INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

Supplement 21

(Series P)

(03/93)

TELEPHONE TRANSMISSION QUALITY

SUBSCRIBERS' LINES AND SETS

THE PRINCIPLES OF A COMPOSITE SOURCE SIGNAL AS AN EXAMPLE OF A MEASUREMENT SIGNAL TO DETERMINE THE TRANSFER CHARACTERISTICS OF TERMINAL EQUIPMENT

Supplement 21 to

ITU-T Series P Recommendations

(Previously "CCITT Recommendations")

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.

Supplement 21 to ITU-T Series P Recommendations was prepared 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 Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.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

Supplement 21

THE PRINCIPLES OF A COMPOSITE SOURCE SIGNAL AS AN EXAMPLE OF A MEASUREMENT SIGNAL TO DETERMINE THE TRANSFER CHARACTERISTICS OF TERMINAL EQUIPMENT

(Helsinki, 1993)

(referred to in Series P Recommendations)

1 Introduction

Non-linear processes such as Acoustic Echo Control (AEC), Automatic Gain Control (AGC), compression systems, etc. are increasingly used in terminal equipment providing telephony, e.g. hands-free telephones or mobile telephone systems to improve speech transmission quality. Echo cancellers or echo suppressors also are used for echo reduction in the network in the case of long-distance calls. The transfer characteristics of all these types of equipment can not be described as linear and time-invariant (LTI). As a consequence, standard measurement procedures using stationary measurement signals can not be used to determine the transfer characteristics of such devices.

A large variety of more suitable non-stationary measurement signals exist, ranging from chirps to complex modulated signals[1)]. In this supplement the principles of a special measurement signal called the composite source signal (CSS) are given as an example of a non-stationary measurement signal.

2 Requirements for measurement signals and analysis procedures

Measurement procedures yielding defined and reproducible results are required to determine transfer functions for tests and certification purposes.

On the one hand, such a signal allowing the determination of the transfer characteristics of these systems must simulate essential properties of real voice adequately (i.e. provide voiced sound, unvoiced sound, modulation, temporal structure, etc.). On the other hand, such a signal must be specified so that not only the transfer function in different operating modes can be measured but also the switching characteristics between modes and the behaviour of such systems in duplex operation. Also the echo return loss and especially the temporal behaviour of echo cancelling equipment should be measured.

In order to meet these conflicting requirements, the composite source signal CSS was defined as a suitable compromise. The CSS as well as its calibration consists basically of different types of signals described in clause 3 which can be composed in different manners. The exact definition of this composition has not yet been fixed,

3 Composite source signal

When composing the composite source signal, the following three components were judged important:

– voiced signal to simulate voice properties;

– deterministic signal for measuring the transfer functions without statistical errors with
constant power density spectrum of the excitation signal in the frequency domain to be measured;

---

1) Different kinds of measurement signals exist. These may be divided into stationary and non-stationary categories.

Stationary signals:	sine wave, multi-sine wave, noise, maximum length (MLS), pseudo-noise (PN) sequences
Non-stationary signals:	chirps
with simple modulated signals:	impulse trains, bursts of tone/noises/chirps, CSS
with complex modulated signals:	IEC RASTI, P.50 artificial voices, P.59 artificial conversational voices, real voices.

– pause “signal” providing amplitude modulation. The following features result:

i) short period of measurement;

ii) feeding-in possibility of the test signal for the talking and listening direction at the same time (duplex operation).

The basic idea for using such a signal is placing the device under test in a well defined, reproducible state for the period of measurement and to secure that the transfer functions of the device do not change appreciably during the actual measurement (quasi-stationarity). More precisely, the composite source signal (see Figure 1) consists of the following components:

a) *Voiced sound produced from the “artificial voice” signal according to Recommendation P.50*

The voiced sound of the CSS is the conditionning signal intended to activate possible speech detectors in voice-controlled systems. The reason why the voiced sound has been chosen is that presumably all future hands-free telephones will quickly respond to a voiced sound. This signal is to activate a hands-free telephone for the direction of transmission to be measured. As the duration, beginning and end of the voiced sound are known exactly, this signal can also be used to measure the switching time for the direction of transmission under test. By means of the signal shape in the time domain the switching time and delay time of the entire system can be determined according to Recommendation P.34. The duration of the signal amounts to 50 ms. Within this period each speech detector must have recognized voice and activated the system.

b) *Pseudo noise signal*

The actual test signal is the Pseudo Noise (PN) signal presented after the voiced artificial speech sound. This signal has certain noise like features. The magnitude of its Fourier transform is constant in frequency while the phase is changing. For measuring hands-free telephones, usually only the magnitude of the transfer function is of interest, the phase is not that important but can be determined as well.

The signal may be produced as follows:

First a complex spectrum is produced in the frequency domain according to the following equation:

$$\begin{cases} S(k) = W(k)\,e^{ji_k\pi} & ;i_k \ E\,\{0,1\}, \text{random}, 0 \le k \ge \dfrac{M}{2} \\ S(M-k) = S^*(k) & ;0 \le k \le \dfrac{M}{2} \end{cases}$$

Note that S(0) and $S\left(\frac{M}{2}\right)$ must be real numbers.

Index M is adjusted to the chosen FFT size (e.g. 2048 points). The equation shows that the amount of the produced complex spectrum is constant for all frequencies if W(k) is chosen equal to 1 for all frequencies, whereas the phase may be $+\pi$ or $-\pi$ for each frequency corresponding to a random sequence. However, to produce a different weighting in the frequency domain, W(k) can easily be adjusted in order to produce different spectra for the duration of the PN-sequence. Then, this spectrum will be transformed into the time domain by means of the inverse Fourier transform producing the following signal:

$$s(n) = \frac{1}{M} \sum_{k=-\frac{M}{2}}^{\frac{M}{2}-1} W(k)\,e^{ji_k\pi}\ e^{j2\pi nk/M} \quad ; \quad -\frac{M}{2} \le n \le \frac{M}{2} - 1$$

Thus, a signal is produced which is limited in time (corresponding to the chosen length of the Fourier transform) and which is adjusted to the chosen FFT size correctly. If a longer time sequence is wanted, the signal can be cycled. This method permits time sequences of any length.

The duration of this test signal may amount to about 200 ms by appropriate choice of M and the sampling rate.

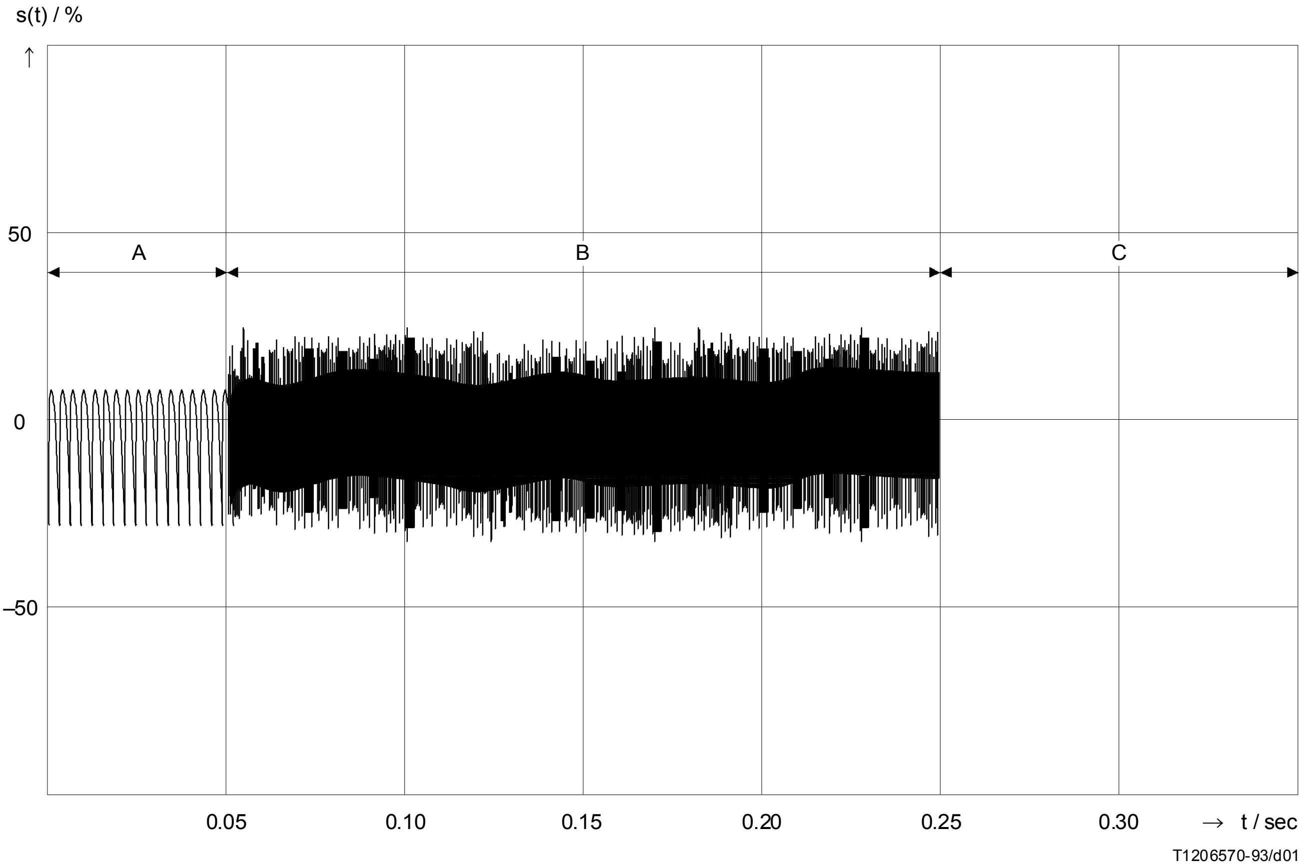
c) *Pause*

The pause has two purposes. An initial pause before applying any measurement signal is necessary to put systems with time-variant transfer functions into a defined initial state. To this end, the pause should be as long as possible (> 1 s). If, however, the system is to be put into a constantly activated state (running speech like), the intermediate pauses should be shorter (about 100 ms) to provide suitable amplitude modulation to the composite signal.

For the measurement, a system is required that is able to extract the measurement signal components from the composite signal, i.e. the PN signal described above. The transfer function of the measured device can be determined relatively simply by means of the FFT. The transfer function of the device under test can be estimated from the ratio of the power density spectrum of the output signal to the power density spectrum of the input signal according to the following equation:

$$H(k) = 20 \log \frac{|F\{So(n)\}|}{|F\{Si(n)\}|}$$

If the measurement is made with the test equipment described above and if the test signal levels are chosen according to the CCITT Recommendations, such a measurement is compatible with all measurements according to CCITT Recommendations made up to now as far as LTI devices under test are concerned. To determine the short-term level just the measurement signal, i.e. the PN signal should be used, for long-term measurements the long-term level of the entire CSS is required.



A Voiced sound of P.50 artifical voice
B Pseudo random noise
C Pause

FIGURE 1

Composite Source Signal

4 Further possibilities of the composite source signal

Instead of the PN signal depicted above, other signal shapes in the composite source signal may be used as well. For example, if distortion is to be measured, a different signal is needed. Then, e.g., a pure sine signal may be used instead of a PN signal. Such a measurement would be compatible with the CCITT Recommendations if the measurement results are calculated from the signal part inserted instead of the PN sequence. However, it must be considered that hands-free telephones might possibly detect a pure signal as singing and interrupt the system accordingly. This could be seen e.g. by observation of the signal in the time domain. To perform this measurement, a different test signal, e.g. narrow-band noise, has to be used.

Different applications of the composite source signal have already been reported in the reference listed below. Future studies are planned to describe more possibilities using composite source signals.

References

[1] SCHRÖDER (M. R.): Synthesis of Low Peak Factor Signals and Binary Sequences with low Autocorrelation, *IEEE Transactions on Information Theory*, IT-6, pp. 85-89, 1970.

[2] Measurement of the Transfer Functions of Hands-free Telephones, ETSI TE4, Oslo, TD No. 49, 1990.

[3] CCITT COM XII-67, Measurement of the Transfer Functions of Hands-free Telephones, Comparison Between Results Measured with Artificial Voice and a Composite Source Signal, (FRG), August 1990.

[4] CCITT COM XII-68, Measurement of Time Constants on Hands-free Telephones in Single-talk and Double-talk Operation, (FRG), August 1990.

[5] CCITT COM XII-D.74, Use of the Composite Source Signal, (FRG), August 1990.

[6] GIERLICH (H. W.): A Measurement Technique to Determine the Transfer Characteristics of Hands-free Telephones, *Signal Processing*, Vol. 26, No. 2, 1992.

[7] CCITT COM XII-D.109, Uses of the Composite Source Signal: Results of Comparison Measurements in Different Test Laboratories, (FRG), September 1991.

[8] CCITT COM XII-D.113, Comments on test signals and signal processing for non LTI (linear time-invariant) measuring objects, *Brüel & Kjaer*, September 1991.

[9] HEYSER (R. C.): Acoustical measurements by Time Delay Spectrometry, *JAES*, Vol. 15, 1967.

[10] STEENEKEN (H. J. M.) and HOUTGAST (T.): A physical method for measuring speech transmission quality, *JASA*, Vol. 67, p. 318, 1980.

[11] Sound system equipment: the objective rating of speech intelligibility in auditoria by the RASTI method, First edition, *International Electronical Committee*, Geneva, IEC-268-16 (1988).

Printed in Switzerland
Geneva, 1994