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

Methods for objective and subjective assessment of  
quality

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**Application guide for objective quality  
measurement based on Recommendations  
P.862, P.862.1 and P.862.2**

ITU-T Recommendation P.862.3



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## **ITU-T Recommendation P.862.3**

### **Application guide for objective quality measurement based on Recommendations P.862, P.862.1 and P.862.2**

#### **Summary**

ITU-T Recommendation P.862.3 provides some important remarks that should be taken into account in the objective quality evaluation of speech conforming to ITU-T Recommendations P.862, P.862.1 and P.862.2. Users of ITU-T Recommendation P.862 should understand and follow the guidance given in this Recommendation.

This Recommendation forms a supplementary guide for users of ITU-T Recommendation P.862, which recommends a means of estimating listening speech quality by using reference and degraded speech samples. The scope of ITU-T Recommendation P.862 is clearly defined in itself. This Recommendation does not extend or narrow the scope, but provides necessary and important information for obtaining stable, reliable and meaningful objective measurement results in practice.

#### **Source**

ITU-T Recommendation P.862.3 was approved on 13 November 2007 by ITU-T Study Group 12 (2005-2008) under the ITU-T Recommendation A.8 procedure.

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## ITU-T Recommendation P.862.3

### Application guide for objective quality measurement based on Recommendations P.862, P.862.1 and P.862.2

#### 1 Scope

This Recommendation provides some important remarks that should be taken into account in the objective quality evaluation of speech conforming to [ITU-T P.862], [ITU-T P.862.1] and [ITU-T P.862.2]. Users of [ITU-T P.862] should understand and follow the guidance given in this Recommendation.

This Recommendation forms a supplementary guide for users of [ITU-T P.862], which recommends a means of estimating listening speech quality by using reference and degraded speech samples. It cannot be used for the assessment of talking quality or interaction quality. It assumes that an objective quality estimation algorithm strictly conforms to [ITU-T P.862]. This can be confirmed by the conformance test provided as an annex to [ITU-T P.862].

The scope of [ITU-T P.862] is clearly defined in itself. This Recommendation does not extend or narrow the scope, but provides necessary and important information for obtaining stable, reliable and meaningful objective measurement results in practice.

Applications and limitations associated with the wideband extension to P.862 defined in [ITU-T P.862.2] are discussed in clause 12.

#### 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.50] ITU-T Recommendation P.50 (1999), *Artificial voices*.
- [ITU-T P.56] ITU-T Recommendation P.56 (1993), *Objective measurement of active speech level*.
- [ITU-T P.501] ITU-T Recommendation P.501 (2007), *Test signals for use in telephony*.
- [ITU-T P.800] ITU-T Recommendation P.800 (1996), *Methods for subjective determination of transmission quality*.
- [ITU-T P.830] ITU-T Recommendation P.830 (1996), *Subjective performance assessment of telephone-band and wideband digital codecs*.
- [ITU-T P.862] ITU-T Recommendation 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*.
- [ITU-T P.862.1] ITU-T Recommendation P.862.1 (2003), *Mapping function for transforming P.862 raw result scores to MOS-LQO*.
- [ITU-T P.862.2] ITU-T Recommendation P.862.2 (2007), *Wideband extension to Recommendation P.862 for the assessment of wideband telephone networks and speech codecs*.

### 3 Definitions

This Recommendation defines the following terms:

**3.1 source speech/signal:** The original speech signal without any degradation. This should be recorded and stored conforming to [ITU-T P.830]. It may or may not be the same as reference speech defined below.

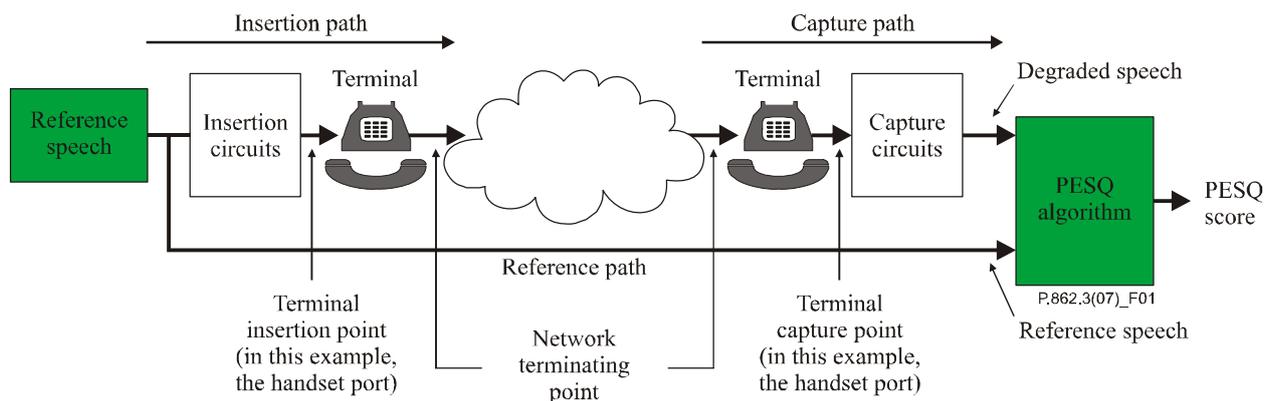
**3.2 reference speech/signal:** The speech signal to be used by the [ITU-T P.862] algorithm as a reference against which the effects of the system under test are revealed.

**3.3 input speech/signal:** The signal fed into the system under test at the signal insertion point. It is derived from the reference speech signal. It may be identical to the reference signal or it may be processed by e.g., overlaying it with noise. Further information is provided in clause 7.10.

**3.4 degraded speech/signal:** The reference speech that has passed through the system under test.

**3.5 signal insertion path:** Consists of the connection path (wiring, electronics, etc.) between the reference signal to the [ITU-T P.862] algorithm and the input interface, called the insertion point.

Figure 1 divides an example test circuit into insertion path, system under test, and capture path, and shows possible insertion and capture points *in the case of hardware measurements*. The particular insertion and capture points will depend on the specific system under test and the configuration of the test set-up.



NOTE – Depending on how the signals are inserted, captured and stored, and whether the ITU-T Rec. P.862 measurement computations are done in real-time or deferred, these paths may be continuous physical electrical pathways, or they may be logical paths, such as when an output sample is stored for later analysis.

**Figure 1 – Example of measurement set-up and terminology**

**3.6 signal capture path:** Consists of the connection path between the capture point (output interface with the network under test) and the [ITU-T P.862] algorithm (see Figure 1).

**3.7 dBov:** The value in dB relative to the overload point of a digital system. According to ITU-T Recommendation G.711, 0 dBm0 in analogue representation corresponds to –6.15 dBov and –6.18 dBov for A-law and  $\mu$ -law codecs, respectively.

### 4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

IRS Intermediate Reference System

MOS Mean Opinion Score

MOS-LQO	Mean Opinion Score – Listening Quality Objective (an estimation of subjective listening quality using an objective measurement technique)
MOS-LQS	Mean Opinion Score – Listening Quality Subjective (a direct measurement of listening quality using subjective ratings of samples)
PESQ	Perceptual Evaluation of Speech Quality
RMS	Root Mean Square

## 5 Conventions

It is recommended that raw results from [ITU-T P.862] measurements be converted to MOS-LQO (as defined in ITU-T Recommendation P.800.1) using the relation defined in [ITU-T P.862.1.]<sup>1</sup>. This will prevent potential confusion in comparison and interpretation of results due to the superficial similarity of the two scales.

## 6 General remarks

This clause gives supplementary remarks about the scope of [ITU-T P.862]. The scope itself is summarized in [ITU-T P.862] quite clearly.

The reliability and consistency of the results are dependent on several factors, for example:

- Number of calls.
- Number of measurements.
- Length of speech samples.
- Type of speech used, e.g., natural or artificial.

These factors, and the following considerations, affect the structure and complexity of the tests:

- Purpose of the measurements (e.g., for benchmarking connections, routine monitoring or fault diagnosis).
- Transmission channel characteristics (e.g., do the channel characteristics vary over time such as with mobile or some types of VoIP connection?).
- Time available to make the measurements (which may impact on the scope of the testing).

Where it is suspected that certain connection types may be affected by 'busy-hour' conditions, it may also be important to carry out a number of sequences of measurements at different times throughout the day.

The test structure used should always be quoted in conjunction with processed values of MOS-LQO.

### 6.1 Testing factors

[ITU-T P.862] is validated for the evaluation of test factors, coding technologies and applications, which are listed in Table 1 of [ITU-T P.862]. In particular, care should be taken when one carries out live network testing since there might be some equipment that causes degradation that [ITU-T P.862] cannot handle, e.g., artefacts caused by noise reduction systems, in between the signal insertion point and signal capture point. It is also known that PESQ underestimates severe linear frequency response distortions. This applies especially to, e.g., bandwidth limitations narrower than 300 Hz ... 3.4 kHz.

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<sup>1</sup> The detailed procedure for obtaining MOS-LQO can be found in clause 10.

Use of [ITU-T P.862] with systems that include noise suppression algorithms between the signal insertion point and the signal capture point is not recommended.

## 6.2 Applications

[ITU-T P.862] can be used as a means for live network testing, in which one evaluates the system under live conditions rather than computer-simulated conditions or fixed test set-up in a laboratory environment.

Live field testing will not produce repeatable results due to uncontrolled time-varying transmission channels. The alternatives are controlled network simulations with exactly repeatable results. For the latter condition, averaging should be used.

Live field network testing, such as mobile drive testing, will affect the structure and content of the reference speech signals. In drive testing this is due to the necessity to assess a very highly time-varying quality in order to get accurate geographical quality information.

Live field network testing also presents the need to assess quality on a per-sample basis since per-condition averaging is not possible with live, time-varying network conditions.

Both of the above will have an effect on the stability and possibly on the accuracy of [ITU-T P.862] results. For this reason, users of [ITU-T P.862] in live network testing should be careful to check results and repeat measurements with a view to checking result stability. Performance results are shown in clause 10.2.

If the system under test involves a broadband terminal (such as certain hands-free headsets, or broadband IP phones) then PESQ will predict the quality as it would have been perceived for IRS-type receive filtering.

## 7 Characteristics of the reference signals

Reference signals are defined and used as input signals to the system under test and as the reference input for [ITU-T P.862]. The characteristics of the signal insertion path are discussed separately in clause 9. If the language under evaluation is included in the speech database provided as Annex B of [ITU-T P.501], we recommend using it as test signals to improve the compatibility among different measurements by avoiding the use of different reference signals.

### 7.1 Length of signal

[ITU-T P.862] has been validated in ITU-T for use with signals that are mostly 8-12 s long. However, it is known that [ITU-T P.862] can be applied to speech up to 30 s long [b-ITU-T COM12-D008]. Therefore, it is recommended that each speech sample should be 8-30 s long. This includes any silence before, after and between utterances<sup>2</sup>.

For live field test scenarios, shorter reference signals may be used, however this may not exercise the system as fully as possible. These shorter sentences should use at least the 3.2 s of speech as defined in clause 7.2.

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<sup>2</sup> The reference software provided as Annex A to [ITU-T P.862] has the following limitation with respect to the length of signals, although this limitation is already out of the range determined in this Recommendation: Due to the precision available to the floating point arithmetic in [ITU-T P.862], once the signals being processed reach a certain length, errors will start to be introduced in the signal energy calculation. Analysis suggests that signals with more than about one million samples will start to cause problems. Sixty seconds of a 16 kHz monoaural signal contains 960'000 samples and this would be a sensible threshold at which to apply a warning.

It should be noted that, because of the non-linearity of the [ITU-T P.862] algorithm, the result obtained using concatenated signals will not correspond to the simple arithmetic mean of the results for individual samples.

## **7.2 Active speech**

The speech activity in the reference speech, which can be measured based on [ITU-T P.56]<sup>3</sup>, should be between 40% and 80%. There should be a minimum of 3.2 s active speech in the reference. In combination with the recommended signal file length, this should ensure that [ITU-T P.862] has enough speech to make an accurate prediction and the speech should contain some silence to exercise important elements in the network.

## **7.3 Temporal structure**

Reference speech should comprise utterances separated by silent periods representative of natural pauses in speech. Most of the experiments used in calibrating and validating [ITU-T P.862] contained pairs of sentences separated by silence. Good examples are speech materials included in [ITU-T P-series Supp.23]<sup>4</sup>, which last 8 s and include two short sentences separated by a silent period of at least 1 s, and in Annex B of [ITU-T P.501] as mentioned above. It is recommended that the reference speech includes a few continuous utterances rather than many short utterances of speech such as rapid counting<sup>5</sup>.

## **7.4 Active speech level**

The active speech level referred to in this Recommendation is the equivalent level of the digitally stored reference signal, as measured according to [ITU-T P.56]. The active speech level applied to the signal insertion path of the measurement system is separately described in clause 9. It is recommended that all the reference speech files be stored at a level of –30 dBov to avoid peak clipping. Note that this is the level of source speech stored in the digital format and that the input level to the system under test should be determined separately according to the purpose of objective measurement (see [ITU-T P.830]).<sup>6</sup>

## **7.5 Application of artificial voice**

The application of artificial voice signals needs more investigation, from the viewpoints of language and temporal structure of signal power, and possibly other factors [b-ITU-T COM12-D145].

## **7.6 Requirements for speech recordings**

[ITU-T P.800] and [ITU-T P.830] give guidance for recording speech materials. This Recommendation assumes that source speech is recorded in conformance with this guidance. Note

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<sup>3</sup> ITU-T Recommendation G.191 provides a software called sv56demo.c, which measures the active speech ratio and active speech level conforming to [ITU-T P.56].

<sup>4</sup> Please note that the copyright on [ITU-T P-series Supp.23] does not allow the use of the signals in commercial applications.

<sup>5</sup> The reference software provided as Annex A to [ITU-T P.862] has a limit of 50 as the maximum number of utterances. If reference signals with many utterances are used, it must be verified that the implementation of [ITU-T P.862] used for the test can handle that large a number.

<sup>6</sup> A typical nominal value for active speech level is –20 dBm0, corresponding to approximately –26 dBov. In any specific system to be tested, the mean active speech level in the system under test may be significantly different from the nominal value of –20 dBm0. In such cases, the measured mean value may be used as the input active speech level. When the system response to input level is being assessed, it is appropriate to use a range of active speech values, for example, –14, –26 and –38 dBov (approximately equivalent to –8, –20 and –32 dBm0) as recommended by [ITU-T P.830].

that the reference speech may be the same as this source speech or it may have added low-level noise floor and/or frequency shaping (see clauses 7.9 and 7.10).

### **7.7 Variation in talker and speech content**

Variation due to the talker and speech content can be controlled by using a fixed set of samples for all test cases to be compared. Therefore, it is helpful to use the speech database provided as Annex B to [ITU-T P.501] to facilitate *post-hoc* comparisons and interpretation of results from different laboratories.

For network simulation scenarios, it is recommended that the reference speech should include a minimum of two female and two male talkers, each speaking different sentences. The P.862.1 scores obtained with these different samples should be averaged for a per-condition evaluation afterwards

For live field test scenarios, less speaker variation may be used, however this may not exercise the system as fully as possible. If this format is necessary, multiple speakers might be included in these short reference signals. In case of an intended per-sample evaluation, samples containing more than one speaker's voice will decrease the sample dependency of the derived results.

During the validation of [ITU-T P.862], very little data was available for children's voices and certain speech characteristics (e.g., voice/speech disorders, etc.). With the limited data available, no problems were observed with children's voices. Music must not be used with [ITU-T P.862].

It is also recommended to use several different speech samples (4-10 sentences) per talker to reflect phonetic variations.

### **7.8 Leading and trailing silences**

[ITU-T P.862] uses the RMS level of the reference and degraded signals for level alignment. If long silences are included at the beginning and end of the reference signal, then the level alignment result may be compromised.

A minimum leading and trailing silence of 0.5 s is recommended, as long as the measurement equipment can synchronize the degraded speech with the reference one within that time.

A maximum leading and trailing silence of about 2 s is recommended and may be useful if there is a high level of delay in the system.

### **7.9 Pre-filtering**

The reference speech prepared according to clauses 7.1 to 7.8 should be filtered so that the sending frequency characteristics of a handset are taken into account. It should be noted that [ITU-T P.862] assumes that reference speech reflects such electro-acoustic characteristics appropriately. When one assumes that the reference speech is fed into networks as the output of a handset terminal, ITU-T recommends the use of the modified IRS sending characteristics defined in Annex D of [ITU-T P.830]. Such filtering should be done after clauses 7.1 to 7.8 have been taken into account appropriately.

Care should be taken to coordinate the filtering used with the nominal frequency response of the system under test, because such filtering is dependent on where one feeds the reference speech to equipment and/or networks under test (see clause 9).

### **7.10 Noise floor**

The noise floor in reference speech should be adequately low as expected in recordings conforming to [ITU-T P.800] and [ITU-T P.830]. It is also possible to add complete silence (e.g., a signal having a digital amplitude of zero) so that the reference speech signals have the proper

characteristics defined in clauses 7.1, 7.2, 7.3 and 7.8<sup>7</sup>. This is the case where the reference speech corresponds to the source speech, as described in clause 7.6.

If one anticipates unwanted noise in the measurement paths described in clause 9 or the noise floor in the device under test itself, however, a low noise floor of about  $-75$  dBov, white spectrum, should be intentionally added to the reference signal as mentioned above and stored in the 16-bit linear PCM format. The level of the noise floor should be determined within 0-4000 Hz<sup>8</sup>. Noise at this level will not adversely affect the results based on [ITU-T P.862], but will effectively remove the contribution of such measurement noise to the final score [ITU-T COM12-D011]. It is quite important to add such a noise floor after to the pre-filtering described in clause 7.9.

The active speech level at the signal insertion path of the measurement system described in clause 9 should be calibrated after such pre-filtering<sup>9</sup>.

It should be noted that the proposed insertion of additional noise into the reference signal will lead to more accurate results if the unwanted noise at the receiving path is a continuous noise-floor and it does not solve problems coming up with comfort noise, which is only inserted in speech pauses.

### 7.11 Implementations issues

Many signals included in the P.862 conformance test do not fulfil the requirements set forth above. For the conformance test, this does not matter at all since the sole purpose is to prove the correctness of the implementation. Care must however be taken that the implemented algorithm also produces results in cases of violation of the requirements defined in this Recommendation since otherwise the conformance test cannot be applied.

## 8 Characteristics of the degraded signal to be assessed

Degraded signals are the output of the system under test that correspond to the test input, including any effects due to the measurement interface. This clause describes the characteristics of signals

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<sup>7</sup> Sending a digital silence into a digital insertion circuit, e.g., ISDN phone, and then to a wireless phone link may cause undesirable side effects if GSM or 3GPP codecs are used in the wireless network. Namely, sending a constant pattern of the lowest positive G.711 A-law value of +8 linear (D5 PCM hex) will cause the speech codec to reset at a period of 20 ms, i.e., 50 times/s. This is due to the built-in codec homing procedure for codec testing purposes. Although the resetting during speech pauses is not harmful for the codec itself, measured speech quality may not be the same compared with the normal situation in which codecs are not reset. A side effect would be that the codec would not use the discontinuous transmission (DTX) during speech pauses although this is enabled by the network. Therefore, the comfort noise effect or possible speech clipping by the voice activity detector (VAD) would not be verified at all. To overcome this specific problem, a low-level noise floor of approximately  $-65$  dBm0 should be added to the test sample before it is sent to the network. This will break the constant pattern of digital silence.

<sup>8</sup> When one employs 16 kHz as the sampling rate of reference speech, care should be taken in determining the level of the noise floor.

<sup>9</sup> The noise floor may not be fed into the system under test because the level of the noise floor becomes less than the minimum possible level of the system. For example, the lowest values for A-law coding are  $\pm 8$  in linear 16-bit values. Thus, the lowest level is  $-72$  dBov. This means that, if the input level to the system is calibrated to  $-30$  dBov for instance, the noise floor at  $-75$  dBov cannot go through the system and does not solve the unwanted noise problem at all. If significantly higher input levels than a nominal active speech level of  $-26$  dBov is to be tested, and if a stored noise floor of  $-75$  dBov is applied to the reference sample, possible degradations (e.g., clicks, noise bursts, etc.) on speech pauses may not be estimated by [ITU-T P.862], because a higher noise floor level in the degraded sample may mask low level degradations. By using a nominal active speech level of approximately  $-26$  dBov, however, this problem does not exist.

that are digitally stored as output of the system under test for use in the calculation based on [ITU-T P.862]. The characteristics of the signal capture path are discussed separately in clause 9.

### **8.1 Difference in active speech duration between reference and degraded speech signal**

The active speech duration is defined in [ITU-T P.56].

[ITU-T P.862] uses the RMS levels of the reference and degraded signals for level alignment. This means that the algorithm may give erroneous results if speech is missing or if silence is added to, or taken away, from the degraded signal.

When an utterance has been deleted from the degraded signal, or if one or more large sections of the degraded signal have been muted, the signal will be level-shifted to a value above the actual value.

When silence has been taken out of the degraded signal, the signal will be level-shifted to a value below the actual value.

These concerns will affect the amount of disturbance present in the degraded signal and will therefore affect the objective quality measurement result. If the durations of speech in the reference and degraded signals differ by more than 25%, the effect may be large enough to significantly bias the result. This is especially true if long continuous sections of speech have been replaced by silence.

### **8.2 Active speech level**

The active speech level is defined in [ITU-T P.56].

Although the active speech level is normalized in calculating PESQ values, it is recommended that the digital speech level stored as degraded signals to the PESQ algorithm should be around  $-30$  dBov to avoid clipping and quantization distortion. It should be noted that [ITU-T P.862] cannot be used to evaluate the effects of the receiving/listening level<sup>10</sup>.

### **8.3 Difference in duration of leading and trailing silence between reference and degraded speech**

[ITU-T P.862] uses the RMS level of the reference and degraded signals for level alignment. If long pauses are included at the beginning and end of the degraded signal, then the level alignment process may be sub-optimal. This issue may become a problem if the reference and degraded signal durations differ by more than 20%<sup>11</sup>.

Additionally, [ITU-T P.862] does not take into account any distortion in the degraded signal occurring before the start or after the end of the active speech signal. This active speech signal is determined from the reference signal as the first and last points where the signal level goes above approximately 50 dB SPL.

## **9 Characteristics of signal insertion and capturing paths**

This clause describes the desired characteristics of the signal insertion and capturing paths in hardware measurements. The measurement circuitry and environment can affect [ITU-T P.862]

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<sup>10</sup> If an MNRU conforming to [ITU-T P.810] is being tested, care should be taken in the level equalization process to preserve the actual speech level excluding the noise added by the MNRU.

<sup>11</sup> Empirical observations suggest that [ITU-T P.862] results for EVRC [b-ANSI/TIA-127-A-2004] depend on the particular alignment of the coding frame boundaries with the input PCM data. The result may vary by up to 0.25 depending on where the frame boundaries fall. In the case of EVRC, the method of obtaining a stable result would be to measure each of the 80 possible alignments and average the results. Similar situations may be uncovered for other DSP processes.

results unless precautions are taken to control the factors involved. Noise and interference must be excluded from the insertion and capture paths as much as possible to ensure that they do not affect the results.

### **9.1 Influence of measurement circuits and test configuration in the insertion path**

If the possibility exists to use a well-defined interface like POTS or ISDN, then this is the preferred method and the test equipment should be calibrated to serve such interfaces with the recommended nominal signal levels.

If no such well-defined interface can be used, then the insertion point is often the handset port of an end device, which is a proprietary interface and the required input level is initially unknown. Although standards such as North America's TIA-810-A specify terminal characteristics between acoustic and network interfaces, they do not specify intermediate points such as the handset interface. Gain distribution and filtering will be specific to the vendor, or even the individual terminal. In some cases, these characteristics are configurable in the end device. When the handset port is used as an [ITU-T P.862] test port, the test engineer may intend to measure:

- 1) the performance of the end device and network together;
- 2) the performance of the end device by itself (connected to a reference network); or
- 3) the performance of the network itself, with minimal contribution from the end device.

In all cases, however, we want to eliminate contributions from the measurement set-up.

The test engineer should ensure that the active speech level applied to the proprietary interface is consistent with the desired network level and the dynamic range of the codec. This requires appropriate gain characterization between the insertion point and the interface point of the terminal and the network in both directions of transmission.

When applying speech signals to the proprietary interface, the test engineer should be aware of filtering (e.g., modified IRS and frequency equalization of the transducer) between the acoustic and network interfaces. Terminal vendors are free to implement any combination of acoustic, electronic or digital filtering on either side of the proprietary handset interface. P.862 test equipment may therefore see either complete, partial, or no filtering after the insertion point. To obtain a precise result, the configuration measured must include the filtering appropriate for the test case being observed. Likewise, the filtering applied to the reference signal must match the filtering applied in the end-to-end test circuit.

This paragraph explains an ideal technique that can be used in order to determine the input characteristic for the reference signal when there is the possibility that an input filter used in normal operation of the communication terminal device is not in the measurement path. An artificial mouth (for example that found on a head and torso simulator) should be used to inject the test signal acoustically into the terminal which should be connected to a far-end reference point (e.g., ISDN point). The acoustic level used should represent normal usage for the terminal device, and the background noise level should be below 35 dBA. This normal use may either reflect the usage of the internal microphone or a personal hands-free kit, and depends on the purpose of the scenario to be assessed. The artificial mouth should be calibrated, and the positioning of the terminal should be representative of normal usage. The electrical level and frequency value should be measured at a reference point in the network connection (e.g., the ISDN end point). The process should then be repeated (with the same test signal) using electrical input at the P.862 test injection point, using the equipment used during P.862 testing. The input signal should now be adjusted so that the electrical level and frequency value match those captured during acoustic injection. The technique described

here is the ideal method, and may be approximated for many situations<sup>12</sup>. If this technique is not used, it is recommended that the tester takes special notice of manufacturers' specifications for acoustic and electrical interfaces to the communication terminals.

## **9.2 Influence of measurement circuits and test configuration in the capture path**

Once the reference speech has passed through the system under test, it must be transferred from the capture point to P.862. This capture path can contribute noise and distortion which may affect the result. The capture path may be subject to difficulties such as ground loops, pickup from a.c. power conductors, or other common-mode signals that may be present. In-band pickup may bias the result. In addition, out-of-band noise at sufficiently high levels may alias into the measurement band where insufficient anti-aliasing filtering is used.

To minimize noise contributed by the insertion and capture paths, it is recommended that insertion and capture paths together should contribute less than  $-70$  dBoV, so that the resultant SNR becomes 40 dB and the objective quality measurement result is determined exclusively by the influence of the system under test.

In general, slowly varying sampling rates, time stretching or time compression of the transmitted signal may lead to too pessimistic scores due to improper time alignment.

If analogue transmission is involved, care must be taken that no excessive click drifts between the AD and DA converters occur. This may be the case with consumer equipment, especially if the hardware does not support the required sample rate and a software sample rate conversion by the driver of the sound card is involved.

## **10 Analysis of the results**

### **10.1 Averaging the measurement results**

As highlighted in clause 7.7, one should use at least two female and two male talkers in an objective measurement. Before computing the mean or other statistics, individual measurement results should first be transformed to the MOS-LQO domain (based on [ITU-T P.862.1]) and then averaged over talkers and speech samples. Since the algorithm defined in [ITU-T P.862] is non-linear, results from concatenated samples will not match the mean results from those samples tested individually.

As mentioned in clause 6.2, there are two types of P.862 application, which require different analysis approaches. In the first case, averaging over talkers and speech samples should be performed before continuing with the analysis. This analysis is suitable for controlled network simulations with exactly repeatable results. The case of live field network testing needs a per-sample quality evaluation due to uncontrolled time-varying transmission channels.

### **10.2 Reliability of the PESQ measurements' results**

A large number of databases have been used for P.862 testing, validation and calibration [ITU-T P.862.1]. As described in [ITU-T P.862] and [ITU-T P.862.1], the databases contained speech samples spoken by different talkers and genders, in different languages and representing speech degradations generated by simulated and live network conditions. In addition, the network conditions corresponded to fixed, wireless and VoIP applications. Details regarding the content of the test databases are presented in Appendix II.

It should be noted that the P.862/P.862.1 measurement results are 95% reliable, and exhibit a known and controlled accuracy, when the algorithm is used on the same type of applications as the

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<sup>12</sup> This approximation is measurement-equipment specific. One possible method is described in Appendix IV.

ones on which the algorithm has been trained, tested and validated. In other words, the measurement scenarios need to represent statistically the same type of sample population as the ones on which P.862/P.862.1 has been trained, tested, validated and calibrated in order for the determined accuracy values to remain valid. The results' reliability and accuracy become unknown and uncontrolled once the algorithm is used to evaluate speech quality on new types of technologies and/or using other types of codecs and/or new live networks.

### 10.3 Accuracy values of the PESQ measurements

Three statistical metrics, the correlation coefficient, the prediction error and the residual error distribution, have been used to evaluate P.862/P.862.1 performance on the databases described in clause 10.2. As mentioned in clause 10.1, the analysis approaches differ depending on the application type, controlled network simulations and live/field network testing conditions.

For all simulated network conditions, averages per-condition over at least four talkers, two male and two female, have been used to calculate the statistical metrics. For live network databases, the statistical metrics have been calculated using per-samples objective and subjective scores.

The performance results are presented in Tables 1 and 2. The 95% confidence critical limits for the correlation coefficient and the prediction error are also calculated in order to provide the 95% lower correlation bound and 95% upper prediction error bound.

The results are presented per application type (e.g., simulated wireless and VoIP network conditions and real-life wireless and VoIP network conditions). These accuracy values express therefore PESQ algorithm's performance if used in any of the applications mentioned in clause 10.2.

**Table 1 – Confidence intervals for correlations coefficient and prediction error**

Application	N	Metric	P.862 (raw PESQ)	P.862.1 (calibrated PESQ)
Simulation data (wireless, VoIP and fixed applications)	1357	R	0.956	0.956
		CI95%-lower limit	0.940	0.940
		PE	N/A	N/A
		CI95%-upper limit	N/A	N/A
Field-collected data (Wireless applications: GSM US and EU, CDMA-US, TDMA-US, iDEN-US, AMPS-US; and VoIP application)	1135	R	0.925	0.926
		CI95%-lower limit	0.916	0.917
		PE	0.479	0.462
		CI95%-upper limit	0.492	0.475

**Table 2 – Residual error distribution**

Application	MOS bins	<0.25	<0.5	<0.75	<1	<1.25	<1.5	<1.75	<2
Field-collected data (Wireless applications: GSM US and EU, CDMA-US, TDMA-US, iDEN-US, AMPS-US; and VoIP application)	P.862 CDF (%)	32.51	66.52	90.84	97.97	99.38	99.91	99.91	100
	P.862 prob (%)	32.51	34.09	24.32	7.14	1.41	0.53	0	0
	P.862.1 CDF (%)	40.44	70.48	90.33	97.71	99.3	99.7	99.91	100
	P.862.1 Prob (%)	40.44	30.04	19.82	7.4	1.59	0.44	0.18	0.09

#### 10.4 Interpretation of the accuracy's results

By definition, the P.862/P.862.1 algorithm is an estimator of the subjective opinion on the speech quality provided by the network under test. It should be noted therefore that any speech quality measurement performed by the PESQ algorithm is affected by the accuracy values presented in Tables 1 and 2.

It should be noted that, as mentioned in clause 10.2, the accuracy values remain valid as long as the measurement scenarios represent statistically the same sample population as the ones presented in clause 10.2.

The lower bound of the 95% confidence interval of the correlation coefficient shows that P.862/P.862.1 measurements are expected to exhibit a correlation with the subjective opinion, which is higher or at least equal to the lower limit of the correlation coefficient 95% confidence interval, regardless of whether simulated or live field network conditions are used, and regardless of the tested network type (such as wireless, VoIP and fixed) (Table 1).

The residual error distribution (Table 2) represents the cumulative density function (CDF) of the absolute errors between MOS and P.862/P.862.1 scores and it shows the probability that the absolute error is lower than a value. For example, the probability that the absolute error is lower than 0.5 MOS is higher than 70%, while the probability that the error is lower than 0.75 MOS is higher than 90%. Table 2 also provides the probability density function (PDF) of the absolute error. As expected, in agreement with the CDF, the PDF shows that lower absolute errors have a higher likelihood of occurrence than higher values.

#### 11 Report of results

As mentioned in clause 10.4, depending on the application type, i.e., simulated or live network conditions, the P.862/P.862.1 measurements should be reported based on the algorithm's accuracy presented in clause 10.3 (Tables 1 and 2).

The correlation coefficient is recommended to be used as an informative statistical metric on the P.862/P.862.1 performance for a specified application. The prediction error along with the residual error distribution is recommended to be used to report P.862.1 measurement results for a specified application.

Generally, average, maximum and minimum PESQ values should be reported, as well as the number of measurements used to calculate the average. Some detailed recommendations for reporting PESQ measurements are presented in Appendix III. In addition, the number of measurements achieving a given PESQ score or range of scores can be presented graphically as a frequency distribution. In cases where the system under test delivers relatively stable listening quality, the standard deviation can be used to help decide whether further measurements are

necessary to achieve a specified accuracy. This approach is not valid for highly time varying systems under test (e.g., VoIP or mobile networks).

## **12 Guidance for using P.862.2 wideband extension to P.862**

In principle, the guidance provided in this Recommendation is applicable to both [ITU-T P.862] and its wideband extension P.862.2. Nevertheless, some specific guidance is necessary for the wideband extension to [ITU-T P.862].

The foregoing guidance is mainly referring to the usage of the IRS send characteristic to be applied on the input or reference signal. For the wideband extension, no filtering of either the speech signal or any environmental noise is recommended. This is referred to in clauses 6.2, 7.9 and 7.10.

Regarding speech activity level calculation according to [ITU-T P.56], it is recommended to use the P.56 wideband option. This is referred to in clauses 7.2, 7.4, 8.1 and 8.2.

The proposed insertion of a low noise floor in clause 7.10 is not evaluated for the wideband extension and cannot be recommended.

Clauses 10 and 11 describe the accuracy of the P.862 method. The figures provided are only applicable to the narrow-band P.862.

Within clause 3.7, the 0 dBm0 is described according to ITU-T Recommendation G.711. It has to be stated that this G.711 reference is only available for narrow-band applications.

Both methods, P.862.1 and P.862.2, refer to a MOS-LQO scale as defined in ITU-T Recommendation P.800.1. It has to be taken into account that the term MOS-LQO might be extended by a qualifier relating to the narrow-band and wideband cases in the future. Please note that the results produced by [ITU-T P.862.1] are related to a narrow-band-only context. [ITU-T P.862.2] results apply to wideband or mixed wideband and narrow-band applications. As a result, direct comparisons of the P.862.1 and P.862.2 MOS-LQO results are not possible.

Studies published within ITU-T show that the application of P.862.2 to narrow-band conditions can lead to unexpected results. Test samples might be scored significantly lower than expected by applying P.862.2 to these narrow-band samples compared to scores derived by P.862.1 or human perception. For that reason, it is not recommended to use P.862.2 for narrow-band conditions. The reasons for this behaviour are still under study. It might be caused by dependencies on the talker's characteristic and/or the behaviour of the transmission error in the particular condition.

## **13 Use of P.862.1 and P.862.2 for EVRC type of codecs and evaluation of CDMA networks**

A series of studies was made discussing the prediction performance of P.862.1 and P.862.2 for EVRC-A, EVRC-B and EVRC-WB in relation to subjective scores and the behaviour of AMR-type speech codecs with comparable perceived quality.

It has to be stated that there is a systematic under-prediction of EVRC family codecs by P.862.1/P.862.2 compared to subjective tests and even more to AMR-type codecs. The systematic noticeable under-prediction can be observed for a wide range of bit rates and error patterns [b-ITU-T COM12-C121].

According to formal 3GPP2 MOS tests for EVRC, the perceived EVRC quality is statistically equivalent to AMR at 12.2 kbit/s. In the experiment reported in [b-ITU-T COM12-C121] the subjective MOS for EVRC is 0.08 MOS lower than for AMR at 12.2 kbit/s. However, the P.862.1 score for EVRC is 0.32 MOS lower than the P.862.1 score for AMR at 12.2 kbit/s. The observed under-prediction in this experiment is higher in case of lower bit rates for EVRC-B [b-ITU-T COM12-C121].

Applying P.862.2 to the wideband version EVRC-WB leads to a more pronounced under-prediction compared to the narrow-band version EVRC-B. Note that the relative quality amongst the different EVRC test conditions can be reproduced widely.

Conclusions:

- 1) The direct comparison of P.862.1/P.862.2 scores obtained with AMR-type codecs or other ITU-T speech codecs with the EVRC family of codecs is not recommended. This includes the benchmarking between GSM/UMTS networks and CDMA networks which are usually equipped with EVRC family codecs.
- 2) The comparison of different conditions (e.g., bit rates, error patterns) using EVRC is possible by P.862.1/P.862.2 due to the correct relative ranking of the quality scores within those conditions. Consequently, P.862.1/P.862.2 might be usable for benchmarking of CDMA networks to each other or for optimization efforts within those networks if the same codec is involved. The direct comparison of P.862.1/P.862.2 scores with P.800 subjective listening scores is not appropriate for the EVRC family codecs.

#### **14 Comparing objective with subjective score**

It has to be noted that the mapping function proposed in P.862.1 predicts on a MOS scale. This mapping function converts the raw P.862 scores to MOS-LQO values. However, the applied mapping is derived as an average function across a huge amount of subjectively scored data in different contexts and languages. The mapping function is not supposed to predict the absolute MOS of a single experiment. There might be offsets or different gradients. However, the qualitative rank-order can be reproduced objectively with the given accuracy.

It is known that subjective MOS scales differ from experiment to experiment, depending on the context of the experiment. This context includes, e.g., the user's expectation and culture, used equipment and, most importantly, the quality range included in the experiment.

Naturally, objective quality metrics do not show such behaviour. The need therefore arises to compensate for such systematic differences before comparing subjective and objective scores. One method to do this is to apply a third order, monotonic polynomial to the objective scores which minimizes the root mean square error (RMSE) or maximizes the correlation between the two data sets. The coefficients for this polynomial (the "mapping function") have to be calculated separately for each experiment. Doing this in the exact and correct way requires some special numerical tools which are not easily available.

For the general case, however, it is sufficient to draw a scatter-plot type of chart by an appropriate analysis tool, add a third order regression line and read the correlation given for the regression line. The biggest risk in this case is that the regression line is not monotonic, but in most cases this can be checked visually.

## Appendix I

### Reference values for objective quality derived by ITU-T Recommendation P.862 for ITU-T/GSM standard codecs

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

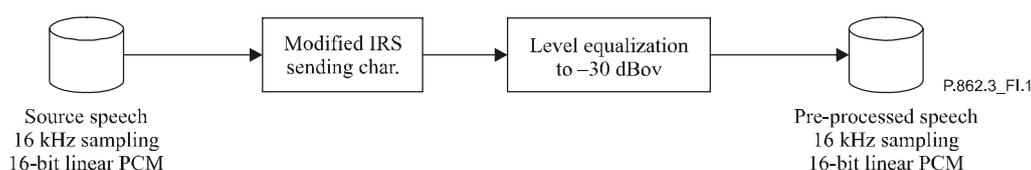
#### I.1 Reference value sources

ITU-T Recommendation P.862.1 reference values were calculated for the following codec/MNRU conditions by using the speech database in Annex B of [ITU-T P.501]:

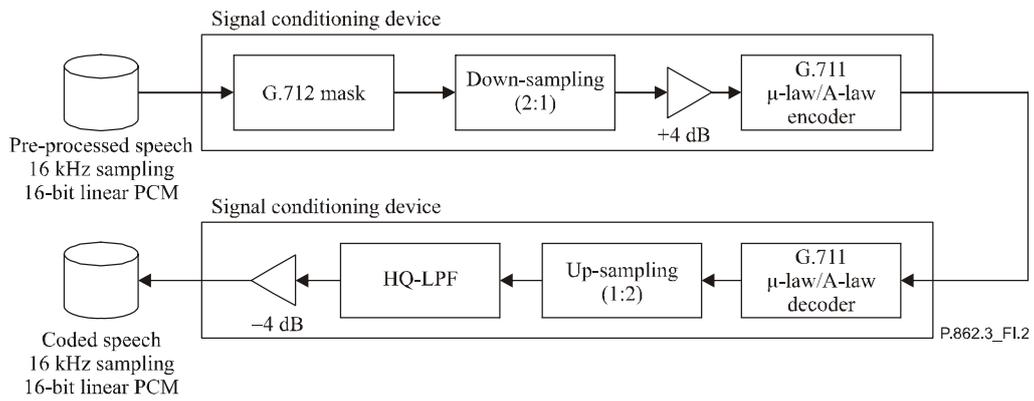
- G.711  $\mu$ -law, A-law;
- G.726 at 16, 24, 32 and 40 kbit/s;
- G.728;
- G.729;
- Annex A/G.729;
- G.723.1 at 5.3 and 6.3 kbit/s;
- GSM-AMR at 4.75, 5.15, 5.9, 6.7, 7.4, 7.95, 10.2 and 12.2 kbit/s;
- GSM-EFR;
- GSM-FR;
- GSM-HR;
- MNRU (Q = 5, 10, 15, 20, 25, 30, 35, 40 and 45 dB).

Figure I.1 shows the pre-processing of the reference speech signals. Figures I.2, I.3, and I.4 show the coding procedures of G.711, G.726 and the other codecs, respectively. Figure I.5 shows the processing procedure of MNRU. Tables I.1, I.2, and I.3 show reference values, which were derived by transforming raw PESQ values to MOS-LQO by using [ITU-T P.862.1], for ITU-T standardized codecs and MNRU. Signal processing of G.711, G.726 and MNRU was done using the software tool provided in ITU-T Recommendation G.191.

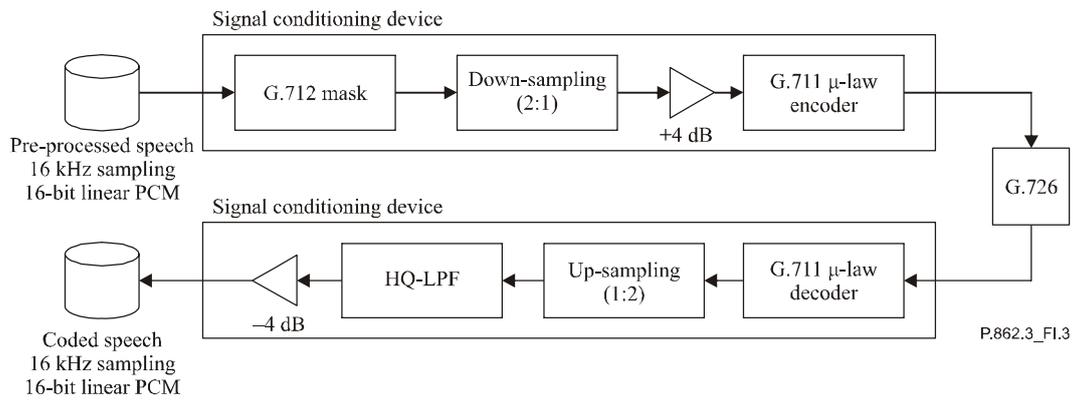
NOTE – In GSM-AMR, GSM-EFR, GSM-FR, GSM-HR and G.711 A-law codecs, raw PESQ values were calculated at a sampling rate of 16 kHz.



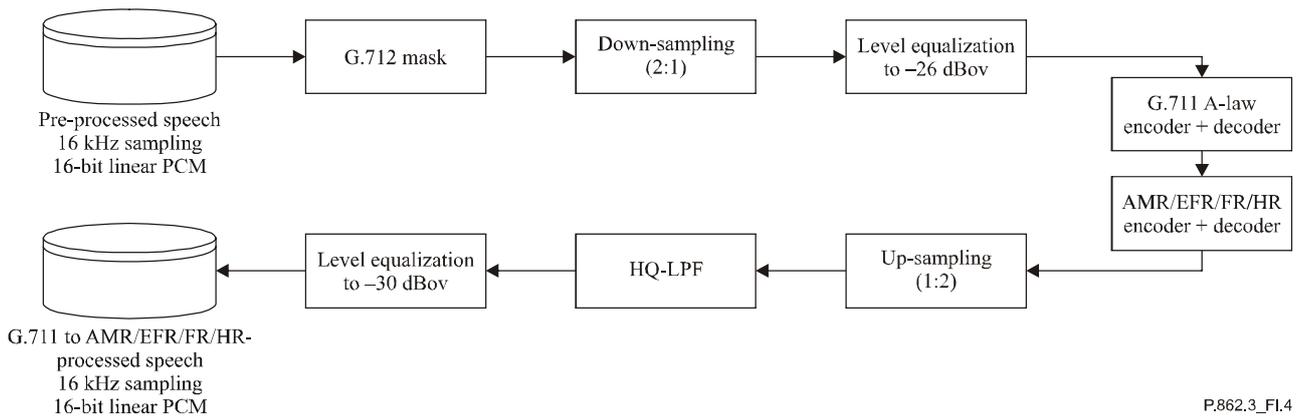
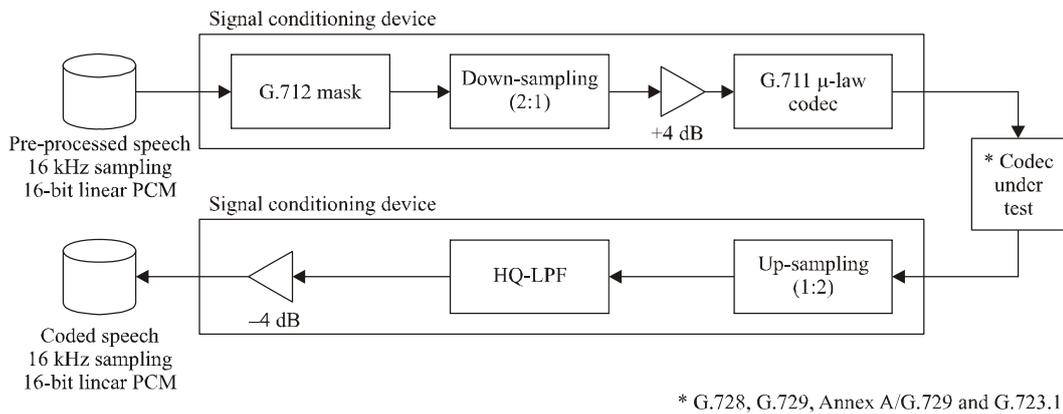
**Figure I.1 – Pre-processing of source speech**



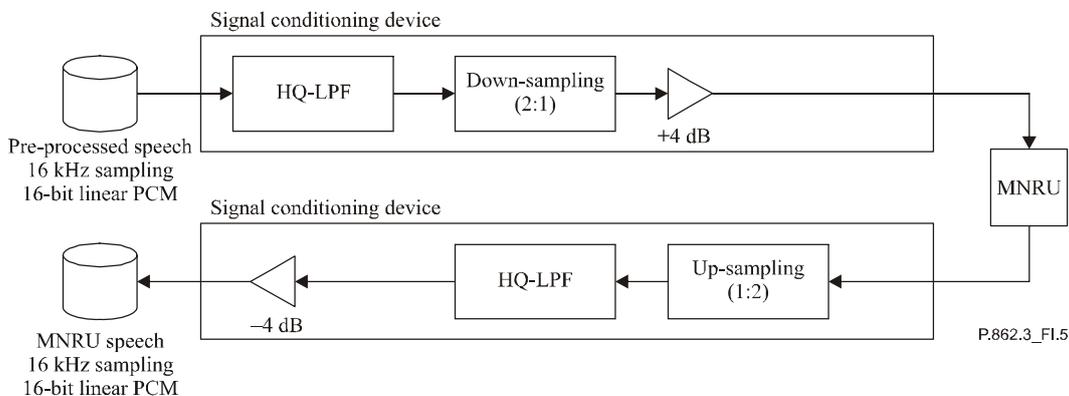
**Figure I.2 – Processing of G.711**



**Figure I.3 – Processing of G.726**



**Figure I.4 – Processing of G.728, G.729, Annex A/G.729 and G.723.1**



**Figure I.5 – Processing of MNRU**

The signal processing procedure in Figures I.1 through I.5 is described below.

## I.2 Pre-processing of source speech

In the pre-processing part, the source speech goes through a transmission-side modified-IRS filter and its speech level is equalized to  $-30$  dBov as illustrated in Figure I.1. Pre-processed speech files are input signals to a signal conditioning device. Using the software tool provided in ITU-T Recommendation G.191, which is called STL2000 hereafter, pre-processed speech can be made by the following commands:

```
$ filter -q mod IRS16 file.inp file.irs
$ sv56demo -q file.irs file.pre 256 1 0 -30
```

### I.3 Processing of G.711

Pre-processed speech is masked according to ITU-T Recommendation G.712 and down-sampled. Down-sampled speech is equalized in level to  $-26$  dBov and its output becomes the input signal to the G.711 encoder. Processing commands are given below:

```
$ filter -q -down PCM file.pre tmpfile1
$ sv56demo -q tmpfile1 g711.inp 256 1 0 -26 8000
```

ITU-T Recommendation G.711  $\mu$ -law encoding and decoding processing is carried out using the following STL2000 command:

```
$ g711demo u lili g711.inp g711.dec
```

Decoded speech is up-sampled from 8 to 16 kHz and its level is equalized to  $-30$  dBov. The G.711 coded speech can be obtained by the following STL2000 commands:

```
$ filter -q -up HQ2 g711.dec tmpfile2
$ scaledemo -q -gain 0.63095 tmpfile2 g711.out
```

### I.4 Processing of G.726

In G.726 processing, the G.711  $\mu$ -law encoder's output becomes the input of the G.726 coder and its output is input to the G.711  $\mu$ -law decoder. G.726-coded speech can be obtained by the following STL2000 commands:

```
$ filter -q -down PCM infile tmpfile1
$ sv56demo -q tmpfile1 tmpfile2 256 1 0 -26
$ g711demo u lilo tmpfile2 tmpfile3
$ g726demo u lolo {40|32|24|16} tmpfile3 tmpfile4
$ g711demo u loli tmpfile4 tmpfile5
$ filter -q -up HQ2 tmpfile5 tmpfile6
$ scaledemo -q -gain 0.63095 tmpfile6 outfile
```

### I.5 Processing of G.728, G.729, Annex A/G.729 and G.723.1

In these codecs, the input signal to the codec under test is the output of a G.711  $\mu$ -law codec, i.e., *g711.dec* as mentioned in Figure I.2. When the outputs of the codec under test are denoted by {*g728|g729|g729A|G7231*}.dec, coded speech is obtained by the following STL2000 commands:

```
$ filter -q -up HQ2 {g728|g729|g729A|g7231}.dec tmpfile2
$ scaledemo -q -gain 0.63095 tmpfile2 {g728|g729|g729A|g7231}.out
```

### I.6 Processing of MNRU

Pre-processed speech is down-sampled without changing the frequency response of the input signal, its level is equalized to  $-26$  dBov, and its output becomes the input signal to MNRU. The output of MNRU is up-sampled from 8 to 16 kHz and its level is equalized to  $-30$  dBov.  $Q$ -dB MNRU speech can be obtained by the following STL2000 commands:

```
$ filter -q -down HQ2 infile tmpfile1
$ sv56demo -q tmpfile1 tmpfile2 256 1 0 -26 8000
$ mnrdemo tmpfile2 tmpfile3 128 1 0 Q
$ filter -q -up HQ2 tmpfile3 tmpfile4
$ scaldemo -q -gain 0.63095 tmpfile3 mnruQ.out
```

**Table I.1 – The P.862.1 reference values for ITU-T standardized codecs**

Language	File name	G.711		G.726				G.728	G.729	G.729A	G.723.1	
		$\mu$ -law	A-law	16 kbit/s	24 kbit/s	32 kbit/s	40 kbit/s				5.3 kbit/s	6.3 kbit/s
American English	Female 1 (0.00-7.97 s).wav	4.46	4.28	2.50	3.34	3.89	4.18	3.95	3.95	3.80	3.65	3.81
	Female 2 (0.00-8.06 s).wav	4.45	4.42	3.12	3.76	4.07	4.33	4.27	4.08	3.99	3.67	3.80
	Male 1 (0.00-8.44 s).wav	4.49	4.36	2.86	3.82	4.25	4.35	4.19	4.17	4.13	3.90	3.97
	Male 2 (0.00-7.96 s).wav	4.47	4.31	2.97	3.80	4.21	4.40	4.22	4.15	4.06	3.95	4.07
	<b>Average</b>	<b>4.47</b>	<b>4.34</b>	<b>2.86</b>	<b>3.69</b>	<b>4.11</b>	<b>4.32</b>	<b>4.16</b>	<b>4.09</b>	<b>4.00</b>	<b>3.80</b>	<b>3.80</b>
Chinese	Female 1 (0.00-10.87 s).wav	4.46	4.48	2.38	3.42	4.21	4.34	3.86	3.72	3.65	3.10	3.33
	Female 1b (0.00-13.39 s).wav	4.42	4.43	2.29	3.34	4.07	4.26	3.98	3.80	3.75	3.26	3.49
	Female 2 (0.00-13.32 s).wav	4.50	4.50	2.26	3.29	4.02	4.37	4.06	3.88	3.75	3.33	3.53
	Female 2b (0.00-13.39 s).wav	4.50	4.50	2.38	3.30	4.03	4.33	4.03	3.87	3.72	3.39	3.59
	Male 1 (0.00-12.15 s).wav	4.44	4.48	2.64	3.58	4.23	4.28	4.19	3.83	3.75	3.36	3.51
	Male 1a (0.00-12.91 s).wav	4.52	4.51	2.84	3.77	4.21	4.37	4.18	4.06	4.00	3.65	3.89
	Male 2 (0.00-12.50 s).wav	4.49	4.48	2.74	3.74	4.30	4.44	4.18	3.99	3.89	3.62	3.78
	Male 2b (0.00-12.82 s).wav	4.50	4.40	2.89	3.90	4.29	4.43	4.16	3.95	3.89	3.35	3.55
	<b>Average</b>	<b>4.48</b>	<b>4.47</b>	<b>2.55</b>	<b>3.55</b>	<b>4.18</b>	<b>4.35</b>	<b>4.08</b>	<b>3.89</b>	<b>3.80</b>	<b>3.38</b>	<b>3.59</b>
English	Female 1 (0.00-8.00 s).wav	4.50	4.49	2.69	3.37	3.88	4.21	3.89	3.72	3.58	3.42	3.59
	Female 2 (0.00-8.00 s).wav	4.48	4.46	2.81	3.44	3.92	4.24	4.00	3.91	3.80	3.60	3.67
	Male 1 (0.00-8.00 s).wav	4.50	4.45	3.03	3.53	3.96	3.99	4.08	3.88	3.88	3.67	3.81
	Male 2 (0.00-8.00 s).wav	4.51	4.48	2.94	3.79	4.24	4.34	4.05	3.73	3.60	3.64	3.83
	<b>Average</b>	<b>4.50</b>	<b>4.47</b>	<b>2.87</b>	<b>3.54</b>	<b>4.01</b>	<b>4.20</b>	<b>4.01</b>	<b>3.81</b>	<b>3.72</b>	<b>3.59</b>	<b>3.73</b>
French	Female 1 (0.00-10.04 s).wav	4.50	4.47	3.06	3.84	4.28	4.42	4.21	3.85	3.77	3.59	3.69
	Female 2 (0.00-10.04 s).wav	4.51	4.48	2.76	3.64	4.15	4.41	4.03	3.77	3.64	3.39	3.56
	Male 1 (0.00-12.18 s).wav	4.50	4.46	3.09	3.79	4.18	4.32	4.09	3.84	3.82	3.45	3.60
	Male 2 (0.00-10.04 s).wav	4.52	4.48	3.33	3.92	4.28	4.40	4.25	4.00	3.91	3.70	3.88
	<b>Average</b>	<b>4.51</b>	<b>4.47</b>	<b>3.06</b>	<b>3.80</b>	<b>4.22</b>	<b>4.39</b>	<b>4.15</b>	<b>3.87</b>	<b>3.79</b>	<b>3.54</b>	<b>3.69</b>
German	Female 1 (0.00-8.00 s).wav	4.49	4.48	2.68	3.46	4.02	4.27	4.04	3.86	3.69	3.54	3.75
	Female 2 (0.00-8.00 s).wav	4.48	4.46	2.84	3.65	4.24	4.40	4.13	4.07	3.89	3.61	3.82
	Male 1 (0.00-8.00 s).wav	4.50	4.47	2.99	3.72	4.27	4.41	4.09	3.95	3.87	3.56	3.84
	Male 2 (0.00-8.00 s).wav	4.50	4.46	2.86	3.47	4.03	4.35	4.12	4.07	4.01	3.75	3.91
	<b>Average</b>	<b>4.50</b>	<b>4.47</b>	<b>2.84</b>	<b>3.58</b>	<b>4.14</b>	<b>4.36</b>	<b>4.09</b>	<b>3.99</b>	<b>3.87</b>	<b>3.62</b>	<b>3.83</b>
Italian	Female 1 (0.00-20.60 s).wav	4.49	4.40	2.38	3.23	3.81	4.25	3.80	3.75	3.62	3.28	3.49
	Female 2 (0.00-21.78 s).wav	4.48	4.39	2.72	3.68	4.14	4.34	4.16	3.95	3.87	3.57	3.75
	Male 1 (0.00-18.13 s).wav	4.50	4.44	2.61	3.50	4.01	4.33	4.12	3.88	3.81	3.59	3.75
	Male 2 (0.00-20.86 s).wav	4.51	4.43	3.05	3.94	4.28	4.41	4.18	4.12	4.05	3.73	3.95
	<b>Average</b>	<b>4.49</b>	<b>4.41</b>	<b>2.69</b>	<b>3.60</b>	<b>4.07</b>	<b>4.33</b>	<b>4.07</b>	<b>3.93</b>	<b>3.84</b>	<b>3.55</b>	<b>3.74</b>
Japanese	Female 1 (0.00-7.60 s).wav	4.46	4.36	2.22	2.97	3.65	4.11	3.76	3.70	3.61	3.25	3.40
	Female 2 (0.00-7.31 s).wav	4.48	4.38	2.45	3.41	4.12	4.40	3.96	3.82	3.73	3.34	3.59
	Male 1 (0.00-7.13 s).wav	4.47	4.39	2.36	3.06	3.61	4.20	3.89	3.88	3.74	3.38	3.53
	Male 2 (0.00-7.45 s).wav	4.49	4.42	2.95	3.90	4.32	4.45	4.31	4.08	4.00	3.83	3.93
	<b>Average</b>	<b>4.47</b>	<b>4.39</b>	<b>2.49</b>	<b>3.35</b>	<b>3.95</b>	<b>4.30</b>	<b>4.00</b>	<b>3.87</b>	<b>3.78</b>	<b>3.46</b>	<b>3.62</b>
Spanish (US)	Female 1 (0.00-8.00 s).wav	4.47	4.40	2.33	3.05	3.73	4.17	4.02	3.84	3.71	3.42	3.64
	Female 2 (0.00-8.00 s).wav	4.40	4.31	2.30	2.92	3.48	4.04	3.77	3.65	3.43	3.07	3.22
	Male 1 (0.00-8.00 s).wav	4.46	4.30	2.86	3.64	4.19	4.36	4.05	3.83	3.76	3.69	3.77
	Male 2 (0.00-8.00 s).wav	4.49	4.42	2.76	3.72	4.22	4.40	4.09	3.86	3.85	3.60	3.72
	<b>Average</b>	<b>4.46</b>	<b>4.36</b>	<b>2.56</b>	<b>3.34</b>	<b>3.93</b>	<b>4.25</b>	<b>3.99</b>	<b>3.80</b>	<b>3.69</b>	<b>3.45</b>	<b>3.60</b>

NOTE – Grey cells indicate samples that do not meet [ITU-T P.501] requirements.

**Table I.2 – The P.862.1 reference values for MNRU**

Language	File name	MNRU								
		5 dB	10 dB	15 dB	20 dB	25 dB	30 dB	35 dB	40 dB	45 dB
<b>American English</b>	Female 1 (0.00-7.97 s).wav	1.80	2.39	3.03	3.68	4.15	4.37	4.47	4.50	4.52
	Female 2 (0.00-8.06 s).wav	2.09	2.67	3.24	3.80	4.16	4.35	4.41	4.44	4.44
	Male 1 (0.00-8.44 s).wav	1.93	2.58	3.30	4.01	4.32	4.46	4.51	4.52	4.53
	Male 2 (0.00-7.96 s).wav	1.99	2.65	3.34	3.90	4.23	4.39	4.45	4.47	4.48
	<b>Average</b>	<b>1.95</b>	<b>2.57</b>	<b>3.23</b>	<b>3.85</b>	<b>4.22</b>	<b>4.39</b>	<b>4.46</b>	<b>4.48</b>	<b>4.49</b>
<b>Chinese</b>	Female 1 (0.00-10.87 s).wav	1.41	1.84	2.45	3.19	3.83	4.20	4.40	4.49	4.52
	Female 1b (0.00-13.39 s).wav	1.35	1.74	2.36	3.07	3.66	4.02	4.27	4.36	4.38
	Female 2 (0.00-13.32 s).wav	1.46	1.91	2.54	3.33	3.99	4.33	4.46	4.51	4.54
	Female 2b (0.00-13.39 s).wav	1.52	2.02	2.72	3.51	4.10	4.35	4.42	4.44	4.45
	Male 1 (0.00-12.15 s).wav	1.72	2.29	3.02	3.71	4.16	4.33	4.37	4.38	4.38
	Male 1a (0.00-12.91 s).wav	1.78	2.44	3.22	3.91	4.26	4.43	4.51	4.54	4.54
	Male 2 (0.00-12.50 s).wav	1.67	2.22	3.02	3.78	4.19	4.36	4.46	4.51	4.52
	Male 2b (0.00-12.82 s).wav	1.81	2.46	3.25	3.91	4.29	4.42	4.46	4.47	4.47
	<b>Average</b>	<b>1.57</b>	<b>2.10</b>	<b>2.82</b>	<b>3.56</b>	<b>4.08</b>	<b>4.31</b>	<b>4.42</b>	<b>4.47</b>	<b>4.48</b>
<b>English</b>	Female 1 (0.00-8.00 s).wav	1.97	2.60	3.27	3.86	4.22	4.41	4.49	4.52	4.53
	Female 2 (0.00-8.00 s).wav	1.90	2.51	3.16	3.64	4.06	4.34	4.45	4.50	4.52
	Male 1 (0.00-8.00 s).wav	2.41	3.06	3.65	4.11	4.38	4.47	4.51	4.52	4.53
	Male 2 (0.00-8.00 s).wav	1.90	2.55	3.32	3.90	4.27	4.45	4.51	4.54	4.54
	<b>Average</b>	<b>2.03</b>	<b>2.68</b>	<b>3.35</b>	<b>3.89</b>	<b>4.24</b>	<b>4.42</b>	<b>4.49</b>	<b>4.52</b>	<b>4.53</b>
<b>French</b>	Female 1 (0.00-10.04 s).wav	1.93	2.55	3.22	3.82	4.23	4.42	4.50	4.53	4.54
	Female 2 (0.00-10.04 s).wav	1.74	2.31	3.02	3.70	4.15	4.40	4.50	4.53	4.54
	Male 1 (0.00-12.18 s).wav	2.07	2.76	3.47	3.96	4.23	4.41	4.49	4.53	4.54
	Male 2 (0.00-10.04 s).wav	2.36	3.14	3.88	4.30	4.46	4.51	4.53	4.54	4.54
	<b>Average</b>	<b>2.01</b>	<b>2.68</b>	<b>3.41</b>	<b>3.96</b>	<b>4.27</b>	<b>4.44</b>	<b>4.51</b>	<b>4.53</b>	<b>4.54</b>
<b>German</b>	Female 1 (0.00-8.00 s).wav	1.70	2.28	3.00	3.65	4.19	4.41	4.49	4.53	4.54
	Female 2 (0.00-8.00 s).wav	1.74	2.26	2.94	3.64	4.13	4.33	4.47	4.51	4.53
	Male 1 (0.00-8.00 s).wav	1.88	2.47	3.17	3.75	4.20	4.42	4.51	4.53	4.54
	Male 2 (0.00-8.00 s).wav	2.09	2.71	3.40	3.97	4.33	4.46	4.51	4.53	4.54
	<b>Average</b>	<b>1.84</b>	<b>2.43</b>	<b>3.13</b>	<b>3.76</b>	<b>4.21</b>	<b>4.41</b>	<b>4.49</b>	<b>4.53</b>	<b>4.54</b>
<b>Italian</b>	Female 1 (0.00-20.60 s).wav	1.52	2.01	2.69	3.44	4.01	4.33	4.46	4.51	4.53
	Female 2 (0.00-21.78 s).wav	1.73	2.25	2.92	3.58	4.12	4.38	4.47	4.51	4.53
	Male 1 (0.00-18.13 s).wav	1.79	2.43	3.24	3.94	4.33	4.46	4.51	4.53	4.54
	Male 2 (0.00-20.86 s).wav	2.09	2.86	3.65	4.16	4.40	4.49	4.52	4.53	4.54
	<b>Average</b>	<b>1.76</b>	<b>2.37</b>	<b>3.13</b>	<b>3.80</b>	<b>4.23</b>	<b>4.42</b>	<b>4.49</b>	<b>4.52</b>	<b>4.53</b>
<b>Japanese</b>	Female 1 (0.00-7.60 s).wav	1.49	1.94	2.57	3.32	3.92	4.27	4.43	4.50	4.52
	Female 2 (0.00-7.31 s).wav	1.48	1.91	2.51	3.27	3.99	4.35	4.48	4.52	4.53
	Male 1 (0.00-7.13 s).wav	1.55	2.02	2.65	3.39	4.05	4.36	4.47	4.52	4.53
	Male 2 (0.00-7.45 s).wav	1.74	2.27	3.01	3.77	4.24	4.45	4.51	4.53	4.54
	<b>Average</b>	<b>1.56</b>	<b>2.03</b>	<b>2.68</b>	<b>3.44</b>	<b>4.05</b>	<b>4.36</b>	<b>4.47</b>	<b>4.52</b>	<b>4.53</b>
<b>Spanish (US)</b>	Female 1 (0.00-8.00 s).wav	1.49	1.95	2.51	3.16	3.81	4.24	4.43	4.50	4.53
	Female 2 (0.00-8.00 s).wav	1.61	2.08	2.62	3.19	3.81	4.19	4.40	4.49	4.52
	Male 1 (0.00-8.00 s).wav	2.02	2.70	3.53	4.14	4.39	4.48	4.51	4.52	4.53
	Male 2 (0.00-8.00 s).wav	1.65	2.24	3.04	3.73	4.23	4.44	4.51	4.53	4.54
	<b>Average</b>	<b>1.67</b>	<b>2.23</b>	<b>2.92</b>	<b>3.59</b>	<b>4.08</b>	<b>4.35</b>	<b>4.46</b>	<b>4.51</b>	<b>4.53</b>

NOTE – Grey cells indicate samples that do not meet [ITU-T P.501] requirements.

**Table I.3 – The P.862 reference values for GSM standardized codecs**

Language	File name	AMR								FR	HR	EFR
		12.2 kbit/s	10.2 kbit/s	7.95 kbit/s	7.4 kbit/s	6.7 kbit/s	5.9 kbit/s	5.15 kbit/s	4.75 kbit/s			
American English	Female 1 (0.00-7.97 s).wav	3.87	3.75	3.60	3.61	3.52	3.43	3.33	3.18	3.94	3.03	3.20
	Female 2 (0.00-8.06 s).wav	4.13	4.07	3.96	3.92	3.79	3.72	3.62	3.49	4.08	3.70	3.54
	Male 1 (0.00-8.44 s).wav	4.10	4.03	3.94	3.97	3.88	3.84	3.64	3.50	4.19	3.69	3.46
	Male 2 (0.00-7.96 s).wav	4.10	4.06	4.03	4.01	3.91	3.85	3.73	3.70	4.20	3.59	3.72
	<b>Average</b>	<b>4.05</b>	<b>3.98</b>	<b>3.88</b>	<b>3.88</b>	<b>3.78</b>	<b>3.71</b>	<b>3.58</b>	<b>3.47</b>	<b>4.10</b>	<b>3.51</b>	<b>3.48</b>
Chinese	Female 1 (0.00-10.87 s).wav	3.94	3.81	3.46	3.50	3.27	3.15	2.99	3.00	3.98	3.18	2.94
	Female 1b (0.00-13.39 s).wav	3.97	3.82	3.52	3.45	3.36	3.22	3.00	2.95	4.04	3.02	2.95
	Female 2 (0.00-13.32 s).wav	4.08	3.99	3.71	3.62	3.53	3.38	3.21	3.07	4.12	3.07	3.01
	Female 2b (0.00-13.39 s).wav	4.08	4.01	3.74	3.74	3.62	3.40	3.21	3.13	4.10	3.03	3.04
	Male 1 (0.00-12.15 s).wav	3.94	3.84	3.66	3.68	3.54	3.45	3.22	3.22	4.06	3.50	3.26
	Male 1a (0.00-12.91 s).wav	4.23	4.14	3.95	3.92	3.78	3.65	3.34	3.35	4.20	3.65	3.36
	Male 2 (0.00-12.50 s).wav	4.07	3.93	3.78	3.77	3.62	3.46	3.22	3.21	4.15	3.72	3.30
	Male 2b (0.00-12.82 s).wav	4.16	4.15	3.94	3.94	3.80	3.64	3.40	3.27	4.23	3.64	3.44
	<b>Average</b>	<b>4.06</b>	<b>3.96</b>	<b>3.72</b>	<b>3.70</b>	<b>3.57</b>	<b>3.42</b>	<b>3.20</b>	<b>3.15</b>	<b>4.11</b>	<b>3.35</b>	<b>3.16</b>
English	Female 1 (0.00-8.00 s).wav	4.00	3.82	3.63	3.62	3.42	3.31	3.21	3.11	3.99	3.27	3.07
	Female 2 (0.00-8.00 s).wav	3.81	3.78	3.65	3.62	3.56	3.49	3.36	3.34	3.78	3.31	3.28
	Male 1 (0.00-8.00 s).wav	4.01	3.88	3.75	3.67	3.46	3.57	3.23	3.01	4.03	3.54	3.37
	Male 2 (0.00-8.00 s).wav	4.06	3.83	3.73	3.68	3.53	3.48	3.18	2.98	4.10	3.75	3.49
	<b>Average</b>	<b>3.97</b>	<b>3.83</b>	<b>3.69</b>	<b>3.65</b>	<b>3.49</b>	<b>3.46</b>	<b>3.25</b>	<b>3.11</b>	<b>3.98</b>	<b>3.47</b>	<b>3.30</b>
French	Female 1 (0.00-10.04 s).wav	4.11	4.01	3.79	3.83	3.66	3.42	3.33	3.30	4.04	3.49	3.37
	Female 2 (0.00-10.04 s).wav	3.91	3.85	3.57	3.54	3.40	3.31	3.22	2.98	3.83	3.19	3.24
	Male 1 (0.00-12.18 s).wav	4.00	3.88	3.71	3.72	3.54	3.37	3.23	3.14	4.07	3.49	3.32
	Male 2 (0.00-10.04 s).wav	4.11	4.05	3.85	3.91	3.75	3.57	3.37	3.28	4.17	3.84	3.30
	<b>Average</b>	<b>4.03</b>	<b>3.95</b>	<b>3.73</b>	<b>3.75</b>	<b>3.59</b>	<b>3.42</b>	<b>3.29</b>	<b>3.18</b>	<b>4.03</b>	<b>3.50</b>	<b>3.31</b>
German	Female 1 (0.00-8.00 s).wav	4.08	3.98	3.65	3.60	3.54	3.36	3.15	3.06	4.06	3.40	3.20
	Female 2 (0.00-8.00 s).wav	4.21	4.14	3.93	3.88	3.76	3.63	3.53	3.47	4.17	3.54	3.37
	Male 1 (0.00-8.00 s).wav	4.12	4.08	3.90	3.88	3.78	3.66	3.55	3.50	4.19	3.82	3.43
	Male 2 (0.00-8.00 s).wav	4.17	4.07	3.97	3.92	3.77	3.69	3.65	3.56	4.21	3.70	3.38
	<b>Average</b>	<b>4.14</b>	<b>4.07</b>	<b>3.86</b>	<b>3.82</b>	<b>3.72</b>	<b>3.58</b>	<b>3.47</b>	<b>3.40</b>	<b>4.16</b>	<b>3.62</b>	<b>3.35</b>
Italian	Female 1 (0.00-20.60 s).wav	3.80	3.67	3.51	3.41	3.34	3.24	3.12	2.92	3.81	2.83	3.00
	Female 2 (0.00-21.78 s).wav	4.09	4.04	3.88	3.86	3.74	3.61	3.40	3.27	4.14	3.29	3.32
	Male 1 (0.00-18.13 s).wav	4.03	3.95	3.82	3.78	3.64	3.49	3.32	3.20	4.13	3.37	3.17
	Male 2 (0.00-20.86 s).wav	4.23	4.15	4.00	4.07	3.89	3.84	3.60	3.47	4.27	3.58	3.47
	<b>Average</b>	<b>4.04</b>	<b>3.95</b>	<b>3.80</b>	<b>3.78</b>	<b>3.65</b>	<b>3.55</b>	<b>3.36</b>	<b>3.22</b>	<b>4.08</b>	<b>3.27</b>	<b>3.24</b>
Japanese	Female 1 (0.00-7.60 s).wav	3.89	3.75	3.53	3.41	3.40	3.28	3.08	3.10	3.87	2.92	2.92
	Female 2 (0.00-7.31 s).wav	3.92	3.85	3.54	3.62	3.42	3.27	3.20	3.08	3.99	2.82	3.09
	Male 1 (0.00-7.13 s).wav	3.87	3.81	3.59	3.50	3.42	3.28	3.19	3.12	3.91	2.89	2.91
	Male 2 (0.00-7.45 s).wav	4.18	4.08	3.92	3.94	3.84	3.73	3.59	3.44	4.21	3.63	3.55
	<b>Average</b>	<b>3.97</b>	<b>3.87</b>	<b>3.64</b>	<b>3.61</b>	<b>3.52</b>	<b>3.39</b>	<b>3.27</b>	<b>3.19</b>	<b>4.00</b>	<b>3.06</b>	<b>3.12</b>
Spanish (US)	Female 1 (0.00-8.00 s).wav	4.03	3.96	3.63	3.68	3.50	3.30	3.23	3.16	3.96	3.03	3.20
	Female 2 (0.00-8.00 s).wav	3.63	3.48	3.23	3.26	3.05	3.03	2.88	2.80	3.73	2.68	2.72
	Male 1 (0.00-8.00 s).wav	3.98	3.61	3.67	3.49	3.51	3.38	3.23	3.15	4.15	3.44	3.19
	Male 2 (0.00-8.00 s).wav	4.04	3.87	3.71	3.58	3.52	3.28	3.11	3.14	4.09	3.48	3.36
	<b>Average</b>	<b>3.92</b>	<b>3.73</b>	<b>3.56</b>	<b>3.50</b>	<b>3.40</b>	<b>3.25</b>	<b>3.11</b>	<b>3.06</b>	<b>3.99</b>	<b>3.16</b>	<b>3.12</b>

NOTE – Grey cells indicate samples that do not meet [ITU-T P.501] requirements.

## Appendix II

### Test databases for P.862/P.862.1

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

The test databases comprised network conditions which corresponded to fixed, wireless and VoIP applications. A large set of codecs used by different technologies (such as GSM-FR, GSM-EFR, GSM-AMR, CDMA-EVRC, IS136-ACELP, G.711, G.726, G.728, G.729, JDC-HR) has been included in the P.862 and P.862.1 tests and analysis.

Table II.1 summarizes the content of all the databases on which PESQ (P.862 and P.862.1) have been validated.

**Table II.1 – Summary of the database type and content**

Description	Conditions
8 kbit/s characterization interworking with standards, P-suppl. 23, exp 1	enc/dec + trans; codecs: G.711, G.726, G.728, G.729, IS-54, GSM-FR, JDC-HR.
8 kbit/s characterization channel errors and noise, P-suppl. 23, exp 3	enc/dec + trans + ErrorPatterns + BGN; codecs: G.729.
Live wireless networks	<ul style="list-style-type: none"> <li>– Wireless networks: IS-136, CDMA, iDEN, AMPS; GSM-US, GSM-Europe;</li> <li>– Codecs: IS-54, 8 kbit/s ACELP, 13 kbit/s QCELP, GSM-FR and EFR, CDMA-EVRC.</li> </ul>
Codecs, error patterns, transcodings, noise	<ul style="list-style-type: none"> <li>– enc/dec + ErrorPatterns + BGN; codecs: G.711, G.726, G.728, G.729, GSM-FR;</li> <li>– enc/dec + ErrorPatterns (C/I levels) + BGN; codecs: G.711, G.723 + transcodings, 8 kbit/s ACELP, EVRC, GSM-EFR and FR;</li> <li>– enc/dec + trans + ErrorPatterns ("bad" and "good" cond) + BGN, ATM/ISDN/POTS; codecs: G.729, G.728, GSM-FR, GSM-HR.</li> </ul>
Background noise test, GSM and fixed networks	<ul style="list-style-type: none"> <li>– enc/dec + VAD + VQE (NR); codecs: GSM-FR, G.729;</li> <li>– enc/dec + ErrorPatterns; codecs: GSM-FR, GSM-AMR-HR.</li> </ul>
DTX, frame/burst erasure and VAD test, AMR, GSM and fixed networks	enc/dec + ErrorPatterns; codecs: GSM-FR, G.726, G.728, G.729, AMR.
AMR + error patterns on hopping and non-hopping GSM channels	enc/dec + ErrorPatterns; codecs: AMR475, AMR590, AMR740, AMR122.
AMR + error patterns with unequal and equal error protection	enc/dec + ErrorPatterns; codecs: AMR515, AMR740, AMR102, AMR122, AMR475, AMR590, AMR670, AMR795.
VoIP error pattern and packet loss; noise	enc/dec + trans + ErrorPatterns + PacketLoss + BGN.
VoIP packet loss scenarios test	VoIP: PktInsert, PktDelete, PktMute.
VoIP codecs	VoIP: G.723.1, G.728, G.711, G.729, G.726, PDC-HR.
VoIP conditions	PLC and jitter buffer management (G.711, G.729, G.723 at 6.3 and 5.3 kbit/s), variable packet length.
Field VoIP conditions	VoIP: Internet PABX, IP gateways.

## Appendix III

### Report of P.862/P.862.1 measurements

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

The P.862/P.862.1 measurements should be reported based on the algorithm's accuracy presented in clause 10.3 (Tables 1 and 2).

#### III.1 Report and interpretation of the average PESQ results

As mentioned in clause 10.1, it is recommended in the case of controlled simulated network test conditions to perform averages per-condition, using at least four talkers.

The averaged P.862.1 scores are expected to be determined with a prediction error of maximum  $\pm 95\%$  confidence interval upper limit.

In the case of field network testing, there are situations when it is desirable to roughly estimate the speech quality provided by the network within a certain test area, and/or during a time window. In this case, P.862.1 scores are averaged along a test route and/or a time window. The averaged P.862.1 scores are expected to be determined with a prediction error of maximum  $\pm 95\%$  confidence interval upper limit. Therefore, assuming for example that the tested network is a CDMA network, then it is expected that the average prediction error of the averaged P.862.1 measurements is less than or equal to 0.462 MOS.

It should be noted though that for both cases presented above, simulated and field conditions, there is a risk of 5% that the measurement error is higher than the 95% upper limit of the prediction error. In addition, as mentioned in clause 10.2, the measurement scenarios need to be similar to the ones presented in Table 1.

#### III.2 Report and interpretation of individual PESQ measurement results

As mentioned in clause 10.1, in the case of live field network testing a per-sample quality evaluation, due to uncontrolled time-varying transmission channels, is required.

There are two recommended reporting procedures, depending on the type of the PESQ measurements, which are used for the network's speech quality analysis.

The first type is the averaged P.862.1 score, which is discussed in clause III.1 and provides a rough estimate of the speech quality within a certain area or during a time window.

The other type of score is the individual P.862.1 measurement when troubleshooting of the network is required.

In the case of individual measurements, it is recommended to calculate a histogram of the P.862.1 measurements along the MOS scale. As an analysis procedure in this case, it is recommended to consider a subjective speech quality threshold and to impose a minimum required percentage of scores lying above this subjective threshold or equivalently a minimum required probability with which the P.862.1 measurements in the tested network lie above the subjective threshold. In this way, the speech quality based on individual P.862.1 scores is represented by the probability or percentage of scores that lie above the imposed subjective speech quality threshold.

For the evaluation of this probability or percentage, two types of errors affect the results. The first error is caused by the calculation of the probability or the percentage. The other error is determined by the P.862/P.862.1 residual error distribution. Due to the P.862.1 measurement error, not all the points determined to be above (and/or below) the imposed MOS threshold are in reality there. As mentioned in clause 10.4, the residual error distribution (see Table 2) shows that lower absolute errors are more likely than higher absolute errors. Therefore, when a P.862.1 score is on one side of

the MOS threshold based on the actual measurement, the likelihood for it to appear on the other side of this threshold is higher if the actual score is closer to the threshold. For example, let us consider a MOS threshold of MOS=3 and a CDMA test network. P.862.1 scores between 3 and 3.1 have a probability of 40.44% to fall below the threshold. P.862.1 scores between 3.1 and 3.2 have a probability of only 30.04% to fall below the threshold. Similarly, scores between 2.9 and 3 have a probability of 40.44% and scores between 2.8 and 2.9 have probability of only 30.04% to fall above the 3 MOS threshold.

A recommended method for calculating the P.862.1 measurement error when using individual scores now follows.

The P.862.1 measurement error, due to the error of evaluation of the probability that P.862.1 scores are higher than the imposed MOS threshold, is defined as the standard deviation of a binomial distribution with the probability of occurrence:

$$p = \frac{n}{N}$$

where n represents the number of scores above the threshold and N represents the total number of P.862.1 measurements. The error caused by the probability evaluation *Error1* is given therefore by:

$$Error1 = \pm \sqrt{\frac{p \times (1-p)}{N}}$$

and it is characterized by the 95% confidence interval:

$$\pm z_{\alpha} \times Error1$$

where:

$$z_{\alpha} = 2$$

represents the Gaussian quantile for 95% probability (the normal distribution describes quite well the PESQ individual score's behaviour)

The P.862/P.862.1 measurement error due to the residual error of the PESQ algorithm *Error2* is recommended to be calculated as the standard deviation of the binomial distribution that describes the residual error presented in Table 2.

When individual P.862/P.862.1 scores are used, it is therefore recommended to calculate the total measurement error of the speech quality as the square root of the sum of the two above-mentioned squared errors, *Error1* and *Error2*.

## **Appendix IV**

### **Calibration method for proprietary interfaces**

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

NOTE – The method described here is intended for situations where the exact required signal levels are unknown and the methods recommended in clause 9 cannot be applied. It cannot be expected to achieve maximum accuracy by this method. It is also important that all level adjustments are either performed in the analogue domain or using a sufficient word length in the digital domain (at least 24 bits).

#### **IV.1 Calibration of the transmit level (near end) of the test equipment**

Adjust the output level of the test equipment to a level which falls approximately in the middle of the operating range of any potential AGC. This can be done by measuring the signal level at the far end terminal and tuning the attenuation of the signal level at the near end to the middle of the range at which the level at the far end remains constant. Alternatively a phone may be used on the far end and the level at the near end is adjusted to the middle of the range which gives a comfortable listening level on the far end. The second method is also preferable in situations where no AGC is present.

#### **IV.2 Calibration of the receive level (far end) of the test equipment**

The level at the far end should be adjusted in a way that the attenuation between the recorded file and the reference file is close to 0 dB.

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