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TCSM3i Functionalities

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1 Overview of TCSM3i functionalities

TCSM3i Functionalities describes the transcoding function of TCSM3i. The following sections describe the functionalities of TCSM3i:

- *Speech coding*
- *Basic functions and features of a TRAU*
- *Operation and maintenance*
- *Additional features*

1.1 Use of the terms TRAU, TCSM, TCSM3i, TR3E, and TR3A

The transcoding and rate adaptation unit (TRAU) is used to identify the entity responsible for processing one speech/data user channel in the GSM/EDGE network as specified in the 3GPP specifications.

Transcoder submultiplexer (TCSM) is a functional unit of the Nokia BSC responsible for transcoding, data rate adaptation, and submultiplexing the traffic channels carried by a single PCM (E1/T1) circuit between the BSC and the transcoding site. One TCSM contains several TRAU functions. The TCSM units are functional units of the BSC, but they can be physically located either at the BSC or the MSC site.

TCSM3i equipment is the third generation Nokia transcoder submultiplexer. It consists of up to 96 TCSM functional units of one or more BSCs. It implements the TCSM functional units with a group of cartridges and their plug-in units installed in a cabinet. TR3A stands for the American National Standards Institute (ANSI) version and TR3E for the European Telecommunications Standards Institution (ETSI) version of the transcoder plug-in unit.

1.2 Introduction to TCSM3i functionalities

The actual transcoding and rate adaptation function is done in the TR3E (transcoder Plug-in unit for ETSI, 120 channels) or the TR3A (Transcoder Plug-in unit for ANSI, 95 channels). The control unit is integrated to the same plug-in unit. One TR3E/A plug-in unit contains 120/95 transcoding and rate adaptation functions, that is, TRAU, each of which is implemented with a channel process in a Digital Signal Processor (DSP).

Transcoder software for c55xx DSP (T55PRB), that is, TRAU software, is capable of handling 8 kbit/s, 16 kbit/s, 32 kbit/s, 48 kbit/s, and 64 kbit/s Ater traffic channels. These can contain the following traffic channels:

- 16 kbit/s full rate speech (FR and enhanced full rate (EFR) speech coding)
- 16 kbit/s full rate data (FR data: 14.4, 12, 6, 3.6 kbit/s)
- 8 kbit/s half rate speech (HR speech coding)
- 16 kbit/s AMR full rate speech
- 8 kbit/s AMR half rate speech
- 16-32 kbit/s High Speed Circuit Switched Data (HSCSD max 2 × FR data (HS2))
- 16-64 kbit/s High Speed Circuit Switched Data (HSCSD max 4 × FR data (HS4))
- Additional features Acoustic Echo Cancellation (AEC), Tandem Free Operation (TFO), Noise Suppression (NS) and Text Telephony (TTY) are supported.



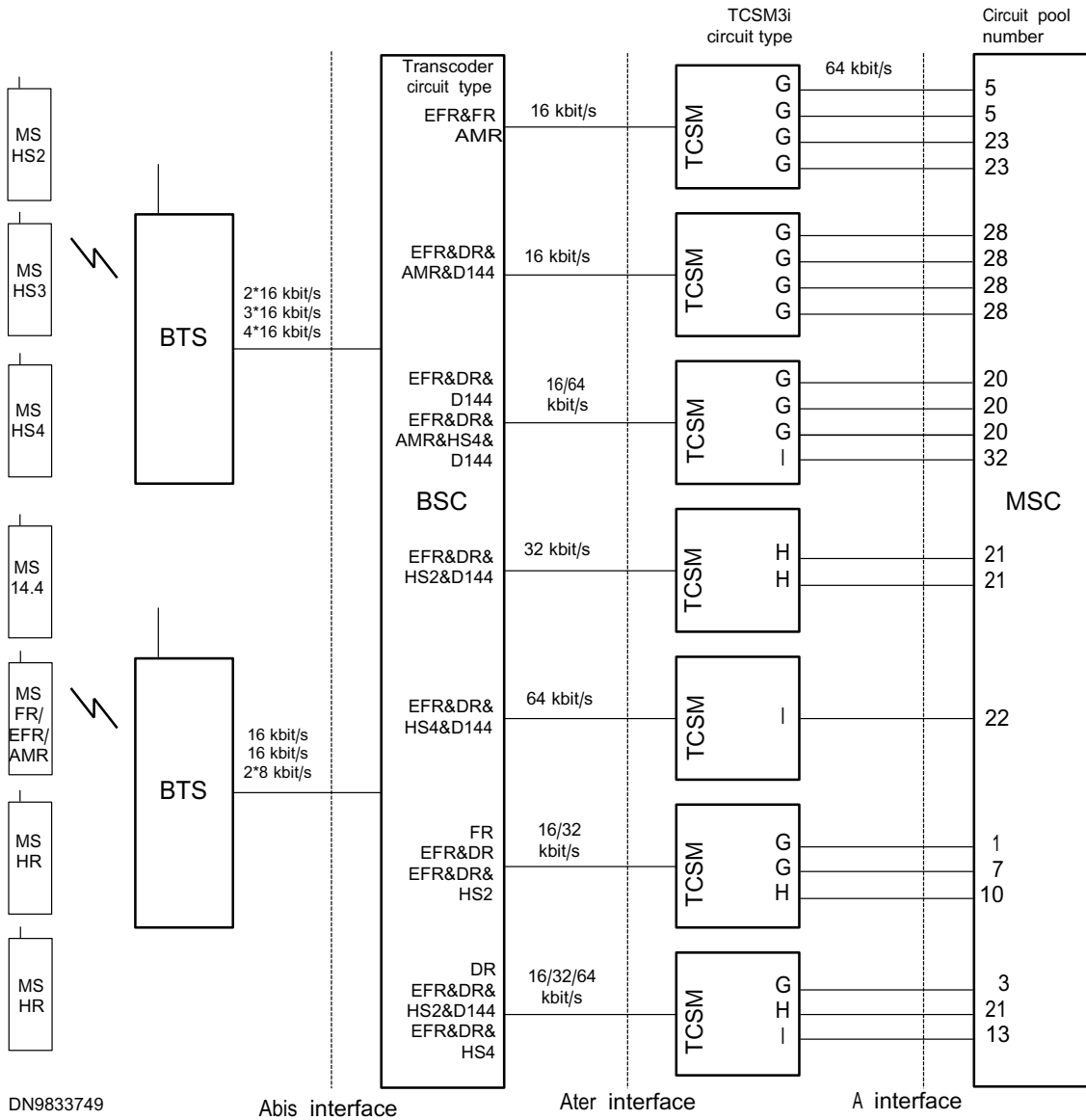
Note

Tandem Free Operation (TFO) for AMR is not supported.

It is possible to configure each A interface PCM of a TCSM unit in TCSM3i with operations and maintenance (O&M) commands to be one of three circuit types (G, H, and I). TRAU has the ability to handle traffic channels below the maximum configured rate in real time, according to the control information received from the Base Station Controller (BSC). The following circuit types are supported:

- “G” circuit type means that the Ater subchannel handled by the TRAU is a 8-16 kbit/s channel, containing FR, EFR, HR or AMR speech or FR data traffic.
- “H” circuit type means that the Ater subchannel handled by the TRAU is an 8-32 kbit/s channel, containing FR, EFR, HR or AMR speech or FR data traffic or HSCSD max 2 × FR data traffic.
- “I” circuit type means that the Ater subchannel handled by the TRAU is an 8-64 kbit/s channel, containing FR, EFR, HR or AMR speech or FR data traffic or HSCSD max 4 × FR data traffic.

There are always two HR traffic channels or one FR traffic channel in one 16 kbit/s subchannel between the BSC and the Base Transceiver Station (BTS), that is, in the Abis interface. In case of circuit types "G, H, and I", the TRAU handles only one HR traffic channel or one FR traffic channel in the lowest 8 or 16 kbit/s subchannel respectively in the Ater interface. This is depicted in the figure: *1 An example of the TCSM3i configurations and different traffic channels.* (The figure is only an example and does not show all possible configurations).



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Figure 1. An example of the TCSM3i configurations and different traffic channels

The 3GPP specifications also recognise that 16 kbit/s submultiplexing with 16 kbit/s HR TRAU frames can be used for HR. This kind of submultiplexing wastes transmission capacity and it has, therefore, not been implemented. Here, the term HR only refers to HR with 8 kbit/s subchannels.

The TRAU software is mainly based on the 3GPP specifications, which determine most of the features required of the software. Some of the features not described in the 3GPP specifications are defined in the ITU-T standards.

Related topics

Product Description of Nokia TCSM3i High Capacity Transcoder Submultiplexer

Commissioning TCSM3i

Engineering for TCSM3i

TCSM3i User Commands

2

Speech coding

For a general introduction, see *Overview of TCSM3i functionalities*.

The speech compression method used in the GSM system for FR is Regular Pulse Excitation - Long Term Prediction (RPE - LTP) coding, which uses 8th order Linear Predictive Coding (LPC) analysis for short term prediction, 1-tap inverse filter for long term prediction, and decimation with a factor of three, resulting in a bit rate of 13 kbit/s.

TheEFR compression method used is 12.2 kbit/s Algebraic Code Excited Linear Prediction (ACELP). The HR speech compression method is also a Code Excited Linear Prediction (CELP) type Vector Sum Excited Linear Prediction (VSELN) coding. HR speech compression uses the analysis by synthesis method, with fixed code books and 10th order LPC for short term prediction and adaptive code book with fractional lags for long-term prediction, resulting in a bit rate of 5.6 kbit/s.

The AMR compression method used is Multi-Rate ACELP (Algebraic Code Excited Linear Prediction). The AMR codec uses eight source codecs with bit rates of 12.2, 10.2, 7.95, 7.40, 6.70, 5.90, 5.15, and 4.75 kbit/s.

The input for all coders is a stream of uniform 13-bit samples that are converted from 8-bit A-law/ μ -law-coded PCM samples. Detailed information about the speech compression methods can be found in 3GPP specifications.

3

Basic functions and features of a TRAU

For a general introduction, see *Overview of TCSM3i functionalities*.

This section describes most of the basic functions of a TRAU. In this section, the term TRAU is used to refer to a channel process in a digital signal processor on a TR3E or TR3A plug-in unit and all its features, according to the 3GPP specifications. Exceptions to these specifications are found in the subsequent text and also in the Nokia Statement of Compliance which also includes some additional exceptions to the 3GPP specifications relating to transcoding.

3.1 Interfaces

A TRAU has a 64 kbit/s PCM interface towards the MSC and an 8, 16, 32, or 64 kbit/s GSM interface towards the BTS. These interfaces are called the A interface and Abis interface, respectively. The terms uplink direction and downlink direction are used when describing the functionalities of a TRAU, as they are used in the 3GPP specifications. The uplink direction is the one coming from the BTS and going towards the MSC. The downlink direction is the opposite of this (MSC to BTS). In addition to the PCM and GSM interfaces, the TRAU has a separate Operation and Maintenance interface to the controller unit of TR3E/A which is used for transmitting O&M messages.

The information is transferred between the BTS and the TRAU in TRAU frames consisting of either 320 bits or 160 bits, and therefore taking 20 milliseconds to transfer one TRAU frame at the transfer speed of 16 kbit/s or 8 kbit/s, respectively. During HSCSD operation, up to four 16 kbit/s data traffic channels are transferred to a TRAU simultaneously.

For the synchronisation of the FR/EFR frames there is a pattern of 16 zeros at the beginning of each frame. Depending on the type of the frame, speech or data, the first bit of the 16-bit or 8-bit words which generate the frame, is a synchronisation bit.

For the synchronisation of the HR frames, there is a pattern of 8 zeros at the beginning of each frame. The first bit of the 8-bit words that generate the frame is a synchronisation bit. The only exception is the third octet, where the first two bits are permanently set to values zero and one.

In addition to the actual data bits containing speech or data information, there are also control bits in all the frames. The control bits contain data about the type of the frame and a varying amount of other type-specific information. In speech frames, the last four bits in FR and the last two bits in HR are reserved for time alignment purposes.

There are six types of FR/EFR frames defined for FR based on the type of information they contain: speech frame, data frame, extended data frames, O&M frame, and idle speech frame. However, the last two frame types have not been implemented in the current transcoding software as they are unnecessary for the operation itself.

There are two types of frames defined for HR depending on the type of information they contain: speech frame, data frame, and O&M frame. The O&M frame is not implemented in the TRAU software. The group of control bits is divided into control bits and extended control bits.

There are four different frame types used for AMR speech from synchronisation point of view. Three of them are used in AMR-HR (AMR in 8 kbit/s traffic channel). Two of them are dedicated frames for AHS 6.70 and AHS 7.40 modes where synchronisation bits are reduced to fit the speech parameter bits in the frame. AHS means AMR-HR and stands for AMR Half Rate Speech

In AMR speech, there are also so called no speech frames, which are used for Discontinuous transmission (DTX) and control purposes. Their role in AHS is significant, as there are very few control and synchronisation bits in AHS speech frames.

3.2 Resource allocation and release

When the TRAU is not performing any operation, it is in idle state. It announces this with an O&M message to the controller unit of TR3E/A. It also transmits idle pattern, that is, 01010101, 0101, or 01 depending on the Ater circuit type, in the downlink direction, and 01010100 (in systems of the ETSI type) or 01111111 (in systems of the ANSI type) in the uplink direction.

When the call is set up, the BSC reserves one TRAU and transmits information about the set-up of the call to the BTS. After receiving the call set-up request, the BTS reserves the capacity required for the transmission of the call from the radio interface and starts to transmit TRAU frames that correspond to the type of the call to the TRAU (in the uplink direction). Upon receiving a frame, the TRAU starts to transmit similar frames in the downlink direction and the call has been set up.

Even during a call, it is possible to change the type of the call by simply changing the type of TRAU frames coming in the uplink direction.

When the call is terminated, the TRAU enters idle state after receiving idle pattern for a duration of three consecutive TRAU frames. In FR, the idle pattern is a 01 or 10 combination in the 16 kbit/s subchannel. In HR, the idle pattern is either all zeros or all ones, depending on the location of HR TRAU frame within the 16 kbit/s subchannel (whether in the first bit or the second). Therefore, the idle pattern of the first 16 kbit/s subchannel is either 01, 10, or 11, because the BSC fills the unused subchannel with ones, and there may be either one or two subchannels in idle state. The idle pattern for circuit types "H" and "I" is 01 or 10 in each subsequent 16 kbit/s subchannel.

According to the 3GPP specifications, the TRAU should enter idle state after receiving the idle bit pattern for one second. However, a shorter time has been chosen because it is possible that a new call starts during the one second period, therefore making it impossible to set speech and state variables and tables to the correct initial values.

3.3 Transmission and reception of idle frames

Even if no speech is received in the downlink direction, encoding continues as usual. According to the 3GPP specifications, idle speech frames should be transferred whenever no speech or data is received in the downlink direction. Such a situation exists when only a stream of continuous ones or an idle bit pattern is received in the downlink direction. However, this is a matter of interpretation. The TRAU software has been implemented in such a way that in the speech mode of operation, the signal received in the downlink direction is encoded and speech frames are sent as usual. In the data mode of operation, in the event that V.110 synchronisation is lost, this is interpreted as "no data in the downlink direction", meaning that idle data frames are sent, that is, data frames in which all data bits have been set to the value one. In case of AMR, the no-speech frames are used as idle frames.

When idle speech frames or bad frames are received in the uplink direction, an attenuation or replacement operation is performed within 120 milliseconds.

When an idle data frame is received, the transmission of data is continued because frame synchronisation has not been lost. As all data bits in the idle data frame are set to binary ones, the frame is coded automatically in the uplink direction as a V110 frame full of binary ones, that is, eight synchronisation zeros and the following 72 bits as ones. This means that also all status and control bits of the V110 frame are transmitted as ones.



Note

Idle speech frames do not exist in HR.

3.4 Time alignment of speech frames

Time alignment is done by means of control bits, time alignment bits, and stop bits that are placed in the space between the frames. The BTS calculates the required adjustment and transmits this information to the TRAU in the TRAU frame control bits reserved for time alignment. Apart from the exceptions mentioned later in this text, the TRAU rewrites the control bits of the next downlink frame with a copy of the adjustment value. It then adjusts the timing of the next frame accordingly.

In the case of HR, the adjustment value of the control bits is not copied to the downlink frame. Instead, the BTS notices that the adjustment has been made by directly monitoring the timing of the downlink frames. If timing is to be delayed, stop bits, that is, binary ones are placed between the frames. In FR, at least eight frames must be received following a change in timing, before the timing can be adjusted again. In FR, time alignment is performed in two different states: initial time alignment state and static time alignment state. In FR, the required adjustment is performed in steps of -250 microseconds, +250 microseconds, or 500 microseconds. In HR, the adjustment is made by combining -250, +250, +500, +1, +3, +6 and +9 microsecond adjustments. Time alignment, however, is not performed at the beginning of the call (during the first 10 seconds, when the TRAU detects the ringing tone of a call) to avoid deterioration of the tone.

In AMR, the time alignment procedure is basically the same as in FR/EFR. In AHS, the no-speech frames have to be used to give the TA command to the TRAU due to lack of control bits in AHS speech frames. TRAU makes the adjustment but does not acknowledge it with a no-speech frame. One of two speech bits have to be stolen if an advance is made. If Time Alignment Extension (TAE) is used with AHS, adjustments can be made to the accuracy of 125 μ s.

The TRAU enters the initial state of time alignment in several cases. The change of state can be caused by a reset of the unit, entry into idle state, loss of synchronisation, or an internal change of the BTS, that is, an internal handover in the BSS. In the initial state, the frames are delayed by 250 microseconds or $N \times 500$ microseconds, where N is between 0 and 39. If the request to advance the timing is received in this state, the frame is delayed by 39×500 microseconds. The state is changed from initial state to static state when one time alignment differing from zero and two consecutive timing adjustments fewer than 500 microseconds has been performed including zero TA command. According to the 3GPP specifications, the state should be changed from initial to static immediately, if two timing adjustments that are less than 500 microseconds have been made. This has been done differently to avoid a change to static state at the beginning of the call when the TRAU receives frames that have a zero time alignment request.

In the static state, an advance or a delay of 250 microseconds can be made. When the TRAU receives a request to advance the timing by 250 microseconds, it does not transmit the last four bits of the frame, that is, the time alignment bits in the uplink direction, and it bypasses two PCM samples in the downlink direction. If the timing is to be delayed by 250 microseconds, four stop bits are added between the frames to be sent and two received PCM samples are repeated. If, in this state, it is requested to delay the timing by more than 250 microseconds, then the timing is delayed by no more than 250 microseconds. If, in this state, a change bigger than 4×250 microseconds is detected in the timing of frames coming in the uplink direction, the TRAU only moves to the initial state and no change is made in the downlink direction. In case of AMR, when a big change command is received in static state, the change is made and the state changed to initial.

3.5 Frame synchronisation

The synchronisation into the frames coming in the uplink direction is done with synchronisation zeros and ones in the frames. The synchronisation is performed continuously and the search window is moved according to the changes in timing. The synchronisation is considered lost when at least

three consecutive frames with at least one synchronisation error in each have been received. According to the 3GPP specifications, the loss of synchronisation should cause the muting of decoded speech in the speech state and after this the TRAU should wait for one second before any other procedure is undertaken. In order to avoid unpleasant sound effects, this has been implemented in the software in a faster way.

If the zero pattern of FR synchronisation is not found, it is still supposed that the frame in question is aligned correctly and decoding is continued normally, except that the speech parameters of the previous frame are used. In case there is a synchronisation error in the synchronisation ones coming after the synchronisation zeroes have been found, the frame is rejected instantly as these frame synchronisation errors are detected and the old speech parameters are used. This situation occurs in almost all intra-BSC handovers. In a handover, a TRAU frame is cut abruptly and bits from another TRAU frame, that is, from the target BTS are received, resulting in the concatenation of the two fragments. The TRAU detects this situation and the corrupted part of the frame is replaced by the bits from the previous TRAU frame.

If FR synchronisation is lost, the decoder is reset and idle speech pattern is immediately set in the uplink direction, that is, muting is performed during only two frames and followed by silence. After this the TRAU searches for a new synchronisation for one second and still sends frames in the downlink direction. If the synchronisation or idle speech pattern is not found during this time, the TRAU transmits an urgent alarm pattern (a continuous zero pattern) in the downlink direction to notify the BTS about the problem.

In HR and EFR, the operation is similar, except that instead of an urgent alarm pattern, the loss of synchronisation is immediately indicated in the control bits of a downlink frame.

If AMR synchronisation is lost, the muting is performed in the speech codec and the UFE (Uplink Framing Error) bit is sent in the downlink frames.

In 14.4 kbit/s data mode, the synchronisation is done in two phases. First normal data frames are used to obtain initial synchronisation. The BTS starts to send these synchronisation frames at the beginning of a call and the TRAU responds with similar frames. When the BTS receives these frames it starts to send extended data frames including user data. In response to these frames, the TRAU also begins to transmit extended data frames to the downlink direction. This procedure is repeated every time synchronisation has been lost either by the BTS or the TRAU. Re-synchronisation is always initiated by the BTS and loss of uplink synchronisation is indicated in the control bits of a downlink frame.

In HSCSD modes, the operation is similar to FR&14.4 kbit/s data traffic channels in each 16 kbit/s subchannel.

If synchronisation is lost in the uplink direction during the data state, the idle pattern is sent in the uplink direction.

If the V110 or ATRAU synchronisation is lost in the data state in the downlink direction, then idle data or idle ETRAU frames are sent in the downlink direction.

3.6 Discontinuous transmission (DTX)

DTX is a mechanism which allows the radio transmitter to be switched off during speech pauses. The purpose is to reduce the power consumption of the transmitter, which is important for mobile phones, and to reduce the overall interference level on the radio channels, which affects the capacity of the network. In what follows, DTX operation is explained for FR only. With the other coding methods the mechanisms are quite similar and, therefore, they are not described in detail here.

Discontinuous transmission is implemented by means of three main structural elements. On the transmitting side a Voice Activity Detection (VAD) is required for detecting whether the signal in question contains speech or just background noise. The VAD function has been defined in the 3GPP specifications and it is mainly based on analysing the energy of the signal and spectral changes. In addition, a function for counting the parameters of background noise is required on the transmitting side. On the receiving side, comfort noise is generated on the basis of noise parameters received from the transmitting side, to avoid disturbing variations of complete silence and speech with background noise for the listener on the receiving side. All parts of the discontinuous transmission are mainly based on internal variables of speech codec.

Functions of the transmitting side (TRAU downlink DTX)

The function taking care of discontinuous transmission on the transmitting side is called TX DTX Handler (Transmit DTX). It delivers speech frames to the radio system continuously. The frames are marked with a flag SP (Speech) which is located in the control bits, if the frame in question contains speech, or if the frame is a SID frame (Silence Descriptor), which contains information about the background noise for generating comfort noise on the receiving side. The SP flag is determined on the basis of the VAD flag (Voice Activity Detection), which is received from the VAD unit. When the value of this flag becomes zero, that is, no speech is detected in the signal, the SP flag is also changed into a zero after the number of

frames required for counting the parameters of background noise. Then the transmission unit of the radio system transmits the frame marked with a zero flag and containing noise parameters, after which the radio transmission is terminated. However, the TX DTX Handler continues to transmit frames containing noise information to the radio system which transmits one of these frames to the radio path at certain intervals, to update the noise parameters of the receiving side. When speech is detected later in the signal, the value of the SP flag is changed to a one and continuous transmission restarts.

On the transmitting side of discontinuous transmission, a function for counting the parameters of background noise is required. The transmitting side generates the parameters for background noise with the encoder which was mentioned earlier. The parameters which describe background noise have been chosen from the normal speech parameters which give information about the level and spectrum of the background noise, that is, block maximums and reflection coefficients which have been changed into LAR coefficients. These are further averaged over four speech frames. One common value is counted for the four block maximums during four speech frames. These parameters are sent to the radio path in the way mentioned earlier. All speech parameters which are normally transmitted are therefore not transmitted and some of the parameters are replaced with a SID code word which is generated from 95 zeros. All the other unnecessary parameters are coded into the value zero.

Functions of the receiving side (TRAU uplink DTX)

The discontinuous transmission of the receiving side is handled by RX DTX Handler (Receive DTX). From the radio system it receives frames which are handled on the basis of three flags received in the control bits.

A BFI flag (Bad Frame Indicator) indicates if the frame in question contains usable information, that is, if a frame, for example, has changed so much on the radio path that it cannot be reconstructed in the radio system of the base station, the frame is marked faulty with this flag. When this kind of a bad frame or a frame marked with value one of the BFI flag is received, the speech parameters of the frame in question are replaced with the speech parameters of the previous frame, before decoding. If there are several consecutive frames marked with the BFI flag, the muting operations according to the 3GPP specifications are performed. The only exception to the handling of the BFI flag is uplink DTX, which means that a valid SID update frame has been received, but no normal speech frame with the BFI of the value zero has been received after that. In this kind of uplink DTX situation, the frame equipped with a BFI flag only means that the generation of comfort noise should be continued. An alternative implementation would be to transmit idle frames.

The two-bit SID flag sent by the radio system is classified to a certain class on the basis of errors in the code word contained by the frame. The decision of how the frame is going to be used is based on this classification. If the SID is of the value two and the BFI of the value zero, the frame is a valid SID frame, which can be used for updating noise parameters.

The TAF (Time Alignment Flag) indicates if the frame in question has been used for signalling outside this subsystem, that is, it usually indicates the time of the next SID updating.

The operations related to the generation of comfort noise, such as muting, are performed from the combinations of the three flags mentioned before, as defined in the 3GPP specifications. In general, the generation of comfort noise is started or the comfort noise is updated when a new valid SID frame has been received.

In a similar way, the generation of comfort noise uses the decoder mentioned earlier. The averaged parameters received from the transmitting side are normally used and they remain unchanged until the next update.

3.7 Data transmission

When remote TRAU is used and therefore sub 64 kbit/s channels (multiplexed to one 64 kbit/channel) are used for the communication between the TRAU and the BTS, an extra conversion for rate adaptation of 16 kbit/s, 32 kbit/s, 48 kbit/s, and 64 kbit/s is required.

These rates are implemented in the A interface 64 kbit/s channel, so that 1, 2, 4, 6, or 8 bits of each 8-bit word are used, depending on the data transmission speed. The rest of the bits are set as binary ones. Within 16 kbit/s subchannels, for other rates than 14.4 kbit/s, the data to be transferred also contains an 80-bit V110 frame structure. This is generated from ten 8-bit words from which the first one contains the synchronisation zeros and the rest of the words contain the actual data bits, so that the first one is always a synchronisation bit. For 14.4 kbit/s a frame structure similar to TRAU frames, ATRAU, is used.

12, 6, and 3.6 kbit/s data

The V110 frames are converted into TRAU frames by removing the first 8-bit word containing synchronisation zeros and by generating TRAU frames from the resulting 72-bit entities. In 16 kbit/s speed, the TRAU frame is generated by setting four of these entities consecutively, after the

synchronisation zeros and control bits of the TRAU frame. At lower speeds, the TRAU frame is generated by setting the data bits in the structure of 16 kbit/s speed at the places of the first and third entity and ones at the places of the second and fourth entity. The conversion to the other direction is performed in a similar way, by removing the TRAU frame structure and by adding synchronisation zeros. The information about the used data rate is transferred in the control bits.

14.4 kbit/s data

The 16 kbit/s ATRAU frames are converted into extended data 16 kbit/s TRAU frames by using Framing Pattern Substitution. The Zero Sequence Position (ZSP) method is used. This is done by substituting all 8 bit ZSP fields present in the ATRAU frame by 8 consecutive zeroes and mapping the resulting data bits to extended data frame. The conversion to the other direction is performed in a similar way, by removing all 8 consecutive zero bit sequences by replacing them with a 8 bit ZSP sequence and mapping the resulting data bits to an ATRAU frame.

High Speed Circuit Switched Data (HSCSD)

During HSCSD operation the TRAU does up to four simultaneous 16 kbit/s data rate adaptation functions either for 3.6, 6, 12, or 14.4 kbit/s FR data. The number of subchannels can vary during the call. These subchannels are always connected to the same 64 kbit/s Ater timeslot and one subchannel is always connected to the first subchannel position by the BSC.

The TRAU has two operating modes for HSCSD. The circuit type "H" supports two simultaneous subchannels and circuit type "I" can handle up to four 16 kbit/s subchannels.

4

Operation and maintenance

For a general introduction, see *Overview of TCSM3i functionalities*.

For implementing the features related to operation and maintenance, the TRAU has a two-directional O&M interface with the controller unit of the TR3E/A plug-in unit. Different types of O&M messages are used for controlling the TRAU feature activation and call status monitoring purposes.

The operation of the TRAU is supervised by a watchdog. In the O&M basic command, the controller unit transmits a check bit, which has to be returned with the same value in order to avoid recovery procedures.

The operation of the DSP processor is also supervised by a self test, in which the program counts a check sum from the instruction codes of the program memory and also checks the operation of the data memory by writing and reading it with suitably chosen values. The core is also tested by using address unit and data unit tests.

4.1 Supervision of interfaces

Ater interface line supervision

The Ater interface line is not supervised in the TRAU software, because the idle pattern in the 8 kbit/s channel is a continuous binary 1 or binary 0 at HR. In 16 kbit/s channel, depending on the subchannel that is switched to the TRAU from the Abis interface, the idle pattern is either 01, 10, or 11. Although the line is not supervised, an alarm is raised when the TRAU frame synchronisation has been lost and new synchronisation or idle pattern is not received within one second.

A interface line supervision

This direction is not supervised in the TRAU software, because there is equipment in the PSTN network such as PBXs, which transmit 64 kbit/s AIS signal, although the situation is not erroneous. This kind of supervision would only cause unnecessary alarms.

Acceptance test

TRAU software can be tested with both 13 bit linear and 8 bit A-law or μ -law test sequences. However, these tests are for Nokia's internal testing only.

4.2 Loop tests

Two other tests have also been implemented in the TRAU. In test state zero, data coming from the 64 kbit/s direction is sent back via encoding and decoding, so that the actual loop is generated with, for example, the ET unit of the 16 kbit/s direction. In this test, unlike usually, the encoder starts before the decoder, and the parts of the control bits which are faulty because of the loop back connection are not taken into account.

Test state one has been improved to verify with very high reliability the correct operation of the speech coding functions of the digital signal processor, that is, to verify the correct operation of the DSP device in speech coding related operations. In the improved test state one, the TRAU unit is tested by transmitting a signal generated from a defined base set of PCM samples to the 64 kbit/s direction. The PCM samples are looped back from the local FPGA unit and encoded with HR speech coding. The resulting HR TRAU frames are sent to the Ater direction, so that HR frames are duplicated to the other subchannels depending on the circuit type, and the whole available part of the timeslot is tested. These TRAU frames are looped back from the FPGA unit and decoded with HR speech coding. This is done for a duration of 1000 frames, after which the last resulting 160 PCM samples are compared with the known correct result. It is enough to compare only these last 160 samples, as the internal states of the encoder and decoder are updated based on the 1000 encodings and decodings. In addition to this, both the A interface and Ater interface loop connections are monitored continuously during the test, in order to detect corrupted bit patterns. If a corrupted bit pattern is detected, an O&M message indicating this and the corrupted direction is given to the controller unit. If the comparison between frames fails, then an O&M message indicating failure in both directions is given.

In test state zero, the speech coding method and the TRAU frame type are dependent on the coding law. The speech coding is FR speech coding and the frames are FR TRAU frames with A-law coding and EFR speech coding and the frames are EFR TRAU frames with μ -law coding.

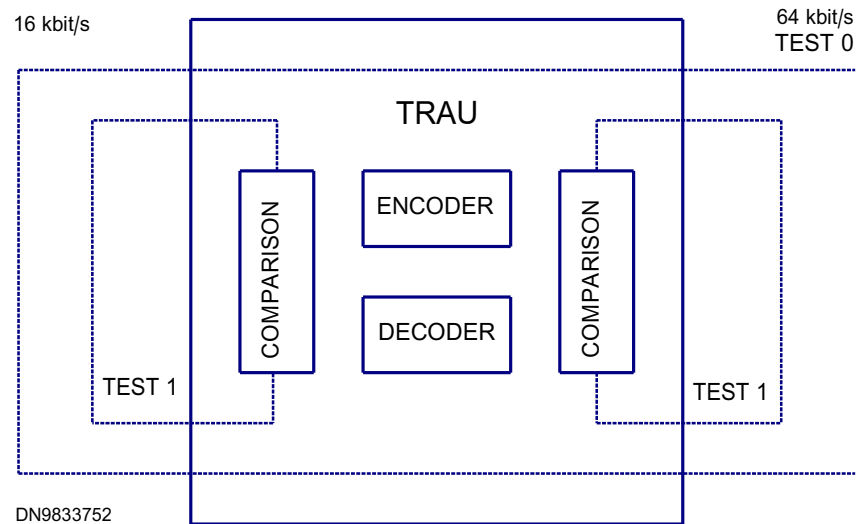


Figure 2. Loop tests.

4.3 Test loops

By means of O&M messages, four kinds of test loops can be set for the TRAU. In test loop one, the data coming from the 16 kbit/s Ater interface direction is sent straight back and in test loop two, the data coming from the same direction goes through decoding and encoding before it is returned. In test loop three, the data coming from the 64 kbit/s A interface direction is transmitted back through encoding and decoding. In test loop four, the 64 kbit/s A interface line is transmitted straight back.

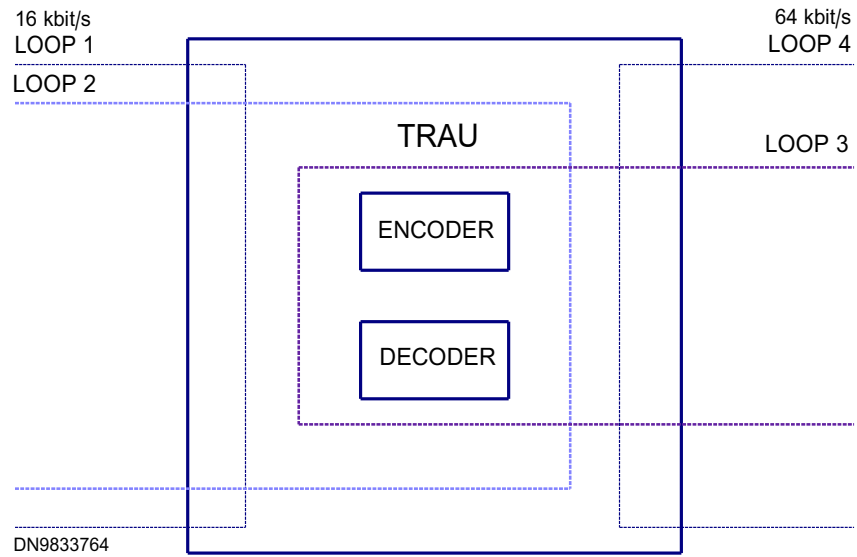


Figure 3. Test loops.

5

Additional features

For a general introduction, see *Overview of TCSM3i functionalities*.

This section describes the additional features of the TRAU software.

Fixed level adjustment

By means of O&M messages, it is possible to choose a fixed level adjustment in the TRAU software in the uplink and downlink directions, between +6 dB and -6 dB, in 1 dB intervals.

Adaptive gain control in the downlink direction

In TRAU software, it is also possible to choose an adaptive gain control in the downlink direction. As an overall solution to the problems caused by a speech volume that is too low in GSM phones and a variable volume level in the PSTN, an adaptive gain has been implemented in the downlink direction to the TRAU, to guarantee sufficient volume level in the Mobile Station (MS). The gain is attenuated immediately in steps of 1 dB, if any clipping is present in the signal. This clipping cannot be heard as the attenuation is done very fast. After the attenuation, there is a short waiting period, after which the amplification is started again slowly in 1 dB steps, if no clipping is present.

The short waiting period is five seconds and the interval in 1 dB steps is one second. If there is no speech present in the downlink direction, then the gain is not increased. The short waiting period is not updated when there is no speech present. This is done to avoid increasing the amplification when the other party of the telephone conversation is silent, as this may cause amplification of acoustic echo in mobile-to-mobile calls.

PCM 8-bit A-law speech samples are converted to linear 13-bit samples as defined in the 3GPP specifications. Amplification is done by multiplying the linear samples by a gain factor. The mechanism detects if the result value of the amplified signal exceeds the range which can be expressed by 13 bits.

Clipping is defined as follows: within a 20 ms period there are 160 samples. Clipping means that two successive values inside a 20 ms period exceed the range. When clipping is detected, the mechanism starts to attenuate the signal. (It decreases the gain factor in 1 dB steps.) Attenuation will be applied in the next 20 ms period.

In adaptive gain control, it is also possible to select the minimum and maximum values within which the gain can vary. The minimum value can vary between 0 dB and 6 dB with integer intervals and the maximum value can vary between 0 dB and 9 dB with integer intervals. The default values are 0 and 6. In the uplink direction the value set for the fixed gain is used.

Speech coding according to μ -law

It is possible to use coding according to A-law or μ -law. This means that the software can be used in an environment according to ETSI or ANSI standards.

64 kbit/s channel through connection

By means of O&M messages, in the TRAU software it is possible to choose a 64 kbit/s through connection. Several continuous through connections can be configured, for example, for wide SS7 signalling channels. Bit rates of 64, 128, 256, and 512 kbit/s are supported.

Time alignment disabling for satellite connections

By means of O&M messages in the TRAU software, it is also possible to set the time alignment off, so that the use of satellite connections is possible, since in the case of long delays the control of time alignment does not work properly.

Bad frame handling improvement (Poor field improvement)

In the 3GPP specifications it is stated that: "Whenever a good speech frame is detected, the DTX handler shall pass it directly onto the speech decoder". This is, however, not a very good solution, for example, when the mobile is in a weak radio field and individual good speech frames are received, surrounded with bad frames. In this case, the good speech frames cause only some arbitrary sounds as they do not contain enough information so that they could be interpreted as speech. It is also more probable that the error detection and correction methods have failed and the frame that has been interpreted as good is actually bad.

Because of this, an improved bad frame handling method is used to prevent short unpleasant sounds. This method works in the uplink direction and it mutes the signal smoothly, if there is evidence of errors and it also recovers smoothly, once good frames are received again.

Handling of frames received with errors (Handover improvement)

This method rejects a frame instantly, if frame synchronisation errors are received, and uses the old speech parameters. This situation occurs in almost all intra-BSC handovers. In a handover, a TRAU frame is cut abruptly and bits from another TRAU frame (from the target BTS) are read, resulting in concatenation of the two fragments. The method detects this situation and the corrupted part of the frame is replaced by the bits from the previous TRAU frame.

Soft comfort noise

In the basic approach presented in the 3GPP specifications, the comfort noise and more specifically, the gain and spectrum of the noise are updated in the receiving part only at 0.5-second intervals when the update is received from the MS. As a result, the comfort noise generated in the TRAU can have fast step-wise changes both in the spectrum and in the volume level. This solution results in unnatural sound that does not resemble real background noise very much. The solution is to interpolate the received comfort noise parameters obtained from the received updating SID frame, to smoothen the comfort noise in the TRAU. This method results in comfort noise that greatly resembles natural background noise. In case of AMR, the SID parameters are updated more frequently and the quality of comfort noise is better.

Acoustic Echo Cancellation (AEC)

Acoustic Echo Cancellation (AEC) is aimed at removing acoustic echo in the uplink direction. The feature operates for FR, HR, AMR, and EFR speech calls, when activated. The AEC support for each codec can be switched on/off using the transcoder MML. Another feature is the possibility to compensate long fixed transmission delays by giving a command from the user interface to adjust the AEC window to the known additional delay (0-620 ms) in 20 ms steps. Additional delay can be controlled by transcoder MML.

Excessive fixed or adaptive downlink gain value settings can reduce the performance of Acoustic Echo Cancellation. Values below 4 dB are recommended with AEC.

The AEC feature is not recommended with transmission that clearly has a one way delay which is longer than 6 ms or with satellite connections in the BSS (where the delay is excessive). The additional delay feature can be used to handle delay caused by any kind of transmission equipment (satellite connections are one example) in AEC operation. However, it may not be sufficient to handle the satellite delay problem for AEC as there can be variable delays in real networks and the estimation of the delay may be difficult for the operator.

The 3GPP specification 43.005 defines a maximum of 188.5 ms for a round trip delay from the PSTN interface to the mobile and back. When we take into account all the delay contributions of different parts of the network, this leaves approximately 6 ms for pure BSS transmission delay. This corresponds to approximately 1000 km of cable transmission equipment. The transcoder and BTS together form a control loop for adjusting the downlink TRAU frame phase for a minimum delay. The stable operation of this controller requires the transmission delay between the transcoder and the BTS to be less than 40 ms. This forms an upper limit for the delay without modifications. The AEC feature has been tuned for maximum performance within the delay limits specified in 3GPP specification 43.005, that is, 6 ms one way (BSS transmission delay) and if the delay is longer than this in the BSS, it can reduce the performance of AEC.

It is not a feasible solution in a GSM network to have AEC performed by normal echo cancellation equipment or by stand-alone AEC equipment. This would cause problems with data calls and would also require additional O&M functionalities. This feature is, therefore, implemented into the TCSM equipment with a software feature and therefore no new equipment is needed.

On the MS (Mobile Station) side, the voice coming from the ear piece of the MS is also picked up by the microphone, that is, there will be an acoustic echo travelling by air and along the body of the MS. 3GPP specification 43.050 states that a handset and a hands-free MS should perform acoustic echo cancellation. In other words, the mobile should have a built-in echo suppressor or canceller and no acoustic echo cancellation should be needed on the network side. However, it seems that some mobiles are not capable of removing the acoustic echo sufficiently and the subscriber may sometimes hear the mobile originated echo. In an MS-to-PSTN call this means that the PSTN subscriber hears his/her own voice as an echo, with a delay of about 200 ms, if the echo cancellation in the MS is not successful. The same problem exists also in MS-to-MS calls although in these cases the delay is normally about 400 ms. In these cases, the echo cancellation should be done by the built-in echo canceller of the MS, as the normal echo cancellation in MSC does not perform any echo cancellation in the uplink direction. Because of the reasons explained above, an AEC feature has been included into the TRAU software.

Tandem Free Operation (TFO)

Tandem Free Operation (TFO) is a procedure, which can be used to enhance speech quality in mobile-to-mobile (MS-MS) calls. In normal MS-MS call the speech signal is encoded in transmitting mobile station and then decoded in transcoding equipment for the fixed part of the network. On the other side of the network another transcoder encodes the speech

samples for transmitting over air interface and the receiving mobile station finally decodes them for listening. From speech coding point of view there are two speech codecs (encoder-decoder pairs) forming a tandem connection in MS-MS call. Because the speech coding methods used in GSM network are lossy, the speech signal is degraded every time it is encoded and decoded. When MS-MS call is formed with TFO, the speech signal is not decoded and encoded between the two transcoders resulting a call in which the speech encoding and decoding is made in mobile stations only.

There are some requirements, which have to be met for a successful TFO call:

- The call is an MS-MS call
- Both transcoders support TFO
- The path between the transcoders is digitally transparent
- The speech codecs used in both radio legs are (or can be changed to be) identical

A digitally transparent path between the two transcoders means that TFO signalling and coded speech samples are not corrupted by any in-path equipment normally used in transmission lines, for example, echo canceller. The in-path equipment must also support TFO, so that they can be commanded to go to a transparent mode for TFO.

In the A interface, the TFO data is exchanged in TFO frames which are as TRAU frames in Abis interface. Some of the control bits are used differently in TFO frames and some of the sync-one bits are changed by embedded TFO messages but the parameterised speech samples are in the same format as in TRAU frames. The bits of the frames are sent in the least significant bits (LSBs) of the PCM samples in A interface (in 2 LSBs in case of Full Rate TFO and in 1 LSB in case of Half Rate TFO). The same speech signal is transmitted in the A interface in the form of ordinary PCM samples and in the TFO frames. The availability of PCM samples guarantees fairly good speech continuity in those situations when TFO discontinues due to local or distant handover. Before the TFO protocol can go to operation state it has to ensure that the requirements of TFO are fulfilled. This is done by sending the so-called in-band signalling messages in A interface. Message bits are sent in the LSB of every 16th PCM sample.

Codec mismatch resolution, which is an optional feature of TFO in the specifications, is not supported in the BSS.

TFO for AMR is not supported. If TFO has been established in a call, the following features do not affect speech even if they are activated: fixed gain, adaptive gain, AEC, and Noise Suppression.

Noise Suppression (NS)

When activated, the Noise Suppression (NS) algorithm processes the uplink and/or downlink speech samples to reduce the background noise level during speech calls. This feature can be activated separately for each speech codec (FR/HR/EFR/AMR) and transmission direction (uplink/downlink) using TCSM3i MML. The level of NS can also be adjusted.

The NS algorithm continuously estimates the background noise power spectrum based on input speech samples and voice activity detection. The speech signal power spectrum is then modified based on the background noise estimate and additional constraints. This results in increased subjective quality and intelligibility in adverse conditions, with no perceptible degradation in clean speech.

The tuning of the NS level is optimised to avoid degradation in tandem situations, for example, when NS is implemented in both mobile station and transcoder or both transcoders in a mobile-to-mobile call.

Text Telephony (TTY)

Persons with hearing impairments (defective hearing, deafness) and/or speech defects have been using a specific Text Telephone (or TTY as it is also called in North America) equipment in the fixed network for many years to transmit text and speech through ordinary speech traffic channels. Modern digital cellular systems, however, do not provide satisfactory character error rates for text transmitted in the speech channel with the traditional modulation developed for the fixed network.

The Text Telephone signal can be adapted for use over the radio interface by a combination of a Text Telephone detector and CTM (Cellular Text telephone Modem) transmitter at one end and a corresponding CTM receiver and Text Telephone regenerator at the other end. The combination of a Text Telephone detector/regenerator and a CTM receiver/transmitter is called a CTM adaptor.

The Text Telephone detector converts the audio signals received from a PSTN Text Telephone device into characters. The CTM transmitter transforms the text telephone characters into a signal that can be transmitted robustly via the speech codec and the radio transmission path

of cellular phone systems. The corresponding CTM receiver decodes the signal back into Text Telephone characters. The Text Telephone regenerator transforms the characters back to audio signals that can be sent to a PSTN Text Telephone.

In Nokia's GSM network, the CTM adaptor is in the transcoder with the speech coding software.