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SERIES P: TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

Subscribers' lines and sets

Speech processing devices for acoustic enhancement

ITU-T Recommendation P.330

ITU-T P-SERIES RECOMMENDATIONS

TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

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ITU-T Recommendation P.330

Speech processing devices for acoustic enhancement

Summary

This Recommendation applies to the generic transmission characteristics, performance and testing principles of Speech Processing Devices for Acoustic enhancement (SPDA) intended for use in terminals, whatever the applications.

A Speech Processing Device for Acoustic enhancement is defined as any signal processing function integrated in terminals that performs voice enhancement. Voice enhancement functions include the control of acoustic echo and noise reduction.

The purpose of this Recommendation is to define a framework for specifying performance constraints for terminals which include SPDA, and when appropriate, to define tests that may be performed on such terminals to verify that these constraints are met. This Recommendation covers generic characteristics that are applicable to both analogue and digital terminals.

Source

ITU-T Recommendation P.330 was prepared by ITU-T Study Group 12 (2001-2004) and approved under the WTSA Resolution 1 procedure on 16 March 2003.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. 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 Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

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

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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

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ITU-T Recommendation P.330

Speech processing devices for acoustic enhancement

1 Scope

This Recommendation applies to the generic transmission characteristics, performance and testing principles of Speech Processing Devices for Acoustic enhancement (SPDA) intended for use in terminals, whatever the applications.

A Speech Processing Device for Acoustic enhancement is defined as any signal processing function integrated in terminals that performs voice enhancement. Voice enhancement functions include the control of acoustic echo and noise reduction. Dereverberation, and any advanced signal processing for multi-channel pick-up and restitution are for further study.

The purpose of this Recommendation is to define a framework for specifying performance constraints for terminals which include SPDA, and when appropriate, to define tests that may be performed on such terminals to verify that these constraints are met. This Recommendation covers generic characteristics that are applicable to both analogue and digital terminals. Requirements that are applicable strictly to hands-free terminals can be found in ITU-T Rec. P.340 [13].

Test methods appropriate for parameters defined in this Recommendation may be found in ITU-T Rec. P.502 [15].

For the use of HATS for testing, ITU-T Rec. P.581 [16] applies.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[1] ITU-T Recommendation G.114 (2003), *One-way transmission time*.

[2] ITU-T Recommendation G.121 (1993), *Loudness ratings (LRs) of national systems*.

[3] ITU-T Recommendation G.122 (1993), *Influence on national systems on stability talker echo in international connections*.

[4] ITU-T Recommendation G.131 (1996), *Control of talker echo*.

[5] ITU-T Recommendation G.167 (1993), *Acoustic echo controllers*.

[6] ITU-T Recommendation G.168 (2002), *Digital network echo cancellers*.

[7] CCITT Recommendation G.223 (1988), *Assumptions for the calculation of noise on hypothetical reference circuits for telephony*.

[8] ITU-T Recommendation P.10 (1998), *Vocabulary of terms on telephone transmission quality and telephone sets*.

[9] ITU-T Recommendation P.50 (1999), *Artificial voices*.

[10] ITU-T Recommendation P.51 (1996), *Artificial mouth*.

[11] ITU-T Recommendation P.78 (1996), *Subjective testing method for determination of loudness ratings in accordance with Recommendation P.76*.

[12] ITU-T Recommendation P.79 (1999), *Calculation of loudness ratings for telephone sets*.

[13] ITU-T Recommendation P.340 (2000), *Transmission characteristics and speech quality parameters of hands-free terminals*.

[14] ITU-T Recommendation P.501 (2000), *Test signals for use in telephonometry*.

[15] ITU-T Recommendation P.502 (2000), *Objective test methods for speech communication systems using complex test signals*.

[16] ITU-T Recommendation P.581 (2000), *Use of head and torso simulator (HATS) for hands-free terminal testing*.

[17] ITU-T Recommendation P.800 (1996), *Methods for subjective determination of transmission quality*.

[18] ITU-T Recommendation P.832 (2000), *Subjective performance evaluation of hands-free terminals*.

[19] ITU-T Handbook on Telephonometry, 1992.

[20] ITU-T Recommendation G.161 (2002), *Interaction aspects of signal processing network equipment*.

[21] ITU-T Recommendation G.108.2 (2003), *Transmission planning aspects of echo cancellers*.

3 Terms and definitions

The relevant definitions given in [8] apply along with the following:

3.1 SPDA: A Speech Processing Device for Acoustic enhancement (SPDA) is defined as any signal processing function integrated in terminals that performs voice enhancement.

3.2 Acoustic Echo (AE): Acoustic echo is the delayed and reflected signal resulting from the acoustic path between the earphone/loudspeaker and microphone of a hand-held or hands-free terminal.

3.3 Acoustic Echo Canceller (AEC): A device which reduces the acoustic echo level with negligible effects on the local and distant users' speech. In order to follow the variation of the acoustic echo path, the acoustic echo control is generally implemented by adaptive identification of the acoustic echo path impulse response.

3.4 loss controller: A device which reduces the acoustic echo level by inserting variable losses on the received and/or transmitted audio signals.

3.5 Non-linear Processor (NLP): A device which reduces the residual echo which is not cancelled by the Acoustic Echo Control. A NLP uses non-linear processing to suppress the echo to a level that is not perceived by the subject at the far-end of the conversation. A centre clipper is a typical device of this kind.

3.6 howling control device: A device which modifies some characteristics of the transmitted and/or received signals in order to improve the stability margin of the terminal. This function is typically implemented by a harmonic processor. To prevent network disturbances, such devices should be avoided in terminals likely to be used on connections, including network electric echo cancellers conforming to ITU-T Rec. G.168 which are not able to work with time-variant echo paths (e.g., frequency shift).

3.7 background noise: The background noise is defined as the signal added to the desired near-end speech signal. The background noise is largely due to the acoustical signal the microphone is detecting other than the near-end speech signal.

3.8 comfort noise: Insertion of a pseudo-random noise during silent periods (no active speech signal) by a SPDA.

3.9 Noise Reduction (NR): A device which reduces the annoying and fatiguing effects of the background noise. In other words, a NR function reduces the level of the background noise so as to improve the overall perceived quality of the transmitted signal.

3.10 noise estimator: A device that computes an estimation of the characteristic of the annoying background noise. For classical systems implemented in the frequency domain, the spectral density of the noise is computed. For systems working in the time domain, the estimated value is the autocorrelation of the noise. Classically, noise estimator operates during non-speech periods (controlled by the VAD device) but other approaches are possible.

3.11 noise filtering: Processing which consists in applying to the input (i.e., the noisy signal) the filter computed by a noise reduction system. This processing is included in the Noise Reduction (NR) device. The application of the filter can be made in the time domain at sample-by-sample rate (convolution) or at block rate in the frequency domain (short-term spectral attenuation).

3.12 Lombard effect: When placed in a high-level background noise, subjects tend to speak louder. This behaviour is called "Lombard effect".

3.13 Voice Activity Detector (VAD): A device which distinguishes between silent periods (no active speech signal), single-talk periods (near-end speech periods or far-end speech periods) and double-talk periods (near-end and far-end speech signals active at the same time).

4 Acoustic echo processing

The components in a terminal used to control acoustic echo include the analogue components (microphone and loudspeaker), the acoustic echo canceller and non-linear devices.

4.1 Analogue components

The location and type of the microphone(s) and loudspeaker(s) determine the acoustic echo level (acoustic coupling) before any signal processing is employed. Without any digital control, the echo return loss (ERL) for a hands-free system for example, measures between 20 dB and –20 dB. In other words, the level of the acoustic echo may be 20 dB higher than the original received signal without any digital echo control device. The ERL may be altered by changing the location and type of microphone (such as using a directional microphone) or loudspeaker(s).

4.2 Functional units of an acoustic echo controller

The functional units of an acoustic echo controller are devices or parts of devices implemented in the processing unit, which contribute to the general function of acoustic echo control. There is no restriction on how to implement them.

The functional units can be combined for better performance. They can use all the available signals in the terminal (for example, the individual signals coming from several microphones arranged in an acoustic array). Moreover, they can be mixed with other functions (for example sub-band speech coding) for efficient implementation, provided that they do not modify the proper characteristics of these functions when they are operating.

A functional block-diagram of a typical processing unit is shown in Figure 1.

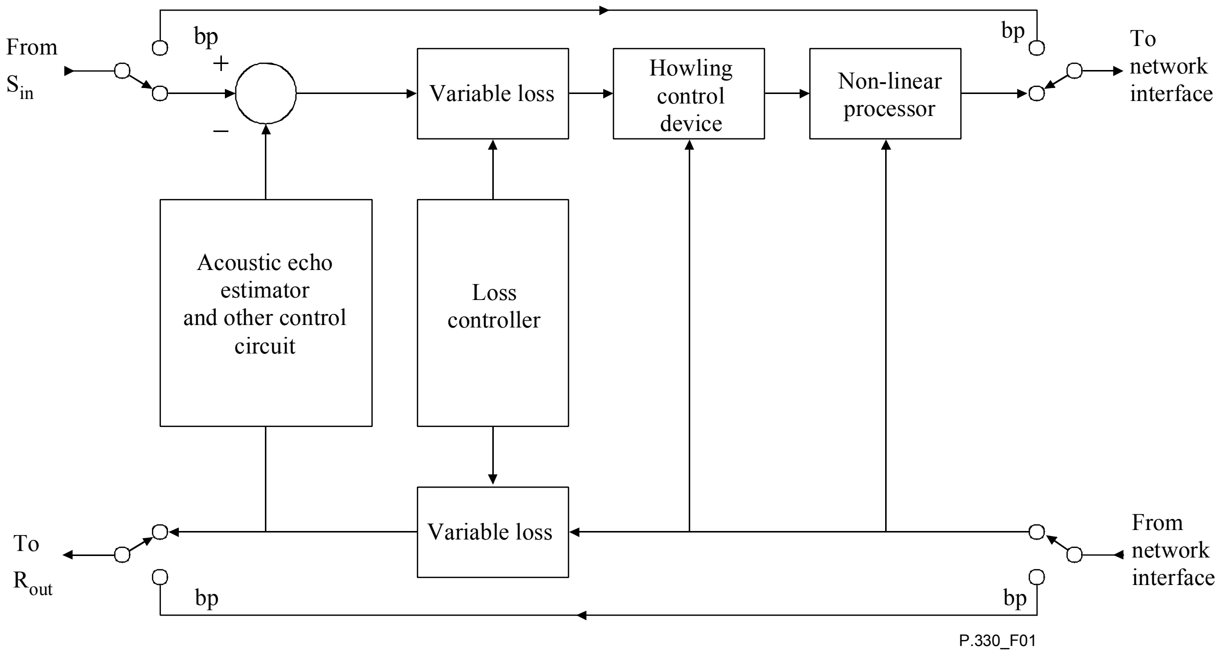

Figure 1/P.330 – Functional block-diagram of a typical processing unit (AEC part) (bp denotes bypass signal paths for testing purposes)

4.3 Interaction between terminal echo control and signal processing network equipment

Acoustic echo control is more easily and effectively done in terminals. However, network equipment may also include acoustic echo control processing. This leads to tandeming issues of acoustic echo control functionalities. The added network component shall prevent any degradation of the overall perceived quality.

Electrical echo control is achieved by network equipment as described in ITU-T Rec. G.168. If not properly controlled by the network, a device may be used in the terminal to partially control the electrical echo. Interactions between both processors are considered in ITU-T Rec. G.161. Transmission planning aspects of echo cancellers are addressed in ITU-T Rec. G.108.2.

4.4 Delay

The values specified below correspond to the extra delay which can result from the AEC processing. In any case, compliance with transmission planning objectives must be achieved.

General information about transmission delays can be found in ITU-T Rec. G.114; ITU-T Rec. G.131 provides rules for echo control in the network.

4.4.1 Processing delay

The echo cancellation processing needs some time. This time creates delay in the terminal, called "processing delay".

4.4.2 Round trip echo path delay (EPDn) – network interface

Echo audibility is dependent upon the round trip delay in the echo path. This requires calculation of the impulse response (echo path). Stationary broadband noise should be used for this measurement. The terminal can be placed in nearly any reverberant or non-reverberant environment, as the first acoustic echo will be due to direct coupling. Acoustic noise should meet the requirements defined in 5.4/P.340. AEC is first trained using a signal at R_{in}.

4.5 Acoustic echo control specifications

Performance requirements can also be found in clause 8/P.340 and test methods in ITU-T Rec. P.502.

4.5.1 Acoustic echo path

The use of real rooms, or enclosures with appropriate acoustic characteristics, is recommended. Echo paths simulated by electronic devices like digital reverberators with non-time-varying reflection patterns can be used as well if the terminal has internal access on the user side. In this latter case, the electronic simulator adjustments should comply with the values recommended for real rooms or enclosures; moreover, the shape of the simulated impulse response envelope should be similar to the real echo path impulse response.

– For teleconference systems, the reverberation time averaged over the transmission bandwidth shall be typically 400 ms; the reverberation time in the lowest octave shall be no more than twice this average value; the reverberation time in the highest octave shall be not less than half this value. The volume of a typical test room shall be of the order of 90 m^3.

– For hands-free terminals and videophones, the reverberation time averaged over the transmission bandwidth shall be typically 500 ms; the reverberation time in the lowest octave shall be no more than twice this average value; the reverberation time in the highest octave shall be not less than half this value. The volume of a typical test room shall be of the order of 50 m^3.

– For mobile radio terminals, an enclosure simulating the interior of a car can be used; a real car can be used as well. A typical average "reverberation time" is 60 ms. The volume of the enclosure shall be of the order of 2.5 m^3.

NOTE – It is recommended to avoid extremely long rooms (Length >> Width, Height) and rooms with extremely low ceilings (Height << Length, Width), and preferably also rooms with all the side dimensions nearly identical.

Large, flat, parallel room-limiting surfaces, and surface areas that provide broadband sound reflection, particularly wall surfaces at an average room height (roughly 0.8 m to 1.8 m above the floor) should be avoided, since they can cause flutter echoes and flutter-echo-like disturbances (echoing, roughness), if the test setup is in an unfavourable position.

Measuring the local frequency-dependent distribution of sound pressure levels within a selected room in the steady state can help to determine the optimum position of the test setup.

As a general suggestion, the minimum distance between the test setup and room limiting surfaces should be 1 m, regardless of the acoustic properties of these surfaces. This can prevent disturbances due to initial reflections and a rise in sound pressure level that can occur locally at low frequencies. The same recommendation applies to geometrically large furniture surfaces that reflect sound.

4.5.2 Parameters and recommended limits

4.5.2.1 Weighted terminal coupling loss – single-talk (TCLwst)

The weighted loss between the R_{in} and S_{out} network interfaces when the AEC is in normal operation, and when there is no signal coming from the local user[1].

Before each test the terminal is switched on.

[1] The weighting is made according to the rule specified in ITU-T Rec. G.122 (computation of talker echo loudness rating). Care must be taken to avoid possible masking of singing effects by the weighting (under study).

The recommended values for each type of hands-free terminal can be found in the relevant ITU-T Recommendations (e.g., ITU-T Rec. P.341 for wideband hands-free terminal and ITU-T Rec. P.342 for digital hands-free terminal).

4.5.2.2 Weighted terminal coupling loss – double-talk (TCLwdt)

The weighted loss between the R_{in} and S_{out} network interfaces when the AEC is in normal operation, and where the local user and the far-end user are active simultaneously[1].

The recommended values for each type of hands-free terminal can be found in clause 8/P.340.

4.5.2.3 Received speech attenuation during double-talk (Ardt)

The received signal attenuation (at the R_{out} point) which is inserted by the AEC during double-talk events.

The frequency response on the receive side during double-talk should ideally be the same as during single-talk conditions. In practice, however, it may not be possible to implement echo cancellation which provides sufficient echo loss during double-talk, without modifying the frequency response.

4.5.2.4 Sent speech attenuation during double-talk (Asdt)

The sent signal attenuation (at the S_{out} point) which is inserted by the AEC during double-talk events.

The frequency response on the send side during double-talk should ideally be the same as during single-talk conditions. In practice, however, it may not be possible to implement echo cancellation which provides sufficient echo loss during double-talk, without modifying the frequency response.

4.5.2.5 Received speech distortion during double-talk (Drdt)

The total non-linear signal distortion at the R_{out} point which can be produced by the AEC during double-talk events.

For all the applications, the supplementary distortion at R_{out} in comparison with single-talk conditions should be low.

4.5.2.6 Sent speech distortion during double-talk (Dsdt)

The total non-linear signal distortion at the S_{out} point which can be produced by the AEC during double-talk events.

For all the applications, the supplementary distortion at S_{out} in comparison with single-talk conditions should be low.

4.5.2.7 Build-up time – single-talk (TRst)

The time interval between the onset of the received signal (similarly the transmitted signal) and the instant when the attenuation on the receive path (similarly on the send path) reaches [3] dB. For this purpose, the other side is quiet.

4.5.2.7.1 Receive side (TRst-r)

For all the applications, TRst-r shall be no more than [20 ms].

4.5.2.7.2 Send side (TRst-s)

For all the applications, TRst-s shall be no more than [20 ms].

4.5.2.8 Build-up time – double-talk (TRdt)

The time interval between the onset of the received signal (similarly the sent signal) and the instant when the attenuation on the receive path (similarly on the send path) reaches the value Ardt (similarly Asdt). For this purpose, the signal in the opposite direction of transmission is held at a specified level.

4.5.2.8.1 Receive side (TRdt-r)

TRdt-r should be less than [20 ms], if the attenuation is more than 6 dB.

4.5.2.8.2 Send side (TRdt-s)

TRdt-s should be less than [20 ms], if the attenuation is more than 6 dB.

4.5.2.9 Convergence time (Tc)

Convergence Time is the time interval between the instant when a specified test signal is applied to the R_{in} port of the terminal (after all the functions of the AEC have been reset and then enabled), and the instant when the returned echo signal at the S_{out} port is attenuated by at least a predefined amount. The local user is not active.

4.5.2.10 Hang-over time after double-talk (THdt)

The time elapsed between the end of a double-talk event and the instant when the attenuation of the echo recovers a specified value (a signal is received continuously from the distant user).

For all the applications, the attenuation of the signal at S_{out} should be at least [20 dB] after THdt = [1] second.

4.5.2.11 Terminal coupling loss temporally weighted – single-talk (TCLtst)

The echo return loss from R_{in} to S_{out} is measured according to the procedure defined for ERLtst in ITU-T Rec. P.502.

4.5.2.12 Terminal coupling loss temporally weighted echo return loss – double-talk (ERLtdt)

The echo return loss from R_{in} to S_{out} is measured according to the procedure described for ERLtdt in ITU-T Rec. P.502.

5 Noise reduction

The main purpose of a noise reduction (NR) system in a device is to reduce the annoying and fatiguing effects of the transmitted background noise. The techniques used to reduce background noise may be classified as analogue only, digital only, and combined analogue and digital techniques.

5.1 Analogue components

The analogue components of a NR system include the microphone and any analogue circuitry connecting the microphone to the CODEC (analogue-to-digital converter). There are several techniques used in reducing background noise that only rely on analogue components:

a) The proximity of the microphone relative to the talker's mouth is a major factor in determining the SNR. Moving the microphone close to the talker's mouth produces an obvious but significant SNR enhancement (SNRE). Additional microphones may be used to improve the SNR.

b) The analogue signal path is typically designed to have a high-pass filter response. When noise has strong low-frequency components (example: automobile noise), this filter technique will enhance the SNR (as measured over the full band). The side effect is a noticeable loss of timbre in speech quality (especially in male voices).

c) Microphones may be designed to provide passive directional gain. The most common type used in automobiles is a first-order differential microphone. This microphone can be designed with a single transducer using two ports. For a diffuse noise field and the correct microphone orientation, a hypercardioid first-order differential microphone array will enhance the SNR by 6 dB compared to an omni-directional microphone. Higher order differential microphones are possible. In addition, microphone arrays using passive only techniques are possible but unlikely to be widely used because these arrays need to be very large to have an impact on the low frequencies components of speech. They may contain as many as 16 elements that can provide a directional gain of approximately 20 dB at some frequencies.

5.2 Functional units of a noise reduction system

The functional units of noise reduction system are devices or parts of devices implemented in the processing unit, which contribute to the general function of noise reduction. There is no restriction on how to implement them.

A functional block-diagram of a typical processing unit is shown in Figure 2.

Two common types of digital noise reduction techniques may be implemented in a digital signal processor (DSP) or other type of microprocessor within a terminal using a single microphone. The following techniques are commonly used:

a) Full-band noise suppression: During the pauses in speech, the noise is reduced significantly as long as its energy is below a threshold level. During active speech, the attenuation is removed allowing both speech and noise to pass. This produces an undesirable noise pumping effect if the attenuation is set too high.

b) Sub-band noise suppression: The transmitted signal is broken into sub-bands using an Fast Fourier Transform (FFT) algorithm. Only the frequency bands with stationary noise are attenuated while the bands with speech signal are unaltered. The well-known method of spectral subtraction is one such technique. In practice, noise suppression levels range from 6 to 15 dB. The drawback of these techniques is the existence of a compromise between the level of noise reduction and the distortion of the original speech signal. Hence, it is difficult to find a tuning that works in all conditions of noise (SNR and type of noise). Under low SNR conditions, however, the speech signal is degraded somewhat if high levels of noise suppression are used.

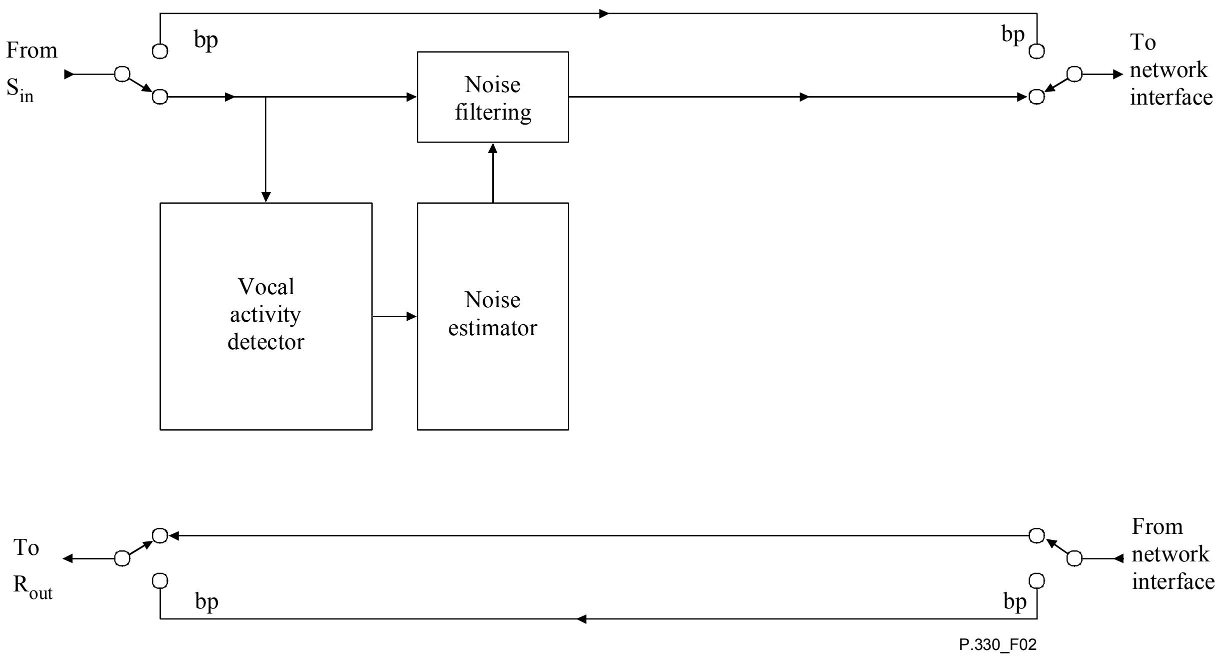

**Figure 2/P.330 – Functional block-diagram of a typical processing unit (NR part)
(bp denotes bypass signal paths for testing purposes)**

5.3 Interaction between terminal noise reduction and signal processing network equipment

Network equipment may also include noise reduction processing. This leads to tandeming issues of noise reduction functionalities. The added network component shall prevent any degradation of the overall perceived quality.

5.4 Noise reduction processing delay

The noise reduction processing delay is highly dependent on the technique used by the noise filtering. In any case, compliance with transmission planning objectives must be achieved.

General information about transmission delays can be found in ITU-T Rec. G.114.

5.5 Noise reduction system specifications

5.5.1 Noise environment

Acoustic characteristics of the test environment are described in 4.5.1. The impact of the environment should be taken into account in the case of distant sound pick-up: in this case, the reverberation of background noise has to be considered as an additional degradation.

Broadcasting of background noise test signals is described in 7.10/P.340.

Background noise test signals should include real signals such as babble noise, office room noise, street noise, car noise (engine, driving conditions at different speeds) and other simulated background noise signals (depending on the uses of the equipment).

Levels of background noise test signals should be varied so as to obtain SNRs in the range [–3 dB, 30 dB].

NOTE – Some of the corresponding test signals are in ITU-T Rec. P.501 and the test methods described in ITU-T Rec. P.502. Additional test signals are under study.

5.5.2 Parameters and recommended limits

All specified parameters should be measured for different SNR values in the range [–3 dB, 30 dB].

For parameters which consist in measuring a signal level attenuation or a delay, measurement procedures (methods and stimulus) can be found in ITU-T Recs P.501, P.502 and P.340, which apply with the following restrictions:

- the speech signal level must be at least 10 dB higher than the noise level,
- the noise must be stationary.

For all other cases (non-stationary noise, low SNR values, distortion measurement), test methods are under study.

All parameters defined below correspond to single-talk conditions. Due to possible interactions between the AEC and the NR processing integrated in the terminal, parameters under double-talk conditions must be considered also (under study).

5.5.2.1 Sent speech attenuation in quiet conditions (Asqc)

The sent signal attenuation (at the S_{out} point) which is inserted by the NR in quiet conditions.

5.5.2.2 Sent speech distortion in quiet conditions (Dsqc)

The total non-linear signal distortion at the S_{out} point which can be produced by the NR in quiet conditions.

For all the applications, the supplementary distortion at S_{out} in comparison with S_{in} should be as low as possible. Ideally, no additional distortion should be introduced by the NR.

5.5.2.3 Sent speech attenuation during noisy conditions (Asnc)

The sent signal attenuation (at the S_{out} point) which is inserted by the NR during noisy conditions.

Ideally, the frequency response on the send side should not change when the NR is activated.

5.5.2.4 Sent speech distortion during noisy conditions (Dsnc)

The total non-linear signal distortion at the S_{out} point which can be produced by the NR during noisy events.

For all the applications, the supplementary distortion at S_{out} in comparison with S_{in} should be as low as possible. Ideally, no additional distortion should be introduced by the NR

5.5.2.5 Adaptation time (TA)

Adaptation time is the time interval between the instant when a specified noise test signal is applied to the S_{in} port of the terminal (after all the functions of the NR have been reset and then enabled), and the instant when the returned noise test signal at the S_{out} port is stable within ±1dB compared with the long term reduced noise level (see Figure 3). The local and distant user are not active.

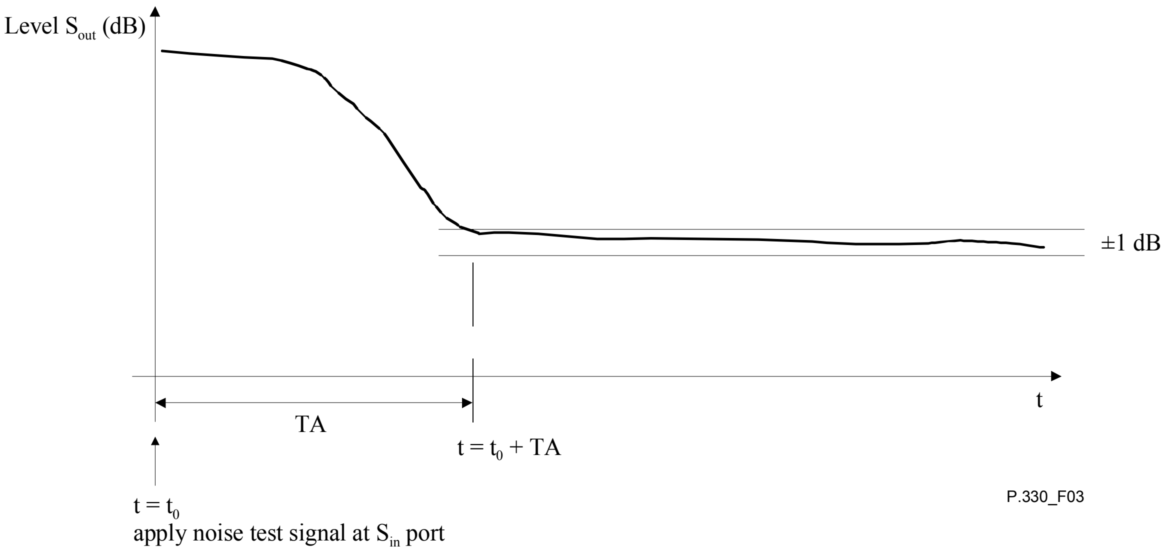

Figure 3/P.330 – Definition of adaptation time (TA)

5.5.2.6 Adaptation time after speech event (TAse)

The time elapsed between the end of a speech event and the instant when the attenuation of the noise recovers a specified value.

For all the applications, with high levels of background noise [–3 dB < SNR < 15 dB], the attenuation of the noise signal at S_{out} should be at least [6 dB] after TAse = [100] millisecond.

5.5.2.7 Terminal noise attenuation – no speech (TNAtns)

The terminal noise attenuation from S_{in} to S_{out} which is inserted by the NR on the background noise signal when no speech signal is present.

5.5.2.8 Noise distortion – no speech (Dnns)

The total non-linear signal distortion at the S_{out} point which can be produced by the NR on the background noise signal when no speech signal is present.

For all the applications, the distortion at S_{out} in comparison with S_{in} should be negligible.

5.5.2.9 Terminal noise attenuation – in the presence of speech (TNAtps)

The terminal noise attenuation from S_{in} to S_{out} which is inserted by the NR on the background noise signal in the presence of speech (measurement of the Signal to Noise Ratio enhancement).

5.5.2.10 Comfort noise level and spectrum matching – no speech (CNLMns and CNSMns)

The comfort noise at the S_{out} point which can be inserted by the NR when no speech is present should match in level and spectrum the background noise present at S_{in} point.

5.5.2.11 Comfort noise level and spectrum matching – in the presence of speech (CNLMps and CNSMps)

The comfort noise at the S_{out} point which can be inserted by the NR in the presence of speech should match in level and spectrum the background noise present at S_{in} point.

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