32 Testing of speech transcoding functions

The test sequences for speech transcoding and DTX tests, both for input and required output, are defined in GSM 06.10, section 5, and GSM 06.32, section 4 for the full rate speech codec. For the half rate speech codec the test sequences are defined in GSM 06.20, section 5 and GSM 06.42 section 7. They are available on floppy disks in IBM/AT MS-DOS format from ETSI publications department.

The Digital Audio Interface (DAI) is described in 36.4.

NOTE: For a definition of the term "traffic frame" used in this chapter, refer to GSM 06.32 and GSM 06.42.

32.1 Full Rate Downlink speech transcoding

32.1.1 Definition and applicability

Downlink speech transcoding transforms the 13 kbit/s net bit stream obtained by channel decoding the incoming bit stream from the air interface to 13 bit linear PCM.

The requirements and this test apply to all MS supporting TCH/FS.

32.1.2 Conformance requirement

The output bit stream from the speech transcoder shall be continuous and bit by bit exactly the same as the predefined output sequence (SEQ01.OUT, SEQ03.OUT, SEQ04.OUT and SEQ05.OUT).

GSM 06.01, 2; GSM 06.10, 5.2/5.2.2.

32.1.3 Test purpose

To verify that the speech transcoding of the MS can transform all predefined sequences (SEQ01.OUT, SEQ03.OUT, SEQ04.OUT and SEQ05.OUT) at 13 kbits/s level to 104 kbit/s (13 bit linear PCM at 8 kHz) level correctly.

32.1.4 Method of test

32.1.4.1 Initial conditions

DTX is off.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech decoder/DTX functions (downlink)".

32.1.4.2 Procedure

a) The SS resets the speech decoder of the MS via the DAI.

b) The SS sends test sequence SEQ01.COD at 13 kbit/s to the MS via the air interface after passing it through the SS channel encoder.

NOTE: These test sequence files contain 16 bit words for all speech encoded parameters and are justified as described in GSM 06.10 table 5.1. 76 words are used as input in a period of 20 ms.

c) The SS records the 104 kbit/s output bit stream from the MS on the digital audio interface.

d) The test is repeated using the test sequences SEQ03.COD, SEQ04.COD and SEQ05.COD.

32.1.5 Test requirements

The bit stream output shall be continuous and bit by bit exactly the same as the sequence given in the files SEQ01.OUT, SEQ03.OUT, SEQ04.OUT and SEQ05.OUT.

NOTE: These files contain 16 bit words of 13 bit linear PCM left justified.

32.2 Full Rate Downlink receiver DTX functions

32.2.1 Definition and applicability

The DTX receiver functions consist of a SID frame detector, comfort noise generator functions and lost frame substitution and muting functions.

The requirements and this test apply to all MS supporting TCH/FS.

32.2.2 Conformance requirement

1) The output level of the decoder has to be constant for an input signal consisting of identical speech frames.

   GSM 06.10.

2/3) When, after the first lost speech frame subsequent speech frames are lost, a muting technique shall be used that will gradually decrease the output level, resulting in the silencing of the output after a maximum of 320 ms. Speech frames with the FACCH flag set provoke a Bad Frame Indication (BFI = 1) and are hence regarded as lost speech frames.

   GSM 06.01, 6; GSM 06.11, 2.1/2.2/3 for requirement 2 (first part).

   GSM 06.01, 6; GSM 06.11, 2.1/2.2/3; GSM 06.31, 1.2.2/3.1.1 for requirement 3 (second part).

4/5) A valid SID-frame followed by a sequence of lost speech frames shall result in comfort noise generation with constant block amplitude parameters. Speech frames with the FACCH flag set provoke a Bad Frame Indication (BFI = 1) and are hence regarded as lost speech frames.

   GSM 06.01, 3/5; GSM 06.12, 3/3.1;
   GSM 06.31, 1.2.2/3.1/3.1.1/3.1.2 for requirement 4 (first part).

   GSM 06.01, 3/5/6; GSM 06.11, 2.1/2.2; GSM 06.12, 3/3.1;
   GSM 06.31, 1.2.2/3.1/3.1.1/3.1.2 for requirement 5 (second part).

6/7) An invalid SID-frame followed by a sequence of lost speech frames shall result in comfort noise generation, using the set of parameters from the last valid SID-frame. Speech frames with the FACCH flag set provoke a Bad Frame Indication (BFI = 1) and are hence regarded as lost speech frames.

   GSM 06.01, 3/5; GSM 06.12, 3/3.1;
   GSM 06.31, 1.2.2/3.1/3.1.1/3.1.2 for requirement 6 (first part).

   GSM 06.01, 3/5/6; GSM 06.11, 2.1/2.2; GSM 06.12, 3/3.1;
   GSM 06.31, 1.2.2/3.1/3.1.1/3.1.2 for requirement 7 (second part).

8) The energy of the output signal is controlled by the block amplitude parameter, xmaxc.

   GSM 06.10, 3.1.20/3.1.21/3.2.1;

9/10) The first SID-frame that is expected and not received shall be substituted by the last valid SID-frame and the procedure for valid SID-frames shall be applied. For the second lost SID-frame, a muting technique shall be used that will gradually decrease the output level, resulting in silencing the output after a maximum of 320 ms. Speech frames with the FACCH flag set provoke a Bad Frame Indication (BFI = 1) and are hence regarded as lost speech frames.

GSM 05.08, 8.3; GSM 06.01, 6; GSM 06.11, 2.3/2.4;
GSM 06.31, 1.2.2/3.1.1/3.1.2.

32.2.3 Test purpose

1) To verify that the signal energy at the output of the decoder is constant with a tolerance of +/- 3 dB if a sequence of identical speech frames is applied at the receiver input.

2) To verify that the muting function of the receiver is within the required limits if a sequence of lost speech frames is applied at the receiver input.

3) To verify that the muting function of the receiver is within the required limits if a sequence of speech frames with the FACCH flag set is applied at the receiver input.

4) To verify the function of comfort noise generation when a valid SID-frame is received followed by a sequence of lost speech frames. The signal energy at the output of the decoder shall be constant with a tolerance of +/- 3 dB.

5) To verify the function of comfort noise generation when a valid SID-frame is received followed by a sequence of speech frames with the FACCH flag set. The signal energy at the output of the decoder shall be constant with a tolerance of +/- 3 dB.

6) To verify the function of comfort noise generation when an invalid SID-frame is received followed by a sequence of lost speech frames. The signal energy at the output of the decoder shall be constant with a tolerance of +/- 3 dB.

7) To verify the function of comfort noise generation when an invalid SID-frame is received followed by a sequence of speech frames with the FACCH flag set. The signal energy at the output of the decoder shall be constant with a tolerance of +/- 3 dB.

8) To verify that the signal energy at the output of the decoder depends on the block amplitude xmaxc of the input frames if a sequence of speech frames is applied to the decoder. The signal energy at the output of the decoder shall be constant with a tolerance of +/- 3 dB.

9) To verify the SID-frame substitution and muting functions on the comfort noise, if two consecutive expected SID-frames are lost with the other frames being lost speech frames.

10) To verify the SID-frame substitution and muting functions on the comfort noise, if two consecutive expected SID-frames are lost with the other frames being speech frames with the FACCH flag set.

32.2.4 Method of test

32.2.4.1 Initial conditions

Uplink DTX is off.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech decoder/DTX functions (downlink)".

32.2.4.2 Procedure

a) The SS transmits coded "speech" traffic frames on the air interface after passing them trough the SS channel encoder. They contain a special test signal at 13 kbit/s as defined below. All traffic frames are identical with the exception of some frames which are SID frames as defined in GSM 06.32.

b) The energy of the PCM signal is evaluated (as a mean square average) at the digital audio interface of the MS at 104 kbit/s level (13 bit, 8 kHz linear PCM) and recorded for each block of 20 ms synchronized to the 20 ms speech frame structure.

c) The SS transmission of the TDMA frames of the TCH/FS on the air interface is ramped "on" and "off" on a traffic frame by traffic frame basis, taking into account the block diagonal interleaving scheme defined in GSM 05.03. The first traffic frame in step 1 occurs one frame after the window of the SACCH multiframe (TDMA frame 60 modulo 104), allocated for the SID frame (see GSM 05.02 and GSM 05.08). The SACCH will also be transmitted.

NOTE 1: 8 timeslots in 8 consecutive TCH/FS TDMA frames are seen as one traffic frame, and the next traffic frame starts in the middle of the previous one (i.e. after 4 TDMA frames of the previous one) due to the block diagonal interleaving scheme defined in GSM 05.03.

d) The special test frame is an encoded "speech" traffic frame of 260 bits obtained from white Gaussian noise band limited to 300 - 3 400 Hz. When repeated, the special test frame results in a humming sound with a fairly constant level when decoded, and is defined in table 32-1.

Table 32-1: Table of special test traffic frame for receiver DTX tests

Encoded parameter:	Value:			
LARc(1)	38			
LARc(2)	42			
LARc(3)	24			
LARc(4)	20			
LARc(5)	10			
LARc(6)	9			
LARc(7)	5			
LARc(8)	3			
	Sub-block no:			
	0	1	2	3
Grid position (Mc)	1	3	2	0
Block amplitude (xmaxc)	40	40	40	40
LTP gain (Bc)	0	0	0	0
LTP lag (Nc)	40	120	40	120
RPE pulses (xmc)				
- pulse no 1	4	6	6	6
- pulse no 2	4	5	4	3
- pulse no 3	2	1	3	4
- pulse no 4	6	2	1	3
- pulse no 5	3	6	4	1
- pulse no 6	5	1	6	3
- pulse no 7	5	2	5	5
- pulse no 8	5	6	2	1
- pulse no 9	1	3	4	4
- pulse no 10	3	2	4	3
- pulse no 11	5	5	4	5
- pulse no 12	6	1	2	2
- pulse no 13	1	3	4	3

NOTE 2: The signal energy of the decoded special test frame is controlled with the block amplitude parameter (xmaxc). Reducing xmaxc from 40 to 32 reduces the signal energy by 6 dB, and reducing xmaxc from 40 to 24 reduces the signal energy by 12 dB.

e) The sequence of traffic frames on the air interface is as follows:

e.1) 23 test frames "on".

e.2) 20 frames "off".

e.3) 20 test frames "on".

e.4) 1 SID frame followed by 6 frames "off", another identical SID frame and 23 frames "off". Except for the SID codeword, the SID frames are identical to the test frame.

e.5) 1 different SID frame, however with 2 to 15 errors inserted in the SID codeword, followed by 23 frames "off".

e.6) 20 test frames "on", but with the block amplitude parameter xmaxc = 24.

e.7) 1 SID frame followed by 50 frames "off". Except for the SID codeword, the SID frames are identical to the test frame.

e.8) The whole test is repeated, but the frames "off" are replaced by frames "on" with the FACCH flag set.

32.2.5 Test requirements

1) In step e.1), the signal energy shall be fairly constant within +/- 3 dB.

2) In step e.2), the signal energy shall decrease to less than -60 dBm within 17 frames.

3) In step e.4), comfort noise shall be generated. The same requirements as in step e.1) apply.

4) In step e.5), the same requirements as in step e.4) apply.

5) In step e.6), the same requirements as in step e.1) apply. However, the signal energy shall be 12 dB lower.

6) In step e.7), the signal energy shall be fairly constant within +/- 3 dB for 28 frames. Then the signal energy shall decrease to less than -60 dBm within 16 frames.

7) In step e.8), the same requirements as in all previous steps apply.

32.3 Full Rate Uplink speech transcoding

32.3.1 Definition and applicability

Uplink speech transcoding transforms 13 bit linear PCM to the 13 kbit/s net bit stream. This net bit stream is to be channel encoded for transmission on the air interface.

The requirements and this test apply to all MS supporting TCH/FS.

32.3.2 Conformance requirement

The output bit stream from the speech transcoder shall be bit by bit exactly the same as the predefined output sequence (SEQ01.OUT, SEQ02.OUT, SEQ03.OUT and SEQ04.OUT).

GSM 06.01, 2; GSM 06.10, 5.2/5.2.1.

32.3.3 Test purpose

To verify that the speech transcoder on the MS can transform all predefined sequences (SEQ01.INP, SEQ02.INP, SEQ03.INP and SEQ04.INP) at 104 kbit/s (13 bit linear PCM at 8 kHz) level to 13 kbit/s level correctly.

32.3.4 Method of test

32.3.4.1 Initial conditions

Uplink DTX is off.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech encoder/DTX functions (uplink)".

32.3.4.2 Procedure

a) The SS resets the speech decoder on the MS (see 36.4).

b) The SS sends a test sequence SEQ01.INP to the MS at 104 kbit/s level via the digital audio interface.

NOTE: These files contain 16 bit words for 13 bit linear PCM left justified. See also GSM 06.10 table 5.1.

c) The SS records the 13 kbit/s output bit stream obtained by channel decoding the incoming bit stream from the air interface.

d) The test is repeated using the test sequences SEQ02.INP, SEQ03.INP and SEQ04.INP.

32.3.5 Test requirements

The bit stream output shall be bit by bit exactly the same as the sequence given in the files SEQ01.COD, SEQ02.COD, SEQ03.COD and SEQ04.COD.

NOTE: These files contain 16 bit words of all the 76 parameters in a speech frame justified as in GSM 06.10 table 5.1. 76 codewords shall occur in a frame of 20 ms.

32.4 Full Rate Uplink transmitter DTX functions

32.4.1 Definition and applicability

The VAD/DTX transmitter functions consist of a Voice Activity Detector (VAD) and a surrounding Discontinuous Transmission (DTX) system introducing additional "speech" traffic frames on the air interface compared to those the VAD itself would classify as speech frames containing real speech. The additional traffic frames on the air are introduced due to:

1) A "hangover" period at the end of speech bursts in order to be certain that the traffic frames contain only noise and to evaluate the background acoustic noise characteristics when no real speech is present.

2) Special traffic frames (SID frames) added on the air at regular intervals containing only the evaluated background acoustic noise characteristics. These frames are used for generation of comfort noise in speaker silence periods on the receiving side.

The requirements and this test apply to all MS supporting TCH/FS.

32.4.2 Conformance requirement

The MS VAD and DTX function allows only those frames to be transmitted that are either marked with SP = 1 or that are properly positioned SID-frames.

GSM 05.08, 8.3; GSM 06.01, 3/4; GSM 06.31, 2.1/2.1.1/2.1.2; GSM 06.32, 1/2/2.1/2.2.8.

32.4.3 Test purpose

To verify that the combination of VAD and DTX operates correctly.

32.4.4 Method of test

32.4.4.1 Initial conditions

A call is set up on a TCH/FS according to the generic call set-up procedure.

Uplink DTX is on.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech encoder/DTX functions (uplink)".

32.4.4.2 Procedure.

a) The SS sends a test sequence SPEC_A1.INP of PCM samples, which are grouped into frames of 20 ms synchronized to the TDMA and traffic frame structure on the air interface, on the digital audio interface in the MS at 104 kbits/s (13 bit, 8 kHz linear PCM).
The start of the test sequences is synchronized with the radio transmission on the air interface so that the first traffic frame on the air occurs just after the traffic frame allocated for the SID frame (TDMA frame 56 modulo 104, see GSM 05.02 and GSM 05.08).

NOTE: 8 timeslots in 8 consecutive TCH/FS TDMA frames are seen as one traffic frame and the next traffic frame starts in the middle of the previous one (i.e. after 4 TDMA frames of the previous one) due to the block diagonal interleaving scheme defined in GSM 05.03.

b) The SS detects whether or not there is any power transmitted over the radio path on a timeslot basis excluding SACCH frames. The speech frame by speech frame on/off transmission (on = 1) is recorded.

c) The test is repeated for all test sequences *.INP described in GSM 06.32 section 4.

32.4.5 Test requirements

1) In step b), the traffic frame on/off sequence recorded shall be bit exact like the sequence of SP flags stored as bit 15 of LAR(2) on the respective reference files *.COD described in GSM 06.32, with the following exceptions:

1.1) The occurrence of a SID frame in its allowed window within the SACCH multiframe as defined in GSM 05.08.

1.2) The occurrence of a SID frame after 1 or more real speech frames consecutively transmitted on the air.

32.5 Full Rate Speech channel transmission delay

32.5.1 Definition and applicability

The total transmission delay within the various elements of a GSM system are specified as round trip delays. For the MS this would be equivalent to applying an RF equivalent of a speech signal to the MS receiver, closing an acoustic path from the ERP to the MRP, detecting the corresponding RF signal at the MS transmitter output and measuring the time interval between the signal originally fed to the MS receiver and that transmitted by the MS transmitter.

This simple approach cannot be demonstrated to be accurate due to the inherent non linear characteristic of the speech transcoder. The overall delay therefore is split into four identifiable and measurable delays. The delays are respectively:

- the downlink delay from RF input to DAI output,
- DAI output to ERP,
- MRP to DAI output, and
- DAI to uplink RF output.

Each delay is defined and its method of test described in the following sections.

The requirements and this test apply to all MS supporting TCH/FS.

32.5.2 Conformance requirement

The overall speech channel transmission delay shall be less than 143,9 ms.

GSM 03.50, section 3.3.6.1.

32.5.3 Test purpose

To verify that the round trip delay, of a speech channel for a MS, which consists of the sum of:

- the downlink delay from RF input to DAI output,
- DAI output to ERP,
- MRP to DAI output, and
- DAI to uplink RF output

meets the requirements when using the predefined test sequences SEQ01.COD, SEQ03.COD, SEQ04.COD and SEQ05.COD.

32.5.4 Downlink processing delay

32.5.4.1 Definition and applicability

The downlink processing delay is the delay from the first bit of a speech block transmitted from the RF output of the SS up to the last bit of the corresponding speech block received at the DAI on the output of the speech transcoder.

The requirements and this test apply to all MS supporting TCH/FS.

32.5.4.2 Method of test

32.5.4.2.1 Initial conditions

DTX is off.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech decoder/DTX functions (downlink)".

32.5.4.2.2 Procedure

a) The test set up is that described in 32.1 for downlink speech transcoding.

b) The SS transmits one of the test patterns SEQ01.COD, SEQ03.COD, SEQ04.COD or SEQ05.COD to the MS.

c) The SS measures for each speech block it transmits the time between the first bit at the air interface and the last bit of that speech block on the DAI. This difference is the delay measured.

d) Step c) is repeated 20 times and the maximum delay measured in ms is the downlink processing delay TDP.

NOTE: This is to account for the fact that the processing time may not be constant.

32.5.5 Downlink coding delay

32.5.5.1 Definition and applicability

The downlink coding delay is defined as the delay between the digital representation of an acoustic signal on the DAI and the corresponding acoustic signal at the ERP.

The requirements and this test apply to all MS supporting TCH/FS.

32.5.5.2 Method of test

32.5.5.2.1 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 recommendation P.76) and the earpiece is sealed to the knife edge of the artificial ear conforming to CCITT recommendation P.51.

32.5.5.2.2 Procedure

a) The SS generates on the DAI a digital representation of a sine wave with a frequency of 1 000 Hz.

b) The SS measures the "phase shift" $\varnothing_1$, in the range of 0 to 360 degrees, between the equivalent sine wave generated at the DAI and the sine wave at the input to the artificial ear.

c) The frequency is increased to 1100 Hz and the resulting phase shift $\varnothing_2$ noted.

d) The downlink coding delay TDC is calculated from either:

$TDC = (\varnothing_2 - \varnothing_1)/36$ ms for $\varnothing_2 > \varnothing_1$

or

$TDC = (\varnothing_2 + 360 - \varnothing_1)/36$ ms for $\varnothing_2 < \varnothing_1$

32.5.6 Uplink processing delay

32.5.6.1 Definition and applicability

The uplink processing delay is the delay from the first bit of a speech block on the DAI to the last bit of that speech block being transmitted on the air interface of the MS.

The requirements and this test apply to all MS supporting TCH/FS.

32.5.6.2 Method of test

32.5.6.2.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech encoder/DTX functions (uplink)".

32.5.6.2.2 Procedure

a) The test set up is that described in 32.3 for uplink speech transcoding.

b) The SS sends one of the test patterns SEQ01.INP, SEQ03.INP, SEQ04.INP or SEQ05.INP to the DAI of the MS.

c) The SS measures the time between the first bit on the DAI, and the last transmitted bit of the block at the air interface for each speech block the SS sends on the DAI. This time difference is the delay measured.

d) Step c) is repeated 20 times. The maximum delay measured in ms is the uplink coding delay TUP.

NOTE: This is to account for the fact that the processing time may not be constant.

32.5.7 Uplink coding delay

32.5.7.1 Definition and applicability

The uplink coding delay is defined as the delay between an acoustic signal at the MRP and the digital representation of that signal on the DAI.

The requirements and this test apply to all MS supporting TCH/FS.

32.5.7.2 Method of test

32.5.7.2.1 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 recommendation P.76) and the earpiece is sealed to the knife edge of the artificial ear conforming to CCITT recommendation P.51.

32.5.7.2.2 Procedure

a) The SS generates an acoustic signal at the artificial mouth of the LRGP, being a pure sine wave with a frequency of 1 000 Hz.

b) The SS measures the "phase shift" $\varnothing_1$, in the range of 0 to 360 degrees, between the signal at the MRP and its digital representation on the DAI.

c) The SS set the generated frequency to 1 100 Hz and measures the resulting phase shift $\varnothing_2$.

d) The uplink coding delay TUC is calculated from either:

$TUC = (\varnothing_2 - \varnothing_1)/36$ ms for $\varnothing_2 > \varnothing_1$

or

$TUC = (\varnothing_2 + 360 - \varnothing_1)/36$ ms for $\varnothing_2 < \varnothing_1$

32.5.8 Test requirement

The sum of the delays {TDP + TDC + TUP + TUC} shall be less than 144,9 ms.

NOTE 1: This limit includes an allowance of 4*0,25 ms delay from the DAI to the MS transmission path.

NOTE 2: No allowances have been made for any delays within the measurement system. These must either be calibrated out or subtracted from the individual delays before performing the sum above.

32.6 Half Rate Downlink speech transcoding

32.6.1 Definition and applicability

Downlink speech transcoding transforms the 5,6 kbit/s net bit stream obtained by channel decoding the incoming bit stream from the air interface to 104 kbit/s (13 bit linear PCM at 8 kHz) level.

The requirements and this test apply to all MS supporting TCH/HS.

32.6.2 Conformance requirement:

The output bit stream from the speech transcoder shall be continuous and bit by bit exactly the same as the predefined output sequences contained in SEQ01.OUT, SEQ02.OUT, SEQ03.OUT and SEQ04.OUT.

GSM 06.02, 5; GSM 06.20.

32.6.3 Test purpose:

To verify that the speech transcoder of the MS can transform all the predefined sequences (SEQ01.DEC, SEQ02.DEC, SEQ03.DEC and SEQ04.DEC) at 5,6 kbit/s level to 104 kbit/s (13 bit linear PCM at 8 kHz) level correctly.

32.6.4 Method of test

32.6.4.1 Initial conditions

Uplink DTX is off.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech decoder/DTX functions (downlink)".

Frequency hopping is on, where the BCCH carrier is part of the hopping sequence. Frequency hopping shall be performed over four carriers using random frequency hopping. Downlink power control shall be activated and a difference of 30 dB between the level of the BCCH carrier and the other carriers adjusted.

NOTE: Frequency hopping is used to ensure that the MS can cope with the reception of bursts (on the BCCH carrier) that have a power level that is different from the rest of the bursts.

32.6.4.2 Procedure

a) The SS sends a reset pulse to the MS on the digital audio interface. This reset pulse will start the clock output of the MS at 104 kHz (pin 24 of the DAI).

b) The SS sends test sequence SEQ01.DEC at 5,6 kbit/s to the MS via the air interface after passing it through the SS channel encoder. The speech decoder of the MS is reset by the special reset sequence which is at the beginning of the test sequence.

c) The SS records the 104 kbit/s output bit stream from the MS on the digital audio interface. The recording shall be triggered by the reception of the encoder homing frame. The encoder homing frame itself shall not be recorded.

d) The test is repeated using test sequences SEQ02.DEC, SEQ03.DEC and SEQ04.DEC.

32.6.5 Test requirement

The bit stream output shall be continuous and bit by bit exactly the same as the sequence describing the speech data contained in the files SEQ01.OUT, SEQ02.OUT, SEQ03.OUT and SEQ04.OUT. The two encoder homing frames at the beginning of each test sequence *.OUT shall be disregarded for this comparison.

32.7 Half Rate Downlink receiver DTX functions

32.7.1 Definition and applicability

The DTX receiver functions consist of a SID frame detector, comfort noise generator functions and lost frame substitution and muting functions.

The requirements and this test apply to all MS supporting TCH/HS.

32.7.2 Conformance requirement

The output bit stream from the speech transcoder shall be continuous and bit by bit exactly the same as the predefined output sequences contained in DTX*.OUT described in GSM 06.07 subclause 7.

GSM 06.02, 6/8; GSM 06.22; GSM 06.41.

32.7.3 Test purpose

To verify that the MS generates comfort noise correctly.

32.7.4 Method of test

32.7.4.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech decoder/DTX functions (downlink)".

Frequency Hopping is on, where the BCCH carrier is part of the hopping sequence. Frequency Hopping shall be done over four carriers using random Frequency Hopping.

NOTE: Frequency Hopping is used to ensure that the MS can cope with the reception of dummy bursts (on the BCCH frequency) during DTX.

32.7.4.2 Procedure

a) The SS sends a reset pulse to the MS on the digital audio interface. This reset pulse will start the clock output of the MS at 104 kHz (pin 24 of the DAI).

b) The SS sends test sequence DTX01.DEC at 5,6 kbit/s to the MS via the air interface after passing it through the SS channel encoder. The speech decoder of the MS will be reset by the special reset sequence which is at the beginning of the test sequence.

c) The SS transmission of the TDMA frames of the TCH/HS on the air interface is ramped "on" and "off" on a traffic frame by traffic frame basis, taking into account the block diagonal interleaving scheme defined in GSM 05.03. The first traffic frame in step b occurs one frame after the window of the SACCH multiframe (TDMA frame 0 or 52 modulo 104 for subchannel 0 and TDMA frame 1 or 53 modulo 104 for subchannel 1), allocated for the SID frame (see GSM 05.02 and GSM 05.08). The SACCH will also be transmitted.

d) The information whether to ramp the transmitter of the SS "on" or "off" is derived from the sequence of SP-flags contained in the file DTX01.COD (see file format description in GSM 06.07 subclause 5 for the position of the SP-flag).

e) The SS records the 104 kbit/s output bit stream from the MS on the digital audio interface. The recording shall be triggered by the reception of the encoder homing frame. The encoder homing frame itself is not recorded.

f) The test is repeated using test sequences *.DEC described in GSM 06.07 subclause 7.

32.7.5 Test requirement

The bit stream output shall be continuous and bit by bit exactly the same as the sequence describing the speech data contained in the files DTX*.OUT described in GSM 06.07 subclause 7. The two encoder homing frames at the beginning of each test sequence *.OUT shall be disregarded for this comparison.

32.8 Half Rate Uplink speech transcoding

32.8.1 Definition and applicability

Uplink speech transcoding transforms 104 kbit/s (13 bit linear PCM at 8 kHz) level to the 5,6 kbit/s net bit stream. This net bit stream is to be channel encoded for transmission on the air interface.

The requirements and this test applies to all MS supporting TCH/HS.

32.8.2 Conformance requirement

The output bit stream from the speech transcoder shall be bit by bit exactly the same as the predefined sequences contained in SEQ01.COD, SEQ02.COD and SEQ03.COD described in GSM 06.07 subclause 6.

GSM 06.02, 5; GSM 06.20.

32.8.3 Test purpose

To verify that the speech transcoder of the MS can transform all the predefined sequences SEQ01.INP, SEQ02.INP and SEQ03.INP at 104 kbit/s (13 bit linear PCM at 8 kHz) level to 5,6 kbit/s level correctly.

32.8.4 Method of test

32.8.4.1 Initial conditions

Uplink DTX is off.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech encoder/DTX functions (uplink)".

Frequency hopping is on.

32.8.4.2 Procedure

a) The SS sends a reset pulse to the MS on the digital audio interface. This reset pulse will start the clock output of the MS at 104 kHz (pin 24 of the DAI).

b) The SS synchronizes the input of the test sequences via the digital audio interface to the framing of the MS in the uplink. This can be done in two steps as follows:

b.1) The SS sends to the MS at 104 kbit/s level via the digital audio interface 13 triplets of input frames, each triplet consisting of 480 samples. The 480 samples of one triplet shall all be identical. The 13 bits of one sample shall all be set to "zero" except for one which is set to "one". The position of the bit within the 13 bits of a sample that is set to "one" shall vary in such a way, that all possible 13 positions are exercised within the 13 triplets of input frames. An example for such a sequence is given in test sequence BITSYNC.INP described in GSM 06.07 subclause 8. The SS records the 5,6 kbit/s output bit stream obtained by channel decoding the incoming bit stream from the air interface. As soon as the decoder homing frame is detected at the output, the framing of the MS with respect to the 13 bit long input words is known by looking at the corresponding input frame that has caused the decoder homing frame at the output.

NOTE: The encoder homing frame consists of 160 identical samples, each 13 bit long left justified, with the least significant bit set to "one" and all other bits set to "zero" (0008 hex). The speech encoder will go to its predefined home state at the end of the first received encoder homing frame. Consecutive encoder homing frames will produce the decoder homing frame at the output of the speech encoder.

b.2) Synchronized to the 13 bit framing of the MS, the SS now sends test sequence SEQSYNC.INP described in GSM 06.07 subclause 8 to the MS at 104 kbit/s level via the digital audio interface. The SS records the 5,6 kbit/s output bit stream obtained by channel decoding the incoming bit stream from the air interface. The recording shall be triggered by the reception of the decoder homing frame. By comparing the first recorded frame that is not a decoder homing frame with the 160 possible output frames contained in sequences SYNC*.COD, the offset of the input to the 20 ms framing of the MS is known.

c) Synchronized to the 20 ms framing of the MS, the SS sends a test sequence SEQ01.INP to the MS at 104 kbit/s level via the digital audio interface. The speech encoder of the MS is reset by the special homing sequence which is at the beginning of the test sequence.

d) The SS records the 5,6 kbit/s output bit stream obtained by channel decoding the incoming bit stream from the air interface. The recording shall be triggered by the reception of the decoder homing frame. The decoder homing frame itself is not recorded.

e) The test is repeated using test sequences SEQ02.INP and SEQ03.INP.

32.8.5 Test requirements

The bit stream output shall be bit by bit exactly the same as the sequences describing the speech parameters contained in the files SEQ01.COD, SEQ02.COD and SEQ03.COD. The two decoder homing frames at the beginning of each test sequence *.COD shall be disregarded for this comparison.

32.9 Half Rate Uplink transmitter DTX functions

32.9.1 Definition and applicability

The VAD/DTX transmitter functions consist of a Voice Activity Detector (VAD) that inhibits the transmitter during speech pauses, and a surrounding Discontinuous Transmission (DTX) system introducing Silence Descriptor (SID) frames on the air interface.

The requirements and this test apply to all MS supporting TCH/HS.

32.9.2 Conformance requirement

The MS VAD and DTX function allow only those frames to be transmitted that are either properly positioned SID-frames, SACCH-frames or frames marked with SP-flag = 1.

For the transmitted frames, the output bit stream from the speech transcoder shall be bit by bit exactly the same as the predefined sequences contained in DTX*.COD described in GSM 06.07 subclause 6.

GSM 05.08, 8.3; GSM 06.02, 6/7; GSM 06.41; GSM 06.42.

32.9.3 Test purpose

To verify that the combination of VAD and DTX operates correctly.

32.9.4 Method of test

32.9.4.1 Initial conditions

Uplink DTX is on.

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech encoder/DTX functions (uplink)".

Frequency Hopping is on.

32.9.4.2 Procedure

a) The SS sends a reset pulse to the MS on the digital audio interface. This reset pulse will start the clock output of the MS at 104 kHz (pin 24 of the DAI).

b) The SS synchronizes the input of the test sequences via the digital audio interface to the framing of the MS in the uplink. This can be done in two steps as follows:

b.1) The SS sends to the MS at 104 kbit/s level via the digital audio interface 13 triplets of input frames, each triplet consisting of 480 samples. The 480 samples of one triplet shall all be identical. The 13 bits of one sample shall all be set to "zero" except for one which is set to "one". The position of the bit within the 13 bits of a sample that is set to "one" shall vary in such a way, that all possible 13 positions are exercised within the 13 triplets of input frames. An example for such a sequence is given in test sequence BITSYNC.INP described in GSM 06.07 subclause 8. The SS records the 5,6 kbit/s output bit stream obtained by channel decoding the incoming bit stream from the air interface. As soon as the decoder homing frame is detected at the output, the framing of the MS with respect to the 13 bit long input words is known by looking at the corresponding input frame that has caused the decoder homing frame at the output.

NOTE: The encoder homing frame consists of 160 identical samples, each 13 bit long left justified, with the least significant bit set to "one" and all other bits set to "zero" (0008 hex). The speech encoder will go to its predefined home state at the end of the first received encoder homing frame. Consecutive encoder homing frames will produce the decoder homing frame at the output of the speech encoder.

b.2) Synchronized to the 13 bit framing of the MS, the SS now sends test sequence SEQSYNC.INP described in GSM 06.07 subclause 8 to the MS at 104 kbit/s level via the digital audio interface. The SS records the 5,6 kbit/s output bit stream obtained by channel decoding the incoming bit stream from the air interface. The recording shall be triggered by the reception of the decoder homing frame. By comparing the first recorded frame that is not a decoder homing frame with the 160 possible output frames contained in sequences SYNC*.COD, the offset of the input to the 20 ms framing of the MS is known.

c) The SS sends test sequence DTX01.INP of PCM samples described in GSM 06.07 subclause 7 on the digital audio interface in the MS at 104 kbit/s (13 bit linear PCM at 8 kHz). The speech encoder of the MS will be reset by the special homing sequence which is at the beginning of the test sequence.

d) The start of the test sequence is synchronized with the radio transmission on the air interface so that the first traffic frame on the air caused by the first encoder homing frame in the test sequence occurs just after the traffic frame allocated for the SID frame (TDMA frame 0 or 52 modulo 104 for subchannel 0 and TDMA frame 1 or 53 modulo 104 for subchannel 1), allocated for the SID frame (see GSM 05.02 and GSM 05.08).

e) The SS detects whether or not there is any power transmitted over the radio path on a time slot basis excluding SACCH frames. The speech frame by speech frame on/off transmission (on = 1) is calculated and recorded. The recording shall be triggered by the reception of the decoder homing frame. The flag marking the decoder homing frame itself is not recorded.

f) The SS records the 5,6 kbit/s output bit stream obtained by channel decoding the incoming bit stream from the air interface. The recording shall be triggered by the reception of the decoder homing frame. The decoder homing frame itself is not recorded.

g) The test is repeated for all test sequences DTX*.INP described in GSM 06.07 subclause 7.

32.9.5 Test requirements

1) The bit stream recorded in step e) shall be continuous and bit by bit exactly the same as the sequence of SP-flags contained in the files DTX*.COD (see file format description in GSM 06.07 subclause 5 for the position of the SP-flag), except for the bits marking those frames that are SID frames scheduled for transmission according to GSM 06.41. The first two frames in the reference files *.COD shall be disregarded for this comparison.

2) The bit stream recorded in step f) shall be continuous and bit by bit exactly the same as the sequence describing the speech parameters contained in the files *.COD described in GSM 06.07 subclause 7, except for the bits of the speech frames marked with SP-flag=0. The two decoder homing frames at the beginning of each test sequence *.COD shall be disregarded for this comparison.

32.10 Half Rate Speech channel transmission delay

32.10.1 Definition and applicability

The total transmission delay within the various elements of a GSM system are specified as round trip delays. For the MS this would be equivalent to applying an RF equivalent of a speech signal to the MS receiver, closing an acoustic path from the ERP to the MRP, detecting the corresponding RF signal at the MS transmitter output and measuring the time interval between the signal originally fed to the MS receiver and that transmitted by the MS transmitter.

This simple approach cannot be demonstrated to be accurate due to the inherent non linear characteristic of the speech transcoder. The overall delay therefore is split into four identifiable and measurable delays. The delays are respectively:

- the downlink delay from RF input to DAI output,
- DAI output to ERP,
- MRP to DAI output, and
- DAI to uplink RF output.

Each delay is defined and its method of test described in the following sections.

The requirements and this test applies to all MS supporting TCH/HS.

32.10.2 Conformance requirement

The overall speech channel transmission delay shall be less than 143,9 ms.

GSM 03.50 section 3.3.6.2.

32.10.3 Test purpose

To verify that the round trip delay of a speech channel for a MS which consists of the sum of:

- the downlink delay from RF input to DAI output,
- DAI output to ERP,
- MRP to DAI output, and
- DAI to uplink RF output

meets the requirements when using the predefined test sequences SEQ01.INP and SEQ01.DEC.

32.10.4 Downlink processing delay

32.10.4.1 Definition and applicability

The downlink processing delay is the delay from the first bit of a speech block transmitted from the RF output of the SS up to the last bit of the corresponding speech block received at the DAI on the output of the speech transcoder.

The requirements and this test apply to all MS supporting TCH/HS.

32.10.4.2 Method of test

32.10.4.2.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech decoder/DTX functions (downlink)".

32.10.4.2.2 Procedure

a) The test set up is that described in section 32.6.4.2 for downlink speech transcoding.

b) The SS transmits the test pattern SEQ01.DEC described in GSM 06.07 subclause 6 to the MS.

c) The SS measures for each speech block it transmits the time between the first bit at the air interface and the last bit of that speech block on the DAI. This time difference is the delay measured.

d) Step c) is repeated 20 times and the maximum delay measured in ms is the downlink processing delay TDP.

NOTE: This is to account for the fact that the processing time may not be constant.

32.10.5 Downlink coding delay

32.10.5.1 Definition and applicability

The downlink coding delay is defined as the delay between the digital representation of an acoustic signal on the DAI and the corresponding acoustic signal at the ERP.

The requirements and this test applies to all MS supporting TCH/HS.

32.10.5.2 Method of test

32.10.5.2.1 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 recommendation P.76) and the earpiece is sealed to the knife edge of the artificial ear conforming to CCITT recommendation P.51.

32.10.5.2.2 Procedure

a) The SS generates on the DAI a digital representation of a sine wave with a frequency of 1 000 Hz.

b) The SS measures the "phase shift" $\varnothing_1$, in the range of 0 to 360 degrees, between the equivalent sine wave generated at the DAI and the sine wave at the input to the artificial ear.

c) The frequency is increased to 1 100 Hz and the resulting phase shift $\varnothing_2$ noted.

d) The downlink coding delay TDC is calculated from either:

$TDC = (\varnothing_2 - \varnothing_1)$ ms/36 for $\varnothing_2 > \varnothing_1$

or

$TDC = (\varnothing_2 + 360 - \varnothing_1)$ ms/36 for $\varnothing_2 < \varnothing_1$

32.10.6 Uplink processing delay

32.10.6.1 Definition and applicability

The uplink processing delay is the delay from the first bit of a speech block on the DAI to the last bit of that speech block being transmitted on the air interface of the MS.

The requirements and this test apply to all MS supporting TCH/HS.

32.10.6.2 Method of test

32.10.6.2.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Test of speech encoder/DTX functions (uplink)".

32.10.6.2.2 Procedure

a) The test set up is that described in section 32.8.4.2 for uplink speech transcoding.

b) The SS sends one of the test patterns SEQ01.INP described in GSM 06.07 subclause 6 to the DAI of the MS.

c) The SS measures the time between the first bit on the DAI, and the last transmitted bit of the block at the air interface for each speech block the SS sends on the DAI. This time difference is the delay measured.

d) Step c) is repeated 20 times. The maximum delay measured in ms is the uplink coding delay TUP.

NOTE: This is to account for the fact that the processing time may not be constant.

32.10.7 Uplink coding delay

32.10.7.1 Definition and applicability

The uplink coding delay is defined as the delay between an acoustic signal at the MRP and the digital representation of that signal on the DAI.

The requirements and this test applies to all MS supporting TCH/HS.

32.10.7.2 Method of test

32.10.7.2.1 Initial conditions

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

32.10.7.2.2 Procedure

a) The SS generates an acoustic signal at the artificial mouth of the LRGP, being a pure sine wave with a frequency of 1 000 Hz.

b) The SS measures the "phase shift" $\varnothing_1$, in the range of 0 to 360 degrees, between the signal at the MRP and its digital representation on the DAI.

c) The SS sets the generated frequency to 1 100 Hz, and measures the resulting phase shift $\varnothing_2$.

d) The uplink coding delay TUC is calculated from either:

$TDC = (\varnothing_2 - \varnothing_1)$ ms/36 for $\varnothing_2 > \varnothing_1$

or

$TDC = (\varnothing_2 + 360 - \varnothing_1)$ ms/36 for $\varnothing_2 < \varnothing_1$

32.10.8 Test requirement

The sum of the delays TDP, TDC, TUP, and TUC shall be less than 144,9 ms.

NOTE: This limit includes an allowance of 4*0,25 ms delay from the DAI to the MS transmission path.

32.11 Intra cell channel change from a TCH/HS to a TCH/FS

32.11.1 Definition and applicability

Dual rate MSs support an intra cell channel change from a TCH/HS to a TCH/FS by switching the Speech and channel codec used from HR to FR.

The requirements and this test apply to all MS supporting TCH/HS.

32.11.2 Conformance requirement:

1) When commanded to perform an intra cell channel change from a TCH/HS to a TCH/FS, the MS shall switch channels from HR to FR. The maximum time allowed for the MS to perform this switch in rates is 20 ms.

   GSM 05.10, section 6.8.

2) For an intra cell channel change, the time between the end of the last complete speech or data frame or message block sent on the old channel and the time the MS is ready to transmit on the new channel shall be less than 20 ms.

   GSM 05.10, section 6.8.

32.11.3 Test purpose:

1) To verify that the MS encodes speech correctly after performing an intra cell channel change from a TCH/HS to a TCH/FS.

2) To verify that the MS, when commanded to perform an intracell channel change to a new ARFCN and/or timeslot number within the same cell, if the starting time is not used in the ASSIGNMENT COMMAND, is ready to transmit on the new channel within 20 ms of the last complete speech or data frame or message block sent on the old channel.

32.11.4 Method of test

32.11.4.1 Initial conditions

Uplink DTX is off.

The SS sets up a call according to the generic call set up procedure on a HR channel in the low ARFCN range on timeslot 1.

32.11.4.2 Procedure

a) The SS records the sequence of BFI flags obtained by channel decoding the incoming bit stream from the uplink air interface using the HR channel decoder on the old channel and at the same time records the sequence of BFI flags obtained by channel decoding the incoming bit stream from the uplink air interface using the FR channel decoder on the channel to which the channel change will take place.

b) The SS sends an ASSIGNMENT COMMAND to the MS allocating a FR channel in the high ARFCN range on timeslot 2, and with a power command of 7. These old and new carriers have a relative frequency tolerance of 0, and a relative timing tolerance of 1/4 bit.

c) The time at which the sequence of BFI flags at the output of the HR channel decoder performs the first transition from 0 to 1 is registered (t_1). In case of occurrence of speech frames after an RR frame, the next transition of the BFI flag from 0 to 1 after the reception of the RR frame is defined as t_1.

d) The time values at which the sequence of BFI flags at the output of the FR channel decoder performs transitions from 1 to 0 are registered. The time t_2 is defined as the time where the BFI flag at the output of the FR channel decoder toggles from 1 to 0 due to a correctly received speech traffic frame received at the channel decoder. Transitions due to the occurrence of an ASSIGNMENT COMPLETE frame or an SABM frame after the reception of good speech frames shall not be considered. If the first frame sent on the new traffic channel was an SABM frame, t_2 is defined as the time the BFI flag toggles from 1 to 0 due to a correctly received speech traffic frame after the reception of the SABM frame.

   NOTE: There shall be an allowance of at maximum two transitions for this BFI flag from 0 to 1 and back to zero again after t_2. These transitions are caused by the SABM frame if it was not the first frame to be sent on the new TCH, or the ASSIGNMENT COMPLETE frame, or both. Since both frames are FACCH frames, each would cause exactly one BFI=1 indications.

e) The time difference $Dt = t_2 - t_1$ shall be calculated.

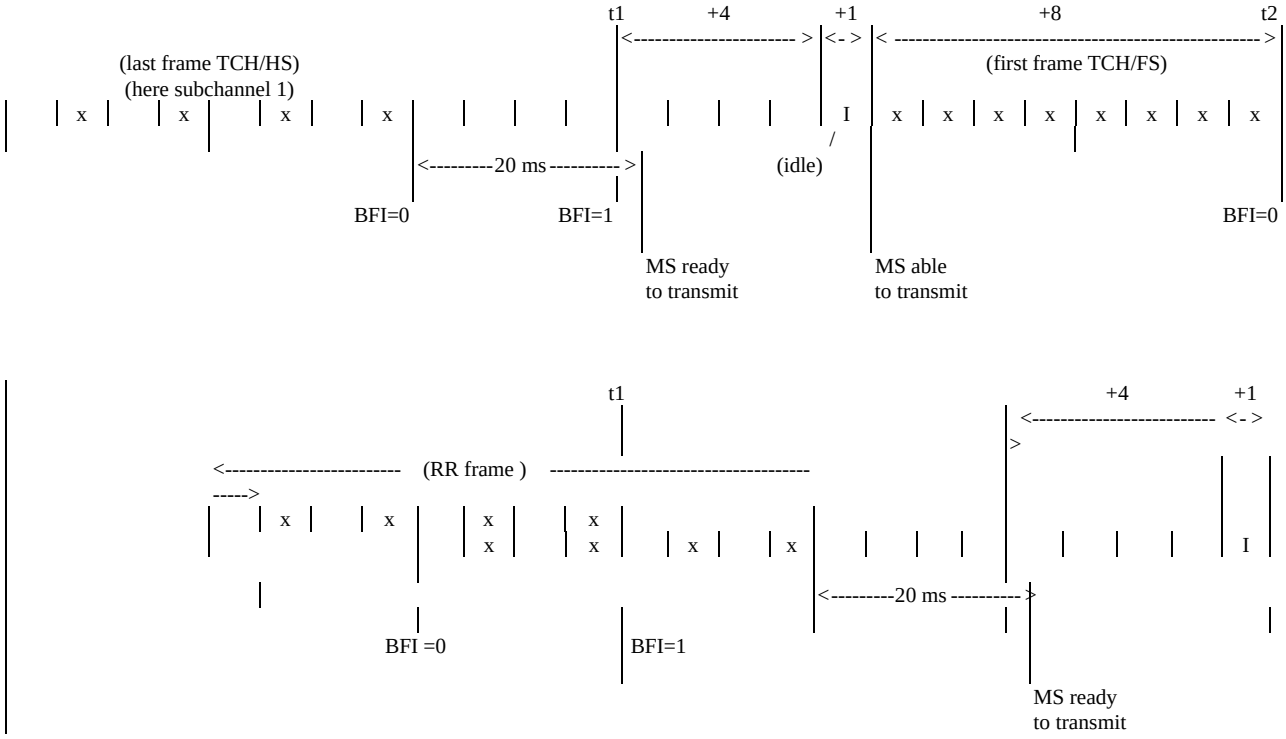
32.11.5 Test requirement

1) The last transition of the BFI flag at the output of the FR channel decoder from 1 to 0 shall be followed by a sequence of at least 50 zeroes, interrupted by at maximum two transitions to 1, each interruption containing exactly one BFI=1 flag, caused by the SABM or the ASSIGNMENT COMPLETE frames.

2) The calculated time difference Dt shall not exceed 13 TDMA frames. If the first frame sent on the new channel was an SABM frame, an additional time difference of 4 frames is allowed. If the last frame sent on the old channel was an RR frame, an additional time difference of 9 frames is allowed.

NOTE: The BFI of the old channel will toggle from 0 to 1 only four frames after the reception of the last bit of the speech frame sent on the old channel. The time between the last bit of the last complete speech or data frame or message block sent on the old channel and the time the MS is ready to transmit on the new channel shall be less than 20 ms (5.10, section 6.8). This time will expire 4 frames and 3 timeslots after the sending of the last bit of the last complete speech frame on the old channel, i.e. the MS may not be able to transmit in the corresponding timeslot in the current frame, but must wait approx. 4 frames until the next allowed frame (FN mod 13 = 0, 4 or 8) is reached.
The next frame could be an idle frame, so the MS must wait for another frame. This equates to 5 frames, after which the MS is able to start transmission on the new channel.
Additionally, 8 frames will be needed due to interleaving until the last bit of the first speech frame on the new channel is received and the BFI flag toggles from 1 to 0.
This makes a total of 13 frames or 60 ms between the frame number when the BFI toggles from 0 to 1 on the old channel and the frame number when the BFI toggles from 1 to 0 on the new channel. See diagram below.
If SABM is the first frame received on the new channel, 4 more frames are allowed.

If RR is the last frame sent on the old channel, additionally 9 more frames are allowed (RR frames plus an idle frame).



32.12 Intra cell channel change from a TCH/FS to a TCH/HS

32.12.1 Definition and applicability

Dual rate MSs support an intra cell channel change from a TCH/FS to a TCH/HS by switching the Speech and channel codec used from FR to HR.

The requirements and this test apply to all MS supporting TCH/HS.

32.12.2 Conformance requirement:

1) When commanded to perform an intra cell channel change from a TCH/FS to a TCH/HS, the MS shall switch channels from FR to HR.

GSM 05.10 section 6.8.

2) For an intra cell channel change, the time between the end of the last complete speech or data frame or message block sent on the old channel and the time the MS is ready to transmit on the new channel shall be less than 20 ms.

GSM 05.10, section 6.8.

32.12.3 Test purpose:

1) To verify that the MS encodes speech correctly after performing an intra cell channel change from a TCH/FS to a TCH/HS.

2) To verify that the MS, when commanded to perform an intra cell channel change to a new ARFCN and/or new timeslot number within the same cell, if the starting time is not used in the ASSIGNMENT COMMAND, is ready to transmit on the new channel within 20 ms of the last complete speech or data frame or message block sent on an old channel.

32.12.4 Method of test

32.12.4.1 Initial conditions

Uplink DTX is off.

The SS sets up a call according to the generic call set up procedure on a FR channel in the low ARFCN range on timeslot 1.

32.12.4.2 Procedure

a) The SS records the sequence of BFI flags obtained by channel decoding the incoming bit stream from the uplink air interface using the FR channel decoder on the old channel and at the same time the sequence of BFI flags obtained by channel decoding the incoming bit stream from the uplink air interface using the HR channel decoder on the channel to which the channel change will take place.

b) The SS sends an ASSIGNMENT COMMAND to the MS allocating a HR channel in the high ARFCN range on timeslot 2, and with a power command of 7. These old and new carriers have a relative frequency tolerance of 0, and a relative timing tolerance of 1/4 bit.

c) The time at which the sequence of BFI flags at the output of the FR channel decoder performs the first transition from 0 to 1 is registered (t_1). In case of occurrence of speech frames after an RR frame, the next transition of the BFI flag from 0 to 1 after the reception of the RR frame is defined as t_1.

d) The time values at which the sequence of BFI flags at the output of the HR channel decoder performs transitions from 1 to 0 are registered. The time t_2 is defined as the time where the BFI flag at the output of the FR channel decoder toggles from 1 to 0 due to a correctly encoded speech traffic frame received at the channel decoder. Transitions due to the occurrence of an ASSIGNMENT COMPLETE frame or and SABM frame after the reception of good speech frames shall not be considered. If the first frame sent on the new traffic channel was an SABM frame, t_2 is defined as the time the BFI flag toggles from 1 to 0 due to a correctly received speech traffic frame after the reception of the SABM frame.

   NOTE: There shall be an allowance of at maximum two transitions for this BFI flag from 0 to 1 and back to zero again after t_2. These transitions are caused by the SABM frame if it was not the first frame to be sent on the new TCH, or the ASSIGNMENT COMPLETE frame, or both. Since both frames are FACCH frames, each would cause exactly two BFI=1 indications.

e) The time difference $Dt = t_2 - t_1$ shall be calculated.

32.12.5 Test requirement

1) The last transition of the BFI flag at the output of the HR channel decoder from 1 to 0 shall be followed by a sequence of at least 50 zeroes, interrupted by at maximum two transitions to 1, each interruption containing exactly two BFI=1 flags, caused by the SABM or the ASSIGNMENT COMPLETE frames.

2) The calculated time difference Dt shall not exceed 12 TDMA frames. if the first frame sent on the new channel was an SABM frame, an additional time difference of 9 frames is allowed. If the last frame sent on the old channel was an RR frame, an additional time difference of 5 frames is allowed.

NOTE: The BFI of the old channel will toggle from 0 to 1 only 4 frames after the reception of the last bit of the last speech frame sent on the old channel. The BFI on the old channel will toggle from 0 to 1 only four frames after the reception of the last bit of the last complete speech or data frame or message block sent on the old channel.

The time between the last bit of the last complete speech frame sent on the old channel and the time the MS is ready to transmit on the new channel shall be less than 20 ms (5.10, section 6.8). This time will expire 4 frames and 3 timeslots after the sending of the last bit of the last complete speech frame on the old channel, i.e. the MS may not be able to transmit in the corresponding new timeslot in its current frame, but must wait approx. 4 frames until the next allowed frame (FN mod 13 = 0, 4 or 8) is reached.

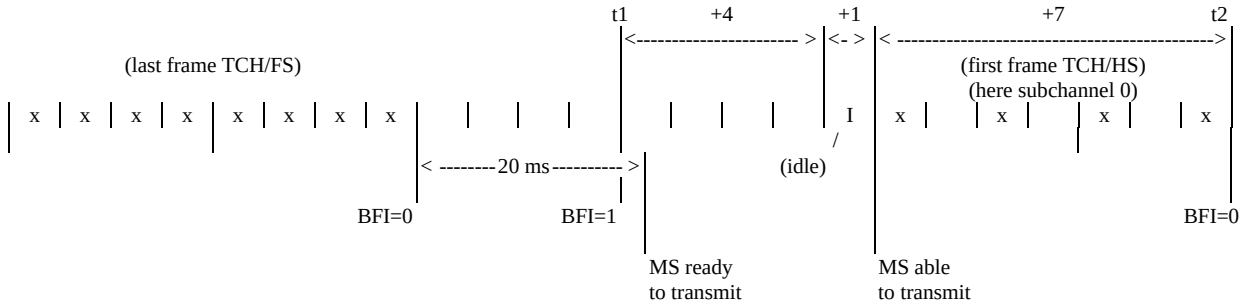
The next frame could be an idle frame, so the MS must wait for another frame. This equates to 5 frames, after which the MS is able to start transmission on the new channel.

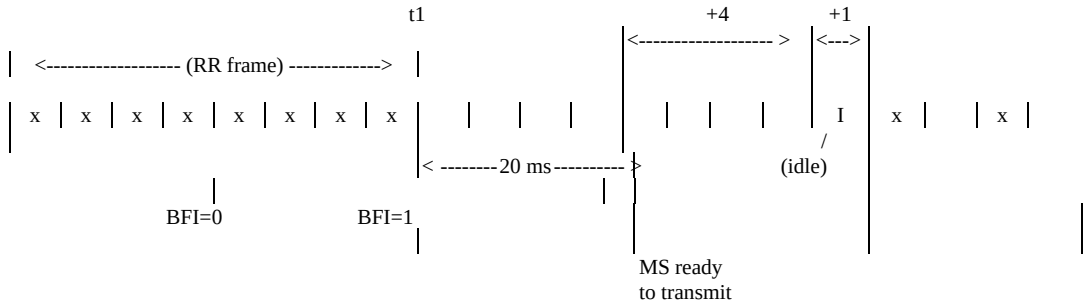
Additionally, 7 frames will be needed due to interleaving until the last bit of the first speech frame on the new channel is received and the BFI flag toggles from 1 to 0.

This makes a total of 12 frames or 55,4 ms between the frame number when the BFI toggles from 0 to 1 on the old channel and the frame number when the BFI toggles from 1 to 0 on the new channel. See diagram below.

If SABM is the first frame received on the new channel, 9 more frames are allowed.

If RR is the last frame sent on the old channel, 5 more frames are allowed.


t1
+4
+1
+7
t2
(last frame TCH/FS)
(first frame TCH/HS)
(here subchannel 0)
x x x x x x x x I x x x x
(idle)
< --------20 ms---------- >
BFI=0
BFI=1
BFI=0
MS ready
to transmit
MS able
to transmit


t1
+4
+1
<---------- (RR frame) ---------->
x x x x x x x x I x x
(idle)
< --------20 ms---------- >
BFI=0
BFI=1
MS ready
to transmit