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SERIES P: TERMINALS AND SUBJECTIVE AND  
OBJECTIVE ASSESSMENT METHODS

Methods for objective and subjective assessment of  
speech quality

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**Application guide for Recommendation ITU-T  
P.863**

Recommendation ITU-T P.863.1

ITU-T



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**TERMINALS AND SUBJECTIVE AND OBJECTIVE ASSESSMENT METHODS**

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# Recommendation ITU-T P.863.1

## Application guide for Recommendation ITU-T P.863

### Summary

Recommendation ITU-T P.863.1 provides important remarks that should be taken into account in the objective quality evaluation of speech conforming to Recommendation ITU-T P.863. Users of ITU-T P.863 should understand and follow the guidance given in this Recommendation.

This Recommendation is a supplementary guide for users of Recommendation ITU-T P.863, which describes a means of estimating listening speech quality by using reference and degraded speech samples. The scope of Recommendation ITU-T P.863 is clearly defined in itself. This Recommendation does not extend or narrow that scope; rather, it provides necessary and important information for obtaining stable, reliable and meaningful objective measurement results in practice.

### History

Edition	Recommendation	Approval	Study Group	Unique ID*
1.0	ITU-T P.863.1	2013-05-14	12	<a href="http://handle.itu.int/11.1002/1000/11935">11.1002/1000/11935</a>
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\* To access the Recommendation, type the URL <http://handle.itu.int/> in the address field of your web browser, followed by the Recommendation's unique ID. For example, <http://handle.itu.int/11.1002/1000/11830-en>.

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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.

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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, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at <http://www.itu.int/ITU-T/ipr/>.

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# Recommendation ITU-T P.863.1

## Application guide for Recommendation ITU-T P.863

### 1 Scope

This Recommendation is a supplementary guide for users of [ITU-T P.863], which describes a means of estimating listening speech quality by using reference and degraded speech samples. It assumes that an objective quality algorithm strictly conforms to [ITU-T P.863]. This can be confirmed by the conformance test provided in [ITU-T P. 863].

This Recommendation does not extend or narrow the scope of [ITU-T P.863]; rather, it provides necessary and important information for obtaining stable, reliable and meaningful objective measurement results in practice. It also provides example results for common situations and explanations for how certain degradations impact the score.

Additional characterization results will be added to this Recommendation, in the form of appendices, as this information becomes available.

### 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.

- [ITU-T P.10] Recommendation ITU-T P.10/G.100 (2006), *Vocabulary for performance and quality of service*; plus its amendments.
- [ITU-T P.863] Recommendation ITU-T P.863 (2014), *Perceptual Objective Listening Quality Assessment*.

### 3 Definitions

#### 3.1 Terms defined elsewhere

This Recommendation uses the terms and definitions defined in [ITU-T P.10].

#### 3.2 Terms defined in this Recommendation

None.

### 4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

- ACR Absolute Category Rating  
AGC Automatic Gain Control  
ASL Active Speech Level  
IRS Intermediate Reference System  
MIRS Modified Intermediate Reference System

MOS	Mean Opinion Score
MOS-LQO	Mean Opinion Score – Listening Quality Objective
NB	Narrowband
PCM	Pulse Code Modulation
PESQ	Perceptual Evaluation of Speech Quality
PSQM	Perceptual Speech Quality Measure
PSTN	Public Switched Telephone Network
SPL	Sound Pressure Level
SWB	Super-Wideband
VoIP	Voice over IP

## 5 Conventions

None.

## 6 Introduction to ITU-T P.863

[ITU-T P.863] provides an objective speech quality measurement algorithm for measuring the voice quality of narrowband (NB), wideband and super-wideband (SWB) networks.

### 6.1 History of objective speech quality

Speech quality has no physical definition. Inherently, people just have an opinion about when something sounds good or bad. However, stakeholders need to be able to quantify the quality delivered by a telephone system in order to maximize investment and ensure adequate service is provided to customers. For many years, the only effective manner by which to determine the quality of a telephone network was to perform a subjective test. Subject testing, explained more in the next clause, involves asking a panel of users what they think of a recording or connection. The panel typically vote on a five point scale, and the average of the votes is deemed to be the quality of the connection. This number is called the mean opinion score (MOS).

The running of subjective tests is time consuming and costly. During the late 1980s, compression technologies were introduced in digital networks to increase capacity while reducing costs. Before their introduction, it was generally possible to determine the performance of a network using simple tone-based measurements. With the introduction of new speech processing technologies, it was found that results from tone-based techniques could contradict users' experiences. A new measurement methodology was required. The increased availability of general purpose computing allowed the development of computer programs capable of modelling the results of subjective tests.

In 1996, Recommendation ITU-T P.861 (perceptual speech quality measure (PSQM)) [b-ITU-T P.861] was published. The core concept introduced in this first generation algorithm was that human hearing could be modelled to extract a representation of audible differences between a reference and a degraded pair of signals, and that these differences could be mapped to the scores of subjective tests.

Shortly after [b-ITU-T P.861] was published, work was started to address practical limitations of the first generation model in terms of its applicability for testing networks. This work led to the publishing of a significantly improved model called perceptual evaluation of speech quality (PESQ), which was published as [b-ITU-T P.862] in 2001, together with the withdrawal of [b-ITU-T P.861]. Work continued on [b-ITU-T P.862] for a number of years, for example, with the introduction of a wideband



extension in 2005. However, as more complex signal processing was added to the telephone network, it became clear that a new model was required.

In 2006 ITU-T initiated a new activity for the development of the third generation model. The intention was to provide a backward-compatible model that could also assess new speech signal processing technologies as well as the anticipated move to super-wideband networks. The result of this work was published as [ITU-T P.863] at the beginning of 2011. It should be noted that the introduction of [ITU-T P.863] does not deprecate [b-ITU-T P.862].

## 6.2 Basics of a subjective test

It is important to consider subjective testing when discussing objective speech quality assessment. Essentially, subjective testing underlies all three generations of objective models described above. The intention for this type of objective model is to predict the result of a listening quality subjective test for a given reference and degraded file pair.

ITU provides various Recommendations for performing subjective tests; the most relevant for [ITU-T P.863] are [b-ITU-T P.800] and [b-ITU-T P.830].

A subjective test aims to find the average user opinion of a system's speech quality by asking a panel of users a directed question and providing a limited-response choice. For example, to determine listening quality, users are asked to rate 'the quality of the speech' by selecting from the scale shown in Table 1.

**Table 1 – MOS**

<b>Opinion</b>	<b>Score</b>
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

A MOS is calculated for a particular condition by averaging the votes of all subjects. This type of test is described as an absolute category rating (ACR) experiment.

The results of a subjective test are influenced by many factors and a great deal of effort is made, in the planning of the subjective test, to control these factors. Regardless of the planning effort, a score for a given condition from a subjective test is not a definitive result; run the same test with another set of subjects and a different result will be received. The expected range of results for a given condition can be expressed as a confidence interval. Some subjective tests provide results with a smaller confidence interval than others. Place the same condition in another subjective test, with a different quality bias, and it will more than likely be found that the score compared to the first test systematically moves up or down. There is a tendency to see subjective tests maintain rank ordering of conditions, but not absolute scores.

It is difficult to cope with these inconsistencies. Thus, it is not uncommon for companies to claim that a particular coding technology is superior to another technology and provide subjective test results to support these claims, while others companies are able to demonstrate a contrary conclusion.

Over the past two decades, the importance of normalizing subjective scores onto a 'common' scale, for the purpose of developing and evaluating objective measurement algorithms, has absorbed significant effort. [b-ITU-T P.1401] describes this process in detail. Later in this Recommendation, a procedure is presented for removing simple subjective-test condition biases using linear-regression;

this step represents a minimum level of normalization that should be applied before comparing subjective and objective results.

### **6.3 Benefit of using Recommendation ITU-T P.863**

The two main reasons for using an ITU recommended objective speech quality algorithm are:

1. the ability to compare results between different organizations;
2. model development represents a small part of the effort; validation over many thousands of conditions represents the bigger effort.

### **6.4 Relationship with other Recommendations**

- [b-ITU-T P.862] – predecessor to [ITU-T P.863] for objectively measuring listening speech quality of narrowband and wideband networks. This Recommendation has not been deprecated by the introduction of [ITU-T P.863].
- [b-ITU-T P.563] – a no-reference, single-ended, objective listening speech quality method for predicting the subjective quality of narrowband networks.
- [b-ITU-T G.107] (E-model) – a planning tool that combines various network performance parameters to determine a quality score on a 100 point scale. This score is referred to as the R-factor.
- [b-ITU-R BS.1387] – an objective tool for measuring perceived audio quality.
- [b-ITU-T P.800.1] – a Recommendation defining MOS terminology. Based on the definitions in this Recommendation, [ITU-T P.863] scores should be labelled as mean opinion score – listening quality objective (MOS-LQO) if presented alongside subjective or estimated scores.

### **6.5 Moving from ITU-T P.862 to ITU-T P.863**

[ITU-T P.863] narrowband mode was developed to provide backward compatibility with [b-ITU-T P.862]. This backward compatibility means that [ITU-T P.863] can be used wherever [b-ITU-T P.862] is currently being used. It also means that the predictions of [ITU-T P.863] can be directly compared to both new and old narrowband subjective tests.

Although built to be backward compatible, different scores will be seen when testing a connection with [ITU-T P.863] in place of [b-ITU-T P.862]. Different scores will be seen because there are fundamental differences between how [b-ITU-T P.862] and [ITU-T P.863] process the signals. In addition, a significantly larger database of subjective test results has been used in the development and calibration of [ITU-T P.863] as compared to [b-ITU-T P.862]. This increased data provides a more complete view of how people judge the full range of speech quality delivered by modern telephone networks. For this reason, and because [ITU-T P.863] achieves better performance across all databases compared to [b-ITU-T P.862], predictions from [ITU-T P.863] should be viewed as more accurate than [b-ITU-T P.862].

It is recommended that [b-ITU-T P.862] be run in parallel with [ITU-T P.863] until an appropriate feeling is developed for [ITU-T P.863] scores. Clause 12 provides additional information on the differences between [b-ITU-T P.862] and [ITU-T P.863].

### **6.6 Challenges with ITU-T P.863**

The biggest challenge of using [ITU-T P.863] is in selecting appropriate reference material. [ITU-T P.863] does not always produce a perfect score when a reference file is compared with itself. This is described as a reference transparency issue.

This transparency issue arises because [ITU-T P.863] assumes that a reference signal will have a balanced timbre and it judges deviations from a balanced timbre as degradation. If the timbre in the reference is not balanced, for example, because of a bass-boost or a lot of sibilance, a reference-

reference comparison with no impairments will not return the maximum score for the scale. Typically, one will see a drop of a 0.1 or 0.2 MOS; however, it can be larger.

In addition to ensuring reference transparency, further constraints have been placed on the reference's content structure to ensure that reported scores are representative of user perception. These additional constraints have been added following practical experience because [b-ITU-T P.862] has been misused over the years.

It should be noted that [ITU-T P.863] does not enforce any reference content constraints, thus it is still possible to use non-conforming reference material. This may produce interesting results in internal studies, but values should not be reported publically if the reference signal does not follow the constraints.

## **6.7 MOS misconceptions**

A number of minor misconceptions exist around objective listening speech quality assessment and what a MOS value represents. Some misconceptions are:

- The effect of a constant end-to-end delay is not assessed by subjective or objective listening quality test methods. The presence of delay variation may affect the score, particularly during periods of active speech.
- The effect of talker echo and echo delay are not assessed by subjective or objective listening test methods. A listener cannot hear talker echo.
- Objective listening test methods cannot assess mixed signals, such as in the double talk condition where there is sidetone at the listener end, or multiple concurrent speakers on a conference bridge.
- MOS is not only a measure of voice quality. MOS is the acronym of Mean Opinion Score. A MOS is the result of any subjective test where subjects are asked to vote on a scale (discrete or continuous) and the votes are averaged across subjects to determine the mean opinion of the subjects. It is equally valid to have a MOS prediction for video and audiovisual content as it is for speech content.

## **7 Operational modes**

[ITU-T P.863] provides two operational modes:

1. narrowband: predicts quality as it would be perceived through an intermediate reference system (IRS) type receive filtered monaural handset;
2. super-wideband: predicts the quality as it would be perceived through a diffuse field equalized Hi-Fi headphone for diotic (same signal in both ears) listening.

[ITU-T P.863] always applies an IRS receive filter to the reference and degraded input signals in narrowband mode. This is not the case in super-wideband mode. [ITU-T P.863], regardless of mode, predicts the perceived quality of a recording on a five point ACR scale.

### **7.1 Why two operational modes?**

Telephone networks contain a mix of audio-bandwidths, including traditional 300-3400 Hz narrowband, wideband up to 7 kHz and super-wideband from 50-14 kHz. Until now, most telephone networks have provided only narrowband connections and users' quality expectations are biased for this bandwidth-range. Adding wider bandwidth conditions to a subjective test can influence the nominal scores achieved for the narrowband conditions. The narrowband [ITU-T P.863] scale is provided to give a scale that is compatible with the majority of subjective testing performed in the previous years and objective tools such as PESQ. It ensures that [ITU-T P.863] can be compared to results from narrowband subjective testing.

## **7.2 When should the narrowband mode be used?**

Use the narrowband mode when assessing a narrowband system, analysing results from a narrowband subjective test, or when interested in how quality would be perceived in a traditional telephone network. In addition, use the narrowband mode if comparing [b-ITU-T P.862.1] results with [ITU-T P.863] results.

## **7.3 When should the super-wideband mode be used?**

The super-wideband mode should be used in all other cases (not covered by clause 7.2). Super-wideband mode is also appropriate for assessing narrowband connections. When assessing narrowband connections in super-wideband mode, a super-wideband reference file is also required as an input to [ITU-T P.863]. [ITU-T P.863] is then able to determine the narrowing of the signal's bandwidth and includes this information in its prediction of quality.

## **7.4 Why no wideband (up to 7 kHz) mode?**

The super-wideband scale should be used when assessing a wideband system; the model will account for the 7 kHz bandwidth in its quality assessment. A wideband scale was not provided because the long-term intention is that all quality scores are reported on the super-wideband scale. It is intended that [ITU-T P.863] super-wideband mode represents a single voice quality scale for all systems regardless of bandwidth limitation and that bandwidth limitation is treated as simply a form of degradation. The narrowband mode has been provided for backward compatibility for [b-ITU-T P.862.1], but backward compatibility was felt to be less important for [b-ITU-T P.862.2] because far less testing is currently performed in wideband than narrowband.

## **7.5 Are narrowband signals scored equally in narrowband and super-wideband modes?**

No. Although it is possible to apply an external IRS filter to the signals before passing to [ITU-T P.863] in super-wideband mode, the same score is not expected from the super-wideband and narrowband modes because of different optimizations used in the perceptual modelling. A narrowband signal with the maximum narrowband mode score of 4.5 will, by design, score around 3.8 in super-wideband mode.

## **7.6 Can I map a narrowband score to a super-wideband score?**

No. The super-wideband scale is not a simple extension of the narrowband or wideband scales to 4.75. It is a completely new scale on which nominal scores for narrowband and wideband codecs are different. A study performed as part of the [ITU-T P.863] characterization phase found that [ITU-T P.863] in super-wideband mode will produce relatively lower scores for narrowband and wideband codecs compared to the scores observed in narrowband and wideband subjective tests.

The maximum super-wideband scale score of 4.75 was chosen to indicate that super-wideband connections offer potentially higher quality than narrowband connections. The narrowband scale produces a maximum score of 4.5.

## **7.7 Is it recommended to mix bandwidths in subjective testing?**

Generally, it is not recommended to mix different bandwidths in a subjective test. When different bandwidth signals are mixed in a single test the bandwidth differences can dominate the resulting mean opinion scores. The [ITU-T P.863] super-wideband scale has been engineered to enable easy comparison of signals with different bandwidths which may be difficult to achieve in subjective tests.

## **8 Influence of reference speech on scores**

The quality and content of the reference speech material has an impact on the robustness of the speech quality measurement process. The [ITU-T P.863] model transforms the reference signal to an idealized form; this process is sensitive to voice timbre and background noise.

### **8.1 What characteristics should a reference signal contain?**

Different conditions exist for a super-wideband reference signal and a narrowband reference signal. The list below describes a common set of required characteristics for both super-wideband and narrowband reference test signals:

- at least three seconds of active speech;
- at least 500 ms of silence between active speech periods;
- no more than six seconds of active speech;
- total length of test sample, including silence, should be no more than 12 seconds;
- active speech level of  $-26$  dBov;
- 16-bit linear pulse code modulation (PCM) encoded;
- noise floor  $< -75$  dBov (A).

Additional characteristics for [ITU-T P.863] super-wideband reference test signals are:

- 48 k sample rate;
- filtered 50 Hz to 14 kHz.

Additional characteristics for [ITU-T P.863] narrowband reference test signals are:

- 8 k sample rate;
- filtered 100 Hz to 3.8 kHz;
- noise floor  $< -80$  dBov (A) after downsampling.

### **8.2 Are there constraints on the recording environment?**

The room where reference material is recorded must have a reverberation time below 300 ms above 200 Hz (e.g., an anechoic chamber). Recordings must be made using omni-directional microphones. The distance to the microphone must be approximately 10 cm. Background noise must be below 30 dB sound pressure level (SPL)(A). Directional microphones are allowed on the condition that the spectral balance frequency response is the same as with the omni-directional microphones.

### **8.3 Can we use artificial speech signals?**

An artificial voice signal, such as that defined in [b-ITU-T P.50], is not recommended. Speech signals generated using a text-to-speech system are better, but have had limited validation.

### **8.4 What filter specification should be used?**

A reference signal should be filtered before presenting it to the [ITU-T P.863] model. A different filter is required for the super-wideband and narrowband modes. The filter definitions are provided in Tables 2 and 3.

The super-wideband filter definition is approximately described in Table 2; the specific filter definition can be found in [b-ITU-T G.191].

**Table 2 – Super-wideband filter definition**

Frequency (Hz)	Super-wideband gain (dB)
20	–20 (max)
50	–3
60	0
13'500	0
14'000	–3
15'000	–40 (max)
24'000	–50 (max)

The approximate narrowband filter definition is given in Table 3.

**Table 3 – Narrowband filter definition**

Frequency (Hz)	Narrowband gain (dB)
20	–20 (max)
100	–3
110	0
3'750	0
3'800	–3
4'000	–40 (max)

### **8.5 Does reference material influence scores?**

Reference speech material has a direct influence on the resulting score. This is true in subjective testing as well as objective testing. A study of 400 speech recordings, from four different speakers, processed through various codec simulations, was found to produce a [ITU-T P.863] super-wideband (SWB) condition score range of 1.2 to 1.4 MOS. It is therefore important to use a variety of material when testing a condition's speech quality.

### **8.6 How much reference material should I use?**

[ITU-T P.863] recommends two samples from each of two male and two female speakers, i.e., eight sentence pairs. Some applications may only permit shorter test durations. Mixing of speakers or genders in a single test sample may limit the maximum score that can be achieved.

Assessing multiple speakers and sentence pairs is required to remove material-specific result bias. It has been found that for some low bit-rate codecs the score can differ by as much as 1.4 depending on the selected talker and sentence pair. This material dependency is only removed by ensuring that the scores, of multiple talker and sentence pairs, are averaged.

The factors in the reference material that influence the scores include, for example, talker gender, talker language and signal duration.

### **8.7 What is the impact of silence padding?**

Silence padding is the addition of silence to the front or back of the reference signal. Typically, sentence material in subjective tests has a 0.5 second silence lead in, two sentences, and then a 0.5 second silence at the end of the signal. If these lead-in and end silences are changed, a slight change in score (approximately 0.1) may be seen when comparing the reference signal with itself. This change indicates that the [ITU-T P.863] idealization processing is sensitive for this reference

signal. The slight change in score is not a significant problem, but if this effect is seen, another reference recording may be selected to increase absolute consistency in the results.

## 8.8 How do I validate a reference signal?

One of the main differences between [b-ITU-T P.862] and [ITU-T P.863] is that the [ITU-T P.863] algorithm takes into account signal level, potential acoustic reverberations and bandwidth limiting. This means that the reference signals used with [ITU-T P.863] should not be corrupted by large level variations, a presence of reverberation, or an imbalanced timbre. A simple way to check this is by comparing a candidate reference recording with itself. If the result is not equal to the maximum theoretical score of the model, then the tested recording is not suitable for use as a reference signal for [ITU-T P.863]. In narrowband mode, a reference-to-reference comparison should score 4.5. In super-wideband mode, the reference-to-reference comparison should score 4.75.

To further confirm the appropriateness of a reference signal, tests should be performed after adding small offset changes (10 ms and 15 ms) to the start of the clean reference before presenting it to [ITU-T P.863]. The predicted score should remain equal to the maximum theoretical score of the model. A full list of tests required to validate a reference recording is given in Table 4 below.

**Table 4 – List of tests required to validate a reference recording**

Test ID	Description
1	ITU-T P.56 active speech level = $-26$ dBov ( $\pm 2$ )
2	Noise floor during non-speech intervals $-75$ to $-90$ dBov for super-wideband reference samples, $< -80$ dBov for narrowband reference samples after down-sampling, S/N in active speech intervals $> 40$ dB
3	Total length, including silences $\leq 12$ s
4	Minimum duration of active speech $\geq 3$ s
5	Maximum duration of active speech $\leq 6$ s
6	Number of speech segments $\geq 2$
7	Silence at start $\geq 0.25$ s ( $\geq 0.5$ s for testing background noise conditions)
8	Silence at end $\geq 0.25$ s ( $\geq 0.5$ s for testing background noise conditions)
9 a	At least one silence interval between speech segments (sentences) $\geq 0.5$ s
9 b	For testing background noise conditions: At least one silence interval between speech segments (sentences) $\geq 1.0$ s and $\leq 2.0$ s
10	Bandpass upper frequency cut-off $\geq 14'000$ Hz
11	Super-wideband score
12	Super-wideband score with 10 ms offset
13	Super-wideband score with 15 ms offset
14	Narrowband score
15	Narrowband score with 10 ms offset
16	Narrowband score with 15 ms offset

A validation of the speech material in [b-ITU-T P.501] using these test criteria is presented in Appendix I.

## 8.9 How do I use a 10 minute reference file?

To use a reference speech sample longer than the recommended maximum 6 seconds of active speech, it is recommended that the signal is split into multiple 3 to 6 second active speech sections, and a

separate score be computed for each section. Average the scores to determine a single score for the complete reference signal.

## **9 How to inject test signals**

### **9.1 Over which interfaces can signals be injected into a network?**

It is possible to inject a test signal into a network via an acoustic, electrical or digital connection. It is equally possible to inject the test signals digitally into software emulations. Regardless of how the signal is injected into the system under test, filtering of the test signal prior to injection and the level of the signal within the system need to be carefully managed.

### **9.2 Should the test signal be filtered?**

Yes. The signal must be filtered for direct input into the system under test. For example, it could be appropriate to apply an IRS ([b-ITU-T P.48]) filter to a signal applied to a public switched telephone network (PSTN) connection, or to apply a modified IRS (MIRS) filter to a signal applied to a mobile network connection. A signal applied to a wideband network should be low-pass filtered to 7.8 kHz. A signal applied to a super-wideband connection may not require any filtering.

Mixed bandwidths might be required for some applications such as conference bridge testing. It may be appropriate to apply IRS, MIRS, wideband and super-wideband signals to the different connections into the bridge.

Speech signals applied acoustically via an artificial mouth should not be filtered.

### **9.3 What level should be used?**

The mean active speech level of the signal must be adjusted to the requirements for the system under test.

For example, an electrical signal may be presented at  $-48$  dBm at a microphone input of a voice over IP (VoIP) phone or mobile phone, or it could be at  $-20$  dBm for a PSTN line. The level adjustment must be made after any sample rate changes and filtering are applied.

The effect of different speech levels upon the system under test can be evaluated in super-wideband mode. Testing at lower speech levels can be expected to yield lower scores.

## **10 Influence of recording process on scores**

How tests are performed, and in particular how the recorded speech is stored in a file, can influence [ITU-T P.863] results. This clause discusses various aspects that should be controlled when capturing a signal in a test.

### **10.1 What influence does file speech level have on score?**

[ITU-T P.863] was designed to identify the effect of speech level upon speech quality, thus the score will be lower if the speech level is lower when using the super-wideband mode.

The score predicted by [ITU-T P.863] in the super-wideband mode will be almost unaffected in the range  $-20$  dBov to  $-36$  dBov. The score reduces by approximately 1 MOS between  $-36$  dBov and  $-46$  dBov. [ITU-T P.863] will be less consistent outside this range and is not recommended for signals below  $-56$  dBov.

The score predicted by [ITU-T P.863] in the narrowband mode will be consistent between samples if the signal level is in the range  $-26$  dBov to  $-46$  dBov.

If the mean active speech level of the signal in the degraded file is very low or very high then the score may be affected by the noise floor or amplitude clipping.



## **10.2 Should we pre-align signal levels?**

Level pre-alignment, a process not defined in [ITU-T P.863], describes how a recorded signal's active speech level is normalized to a value of  $-26$  dBov prior to submission to the [ITU-T P.863] algorithm. When applied, level pre-alignment removes the influence of listening level on score prediction.

[ITU-T P.863] assumes that a digital file level of  $-26$  dBov equates to an acoustic listening level of 79 dB SPL in narrowband mode and 73 dB SPL in super-wideband mode. The degraded file obtained from a test will rarely have a defined relationship to these sound pressure levels. It is therefore desirable to adjust the level of the recording to  $-26$  dBov so that the influence of speech level on score is eliminated.

If the objective of the test is to evaluate the performance of the system under test at different speech levels, then level pre-alignment should not be applied and the super-wideband mode should be used.

Level pre-alignment should always be applied in the [ITU-T P.863] narrowband mode, as the score will be affected and the results will not be predictable. The [ITU-T P.863] narrowband mode validation assumes that all file levels are at  $-26$  dBov. In addition, [b-ITU-T P.862] automatically applied level pre-alignment, thus level pre-alignment should be applied in order to maintain backward compatibility with [b-ITU-T P.862].

## **10.3 Should we apply a filter to the recording?**

[ITU-T P.863] in super-wideband mode applies an internal filter that is equivalent to a diffuse field Hi-Fi headset, while in narrowband mode an IRS receive filter is applied. As such, it is not necessary to further filter the degraded file before processing it through [ITU-T P.863]. It is important to manage level and spectral response in the conversion of an analogue signal collected from the network interface for [ITU-T P.863]. The circuits should not filter the signal below 14 kHz or impose any distortion or noise.

## **10.4 Does silence padding affect the ITU-T P.863 scores?**

When recording test signals on a test network, it is common to capture additional silence before and after the test signal. This occurs, for example, when testing between two unsynchronized locations and using large capture windows to compensate for the lack of synchronization. These additional silences at the start and end of the recorded signal should have no influence on the [ITU-T P.863] result.

## **10.5 Can I assess a narrowband recording in super-wideband mode?**

[ITU-T P.863] will evaluate a narrowband degraded speech file in super-wideband mode and produce a score prediction that takes into account the lower bandwidth. The maximum score will be around 3.8 on the super-wideband scale. The [ITU-T P.863] super-wideband mode was designed to enable the evaluation of narrowband, wideband and super-wideband signals on the same scale. It relies upon detecting the absence of any speech energy above 3.8 kHz to indicate a narrowband degraded file and the absence of any speech energy above 7 kHz to indicate a wideband degraded file.

## 11 Behaviour of ITU-T P.863

This clause describes how [ITU-T P.863] scores are affected by various network degrading processes. It is intended that this clause will help in the analysis of [ITU-T P.863] results by providing reference information on how specific degradations influence the final score. It should be noted that in actual network testing it is likely that multiple degrading processes will be responsible for a drop in score and the interaction between these combined degradations may not be linear.

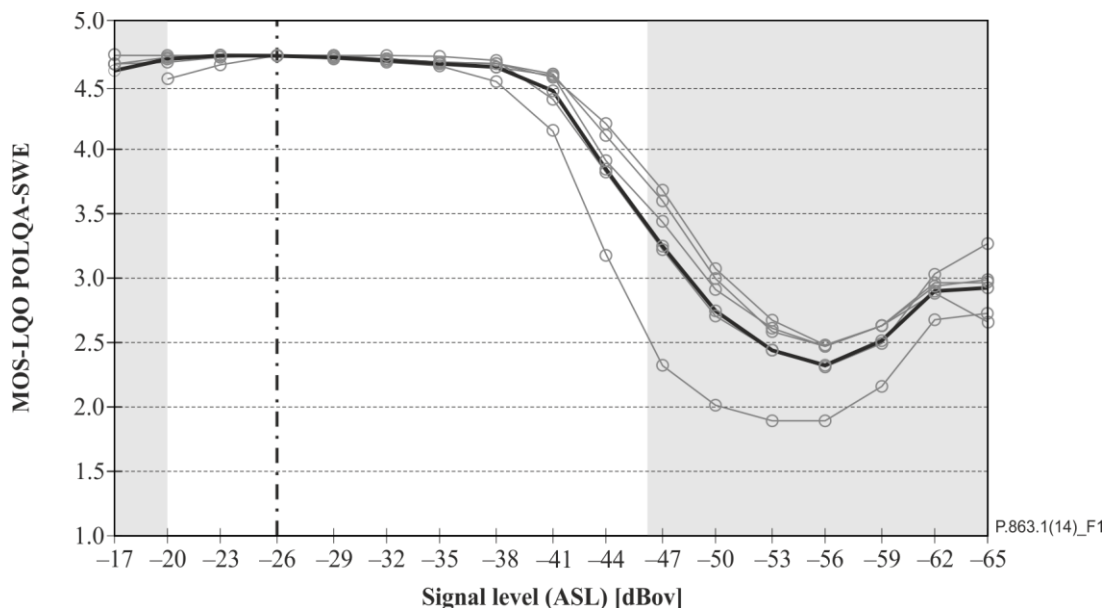
### 11.1 Global level changes

This clause assumes that level pre-alignment described in clause 10.2 is not applied.

[ITU-T P.863] considers the presentation level of the speech signals in its prediction. Within the specification, and in the evaluation phase, a level range of +5 dB to –20 dB relative to the nominal level was specified and tested.

Figure 1 illustrates the dependency on the presentation level for five different super-wideband samples. The average for all five samples is given as a solid bold line.

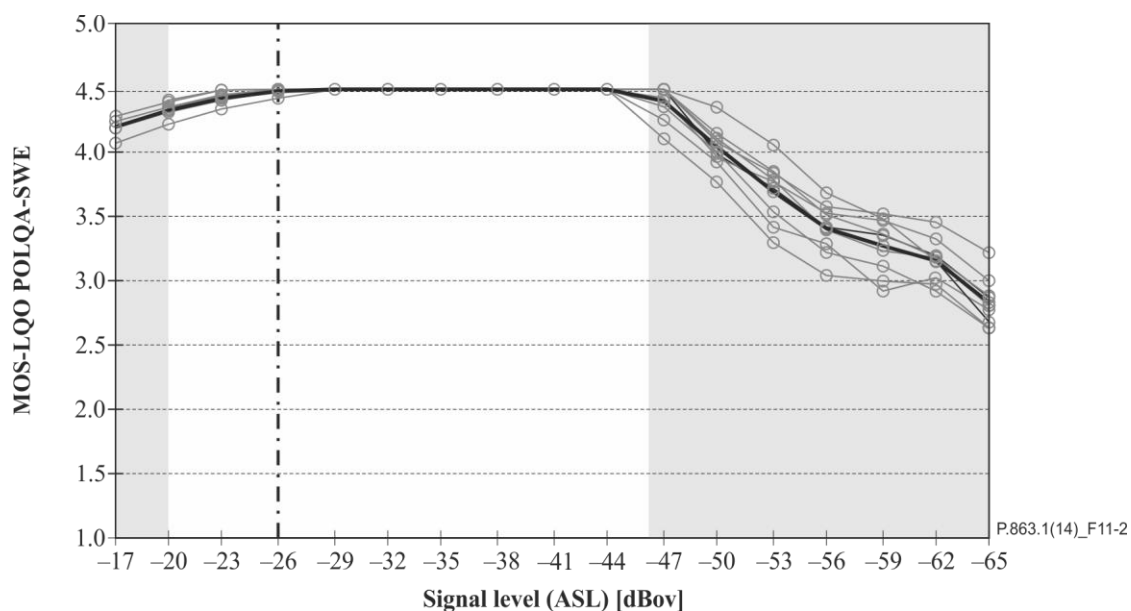
The nominal level is marked by a vertical dashed line at –26 dBov that corresponds to 73 dB SPL at each ear in a diotic presentation. The white area is the specified range; the grey shaded area is the level range outside of the specification for [ITU-T P.863].



**Figure 1 – Dependency of the MOS-LQO<sub>SWB</sub> on the presentation level**

It is recommended that the signal levels in super-wideband mode of [ITU-T P.863] neither exceed –20 dBov nor fall below –46 dBov. [ITU-T P.863] must not be used to predict signals of an active speech level (ASL) below –56 dBov.

Presentation level was not required for [ITU-T P.863] narrowband mode and all databases were normalized to –26 dBov. However, for completeness an equivalent graph is shown in Figure 2 for the influence of presentation level in narrowband mode.



**Figure 2 – Influence of the presentation level on MOS-LQO<sub>NB</sub>**

It can be seen from the graph that [ITU-T P.863] narrowband mode exhibits almost no level dependency in the range  $-26$  to  $-46$  dBov.

## 11.2 Local level changes

Discussions of local level changes (automatic gain control (AGC)) are for further study.

## 11.3 Bandwidth limitations

The score in super-wideband mode is affected by bandwidth differences between a degraded signal and its super-wideband reference signal. Degraded signals that have some spectral loss between 8 kHz and 14 kHz will have scores below the scale's 4.75 maximum. The super-wideband scale will produce a score of approximately 3.8 for a narrowband signal with no other impairment and yield a score around 4.5 for a wideband signal.

It should be noted that [ITU-T P.863] scores may be influenced by the spectral content of the talker's speech in the reference file. If the spectral content is unbalanced, the maximum score for the scale will not be reached.

## 11.4 Stretching and compressing speech

Temporal compression or stretching in the context of speech coding and voice quality measurement is synonymous with "time scaling". From the perspective of [ITU-T P.863], this can be seen as four distinct categories:

1. Uniform time scaling without pitch preservation.
2. Uniform time scaling with pitch preservation.
3. Non-uniform time scaling without pitch preservation.
4. Non-uniform time scaling with pitch preservation.

A simple modelling of uniform time scaling without pitch preservation can be performed by varying the actual sample rate of a signal while keeping its nominal sample rate constant (i.e., resample a 48 kHz to 48.005 kHz signal and claim that it still has a 48 kHz sample rate). Non-uniform time scaling, is where small sections of the signal are stretched or compressed, or multiple compression and stretching factors are applied to different sections of the speech signal.

Time scaling is frequently observed in modern applications, but may also be caused by the test equipment itself. The sources for time scaling are manifold, but the most frequent ones are:

- advanced jitter buffer adaptation algorithms in VoIP systems;
- packet loss/error concealment methods;
- poor clock generators in A/D or D/A converters.

#### **11.4.1 Uniform time scaling without pitch preservation**

A consequence of uniform time scaling is that [ITU-T P.863] sees a continuously increasing or decreasing delay between the reference and the degraded signal. In [ITU-T P.863], this is overcome by an algorithm which estimates the amount of time scaling and subsequently resamples the input signals in order to eliminate the temporal compression or stretching.

Within the  $\pm 3\%$  difference in uniform time scaling between reference and degraded speech samples, [ITU-T P.863] exhibits a score deviation of less than  $\pm 0.1$  MOS in 50 per cent of the cases and  $\pm 0.3$  MOS for 90 per cent of the cases.

Uniform time scaling without pitch preservation outside of the  $\pm 3\%$  range is not expected to be encountered in the field. However, if presented with temporal modification exceeding this  $\pm 3\%$  range, [ITU-T P.863] compensates without taking into account the perceptual impact.

#### **11.4.2 Uniform time scaling with pitch preservation**

Discussions of uniform time scaling with pitch preservation are for further study.

#### **11.4.3 Non-uniform time scaling without pitch preservation**

This type of degradation is unlikely in live networks. As such no studies investigating this topic have been performed.

#### **11.4.4 Non-uniform time scaling with pitch preservation**

Studies into this area indicate that [ITU-T P.863] has more difficulty in predicting scores where high levels of non-uniform compression occur, than when non-uniform stretching occurs. With stretching, the scores typically appear to be within 0.5 of the subject test scores, while with compression the scores appear to be as much as 2.0 adrift. In all cases [ITU-T P.863] has a tendency to underestimate the MOS returned from a subjective test.

### **11.5 Step delay changes**

Step changes in delay between reference and degraded speech files result from adaptive jitter buffer implementations which adjust the silences between talk spurts. Such adjustments have a negligible effect in subjective test scores. [ITU-T P.863] will generally detect and account for step changes, but may fail to cope with detecting and eliminating all time-shifting, and thus return low scores. The ability to compensate for time-shifting depends on the magnitude of the time-shift and also the location within the sample. Results also seem to differ between male and female samples.

### **11.6 Long-term gaps in speech**

[ITU-T P.863] identifies missing speech, whether at the beginning, the middle or the end of a sentence. The score will be reduced whenever speech is missing.

Periods of silence might be in the range 200-500 ms. The cause of such silences might be an inter-system handover (from 3G to 2G) in mobile connections or sustained packet loss due to re-buffering in VoIP or streaming applications. The subjective effect will depend on whether the silence occurs at the beginning, middle or end of a sentence. Loss of speech in the middle of a sentence will affect the score more than a loss at the beginning or end of a sentence.

## 11.7 Short-term gaps in speech

[ITU-T P.863] considers the amount of lost speech in its MOS prediction. The subjective perception of lost speech is highly dependent on the occurrence of the loss with regard to the speech information content. It is dependent on the structure of the loss pattern in conjunction with the temporal structure of the particular speech file.

Short-term loss of speech might be caused by packet loss giving rise to occasional losses of 20 ms or more.

[ITU-T P.863] indicates a reduction in score for small loss rates. There is some dependency on different speech content of the reference material. Loss rates of 25 per cent and above result in a range close to a score of 1, which was a requirement of the specification for [ITU-T P.863].

## 11.8 Cellular handovers

Discussions of cellular handovers are for further study.

## 12 Comparison of ITU-T P.862.1/ITU-T P.862.2 and ITU-T P.863

### 12.1 Scale differences

This clause provides guidelines on how to compare existing speech quality results obtained with [b-ITU-T P.862.1] and [ITU-T P.863] narrowband mode, as well as how to compare existing results from [b-ITU-T P.862.2] and [ITU-T P.863] super-wideband mode.

[ITU-T P.863] will return scores very similar to PESQ [b-ITU-T P.862.1] in the narrowband mode with simple codecs such as ITU-T G.711. Tests with more sophisticated codecs and transmission techniques may yield different scores as [ITU-T P.863] addresses the objective assessment limitations of PESQ.

It is more difficult to compare [ITU-T P.863] super-wideband mode with PESQ [b-ITU-T P.862.2] because most wideband experiments were performed with 16 k sample rate material. [ITU-T P.863] super-wideband mode requires a 48 k sample rate reference file. The [ITU-T P.863] results with a 16 k sample rate reference file or an up-sampled 16 k reference file will be wrong, as there will not be any speech energy above 8 kHz.

### 12.2 Degradation handling differences

Table 5 provides a brief explanation of what can be expected from [ITU-T P.863] for certain types of degradation compared to the PESQ processing model.

**Table 5 – Brief explanation**

<b>Degradation</b>	<b>[ITU-T P.863]</b>	<b>[b-ITU-T P.862]/[b-ITU-T P.862.1]</b>
Bandwidth limitations	[ITU-T P.863] in super-wideband mode takes into account bandwidth limitations by detecting the absence of any speech energy above 3.8 kHz to indicate a narrowband degraded file and the absence of any speech energy above 7 kHz to indicate a wideband degraded file. With a narrowband signal the maximum achievable score is 3.8. Change in bandwidth between speakers (different gender or talker) in a single test may lead to a lower than expected score.	PESQ applies a linear frequency equalization stage before presenting the signals to the psycho-acoustic model. This effectively removes frequency response influences from being detected in the model. This is useful for small degrees of frequency shaping, but PESQ underestimates severe linear frequency response distortions.

**Table 5 – Brief explanation**

Degradation	[ITU-T P.863]	[b-ITU-T P.862]/[b-ITU-T P.862.1]
Short interrupts (e.g., packet loss)	The predicted quality score tends to a MOS value of 1.0 as frame loss increases up to 30%. Even small loss rates cause a drop in measured speech quality. Results show that the super-wideband mode is slightly more sensitive to short interrupts than the narrowband mode.	PESQ behaves in a similar way to [ITU-T P.863] with scores tending to a MOS of 1.0 as loss rate increases to 30%.
Long interrupts (e.g., VAD clipping, inter-system handovers)	Long interrupts describe muting of speech for 200 ms or more at the front, in the middle or the end of a speech sentence. Loss in the middle of speech leads to the largest drop in quality, followed by front-end loss with losses at the end of a sentence having least impact on score.	It has been claimed that PESQ reacts unexpectedly to lost speech. For small interrupts [ITU-T P.863] and PESQ produce consistent scores, but with longer interrupts PESQ predictions are significantly more optimistic than expected.

### 12.3 Standard codecs

Table 6 presents average scores for a number of narrowband codecs for [b-ITU-T P.862.1] and [ITU-T P.863]. An IRSsend mod filter was applied before processing. Additionally, average [ITU-T P.863] SWB scores for wideband codecs are given and super-wideband references have been used without IRS pre-filtering.

The source material is taken from [b-ITU-T P.501] and averaged across the provided samples and languages. The samples have been used regardless of whether or not they meet the transparency criteria. Please note that the scores are only presented to one decimal place because users are unable to distinguish differences of less than one decimal place. Note that these values are averages; for individual samples the scores may differ.

**Table 6 – Average scores**

Codec	[b-ITU-T P.862.1]	[ITU-T P.863] NB	[ITU-T P.863] SWB
Transparent	–	–	4.75
50-7 600 Hz	–	–	4.6
P.341	–	–	4.5
AMR-WB 23.85	–	–	4.0
AMR-WB 15.85	–	–	4.0
AMR-WB 12.65	–	–	3.9
AMR-WB 8.85	–	–	3.6
AMR-WB 6.60	–	–	3.2
NB only (50-3'800 Hz)	4.5	4.5	3.7
IRS only	4.5	4.4	3.7
IRS G.711	4.4	4.3	3.6
IRS G.729 A	3.6	3.9	3.0
IRS G.723 6.3 kbit/s	3.6	3.8	2.9
IRS G.723 5.3 kbit/s	3.4	3.6	2.8
IRS AMR 12.2 kbit/s	4.0	4.2	3.3

**Table 6 – Average scores**

Codec	[b-ITU-T P.862.1]	[ITU-T P.863] NB	[ITU-T P.863] SWB
IRS AMR 10.2 kbit/s	3.9	4.1	3.2
IRS AMR 7.95 kbit/s	3.7	3.9	3.1
IRS AMR 7.4 kbit/s	3.7	3.9	3.1
IRS AMR 6.7 kbit/s	3.6	3.8	3.0
IRS AMR 5.9 kbit/s	3.4	3.6	2.9
IRS AMR 5.15 kbit/s	3.3	3.5	2.8
IRS AMR 4.75 kbit/s	3.2	3.5	2.7
IRS EVRC-A	3.7	3.9	3.2
IRS EVRC-Bop0.	3.7	4.0	3.2
IRS QCELP 13 kbit/s	3.9	4.0	3.1
<p>NOTE 1 – IRS stands for IRS mod (send); 32 P.501 samples (SWB G.191 filter, –26 dB SPL acc. P.56, cut to 8 s, initial pause 500 ms, noise floor –85 dB).</p> <p>NOTE 2 – ITU-T P.863 in SWB mode scores transparent SWB speech samples at 4.75 MOS-LQO. Since, bandwidth limitations relative to SWB signals are treated as degradations, any limitation leads to lower scores. On average, a plain wideband signal (50-7'600 Hz) is scored at 4.6 MOS-LQO and a plain narrowband signal (50-3'800 Hz) at 3.7. This means that in narrowband networks scores of 3.7 MOS-LQO are not exceeded on average, and real connections are usually below that value due to additional degradations.</p>			

### 13 Procedure for comparing subjective test results to ITU-T P.863 results

It has never been possible to directly compare the results of two or more subjective tests because too many variables influence the results for a specific condition. Even if the same subjective test is run in the same subjective test laboratory with two different groups of people, slight differences in condition scores are always expected. The reason is that one group will never vote exactly the same way as a second group. When further variables are added, such as a different condition order, or different types of degradation, the absolute score achieved for a given condition will change.

Naturally, from a business perspective it is important to be able to compare the results of a new test with results from previous tests, and subjective test experts have developed various mechanisms to support this interpretation. These include the use of common reference conditions between tests, and also careful balancing of the test design to ensure a reasonable range of quality. Despite this effort, different subjective tests will always exhibit a broad range of scores for a given condition.

The very nature of an objective tool is that it always reports the same absolute score for a given speech recording. This is opposite to how a subjective test behaves. Therefore, a process must be used to compensate for the methodology mismatch between a subjective and an objective result set.

Before comparing results of a subjective test with [ITU-T P.863] results, perform the following simple linear-normalization process:

1. Calculate the linear-regression function between subjective test condition average scores (Subj) and [ITU-T P.863] condition average scores (P863) using the formula below:
  - a. *regression formula*(y) = a + bx
  - b.  $slope(b) = \frac{[N * \sum(Subj * P863) - \sum(Subj) * \sum(P863)]}{[N * \sum(Subj^2) - (\sum Subj)^2]}$
  - c.  $intercept(a) = \frac{[\sum(P863) - b * \sum(Subj)]}{N}$

2. Apply the linear-regression formula calculated above to each subjective test condition average.

Table 7 shows a worked example:

**Table 7 – Worked example for mapping objective to subjective scores**

Condition	Subj	ITU-T P863	Subj <sup>2</sup>	Subj × ITU-T P863	Mapped-Subj
1	3.52	3.95	12.40	13.89	4.05
2	2.84	3.55	8.09	10.09	3.35
3	2.30	3.02	5.30	6.96	2.80
4	3.39	4.20	11.46	14.22	3.91
5	2.69	3.19	7.22	8.56	3.19
6	3.19	3.79	10.16	12.08	3.70
7	3.56	4.45	12.69	15.87	4.09
8	3.28	3.53	10.77	11.60	3.80
9	3.27	3.51	10.70	11.49	3.79
10	2.69	3.03	7.22	8.14	3.19
11	3.17	3.64	10.03	11.51	3.68
12	2.91	3.43	8.45	9.98	3.42
13	2.64	3.08	6.95	8.11	3.14
14	2.94	3.01	8.63	8.84	3.45
15	2.24	2.88	5.02	6.46	2.73
16	3.29	3.75	10.84	12.34	3.81
17	3.19	3.57	10.16	11.37	3.70
18	2.63	3.34	6.89	8.78	3.13
19	2.78	3.10	7.74	8.61	3.29
20	2.28	2.75	5.20	6.27	2.78
21	2.28	2.83	5.20	6.46	2.78
22	2.38	2.50	5.64	5.95	2.87
23	1.79	2.46	3.21	4.41	2.27
24	2.52	2.96	6.35	7.45	3.02
25	2.32	2.69	5.40	6.25	2.82
26	2.33	3.16	5.44	7.38	2.83
27	2.33	2.69	5.44	6.28	2.83
28	1.75	2.42	3.06	4.23	2.23
29	2.75	3.36	7.56	9.25	3.26
30	2.63	3.07	6.89	8.05	3.13
31	2.53	2.91	6.41	7.37	3.03
32	2.32	2.56	5.40	5.94	2.82
33	2.07	2.51	4.30	5.20	2.56
34	2.60	2.77	6.78	7.22	3.11
35	1.70	2.28	2.88	3.87	2.18
36	1.73	2.37	2.99	4.10	2.21



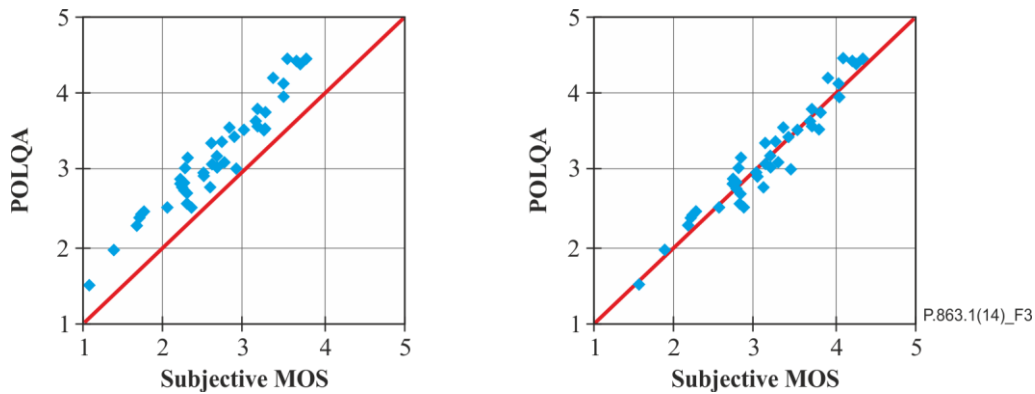
**Table 7 – Worked example for mapping objective to subjective scores**

Condition	Subj	ITU-T P863	Subj <sup>2</sup>	Subj × ITU-T P863	Mapped-Subj
37	1.10	1.50	1.22	1.65	1.57
38	1.42	1.95	2.01	2.76	1.89
39	2.24	2.82	5.02	6.31	2.73
40	3.02	3.52	9.13	10.65	3.53
41	3.51	4.12	12.32	14.47	4.03
42	3.72	4.38	13.83	16.29	4.25
43	3.68	4.42	13.52	16.24	4.21
44	3.80	4.45	14.46	16.93	4.33
sum	117.31	139.46	330.36	389.85	

$$\text{slope}(b) = (44 * 389.85 - 117.31 * 139.46) / (44 * 330.36 - 117.31^2) = 1.0247$$

$$\text{intercept}(a) = (139.46 - 1.0247 * 117.31) / 44 = 0.4376$$

In Figure 3, the graph on the left plots un-normalized subjective test scores against [ITU-T P.863], while the graph on the right illustrates the effect of normalizing/mapping the subjective scores to [ITU-T P.863].



**Figure 3 – Illustration of mapping**

The use of linear regression removes simple test condition biases from the subjective results.

Further normalization is possible by applying third-order monotonic polynomial regression function to remove biases that occur when a subjective test does not have a broad enough spread of quality, but applying this linear regression normalization process is a minimum requirement.

The topic of comparing subjective test results with objective model predictions is more fully discussed in [b-ITU-T P.1401].

## 14 Validation scope

[ITU-T P.863] has been validated across many degradation types that are seen today on telephone networks, or expected to become prevalent over the next 10 years.

### 14.1 Validated

Table 8 provides a list of factors that have been used in the selection and validation phase of the [ITU-T P.863] algorithm.

**Table 8 – Validated factors**

<p><b>Test factors</b></p> <p>Speech input levels to a codec</p> <p>Transmission channel errors</p> <p>Packet loss and packet loss concealment</p> <p>Bit rates if a codec has more than one bit-rate mode</p> <p>Transcoding</p> <p>Acoustic noise in sending environment</p> <p>Effect of varying delay in listening only tests</p> <p>Short-term time warping of audio signal</p> <p>Long-term time warping of audio signal</p> <p>Listening levels between 53 and 78 dB(A) SPL in super-wideband mode</p> <p>Packet loss and packet loss concealment with PCM type codecs</p> <p>Temporal and amplitude clipping of speech</p> <p>Linear distortions, including bandwidth limitations and spectral shaping ('non-flat frequency responses')</p> <p>Frequency response</p>
<p><b>Coding technologies</b></p> <p>ITU-T G.711, ITU-T G.711 PLC, ITU-T G.711.1</p> <p>ITU-T G.718, ITU-T G.719, ITU-T G.722, ITU-T G.722.1, ITU-T G.723.1, ITU-T G.726, ITU-T G.728, ITU-T G.729</p> <p>GSM-FR, GSM-HR, GSM EFR</p> <p>AMR-NB, AMR-WB (ITU-T G.722.2), AMR-WB+</p> <p>PDC-FR, PDC-HR</p> <p>EVRC (ANSI/TIA-127-A), EVRC-B (TIA-718-B)</p> <p>Skype (SILK V3, iLBC, iSAC and ITU-T G.729)</p> <p>Speex, QCELP (TIA-EIA-IS-733), iLBC, CVSD (64 kbit/s, "Bluetooth")</p> <p>MP3, AAC, AAC-LD</p> <p><b>Applications</b></p> <p><b>Test Factors</b></p> <p>Codec evaluation</p> <p>Terminal testing, influence of the acoustic path and the transducer in the sending and receiving direction. (NOTE – Acoustic path in receiving direction only for SWB mode)</p> <p>Bandwidth extensions</p> <p>Live network testing using digital or analogue connection to the network</p> <p>Testing of emulated and prototype networks</p> <p>UMTS, CDMA, GSM, TETRA, WB-DECT, VoIP, POTS, PSTN, Video Telephony, Bluetooth</p> <p>Voice Activity Detection (VAD), Automatic Gain Control (AGC)</p> <p>Voice Enhancement Devices (VED), Noise Reduction (NR)</p> <p>Discontinuous Transmission (DTX), Comfort Noise Insertion</p>

## 14.2 Not yet validated

[ITU-T P.863] has not been validated against the variables given in Table 9.

**Table 9 – Factors not validated**

<b>Test factors</b> Talker dependencies Multiple simultaneous talkers Bit-rate mismatching between an encoder and a decoder if a codec has more than one bit-rate mode Network information signals as input to a codec Artificial speech signals as input to a codec Music as input to a codec Listener echo Coding Technologies Coding technologies operating below 4 kbit/s, EVS
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## 14.3 Outside scope

[ITU-T P.863] is not intended to be used with the variables provided in Table 10.

**Table 10 – Factors outside scope**

<b>Test factors</b> Effect of delay in conversational tests Talker echo Sidetone Acoustic noise in receiving environment Applications Non-intrusive measurements Two-way communications performance
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## Appendix I

### ITU-T P.863 Transparency checks of reference files

(This appendix does not form an integral part of this Recommendation.)

Table I.1 contains [ITU-T P.863] transparency reference validation checks for the speech samples contained in Amendment 1 to [b-ITU-T P.501] (32 P.501 samples (SWB G.191 filter, –26 dB SPL acc. P.56, cut to 8 s, initial pause 500 ms, noise floor –85 dB)).

**Table I.1 – Transparency checks of reference files**

	P.863.1, Shift 0, 10, 15 ms by cutting				
	NB + SWB	Narrowband (8 kHz)		Super-wideband (48 kHz)	
	Transparent	Transparent	Min(0, 10, 15 ms)	Transparent	Min(0, 10, 15 ms)
<b>chinese_f1</b>	Y	Y	4.50	Y	4.75
chinese_f2		Y	4.50		4.70
<b>chinese_m1</b>	Y	Y	4.50	Y	4.75
<b>chinese_m2</b>	Y	Y	4.50	Y	4.75
<b>dutch_f1</b>	(Y)	Y	4.50	(Y)	4.75
<b>dutch_f2</b>	Y	Y	4.50	Y	4.75
<b>dutch_m1</b>	Y	Y	4.50	Y	4.75
<b>dutch_m2</b>	(Y)	(Y)	4.48	Y	4.75
<b>english_f1</b>	Y	Y	4.50	Y	4.75
<b>english_f2</b>	Y	Y	4.50	Y	4.75
<b>english_m1</b>	Y	Y	4.50	Y	4.75
<b>english_m2</b>	Y	Y	4.50	Y	4.75
<b>finnish_f1</b>	Y	Y	4.50	Y	4.75
<b>finnish_f2</b>	(Y)	(Y)	4.46	Y	4.75
finnish_m1			4.41	Y	4.75
<b>finnish_m2</b>	(Y)	(Y)	4.47	Y	4.75
french_f1		Y	4.50		4.58
<b>french_f2</b>	Y	Y	4.50	Y	4.75
<b>french_m1</b>	Y	Y	4.50	Y	4.75
<b>french_m2</b>	(Y)	(Y)	4.47	(Y)	4.74
<b>german_f1</b>	Y	Y	4.50	Y	4.75
<b>german_f2</b>	Y	Y	4.50	Y	4.75
german_m1			4.43	Y	4.75
<b>german_m2</b>	Y	Y	4.50	Y	4.75
<b>italian_f1</b>	Y	Y	4.50	Y	4.75
<b>italian_f2</b>	Y	Y	4.50	Y	4.75
<b>italian_m1</b>	Y	Y	4.50	Y	4.75
italian_m2			4.44	Y	4.75
<b>japanese_f1</b>	Y	Y	4.50	Y	4.75
<b>japanese_f2</b>	Y	Y	4.50	Y	4.75

**Table I.1 – Transparency checks of reference files**

	<b>P.863.1, Shift 0, 10, 15 ms by cutting</b>				
	<b>NB + SWB</b>	<b>Narrowband (8 kHz)</b>		<b>Super-wideband (48 kHz)</b>	
	<b>Transparent</b>	<b>Transparent</b>	<b>Min(0, 10, 15 ms)</b>	<b>Transparent</b>	<b>Min(0, 10, 15 ms)</b>
japanese_m1		Y	4.50		4.70
<b>japanese_m 2</b>	<b>Y</b>	<b>Y</b>	<b>4.50</b>	<b>Y</b>	<b>4.75</b>

## Appendix II

### Typical scores expected from ITU-T P.863 for a given codec

(This appendix does not form an integral part of this Recommendation.)

Tables II.1 and II.2 list typical scores expected from [ITU-T P.863] for a given codec obtained by using a set of 400 samples spoken by two male and two female French talkers. It should be noted that because [ITU-T P.863] demonstrates a significant material dependency, as discussed in clause 8, a single sample may differ from the typical value presented below by as much as 1.4 MOS.

**Table II.1 – Narrowband mode**

Codec name	Details	V 2.3 With level pre-alignment		
		Minimum	Average	Maximum
ITU-T G.711	A law, 64 kbit/s	3.4	4.27	4.5
ITU-T G.711	$\mu$ law, 64 kbit/s	3.89	4.41	4.5
ITU-T G.726	32 kbit/s	3.41	4.28	4.5
ITU-T G.728	16 kbit/s	3.56	4.21	4.5
ITU-T G.729	8 kbit/s	3.84	4.21	4.47
ITU-T G.729A	8 kbit/s	3.48	4	4.34
GSM FR	13 kbit/s	2.7	3.73	4.32
GSM HR	5.6 kbit/s	2.79	3.39	3.68
GSM HR	5.6 kbit/s, DTX on	2.79	3.39	3.68
GSM EFR	12.2 kbit/s	4	4.39	4.5
GSM EFR	12.2 kbit/s, DTX on	4.1	4.42	4.5
AMR	4.75 kbit/s	2.7	3.44	3.95
AMR	4.75 kbit/s, DTX on	2.85	3.46	3.92
AMR	5.15 kbit/s	2.85	3.52	3.93
AMR	5.15 kbit/s, DTX on	2.75	3.53	4.01
AMR	5.9 kbit/s	3.04	3.73	4.16
AMR	5.9 kbit/s, DTX on	3.07	3.71	4.2
AMR	6.7 kbit/s	3.34	3.88	4.28
AMR	6.7 kbit/s, DTX on	3.29	3.86	4.3
AMR	7.4 kbit/s	3.52	4.02	4.39
AMR	7.4 kbit/s, DTX on	3.43	3.98	4.42
AMR	7.95 kbit/s	3.54	4.01	4.36
AMR	7.95 kbit/s, DTX on	3.45	3.98	4.36
AMR	10.2 kbit/s	3.93	4.24	4.5

**Table II.1 – Narrowband mode**

Codec name	Details	V 2.3 With level pre-alignment		
		Minimum	Average	Maximum
AMR	10.2 kbit/s. DTX on	3.76	4.21	4.47
AMR	12.2 kbit/s	3.94	4.32	4.5
AMR	12.2 kbit/s. DTX on	3.86	4.28	4.49

**Table II.2 – Super-wideband mode**

Codec name	Details	V 2.3 With level pre-alignment		
		Minimum	Average	Maximum
ITU-T G.711	A law. 64 kbit/s	2.97	3.5	4.02
ITU-T G.711	$\mu$ law. 64 kbit/s	2.89	3.5	4.04
ITU-T G.726	32 kbit/s	2.9	3.42	3.94
ITU-T G.728	16 kbit/s	2.68	3.19	3.87
ITU-T G.729	8 kbit/s	2.65	3.14	3.9
ITU-T G.729A	8 kbit/s	2.5	2.93	3.82
GSM FR	13 kbit/s	1.81	2.68	3.73
GSM HR	5.6 kbit/s	2.25	2.57	3.3
GSM HR	5.6 kbit/s. DTX on	2.25	2.57	3.3
GSM EFR	12.2 kbit/s	2.8	3.3	4.06
GSM EFR	12.2 kbit/s. DTX on	2.79	3.29	4.07
AMR	4.75 kbit/s	1.79	2.56	3.44
AMR	4.75 kbit/s. DTX on	1.81	2.57	3.48
AMR	5.15 kbit/s	2.03	2.67	3.47
AMR	5.15 kbit/s. DTX on	1.96	2.65	3.45
AMR	5.9 kbit/s	2.19	2.81	3.7
AMR	5.9 kbit/s. DTX on	2.17	2.81	3.74
AMR	6.7 kbit/s	2.29	2.94	3.83
AMR	6.7 kbit/s. DTX on	2.22	2.95	3.86
AMR	7.4 kbit/s	2.51	3.03	3.89
AMR	7.4 kbit/s. DTX on	2.48	3.02	3.91
AMR	7.95 kbit/s	2.5	3.03	3.84

**Table II.2 – Super-wideband mode**

Codec name	Details	V 2.3 With level pre-alignment		
		Minimum	Average	Maximum
AMR	7.95 kbit/s. DTX on	2.51	3.02	3.89
AMR	10.2 kbit/s	2.79	3.25	4.05
AMR	10.2 kbit/s. DTX on	2.77	3.25	4.05
AMR	12.2 kbit/s	2.83	3.33	4.1
AMR	12.2 kbit/s. DTX on	2.86	3.31	4.13
G.711	A law. 64 kbit/s. IRS	2.86	3.4	3.74
G.711	$\mu$ law. 64 kbit/s. IRS	2.72	3.46	3.74
G.726	32 kbit/s. IRS	2.77	3.32	3.68
G.728	16 kbit/s. IRS	2.55	3.09	3.57
G.729	8 kbit/s. IRS	2.52	3.06	3.64
G.729	Annex A. 8 kbit/s. IRS	2.39	2.98	3.57
GSM FR	13 kbit/s. IRS	1.84	2.65	3.23
GSM HR	5.6 kbit/s. IRS	2.09	2.63	3.21
GSM HR	5.6 kbit/s. DTX on. IRS	2.09	2.63	3.21
GSM EFR	12.2 kbit/s. IRS	2.8	3.3	4.06
GSM EFR	12.2 kbit/s. DTX on. IRS	2.61	3.17	3.73
AMR	4.75 kbit/s. IRS	2	2.69	3.31
AMR	4.75 kbit/s. DTX on. IRS	2.08	2.69	3.28
AMR	5.15 kbit/s. IRS	2.12	2.75	3.3
AMR	5.15 kbit/s. DTX on. IRS	2.12	2.72	3.32
AMR	5.9 kbit/s. IRS	2.22	2.86	3.51
AMR	5.9 kbit/s. DTX on. IRS	2.26	2.82	3.48
AMR	6.7 kbit/s. IRS	2.2	2.96	3.61
AMR	6.7 kbit/s. DTX on. IRS	2.27	2.94	3.59
AMR	7.4 kbit/s. IRS	2.4	3.02	3.65
AMR	7.4 kbit/s. DTX on. IRS	2.48	2.99	3.66
AMR	7.95 kbit/s. IRS	2.45	3.02	3.66



**Table II.2 – Super-wideband mode**

Codec name	Details	V 2.3 With level pre-alignment		
		Minimum	Average	Maximum
AMR	7.95 kbit/s. DTX on. IRS	2.41	2.99	3.61
AMR	10.2 kbit/s. IRS	2.56	3.13	3.73
AMR	10.2 kbit/s. DTX on. IRS	2.49	3.1	3.73
AMR	12.2 kbit/s. IRS	2.3	3.18	3.82
AMR	12.2 kbit/s. DTX on. IRS	2.62	3.19	3.77
ITU-T G.722	64 kbit/s	3.96	4.43	4.73
ITU-T G.711.1	A law. 96 kbit/s	4.01	4.45	4.75
ITU-T G.711.1	$\mu$ law. 96 kbit/s	4	4.45	4.75
ITU-T G.718	32 kbit/s	3.86	4.28	4.62
ITU-T G.718	32 kbit/s. DTX on	3.83	4.27	4.61
ITU-T G.722.1	32 kbit/s	3.29	3.91	4.46
ITU-T G.729.1	32 kbit/s	3.86	4.26	4.65
ITU-T G.722.2	6.6 kbit/s	2.41	3.15	3.92
ITU-T G.722.2	6.6 kbit/s. DTX on	2.42	3.18	3.86
ITU-T G.722.2	8.85 kbit/s	2.8	3.54	4.17
ITU-T G.722.2	8.85 kbit/s. DTX on	2.85	3.57	4.2
ITU-T G.722.2	12.65 kbit/s	3.24	3.87	4.41
ITU-T G.722.2	12.65 kbit/s. DTX on	3.18	3.88	4.42
ITU-T G.722.2	14.25 kbit/s	3.37	3.94	4.44
ITU-T G.722.2	14.25 kbit/s. DTX on	3.36	3.98	4.5
ITU-T G.722.2	15.85 kbit/s	3.32	3.99	4.51
ITU-T G.722.2	15.85 kbit/s. DTX on	3.38	3.99	4.54
ITU-T G.722.2	18.25 kbit/s	3.47	4.06	4.57
ITU-T G.722.2	18.25 kbit/s. DTX on	3.47	4.09	4.57

**Table II.2 – Super-wideband mode**

Codec name	Details	V 2.3 With level pre-alignment		
		Minimum	Average	Maximum
ITU-T G.722.2	19.85 kbit/s	3.25	4.08	4.61
ITU-T G.722.2	19.85 kbit/s. DTX on	3.44	4.09	4.61
ITU-T G.722.2	23.05 kbit/s	3.6	4.17	4.66
ITU-T G.722.2	23.05 kbit/s. DTX on	3.63	4.18	4.64
ITU-T G.722.2	23.85 kbit/s	3.36	4.09	4.56
ITU-T G.722.2	23.85 kbit/s. DTX on	3.36	4.09	4.59
ITU-T G.718B	48 kbit/s	4.16	4.57	4.75
ITU-T G.718B	48 kbit/s. DTX on	4.12	4.59	4.75
ITU-T G.711.1D	A law. 128 kbit/s	4.52	4.73	4.75
ITU-T G.711.1D	$\mu$ law. 128 kbit/s	4.51	4.73	4.75
ITU-T G.722B	96 kbit/s	4.4	4.72	4.75
ITU-T G.722.1C	48 kbit/s	3.64	4.21	4.73
ITU-T G.729.1E	64 kbit/s	4.45	4.7	4.75

## **Appendix III**

### **Applications for further investigation**

(This appendix does not form an integral part of this Recommendation.)

Claims have been made that [ITU-T P.863] provides inaccurate predictions of absolute quality for the following:

- Acoustic recordings using free-field microphones without head and torso simulator (HATS) or ear-canal simulation;

These claims need to be validated by future investigations.

## Bibliography

- [b-ITU-T G.107] Recommendation ITU-T G.107 (2011), *The E-model: a computational model for use in transmission planning*.
- [b-ITU-T G.191] Recommendation ITU-T G.191 (2010), *Software tools for speech and audio coding standardization*.
- [b-ITU-T P.48] Recommendation ITU-T P.48 (1988), *Specification for an intermediate reference system*.
- [b-ITU-T P.50] Recommendation ITU-T P.50 (1999), *Artificial voices*.
- [b-ITU-T P.501] Recommendation ITU-T P.501 (2012), *Test signals for use in telephony*.
- [b-ITU-T P.563] Recommendation ITU-T P.563 (2004), *Single-ended method for objective speech quality assessment in narrow-band telephony applications*.
- [b-ITU-T P.800] Recommendation ITU-T P.800 (1996), *Methods for subjective determination of transmission quality*.
- [b-ITU-T P.800.1] Recommendation ITU-T P.800.1 (2006), *Mean Opinion Score (MOS) terminology*.
- [b-ITU-T P.830] Recommendation ITU-T P.830 (1996), *Subjective performance assessment of telephone-band and wideband digital codecs*.
- [b-ITU-T P.861] Recommendation ITU-T P.861 (1998), *Objective quality measurement of telephone-band (300-3400 Hz) speech codecs*. (superseded)
- [b-ITU-T P.862] Recommendation ITU-T P.862 (2001), *Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs*.
- [b-ITU-T P.862.1] Recommendation ITU-T P.862.1 (2003), *Mapping function for transforming P.862 raw result scores to MOS-LQO*.
- [b-ITU-T P.862.2] Recommendation ITU-T P.862.2 (2007), *Wideband extension to Recommendation P.862 for the assessment of wideband telephone networks and speech codecs*.
- [b-ITU-T P.1401] Recommendation ITU-T P.1401 (2012), *Methods, metrics and procedures for statistical evaluation, qualification and comparison of objective quality prediction models*.
- [b-ITU-R BS.1387] Recommendation ITU-R BS.1387 (2001), *Method for objective measurements of perceived audio quality*.
- [b-ETSI TR103138] ETSI TR 103 138 (2013-10), *Speech and multimedia Transmission Quality (STQ); Speech samples and their usage for QoS testing*.



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