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TRANSMISSION PERFORMANCE OF GROUP AUDIO TERMINALS (GATs)

ITU-T Recommendation P.30

(Extract from the *Blue Book*)

NOTES

1 ITU-T Recommendation P.30 was published in Volume V of the *Blue Book*. This file is an extract from the *Blue Book*. While the presentation and layout of the text might be slightly different from the *Blue Book* version, the contents of the file are identical to the *Blue Book* version and copyright conditions remain unchanged (see below).

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

Recommendation P.30

TRANSMISSION PERFORMANCE OF GROUP AUDIO TERMINALS (GATs)

(Melbourne, 1988)

1 Introduction

Group Audio Terminals (GATs) are terminals which have been specifically designed to be used by several users.

GATs cover a wide range of products ranging from the hands-free telephone when it is used by several users, to the sophisticated teleconference studio.

The CCITT recommends that GATs satisfy the specifications[1] in this Recommendation.

GATs must also comply with Recommendation P.34 as far as loudness is concerned, when they are connected to the telephone network. If they use voice-activated circuits, Recommendation P.34 may also be applied. Such terminals are sensitive to the acoustics of the location where they are utilized and they may resort to sophisticated acoustical echo processing devices.

The first generation of GATs will operate mainly on 4-wire digital networks and will make use of the wideband (WB) speech coding algorithm specified in Recommendation G.722. Such terminals urgently need specifications that can be based on the present Recommendation.

A typical GAT configuration is represented in Figure 1/P.30.

Such a terminal includes one or several microphones, one or several loudspeakers, sending and receiving amplification. Optionally, it includes a sound-managing and mixing device to the loudspeakers and from the microphones, a coder-decoder for digital networks, a voice-activated gain processing device and an echo processing device.

The location where the GAT is to be used is very important. Several measurements defined in this Recommendation have to be made at the location where the GAT is to be used. These are referred to as “in situ” measurements. They are to be made with the full complement of equipment in the conference room, but with no conferees present.

The present Recommendation is divided into three parts:

- interconnection specifications,
- transmit specifications,
- near-end specifications.

[1] The specifications in this Recommendation are subject to future enhancement and therefore should be regarded as provisional.

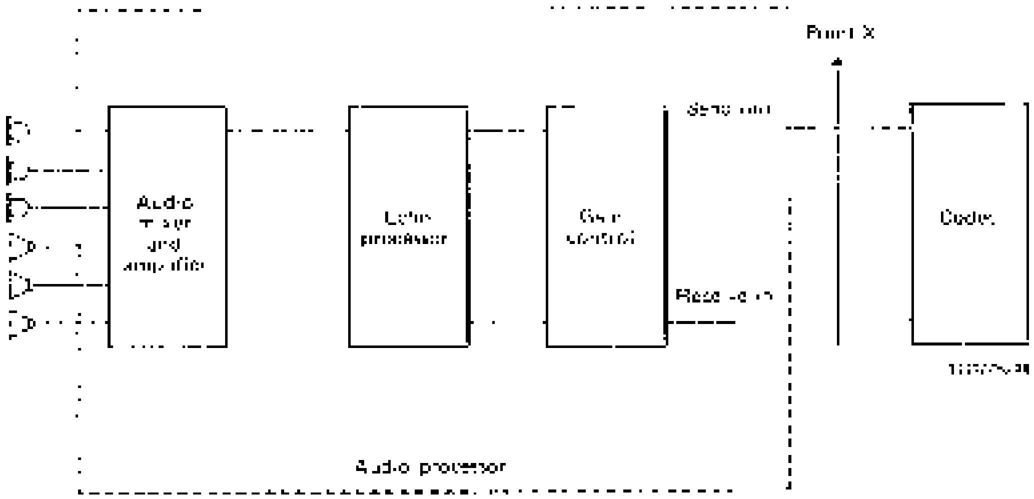

FIGURE 1/P.30

Group audio terminal configuration

Two test signals are used in this Recommendation:

- an acoustic test signal as defined in Recommendation P.50 (see Note): i.e. an artificial voice as defined in Recommendation P.50 produced by a sound source (an artificial mouth) as described in § 2 of Recommendation P.51 and,
- an electric test signal whose long-term spectrum is identical to the acoustic signal; when applied by a source with a matched internal resistive impedance, it provides a level of -22 dBV.

Both test signals are filtered in the transmission system bandwidth.

Note - The preferred acoustic signal to be used in the measurements for the audio alignment is defined in Recommendation P.50. However, other signals such as speech-shaped noise or pink noise may be used in some applications.

2 Interconnection specifications

These specifications are the basic requirements for a GAT to be connected to a network and to allow communication between several locations.

2.1 *Sending sensitivity*

2.1.1 *Wideband GATs*

For wideband applications, the transmission characteristics of the audio-channel shall be in accordance with Recommendation G.722.

2.1.1.1 *Send side alignment*

The sound source is positioned over the edge of the conference table on the centre line of each conferee's position, as defined in Recommendation P.34 (see Figure 3/P.34), and delivers a signal which complies with Recommendation P.64 [i.e. -4.7 dBPa at the mouth reference point (MRP)].

During the send side alignment the microphones of the GAT shall be positioned on the table as in real use.

The microphone gain controls must be adjusted to achieve, for each position of the source, an output line level of - 22 (± 2) dBV at point X (see Figure 1/P.30), assuming the signal recommended in Recommendation P.50 is used. This value takes account of an 18 dB peak factor of the speech signal and 6 dB for the variations between speakers and the variations due to conferees' movements.

2.1.2 *GATs connected to the public switched telephone network*

Such terminals must comply with Recommendation P.34.

2.2 *Stability test*

The GAT shall have a minimum stability margin of 3 dB when the microphone and loudspeaker paths are looped at reference point X in Figure 1/P.30 and the sound source is activated as described in § 2.1.

During the measurement, the volume control shall be in maximum position.

3 Transmit quality specifications

These specifications limit the degradations induced on the network by a GAT.

3.1 *Electro-acoustical specifications*

3.1.1 *Microphone*

The electro-acoustical characteristics of the microphones should conform to IEC Publication 581-5.

3.1.2 *Octave band measurements*

In situ measurement of the overall transmission frequency response characteristic is recommended. It is defined as the difference between the octave spectra of the electrical signal at the X interface and the acoustic excitation at the MRP. The artificial mouth is positioned as in § 2.1.1.

In order to prevent excessive fluctuations of the frequency response of the system, and since the measurements are performed on site, octave band measurements are recommended in the range 125 Hz to 4 kHz.

The sum of the absolute differences between the measured values and their average should be as low as possible. A practical target of 10 dB is achievable.

3.2 *Echo performance*

3.2.1 *Acoustic echo control*

To get satisfactory suppression of acoustic echoes it is necessary to provide the audio processor with either an echo canceller or an echo suppressor. The echo cancellation technology is recommended if highest possible speech quality performance is aimed at. However, it is recommended always to complement echo cancellation with a mild echo suppression, in order to prevent the undue transmission of room background noises when no talkers are active in the room. This condition should particularly be met in multi-conference environments.

3.2.2 *Echo return loss*

The echo return loss of the audio system shall be measured at reference point X of Figure 2/P.30, with the volume control in maximum position. When the electric test signal, as specified in § 1, is applied to the input port (receive in), the level measured at the output port (send out) shall not be higher than -62 dBV.

An acoustic echo loss of 40 dB includes a margin of 5 dB in order to provide an echo return loss of 35 dB when several GATs are used in a conference situation. This value of 35 dB should be understood as a minimum value. The long-term target value for the acoustic echo loss must be considered as being 45 dB (especially, to take into account the case where a handset is connected to a hands-free terminal). This value is known to prevent any subjective degradations due to delayed acoustic echo [1, 2]. The level measured at reference point X will then be -72 dBV.

Note - The echo canceller shall permit double-talk with negligible speech quality degradation (under study with Question 2/XII).

3.3 *Electrical noise*

The electrical noise emitted by the GAT at the reference point X should be less than -55 dBm, within the transmission bandwidth. No component outside the band should exceed 20 dB above the noise level in the band.

The measurement must be done with no conferees in the room and without incoming signals on the receiving side of the equipment in order not to activate the microphone circuits.

The noise emitted by the GAT at the reference point X when the microphones are active should be no more than -50 dBm. It must be measured by forcing the system into the emission mode as if one speaker were active in the room.

3.4 *Reverberated field picked up by the microphone*

For this measurement, the sound source is positioned in order that the distances between the sound source and all the microphones greater than three times the distance between the microphone and the position defined for the send side alignment. It is also recommended that the source be, at least, one meter from the walls. Then the signal measured at point X shall be not more than -29 dBV (this accounts for a direct-field over reverberated-field ratio of 6 dB [3]). It must be measured by forcing the system into the emission mode as if one speaker were active in the room. The test must be performed for each microphone in the room.

Basic requirements for the choice of the conference room, for its acoustical treatment and for the positioning of microphones and loudspeakers can be found in Supplement No. 16.

4 Near-end quality specifications

This part of the Recommendation tests the minimum specifications intended for the local users.

4.1 *Electro-acoustical specifications*

4.1.1 *Loudspeakers*

The electro-acoustical characteristics of the loudspeakers should conform to IEC Publication 581-7.

4.1.2 *Octave band measurements*

In-situ measurement of the overall reception frequency response characteristics is recommended. It is defined as the difference between the octave spectra of the acoustic signal delivered by the loudspeaker(s) at the listening positions and the input electric signal at the X interface.

The sum of the absolute differences between the measured values and their average should be as low as possible. A practical value of 12 dB is achievable.

4.2 *Receiving sensitivity*

4.2.1 *Volume control*

The audio conference terminal shall be provided with a volume control. The gain at maximum position should conform to § 4.2.2. The volume control should ideally be linked to the echo control mechanism.

4.2.2 *Receiving side alignment*

4.2.2.1 *Wideband GATs*

The electrical test signal is connected to the input port of the system. The receiving gain shall be adjusted in order to reach a sound pressure level of at least 65 dB and 20 dB above the acoustical noise level at the MRP. The alignment procedure should be performed with the volume control in the maximum position.

4.2.2.2 *GATs connected to the analogue public switched telephone network*

Such terminals must comply with Recommendation P.34.

References

[1] CCITT - Contribution COM XII-No. 170, Study Period 1985-1988

[2] CCITT - Contribution COM XII-No. 171, Study Period 1985-1988

[3] CCITT - Contribution COM XII-No. 172, Study Period 1985-1988