30 Speech teleservices

When an artificial ear is required, the ITU-T Recommendation P.57 [107] Type 1 artificial ear shall be used.

If requested by the terminal supplier, the ITU-T Recommendation P.57 [107] Type 3.2 artificial ear shall be used. In this case the following apply:

- The low leakage option of Type 3.2 artificial ear shall be adopted;
- The force against the ear shall be as specified in ITU-T Recommendation P.57 [107].
- Sound pressure measurements shall be referred to the ERP as specified in ITU-T Recommendation P.57 [107].
- No leakage correction shall be made in the calculation of RLR (i.e. L_E=0).

If requested by the terminal supplier, the ITU-T Recommendation P.57 [107] Type 3.4 artificial ear may be used for Release 96 MS or later. The positioning is defined in ITU-T Recommendation P.64.

The manufacturer declares in the IXIT statement which type of artificial ear will be used for teleservices speech testings.

NOTE 1: An MS may be either a handset MS, a handsfree MS or a combined handset and handsfree MS. The test description for handsfree operation, however, at the moment only covers the stability margin as no test method could be defined for the other parameter.

NOTE 2: Frequency settings in the following tests are taken from ISO 3, R10 series or R40 series or from table 2 of CCITT P.79. A departure from the nominal frequencies of +5 % below 240 Hz and + 2 % at 240 Hz and above is accepted. Any sub-multiple of the sampling frequency of 8 kHz shall be avoided. In the case of 4 kHz the departure is restricted to -2 %.

NOTE 3: The measurement accuracy for signal level is +/- 0,2 dB and for sound pressure +/-0,6 dB.

NOTE 4: The digital test signals shall be generated as 8 bit A-law companded PCM signals, which internally in the SS are expanded according to CCITT Rec. G.721 (Law=1) to 13 bit linear before being applied to the MS via the DAI.

NOTE 5: When measuring signal levels on the DAI, a digital measuring instrument is connected to the 64 kbit/s output of the A-law compression equipment in the SS, which is in turn connected to the DAI in the MS.

NOTE 6: Measurements shall be possible with and without psophometric weighting according to Rec. CCITT G.223, table 4.

30.1 Sending sensitivity/frequency response

30.1.1 Definition and applicability

The sending sensitivity frequency response is, as a function of the input test signal frequency, the ratio expressed in dB between the output level at the Digital Audio Interface (DAI) or at the audio output of the reference speech decoder of the SS and the input sound pressure in the artificial mouth required to obtain this.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.1.2 Conformance requirement

The sending sensitivity frequency response shall be within the mask given in GSM 03.50.

GSM 03.50; 3.8.1.1, table 1.

30.1.3 Test purpose

To verify that the sending sensitivity frequency response is within the mask given in GSM 03.50, 3.8.1.1, table 1.

30.1.4 Method of test

30.1.4.1 Initial conditions

When measured at the DAI:

a) The handset is mounted in the LRGP (see annex A of CCITT Recommendation P.76). The earpiece is sealed to the knife-edge of the artificial ear.

b) A pure tone with a sound pressure of -4,7 dBPa (in accordance with CCITT Recommendation P.64) is applied at the mouth reference point (MRP) as described in CCITT Recommendation P.64 using an artificial mouth conforming to CCITT Recommendation P.51.

c) A digital measuring instrument, or high quality digital decoder followed by an analogue level measuring set, is connected to the Digital Audio Interface (DAI). The DAI is set to the operating mode "Test of acoustic devices and A/D & D/A ".

When measured at the output of the reference speech decoder of the SS:

a) The handset is mounted in the LRGP and the earpiece is sealed to the knife-edge of an artificial ear.

b) A full rate speech call is set up between the MS and the SS.

c) Artificial speech conforming to ITU-T Recommendation P.50, shall be applied to the MRP, at a wideband sound pressure level of -4,7 dBPa. This implementation could be a real time algorithm producing the artificial speech or a pre-recorded tape of the artificial speech.

d) The artificial speech shall comprise of a concatenation of three 10 s intervals of "male" and "female" voice. The first 10 s interval is not used for measurement purposes but allows any noise/echo cancelling devices in the MS to adapt. The second and third 10 s intervals consist of separately "male" and "female" artificial voice.

30.1.4.2 Procedure

When measured at the DAI:

The SS measures the output level represented by the PCM bit stream at the DAI (pin 23) at one-twelfth-octave intervals as given by the R40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 4 000 Hz inclusive.

When measured at the output of the reference speech decoder of the SS:

The 1/3 octave filtered long-term average spectrum of the signal is measured at the analogue or digital output of the reference speech decoder of the SS and an average for the "male" and "female" voices is obtained. The sending sensitivity/frequency response is calculated as the difference between the 1/3 octave input power and the 1/3 octave output power.

30.1.5 Test requirement

The sending sensitivity/frequency response shall be within a mask given in table 30.1. The mask can be drawn with straight lines between the breaking points in the table on a logarithmic (frequency) vs linear (dB sensitivity) scale.

All sensitivity levels are dB on an arbitrary scale.

Table 30.1

Frequency (Hz)	Upper Limit (dB)	Lower Limit (dB)
100	-12	
200	0	
300	0	-12
1000	0	-6
2000	4	-6
3000	4	-6
3400	4	-9
4000	0	

30.2 Sending loudness rating

30.2.1 Definition and applicability

The Sending Loudness Rating (SLR) is a means of expressing the sending frequency response based on objective measurements in a way which relates to how a speech signal would be perceived by a listener.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.2.2 Conformance requirement

The Sending Loudness Rating (SLR) shall be 8 +/- 3 dB.

GSM 03.50; 3.1.1.

30.2.3 Test Purpose

To verify that the Sending Loudness Rating (SLR) is 8 +/- 3 dB.

30.2.4 Method of test

30.2.4.1 Initial conditions

When measured at the DAI:

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A".

When measured at the output of the reference speech decoder of the SS:

A full rate speech call is set up between the MS and the SS.

30.2.4.2 Procedure

When measured at the DAI:

a) The sending sensitivity is measured at each of the 14 frequencies given in table 2 of CCITT P.79, bands 4 to 17.

b) The sensitivity is expressed in terms of dBV/Pa and the SLR is calculated according to CCITT Recommend ation P.79 formula 4.19 b of CCITT P.79, over bands 4 to 17, using the sending weighting factors from CCITT Recommendation P.79 table 2, adjusted according to table 3 of CCITT Recommendation P.79.

When measured at the output of the reference speech decoder of the SS:

a) The sending sensitivity from the MRP to the analogue or digital output of the reference speech decoder of the SS is determined according to 30.1.4.1, 30.1.4.2.

b) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 formula 2.1, over bands 4 to 17, and using m = 0.175 and the sending weighting factors from ITU-T Recommendation P.79 table 1.

30.2.5 Test requirement

The SLR shall be 8 +/- 3 dB.

30.3 Receiving sensitivity/frequency response

30.3.1 Definition and applicability

The receiving sensitivity frequency response is, as a function of the input test signal frequency, the ratio expressed in dB between the output sound pressure in the artificial ear and the input level, represented by the PCM bit stream at the Digital Audio Interface (DAI) or the level at the SS audio input, required to obtain this.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.3.2 Conformance requirement

The receiving sensitivity frequency response shall be within the mask given in GSM 03.50.

GSM 03.50; 3.8.1.2, table 2.

30.3.3 Test purpose

To verify that the receiving sensitivity frequency response is within the mask given in GSM 03.50; 3.8.1.2, table 2.

30.3.4 Method of test

30.3.4.1 Initial conditions

When measured from the DAI:

a) The handset is mounted in the LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

b) A digital signal generator is connected at the digital interface delivering a signal equivalent to a pure tone level of -16 dBm0, see CCITT Recommendation P.64.

c) The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

When measured from the input of the reference speech encoder of the SS:

a) The handset is mounted in the LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

b) A full rate speech call is set up between the MS and the SS.

c) Artificial speech conforming to ITU-T Recommendation P.50, shall be applied to the analogue or digital input of the reference speech encoder of the SS, at a wideband level of -16 dBm0. This implementation could be a real time algorithm producing the artificial speech or a pre-recorded tape of the artificial speech.

d) The artificial speech shall comprise of a concatenation of three 10 s intervals of "male" and "female" voice. The first 10 s interval is not used for measurement purposes but allows any echo cancellation devices in the MS to adapt. The second and third 10 s intervals consist of separately "male" and "female" artificial voice.

30.3.4.2 Procedure

When measured from the DAI:

Measurements are made at one twelfth-octave intervals as given in the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 4 kHz inclusive. At each frequency, the sound pressure in the artificial ear is measured by connecting a suitable measuring set to the artificial ear.

When measured from the input of the reference speech encoder of the SS:

The 1/3 octave filtered long-term average spectrum of the signal is measured and an average for the "male" and "female" voices is obtained. The receiving sensitivity/frequency response is calculated as the difference between the 1/3 octave input power and the 1/3 octave output power.

30.3.5 Test requirement

The receiving sensitivity/frequency response shall be within the mask given by table 30.2. The mask can be drawn with straight lines between the breaking points in the following table on a logarithmic (frequency) vs linear (dB sensitivity) scale.

All sensitivity levels are dB on an arbitrary scale.

Table 30.2

Frequency (Hz)	Upper Limit (dB)	Lower Limit (dB)
100	-12	
200	0	
300	2	-7
500	*	-5
1000	0	-5
3000	2	-5
3400	2	-10
4000	2	

NOTE: * The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) vs linear (dB) scale.

30.4 Receiving loudness rating

30.4.1 Definition and applicability

The Receiving Loudness Rating (RLR) is a means of expressing the receiving frequency response based on objective measurements in a way which relates to how a speech signal would be perceived by a listener.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.4.2 Conformance requirement

1) The nominal Receiving Loudness Rating (RLR) shall be 2 +/- 3 dB.

   If a user controlled receive volume control is provided the equipment shall meet this nominal value for at least one setting of the control.

   GSM 03.50; 3.1.1.

2) If a user controlled receive volume control is provided the Receive Loudness Rating (RLR) shall not be less than -13 dB when the control is set to maximum.

   GSM 03.50; 3.1.1.

30.4.3 Test purpose

1) To verify that the nominal Receiving Loudness Rating (RLR) is 2 +/- 3 dB.

2) To verify that if a user controlled receive volume control is provided the Receive Loudness Rating (RLR) is not less than -13 dB when the control is set to maximum.

30.4.4 Method of test

30.4.4.1 Initial conditions

When measured at the DAI:

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A".

When measured at the output of the reference speech decoder of the SS:

A full rate speech call is set up between the MS and the SS.

30.4.4.2 Procedure

When measured at the DAI:

a) The receiving sensitivity is measured at each of the 14 frequencies listed in table 2 of CCITT Recommendation P.79, bands 4 to 17.

b) The sensitivity is expressed in terms of dBPa/V and the RLR is calculated according to CCITT Recommendation P.79 formula 4.19 c, over bands 4 to 17, using the receiving weighting factors from table 2 of CCITT Recommendation P.79, adjusted according to table 3 of CCITT Recommendation P.79.

c) The artificial ear sensitivity must be corrected according to the real ear correction of table 4 of CCITT Recommendation P.79.

NOTE: The values of real ear correction in CCITT Recommendation P.79 table 4 were derived for one type of handset conforming to the shape defined in CCITT Recommendation P.35.

These values are used in this EN because there is no measurement method agreed for the real ear correction. If a method of measurement is agreed, it is intended to change this EN to use the values appropriate to each handset.

When measured from the input of the reference speech encoder of the SS:

a) The receiving sensitivity from the analogue or digital input of the reference speech encoder of the SS to the output of the artificial ear is determined according to 30.3.4.1, 30.3.4.2 .

b) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 formula 2.1, over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79.

30.4.5 Test requirement

If no user controlled receive volume control is provided, the RLR shall be 2 +/- 3 dB.

If a user controlled receive volume control is provided, the RLR shall meet this nominal value for (at least) one setting of the receive volume control.

When the receive volume control is set to maximum the RLR shall not be less than (i.e. louder than) -13 dB.

30.5 Side tones

30.5.1 Side Tone Masking Rating (STMR)

30.5.1.1 Definition and applicability

The sidetone loudness ratings are a means of expressing the path loss from the artificial mouth to the artificial ear based on objective single tone measurements in a way that relates to how a speaker will perceive his own voice when speaking (talker sidetone, expressed by the sidetone masking rating - STMR), or how a listener will perceive the background noise picked up by the microphone (listener sidetone rating - LSTR).

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.5.1.2 Conformance requirement

The nominal value of the Side Tone Masking Rating (STMR) shall be 13 +/- 5 dB. Where a user controlled receiving volume control is provided the STMR shall meet the requirement at the setting where the RLR is equal to the nominal value.

GSM 03.50; 3.10.1.

30.5.1.3 Test purpose

1) To verify that the Side Tone Masking Rating (STMR) is 13 +/- 5 dB.

2) To verify that is a user controlled receiving volume control is provided, the STMR is 13 +/- 5 dB at the setting where the RLR is equal to the nominal value.

30.5.1.4 Method of test

30.5.1.4.1 Initial conditions

a) The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A".

b) The handset is mounted in the LRGP (see annex 1 of CCITT P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to CCITT P.51.

30.5.1.4.2 Procedure

a) The SS sends a PCM bit stream coded with the value No 1 over the DAI (pin 25). Or alternatively the activation of the A/D and D/A converters is performed via a call setup, in which case the DAI connection between the MS and SS, and the PCM bit stream are optional.

NOTE: The idle channel noise in the receiving direction is the acoustic sound pressure in the artificial ear when the digital input signal at the DAI is the PCM coded value No. 1.

b) The SS applies a pure tone with a sound pressure of -4,7 dBPa at the mouth reference point as described in CCITT P.64 using an artificial mouth conforming to CCITT P 51.

c) For each frequency given in table 2 of CCITT P.79, bands 4 to 17, the sound pressure in the artificial ear is measured.

d) The sidetone path loss (LmeST) is expressed in dB and the STMR (in dB) is calculated from the formula 8.4 of CCITT Recommendation P.79, using the weighting factors of column (3) in table 6 of CCITT Recommendation P.79 (unsealed), and values of LE in accordance with table 4 of CCITT Recommendation P.79.

30.5.1.5 Test requirement

The STMR shall be 13 +/- 5 dB.

Where a user controlled receive volume control is provided, the STMR shall meet the requirement given above at the setting where the RLR is equal to the nominal value.

30.5.2 Listener Side Tone Rating (LSTR)

30.5.2.1 Definition and applicability

The Listener Sidetone Rating (LSTR) is considered a major parameter affecting the user perception of the system.

The requirements and this test is applicable to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.5.2.2 Conformance requirement

The value of the Listener Sidetone Rating (LSTR) shall not be less than 15 dB.

GSM 03.50, 3.10.1.

30.5.2.3 Test purpose

To verify that the value of LSTR is not less than 15dB.

30.5.2.4 Method of test

30.5.2.4.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A".

The SS sends a PCM bit stream coded with the value No. 1 over the DAI (pin 25) to the MS.

30.5.2.4.2 Procedure

a) The sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within +4 dB/-2 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100 Hz to 8 kHz (bands 1 to 20).

b) A calibrated half-inch microphone is mounted at MRP. The sound field is measured in one-third octave bands. The spectrum shall be "Pink noise" as described in CCITT recommendation P.64 annex B to within +/-1 dB and the level shall be adjusted to 70 dBA (-24 dBPa(A)). The tolerance on this level is +/-1 dB.

c) The artificial mouth and ear are placed in the correct position relative to MRP, the handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

d) Measurements are made in one-third octave bands for the 14 bands centred at 200 Hz to 4 kHz (bands 4 to 17). For each band the sound pressure in the artificial ear shall be measured by connecting a suitable measuring set to the artificial ear.

e) The listener sidetone path loss is expressed in dB and the LSTR shall be calculated from the CCITT Recommendation P.79 formula 8-4, using the weighting factors in column (3) in table 6 of the Recommendation, and the values of LE; in accordance with table 4 of the Recommendation.

30.5.2.5 Test requirement

The LSTR shall not be less than 15 dB.

30.6 Telephone Acoustic coupling Loss (TAL)

30.6.1 Echo Loss (EL)

30.6.1.1 Definition and applicability

The echo loss is the path loss from the input of the reference speech encoder of the SS to the output of the reference speech decoder of the SS.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.6.1.2 Conformance requirement

The echo loss from the input to the output of the reference speech codec in the SS shall be at least 46 dB.

GSM 03.50; 3.4.3.2.

30.6.1.3 Test purpose

To verify that the echo loss from the input to the output of the reference speech codec in the SS is at least 46 dB.

30.6.1.4 Method of test

30.6.1.4.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Normal operation".

The SS sets up a speech call according to the generic call set up procedure.

The handset is mounted in the LRGP (see annex 1 of CCITT P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to CCITT P.51.

Where a user controlled volume control is provided it is set to maximum.

30.6.1.4.2 Procedure

An implementation of the CCITT P.50 artificial speech is connected to the analogue or digital input of the reference speech encoder of the SS. This implementation is either a real time algorithm producing the artificial speech or a pre-recorded tape of artificial speech. Both "male" and "female" artificial speech is required.

A ten second segment of the "male" artificial speech is applied to the analogue or digital input of the reference speech encoder of the SS. The third octave power of the input signal is measured. The echo loss signal is not measured at this stage as the first ten second segment is used to allow any acoustic echo cancellation devices within the MS to adapt to the echo path.

Immediately after a second ten second segment of the "male" artificial speech is applied to the analogue or digital input of the reference speech encoder of the SS. The third octave power of the echo signal is measured at the analogue or digital output of the reference speech decoder of the SS.

The difference between the third octave input power and the third octave output power is entered into the CCITT G.122 TCL algorithm and the acoustic echo loss calculated.

The test is repeated with the "female" artificial speech and the results of both "male" and "female" averaged to give the final result.

30.6.1.5 Test requirement

The echo loss from the input to the output of the reference speech codec in the SS shall be at least 46 dB.

30.6.2 Stability margin

30.6.2.1 Definition and applicability

The receive-transmit stability margin is a measure of the gain that would have to be inserted between the go and return paths of the reference speech coder in the SS for oscillation to occur.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 MS supporting speech.

30.6.2.2 Conformance requirement

The stability margin shall be at least 6 dB.

GSM 03.50; 3.2.

30.6.2.3 Test purpose

To verify that the stability margin is at least 6 dB.

30.6.2.4 Method of test

30.6.2.4.1 Initial conditions

For handset operation the handset is placed on a hard plane surface with the transducers facing the surface.

For handsfree operation the test setup is shown in CCITT P.34 (Fig 3/CCITT P.34), but omitting the test table.

Where a user controlled volume control is provided it is set to maximum.

30.6.2.4.2 Procedure

a) A gain equivalent to the minimum stability margin is inserted in the loop between the go and return paths of the reference speech coder in the SS and any acoustic echo control is enabled.

b) A test signal according to CCITT O.131 is injected into the loop at the analogue or digital input of the reference speech codec of the SS and the stability is measured. The test signal has a level of -10 dBm0 and a duration of 1 s.

30.6.2.5 Test requirement

The minimum stability margin shall be 6 dB and no audible oscillation shall be detected.

30.7 Distortion

30.7.1 Sending

30.7.1.1 Definition and applicability

The transmit signal to total distortion ratio is a measure of the linearity of the transmitter equipment.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.7.1.2 Conformance requirement

The ratio of signal to total distortion power in the sending direction measured with a psophometric filter at the DAI of the MS or at the output of the reference speech decoder of the SS shall be above the limits given in GSM 03.50; 3.9.1, table 3, unless the sound pressure at MRP exceeds +10 dBPa.

GSM 03.50; 3.9.1.

30.7.1.3 Test purpose

To verify that the ratio of signal to total distortion power in the sending direction measured with psophometric filter is above the limits given in GSM 03.50; 3.9.1, table 3.

30.7.1.4 Method of test

30.7.1.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of CCITT P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to CCITT P.51.

When measured at the DAI:

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

When measured at the output of the reference speech decoder of the SS:

A full rate speech call is set up between the MS and the SS.

30.7.1.4.2 Procedure

a) A sine-wave signal with a frequency in the range 1004 Hz to 1025 Hz is applied to the MRP. The level of this signal is adjusted until the level at the DAI output (pin 23) of the MS or at the analogue or digital output of the reference speech decoder of the SS corresponds to -10 dBm0. The level of the signal at the MRP is then the acoustic reference level (ARL).

b) The test signal is applied at the following levels: -35, -30, -25, -20, -15, -10, -5, 0, 5, 10 dB relative to the ARL.

c) The ratio of signal to total distortion power is measured at the DAI of the MS or at the analogue or digital output of the reference speech decoder of the SS with the psophometric noise weighting (see CCITT G.714 and O.132) at each signal level.

NOTE: The measurement is not to be carried out at sound pressures exceeding +10 dBPa.

30.7.1.5 Test requirement

The ratio of signal to total distortion power measured with the psophometric noise weighting (see table 4/CCITT G.223) shall be above the limits given in table 30.3.

Table 30.3

dB relative to ARL	Level ratio
-35 dB	17,5 dB
-30 dB	22,5 dB
-20 dB	30,7 dB
-10 dB	33,3 dB
0 dB	33,7 dB
7 dB	31,7 dB
10 dB	25,5 dB

Limits for the signal to total distortion ratio (sending) when using the sine wave method.

Limits for intermediate levels are found by drawing a straight line between breaking points in a linear (dB signal level) vs linear (dB ratio) scale.

30.7.2 Receiving

30.7.2.1 Definition and applicability

The receive signal to total distortion ratio is a measure of the linearity in the receive equipment (excluding the speech decoder).

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.7.2.2 Conformance requirement

The ratio of signal to total distortion power in the receiving direction measured at the ERP with psophometric filter shall be above the limits given in GSM 03.50; 3.9.2, table 5.

GSM 03.50; 3.9.2.

30.7.2.3 Test purpose

To verify that the ratio of signal to total distortion power in the receiving direction measured at the ERP with psophometric filter is above the limits given in GSM 03.50; 3.9.2, table 5.

30.7.2.4 Method of test

30.7.2.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of CCITT P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to CCITT P.51.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.7.2.4.2 Procedure

a) The SS sends, via the DAI (Pin 25), a PCM bit stream simulating a sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz corresponding to CCITT O.132 at the following levels: -45, -40, -35, -30, -25, -20, -15, -10, -5, 0 dBm0.

b) The ratio of signal to total distortion power is measured with the psophometric noise weighting in the artificial ear (see CCITT G.714 and O.132) at each signal level.

c) The measurement is only carried out at sound pressures between -50 dBPa and +10 dBPa.

30.7.2.5 Test requirement

The ratio of signal to total distortion power measured at the artificial ear with the psophometric noise weightin g (see table 4/CCITT G.223) shall be above the limits given in table 30.4.

Table 30.4

Level at the digital audio interface	Level ratio
-45 dBm0	17,5 dB
-40 dBm0	22,5 dB
-30 dBm0	30,5 dB
-20 dBm0	33,0 dB
-10 dBm0	33,5 dB
-3 dBm0	31,2 dB
0 dBm0	25,5 dB

Limits for the signal to total distortion ratio (receiving) when using the sine wave method.

Limits for intermediate levels are found by drawing a straight line between breaking points in a linear (dB signal level) vs linear (dB ratio) scale.

30.8 Sidetone distortion

30.8.1 Definition and applicability

The sidetone distortion expresses the linearity of the sidetone path in the handset.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.8.2 Conformance requirement

The third harmonic distortion of the sidetone shall not be greater than 10 %.

GSM 03.50; 3.10.2.

30.8.3 Test purpose

To verify that the third harmonic distortion of the sidetone is not greater than 10 %.

30.8.4 Method of test

30.8.4.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A".

The handset is mounted in the LRGP (see annex 1 of CCITT P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to CCITT P.51.

30.8.4.2 Procedure

a) The SS sends the PCM bit stream coded with the value No 1 over the DAI (pin 25) to the MS.

b) An instrument capable of measuring the third harmonic distortion of signals with fundamental frequencies in the range 315 Hz to 1000 Hz is connected to the artificial ear.

c) A pure-tone signal of -4,7 dBPa is applied at the mouth reference point at frequencies of 315 Hz, 500 Hz, and 1 000 Hz. For each frequency the third harmonic distortion is measured in the artificial ear.

30.8.5 Test requirement

The third harmonic distortion generated shall not be greater than 10 %.

30.9 Out-of-band signals

30.9.1 Sending

30.9.1.1 Definition and applicability

The discrimination against out-of-band input signals in the sending direction is a requirement on the in-band image frequencies created by any out-of-band input signals.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.9.1.2 Conformance requirement

With any sine wave signal above 4,6 kHz and up to 8 kHz applied at the MRP at a level of -4,7 dBPa, the level of any image frequency produced at the digital interface shall be below a reference level obtained at 1 kHz (-4,7 dBPa at MRP) by at least the amount (in dB) specified in GSM 03.50; 3.11.1, table 7.

GSM 03.50; 3.11.1.

30.9.1.3 Test purpose

To verify that the conformance requirement is met for input signals with frequencies of 4,65, 5, 6, 6,5, 7 and 7,5 kHz.

30.9.1.4 Method of test

30.9.1.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of CCITT P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to CCITT P.51.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.9.1.4.2 Procedure

a) A pure tone with a sound pressure of -4,7 dBPa is applied at the mouth reference point as described in CCITT P.64 using an artificial mouth conforming to CCITT P 51.

b) For input signals at frequencies of 4,65, 5, 6, 6,5, 7, and 7,5 kHz, the level represented by the PCM bit stream at the DAI (Pin 23) of any image frequency is measured.

30.9.1.5 Test requirement

The level of any image frequency shall be below a reference obtained at 1 kHz by at least the amount as specified in table 30.5.

Table 30.5

Applied sine-wave frequency	Limit (minimum)
4,6 kHz	30 dB
8 kHz	40 dB

Limits for the image frequency discrimination.

The limit at intermediate frequencies lies on a straight line drawn between the given values on a log(frequency) vs linear(dB) scale.

30.9.2 Receiving

30.9.2.1 Definition and applicability

The discrimination against out-of-band signals in the receiving direction is a requirement on the out-of-band signals generated in the artificial ear from in-band input signals.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.9.2.2 Conformance requirement

With a digitally simulated sine wave signal in the frequency range of 300 Hz to 3,4 kHz and at a level of 0 dBm applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4,6 to 8 kHz measured selectively in the artificial ear shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in GSM 03.50; 3.11.2, table 8.

GSM 03.50; 3.11.2.

30.9.2.3 Test purpose

To verify that the conformance requirement is met for input signals at the nominal frequencies 500, 1 000, 2 000, and 3 350 Hz.

30.9.2.4 Method of test

30.9.2.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of CCITT P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to CCITT P.51.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.9.2.4.2 Procedure

a) The SS sends over the DAI (pin 25) a PCM bit stream simulating a sine-wave signal with a level of 0 dBm0.

b) For input signals at the nominal frequencies 500, 1 000, 2 000, and 3 350 Hz (bearing in mind the restriction on sub-multiples of the sampling frequency) the level of any out-of-band signals at frequencies up to 8 kHz is measured in the artificial ear.

30.9.2.5 Test requirement

The level of out-of-band signals shall be lower than the in-band acoustic level obtained by a digital signal at 1 kHz set at the level specified in table 30.6.

Table 30.6

Image signal frequency	Equivalent input signal level
4,6 kHz	-35 dBm0
8 kHz	-45 dBm0

Limits for the image frequency discrimination.

The limit at intermediate frequencies lies on a straight line drawn between the given values on a log(frequency) vs linear(dB) scale.

30.10 Idle channel noise

30.10.1 Sending

30.10.1.1 Definition and applicability

The idle channel noise in the sending direction is the equivalent noise level produced at the DAI, when the mouth reference point is in a quiet environment.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.10.1.2 Conformance requirement

The idle noise in the sending direction shall not exceed - 64 dBm0p at the UPCMI under silent conditions.

GSM 03.50; 3.6.1.

30.10.1.3 Test purpose

To verify that the idle noise in the sending direction does not exceed -64 dBm0p at the UPCMI under silent conditions.

30.10.1.4 Method of test

30.10.1.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of CCITT P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to CCITT P.51 in a quiet environment (ambient noise less than 30 dBA).

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.10.1.4.2 Procedure

The noise level represented by the PCM bit stream output at the DAI (pin 23) is measured with psophometric weighting according to CCITT G.223, table 4.

NOTE: The ambient noise criterion should be met if the ambient noise does not exceed NR20.

30.10.1.5 Test requirement

The noise produced by the MS in the sending direction shall not exceed -64 dBm0p.

30.10.2 Receiving

30.10.2.1 Definition and applicability

The idle channel noise in the receiving direction is the acoustic sound pressure in the artificial ear when the digital input signal at the DAI, is the PCM coded value No 1.

The requirements and this test apply to all types of GSM 400, GSM 900 and DCS 1 800 handset MS supporting speech.

30.10.2.2 Conformance requirement

1. If no user controlled receiving volume control is provided, or if it is provided, at the setting of the user controlled receiving volume at which the RLR is equal to the nominal value, the noise measured in the artificial ear contributed by the receiving equipment alone shall not exceed -57 dBPa (A) when driven by a PCM signal corresponding to the decoder output value No. 1.

   GSM 03.50; 3.6.2.

2. Where a volume control is provided, the measured noise shall not exceed -54 dBPa(A) at the maximum setting of the volume control.

   GSM 03.50; 3.6.2.

30.10.2.3 Test purpose

1. To verify that the idle noise in the receiving direction does not exceed -57 dBPa (A). If a user controlled receive volume control is provided it shall be set to the position where RLR is equal to the nominal value.

2. To verify that if a user controlled receive volume control is provided, the idle noise in the receiving direction does not exceed -54 dBPa(A) when the control is set to maximum.

30.10.2.4 Method of test

30.10.2.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of CCITT P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to CCITT P.51.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.10.2.4.2 Procedure

a) The SS sends a PCM bit stream coded with the value No 1 over the DAI (Pin 25) to the MS.

b) The level of the noise is measured in the artificial ear with any volume control set at the position at which the RLR is equal to the nominal value.

c) Where a volume control is provided, the level of the noise is measured in the artificial ear with the volume control set to maximum.

30.10.2.5 Test requirement

In step b) the measured noise generated by the MS shall not exceed -57 dBPa (A).

In step c) the measured noise shall not exceed -54 dBPa (A).

30.11 Ambient Noise Rejection

30.11.1 Definition and Applicability

An MS that supports speech will typically be operated within an area of high ambient acoustic noise. A level of noise rejection will therefore be required.

The requirements and this test apply to all types of GSM 400, GSM900 and DCS1800 handset R96 MS (or later) supporting full rate speech.

30.11.2 Conformance Requirement

Compliance shall be checked by calculating the single figure DELSM (SFDELSM) according to the following formula, the SFDELSM shall be ≥0dB.

$$SFDELSM = -4/5 \times \sum_{n=1}^{14} Del_n \times 10^{(-0.0175 \times Wm)}$$

where :-

n = the third octave band centre frequencies from 160Hz to 3150Hz inclusive.

Deln = is the 1/3 octave band pressure level centered on the n^{th} frequency.

Wm = is the SLR weighting for the nth 1/3 octave band centre frequency.

GSM 03.50; 3.14.1

30.11.3 Test Purpose

To verify that ambient noise calculated as SFDELSM shall be rejected, by verifying that SFDELSM ≥ 0dB.

30.11.4 Method of Test

30.11.4.1 Initial Conditions

A 1/2 inch pressure microphone is calibrated using a known sound source and mounted at the MRP, without the LRGP head present. A frequency analyser is calibrated to enable the sound pressure levels at the microphone to be determined in 1/3rd octave bands.

Flood the room in which the measurement is to be made with the selected noise file, and adjust the level such that the noise level at the MRP is 70dBA. A single noise file of real noise , covering the various noise environments that the MS could be subjected to, is used. This file is three minutes long and also commences with a three minute signal. Once this tone has been adjusted to a level of 70dBA, the average level of the noise will be 70dBA. The resulting sound spectrum is Prn dBPa, measured in 1/3rd octave bands.

To ensure that the sound field is diffuse enough, the following apply:-

The diffuse sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within +/- 3dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100Hz to 3,15 Khz.

Where more than one loudspeaker is used to produce the desired sound field, the loudspeakers may require to be fed with non-coherent signals to eliminate standing waves and other interference effects.

Position an LRGP in the correct relative position to the MRP and mount the MS under test. Recalibrate the 1/3rd octave frequency analyser using a known voltage source to facilitate the analysis of the Voltage Vrn, where Vrn is the voltage at the audio output of the System Simulator (SS) due to the noise spectrum input.

30.11.4.2 Procedure

Set up a full rate speech path between the MS and the SS.

The SS determines, as a function of frequency, using the frequency analyser, in 1/3rd octave bands, the electrical output Vrn, (expressed as dB rel 1V) at the audio output of the SS for the applied acoustic pressure Prn (expressed as dB rel 1Pa) at the MRP. Since, the MS sending sensitivity is not defined above 3,4kHz and below 300Hz the measurement shall be cut off at 3,4kHz and for the bands below 300Hz. The noise level shall be referenced to the speech level at 300Hz to yield the DELSM.

The room noise sensitivity is defined as :- Smj_{rn} = Vrn (dBV) - Prn (dBPa).

The ambient noise send sensitivity has now been determined.

The MS speech send sensitivity is now required. The required sensitivity is defined as the electrical output from the MS, measured at the audio output of the SS, as a function of the free field sound pressure at the MRP of the artificial mouth.

The measurment is made using an artificial speech source at the MRP of the artificial mouth. The 1/2 inch pressure microphone is calibrated using a known sound source. The frequency analyser is calibrated to measure in 1/3rd octave bands. The artificial mouth output shall be in accordance with the CCITT P.50 male artificial voice. Whilst maintaining the CCITTT P.50 'male' spectrum , the total signal level is adjusted to -4,7 dBPa. The resulting sound spectrum is P_0dBpa, measured in 1/3rd octave bands. The 1/3rd octave frequency analyser shall be re-calibrated, using a known voltage source , to facilitate the analysis of the voltage V_j. Where V_j is the voltage at the audio output of the SS due to the speech spectrum input. A speech path is setup between the MS and the SS. The function of the frequency is determined using the frequency analyser, and in 1/3rd octave bands, the electrical output V_j, (expressed as dB rel. 1V), at the audio output of the SS for the applied acoustic pressure, P_0, (expressed as dB rel. 1Pa/V), at the MRP.

The sending sensitivity is expressed as : -

$$S_{mjs}(dB) = V_j(dBV) - P_o(dBPa) dBrel 1V/Pa$$

The D_{SM} for the MS is determined as :-

$$D_{SM} = Smj_{rn} - S_{mjs}\ (dB).$$

30.11.5 Test Requirement

The MS ambient noise rejection, calculated as a single figure DELSM (SFDELSM) shall be greater than or equal to 0dB.