 to 0.5%	100 to 0.5%
Active-level	– 6.9 ± 0.5 dBV	–26.9 ± 0.5 dBV
Long-term level	– 6.9 ± 0.5 dBV	–26.9 ± 0.5 dBV

8.3.3 Pulsed noise

With the meter having no filter in circuit and the white noise source pulsed at 3 s "ON" and 3 s "OFF" and calibrated to be 0 dBV when "ON", the meter should display the following results for the two settings of the attenuator when applied for 12 + 0.2 s:

	Attenuator = 0 dB	Attenuator = 20 dB
Activity factor	55 ± 1.5%	55 ± 1.5%
Active-level	0 ± 1 dBV	–20 ± 1 dBV
Long-term level	– 2.7 ± 1 dBV	–22.7 ± 1 dBV

NOTE – It is possible that clause 8 could be revised to calibrate both method B and B-equivalent meters when a speech-like signal has been found suitable to perform this function.

Annex A

A method using a speech voltmeter complying with method B in network conditions

(This annex forms an integral part of this Recommendation)

A speech voltmeter complying with method B is not suitable in its present form for speech measurements (see, for example, Recommendation G.223) on real connections since the meter is unable to distinguish between speech coming from one or the other end of the connection. However, if the meter is connected to a 4-wire point in a connection of the type 2-4-2 wire, then measurements may be made using an operator monitoring the beginning and the end of the conversation. The operator can perform this function using earphones (provided the subscriber's permission has been obtained) or by an auxiliary meter (for example conforming to P.52). The circuit arrangement is shown in Figure A.1.

The operator monitors the conversation, using the auxiliary meter or earphones, and then by means of a start/stop button can measure the beginning and end of the relevant conversation.

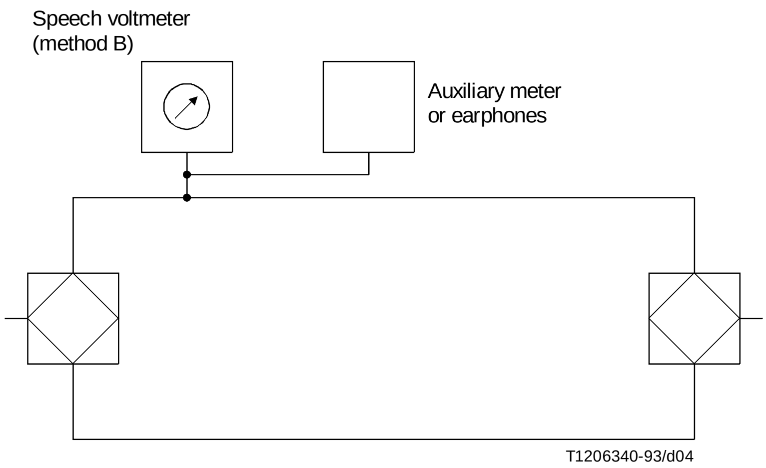

FIGURE A.1/P.56

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