Evaluation Report for ETSI UMTS Terrestrial Radio Access (UTRA) ITU-R RTT Candidate

1. Introduction

This document contains the self-evaluation report, provided by ETSI SMG2, for the ETSI UMTS Terrestrial Radio Access (UTRA) RTT proposal to ITU-R.

The document is structured as follows: Section 2 describes the methodology used for the self-evaluation of UTRA. Section 3 describes the studies of other terrestrial RTT proposals that have been conducted as part of the self-evaluation process. The main changes to the UTRA RTT proposal since the June submission are outlined in Section 4 and a summary of the key characteristics of UTRA is provided in Section 5. The conclusions of this evaluation report are contained in Section 6. The majority of this report, including the Evaluation Template, the updated System Description and Technology Description Template, are contained in a number of attachments and these are detailed in Section 7

2. Self-evaluation of the UTRA RTT proposal to ITU-R

The evaluation methodology is given in section 9 of ITU-R M.1225. The proposed radio transmission technology has been evaluated against each evaluation criterion given in section 6.1 of ITU-R M.1225.

Moreover, a 'summary evaluation' has been provided for each criterion. This summary criterion evaluation has been obtained as the outcome of the discussion within ETSI/SMG2 based on the results of the evaluation against each criterion. Therefore, the summary criterion evaluations reflect the common understanding of ETSI/SMG2 on the specific features and capabilities of the proposal for each criterion.

3. Studies of other terrestrial RTT proposals to ITU-R

ETSI has considered other terrestrial RTT proposals, and a list of the key characteristics is contained in the tables attached to this document. In the light of examining these other RTTs, and of previous experience (the ETSI concept group analysis during the development of UTRA) it is considered inappropriate to make comments or other subjective assessments of the RTT proposals, due to the difficulties inherent in the comparison of different systems, especially those with which the analysts are not familiar.

The tables are intended to be informative and the aim is to:

- facilitate the understanding of the various RTT proposals;
- highlight the commonalities between the various proposals;
- highlight the differences between the various proposals;
- assist in identifying areas where clarification or additional information is needed.

The parameters listed are relevant to layer 1 as it was felt that this is where the potential or limitations towards achieving good performance lie. Additionally, it was felt that not enough information was available for a meaningful comparison at layers 2 and 3.

It should be recognised that these tables present a snapshot taken on 15 September, of documents that were frozen before that date - they do not represent a definitive position but are living items that will change with time as the technologies develop.

It should also be stressed that these tables, in which the information has been drawn from a variety of sources, may need additional review to verify compliance with the source documentation. It is not considered appropriate to attempt to achieve a “perfect” summary at this time.

4. Updated version of UTRA RTT proposal to ITU-R

UTRA is currently being developed within ETSI SMG2 and, as a result of this process, a number of modifications have been made to the RTT that was submitted to ITU-R in June 1998. For this reason, an updated version of the UTRA RTT, reflecting the status of work as of September 1998, is included in this report.

The refinements to the RTT submission document since June 1998 are outlined below:

- The System Description has been updated as the details of UTRA have been refined.
- The Technology Description Template (Annex A of the RTT submission) has received very few minor modifications.
- The Requirements and Objectives Template (Annex B of the RTT submission) has been confirmed and extra comments have been added.
- The Capacity and Coverage Results (Annex C of the RTT submission) has been updated in line with the latest simulation results and clarifications requested by other evaluation groups. In particular, this section now includes results for the TDD mode of operation that were not available for the original submission.

5. Key Characteristics of UTRA

- Chip rate

The chip rate is selected as high as possible within a chosen carrier bandwidth (a 5 MHz carrier spacing between uncoordinated operators). Higher chip rate gives higher transmission rate and higher capacity and coverage.

- Frequency re-use

A re-use of one provides for no need for frequency planning.

- Service flexibility through OVSF codes and service multiplexing techniques

OVSF codes will provide less intra-cell interference in downlink and improve performance while still being able to provide users with simultaneous use of bearers for packet and circuit each having different bit rate and quality of service requirements;

- Dedicated channel pilot symbols

Provides for the use of adaptive antenna techniques and coherent detection.

- Asynchronous base station operation

Provides for easy and robust deployment. There is no need for tight time synchronisation requirements of the radio interface, e.g. needing an external time reference network like GPS.

- Support of handover to other systems and support of hierarchical cellular structures

Introducing measurement slots provides means to measure other carrier frequencies of UTRA and other systems, like GSM radios.

- Power control on uplink and downlink

Fast power control techniques is defined for both uplink and downlink. The power control step is variable, e.g. may depend on environment.

- Support of both TDD and FDD operation with harmonised radio parameters between the modes.

The FDD operation is to be used in the paired IMT2000/UMTS frequency band (1920 – 1980 MHz for the uplink and 2110 – 2170 MHz for the downlink) and the TDD operation is assumed to be used in the unpaired part of the IMT2000/UMTS frequency band (1900 – 1920 MHz and 2010 – 2025 MHz). Further the two modes allows for efficient support of asymmetric services.

6. Summary and Conclusion

This evaluation report shows that the ETSI UMTS Terrestrial Radio Access (UTRA) RTT proposal meets and exceeds the minimum performance capabilities, as well as the full set of requirements and objectives for IMT2000 as defined by ITU-R.

Based on information provided in Attachment 7, it is apparent that UTRA has a lot of common elements with other proposals in other regions, and this is the result of successful global participation of organisations, in ETSI, ITU and other fora.

7. Attachments

The evaluation report includes the following attachments:

Attachment	Description
1	Evaluation Template
2	Updated System Description
3	Updated Technology Description Template
4	Updated Requirements and Objectives Template
5	Updated Capacity and Coverage Results
6	Minimum Performance Capabilities
7	Summary tables of key characteristics for RTT proposals based on CDMA technology and TDMA technology

ATTACHMENT 1

Evaluation Template

Index	Criteria and attributes	Q or q[1]	Related attributes in Annex 1	Comments
A3.1	Spectrum efficiency The following entries are considered in the evaluation of spectrum efficiency:			
A3.1.1	For terrestrial environment			
A3.1.1.1	Voice traffic capacity (E/MHz/cell) in a total available assigned non-contiguous bandwidth of 30 MHz (15 MHz forward/15 MHz reverse) for FDD mode or contiguous bandwidth of 30 MHz for TDD mode. This metric must be used for a common generic continuous voice bearer with characteristics 8 kbit/s data rate and an average BER 1×10^{-3} as well as any other voice bearer included in the proposal which meets the quality requirements (assuming 50% voice activity detection (VAD) if it is used). For comparison purposes, all measures should assume the use of the deployment models in Annex 2, including a 1% call blocking. The descriptions should be consistent with the descriptions under criterion § 6.1.7 – Coverage/power efficiency. Any other assumptions and the background for the calculation should be provided, including details of any optional speech codecs being considered.	Q and q	A1.3.1.5.1	**Simulation Results for FDD Mode** INDOOR (A) Uplink: 33.8 E/MHz/cell Downlink: 18.4 E/MHz/cell PEDESTRIAN (A) Uplink: 30.8 E/MHz/cell Downlink: 31.4 E/MHz/cell VEHICULAR (A) Uplink: 22.4 E/MHz/cell Downlink: 17.8 E/MHz/cell
A3.1.1.2	Information capacity (Mbit/s/MHz/cell) in a total available assigned non-contiguous bandwidth of 30 MHz (15 MHz forward/15 MHz reverse) for FDD mode or contiguous bandwidth of 30 MHz for TDD mode. The information capacity is to be calculated for each test service or traffic mix for the appropriate test environments. This is the only measure that would be used in the case of multimedia, or for classes of services using multiple speech coding bit rates. Information capacity is the instantaneous aggregate user bit rate of all active users over all channels within the system on a per cell basis. If the user traffic (voice and/or	Q and q	A1.3.1.5.2	**Simulation Results for FDD Mode** (all figures are expressed in Mbit/s/MHz/cell) **Speech Service** INDOOR (A) 3km/h

[1] **Q** identifies attributes which can be described quantitatively, while **q** identifies attributes which can be described qualitatively

	data) is asymmetric and the system can take advantage of this characteristic to increase capacity, it should be described qualitatively for the purposes of evaluation.			Uplink: 0.135 Downlink: 0.074 <u>PEDESTRIAN (A)</u> Uplink: 0.123 Downlink: 0.125 <u>VEHICULAR (A)</u> Uplink: 0.090 Downlink: 0.071 **<u>LCD Service</u>** <u>INDOOR (A) 3km/h 2048 Kbps</u> Uplink: 0.176 Downlink: 0.047 <u>PEDESTRIAN (A) 384 kbps Using 4 code sets</u>: Uplink: 0.269 Downlink: 0.461 <u>VEHICULAR (A) 144 kbps Using 2 code sets</u>: Uplink: 0.208 Downlink: 0.210 **<u>UDD Service</u>** <u>INDOOR (A) 3km/h 2048 kbps Using 2 codes</u> Uplink: 0.273 Downlink: 0.453 <u>PEDESTRIAN (A) 384 kbps Using 2 code sets</u> Uplink: 0.449 Downlink: 0.668 <u>VEHICULAR (A) 144 kbps</u> Uplink: 0.202 Downlink: 0.290 **<u>Simulation Results for TDD Mode</u>** (all figures are expressed in Mbit/s/MHz/cell) Since it has been found that the downlink is limiting the system

				capacity, only the **downlink** direction is considered. **SPEECH SERVICE** INDOOR Office : 0.073 OUTDOOR TO INDOOR, PEDESTRIAN: 0.148 VEHICULAR: 0.070 **LCD SERVICE** INDOOR Office (2048 kbps): 0.062 OUTDOOR TO INDOOR, PEDESTRIAN (384 kbps): 0.330 VEHICULAR (144 kbps): 0.201 **UDD SERVICE** INDOOR Office (2048 kbps): 0.400 OUTDOOR TO INDOOR, PEDESTRIAN (384 kbps): 0.642 VEHICULAR (144 kbps): 0.320 **Note**: For LCD and UDD services, antenna diversity is used.
A3.1.2*	For satellite environment These values (§ A3.1.2.1 and A3.1.2.2) assume the use of the simulation conditions in Annex 2. The first definition is valuable for comparing systems with identical user channel rates. The second definition is valuable for comparing systems with different voice and data channel rates.			Not applicable.
A3.1.2.1*	Voice information capacity per required RF bandwidth (bit/s/Hz)	Q	A1.3.2.3.1	Not applicable.
A3.1.2.2*	Voice plus data information capacity per required RF bandwidth (bit/s/Hz)	Q	A1.3.2.3.2	Not applicable.

Comments

The RF Carrier bandwidth is typically between 4.2 to 5 MHz, depending on the operation mode. For the present template, the value used in this atribute is 5 MHz.

Based on simulation results, the spectrum efficiency of UTRA exceeds the one provided by current systems.

In UDD mode, the RTT provides higher capacity in the downlink than in the uplink direction. This performance is in accordance with expected demand in wireless multimedia applications.

For the TDD mode, ODMA can increase capacity by reducing the effective path loss, optimum link adaptation and link diversity thus lowering the transmission power and the associated inter-cell interference.

Summary

The RTT Voice Traffic capacity (FDD mode) presents, for Indoor and Vehicular environments, significantly higher efficiency in the uplink. In the pedestrian environment, the efficiency in both directions

is of the same order. In the TDD mode, the downlink direction is the limiting capacity factor.				
A3.2	Technology complexity – Effect on cost of installation and operation The considerations under criterion § 6.1.2 – Technology complexity apply only to the infrastructure, including BSs (the handportable performance is considered elsewhere).			
A.3.2.1	Need for echo control The need for echo control is affected by the round trip delay, which is calculated as shown in Fig. 6. Referring to Fig. 6, consider the round trip delay with the vocoder (D1, ms) and also without that contributed by the vocoder (D2, ms). NOTE 1 – The delay of the codec should be that specified by ITU-T for the common generic voice bearer and if there are any proposals for optional codecs include the information about those also.	Q	A1.3.7.2 A1.3.7.3	Echo Control is required in the Mobile Station. It is not necessarilly required in the fixed network. It is required towards the PSTN.
A3.2.2	Transmitter power and system linearity requirements			
	NOTE 1 – Satellite e.i.r.p. is not suitable for evaluation and comparison of RTTs because it depends very much on satellite orbit. The RTT attributes in this section impact system cost and complexity, with the resultant desirable effects of improving overall performance in other evaluation criteria. They are as follows.			
A3.2.2.1	Peak transmitter/carrier (P_b) power (not applicable to satellite)	Q	A1.2.16.2.1	Not limited by the RTT.
	Peak transmitter power for the BS should be considered because lower peak power contributes to lower cost. Note that P_b may vary with test environment application. This is the same peak transmitter power assumed in Annex 2, link budget template (Table 6).			Limitations may come from Regulation. See also A.3.2.2.2.
A3.2.2.2	Broadband power amplifier (PA) (not applicable to satellite)	Q	A1.4.10 A1.2.16.2.1	Yes, a Broadband Power Amplifier is required.
	Is a broadband power amplifier used or required? If so, what are the peak and average transmitted power requirements into the antenna as measured in watts.		A1.2.16.2.2 A1.5.5 A1.2.5	The peak power is not limited by the RTT. The maximum power needed in the BS is typically 43 dBm/carrier.
A3.2.2.3	Linear base transmitter and broadband amplifier requirements (not applicable to satellite)			
A3.2.2.3.1	Adjacent channel splatter/emission and intermodulation affect system capacity and performance. Describe these requirements and the linearity and filtering of the base transmitter and broadband PA required to achieve them.	q	A1.4.2 A1.4.10	Adjacent channel emission requirements are set for adjacent carriers at 5 and 10 MHz spacing. For good inter-system interference performance, tentative requirements for Adjacent Channel Protection (ACP) are: **Mobile Station:** 40 dB @ 5 MHz 55 dB @ 10 MHz

				Base Station: 55 dB @ 5 MHz 60 dB @ 10 Mhz The limits for spurious emissions at frequencies greater than +/-250% of the necessary bandwidth would be based on the applicable tables from the ITU-R Recommendation SM.329. Further guidance would be taken from the CEPT ERC recommendations that are currently under progress. The possibility for relaxed ACP requirements is under study. **Linearity** For 2 equal power signals being separated by 200 kHz leading to an output level of 6 dB below nominal output level the resulting intermodulation spectrum shall not exceed relative to peak spectrum: -38 dB at 200 kHz to -90 dB at 800 kHz offset from higher/lower frequency signal (linear decrease) <=-95 dB at 1 MHz from higher/lower frequency signal and above
A3.2.2.3.2	Also state the base transmitter and broadband PA (if one is used) peak to average transmitter output power, as a higher ratio requires greater linearity, heat dissipation and cost.	Q and q	A1.4.10 A1.2.16.2.1 A1.2.16.2.2	A new modulation scheme is currently being adopted. It is not possible to give the Peak-to-Average ratio at this moment.
A3.2.2.4	Receiver linearity requirements (not applicable to satellite)	q	A1.4.11 A1.4.12	A linear receiver is required.
	Is BS receiver linearity required? If so, state the receiver dynamic range required and the impact of signal input variation exceeding this range, e.g., loss of sensitivity and blocking.			The 3rd order intercept point is specified between -10 dBm and -5 dBm. The Automatic Gain Control dynamic range is 80 dB.
A3.2.3	Power control characteristics (not applicable to satellite) Does the proposed RTT utilize transmitter power control? If so, is it used in both forward and reverse links? State the power control range, step size (dB) and required accuracy, number of possible step sizes and number of power controls per second, which are concerned with BS technology complexity.	Q and q	A1.2.22 A1.2.22.1 A1.2.22.2 A1.2.22.3 A1.2.22.4 A1.2.22.5	Yes, the RTT utilizes transmitter power control in both forward and reverse links. FDD Mode Dynamic range (downlink): 30 dB Dynamic link (uplink): 80 dB Step size (downlink): 0.25 - 1.5 dB Step size (uplink): 0.25 - 1.5 dB Number of power control cycles: 1600/sec There are no accuracy constraints, since power levels can be dynamically adjusted to compensate for any overshoots or undershoots.

				<u>TDD Mode</u> Uplink: Open loop power control with dynamic range of 80 dB + optional closed loop power control step size of 0.5 - 3 dB. Downlink: Closed loop power control with dynamic range of 30 dB and step size 0.5 - 3 dB. Number of power controls: 100 to 800/sec. There are no accuracy constraints, since power levels can be dynamically adjusted to compensate for any overshoots or undershoots.
A3.2.4	Transmitter/receiver isolation requirement (not applicable to satellite) If FDD is used, specify the noted requirement and how it is achieved.	q	A1.2.2 A1.2.2.2 A1.2.2.1	<u>FDD Mode</u> Required transmit/receive isolation: 80 dB (BS). The required isolation in the BS can be achieved by separating the transmitter and receiver antennas together with an appropriate receiver filter. <u>TDD Mode</u> No duplexer needed.
A3.2.5	Digital signal processing requirements			
A3.2.5.1	Digital signal processing can be a significant proportion of the hardware for some radio interface proposals. It can contribute to the cost, size, weight and power consumption of the BS and influence secondary factors such as heat management and reliability. Any digital circuitry associated with the network interfaces should not be included. However any special requirements for interfacing with these functions should be included. This section of the evaluation should analyse the detailed description of the digital signal processing requirements, including performance characteristics, architecture and algorithms, in order to estimate the impact on complexity of the BSs. At a minimum the evaluation should review the signal processing estimates (MOPS, memory requirements, gate counts) required for demodulation, equalization, channel coding, error correction, diversity processing (including Rake receivers), adaptive antenna array processing, modulation, A-D and D-A converters and multiplexing as well as some IF and baseband filtering. For new technologies, there may be additional or alternative requirements (such as FFTs). Although specific implementations are likely to vary, good sample descriptions should allow the relative cost, complexity and power consumption to be compared for the candidate RTTs, as well as the size and the weight of the circuitry. The descriptions should allow the evaluators to verify the signal processing requirement metrics, such as MOPS, memory and gate count, provided by the RTT proponent.	Q and q	A1.4.13	<u>FDD mode:</u> 8 kbps – 2048 kbps: 5 – 86 million real multiplications (DSP and correlators included, taken the word length requirements relative to the DSP operation into account). <u>TDD mode:</u> It depends on the implementation, e.g. for a 110 kbps service around 15-20 MIPS is needed. Remark: The convolutional encoding and decoding is not included in the figures as it is the same regardless of the multiple access for the same data rate(s).
A3.2.5.2	What is the channel coding/error handling for	q	A1.2.12 A1.4.13	Default: Convolutional inner code (rate 1/3 or rate 1/2, constraint length K=9).

	both the forward and reverse links? Provide details and ensure that implementation specifics are described and their impact considered in DSP requirements described in § A3.2.5.1.			Optional outer Reed-Solomon code (rate TBD) for BER=10^{-6} circuit-switched services. The use of Turbo codes for high-rate services is under consideration and will most likely be adopted. Preliminary results of performance improvement are available. Special FEC schemes, e.g. unequal error protection can be applied.
A3.2.6	Antenna systems			
	The implementation of specialized antenna systems while potentially increasing the complexity and cost of the overall system can improve spectrum efficiency (e.g. smart antennas), quality (e.g. diversity), and reduce system deployment costs (e.g. remote antennas, leaky feeder antennas).			
	NOTE 1 – For the satellite component, diversity indicates the number of satellites involved; the other antenna attributes do not apply.			
A3.2.6.1	*Diversity* : describe the diversity schemes applied (including micro and macro diversity schemes). Include in this description the degree of improvement expected, and the number of additional antennas and receivers required to implement the proposed diversity design beyond and omni-directional antenna.	Q	A1.2.23 A1.2.23.1 A1.2.23.2	<u>Time diversity</u> Channel coding and interleaving in both uplink and downlink. <u>Multipath diversity</u> RAKE receiver, Joint Detection or similar receiver structures with, typically, maximum ratio combining is used in both BS and MS (implementation dependent). <u>Space diversity</u> Receive antenna diversity with, typically, maximum ratio combining can be used in both uplink and downlink. Transmit antenna diversity is under consideration for downlink. <u>Macro diversity</u> Soft (inter-site) handover with, typically, maximum ratio combining in downlink, selection combining in uplink. Softer (inter-sector) handover with, typically, maximum ratio combining in both uplink and downlink. <u>Frequency diversity</u> Wideband carrier (equivalent to multi-path diversity). <u>Improvement due to Diversity</u> For receiver antenna diversity, the diversity gain is 2.5 - 3.5 dB in required E_b/N_o for BER=10^{-3}. If power control is disabled, the gain is much higher for the low speed cases. On top of the gain in reduced required E_b/N_o there is a gain in decreased transmitted power. This gain can be up to 2.5 dB, depending on the environment. Transmit diversity can also be employed, especially in the downlink. A gain similar to the gain with

				receiver antenna diversity is expected. All other diversity methods are inherent parts of the RTT concept and therefore it is difficult to specify an explicit diversity gain figure in dB.
A3.2.6.2	*Remote antennas* : describe whether and how remote antenna systems can be used to extend coverage to low traffic density areas.	q	A1.3.6	Both FDD and TDD modes support Remote Antennas. Although not required by the RTT, they can provide a means for increased performance through spatial diversity.
A3.2.6.3	*Distributed antennas* : describe whether and how distributed antenna designs are used.	q	A1.3.6	Both FDD and TDD modes support Distributed Antennas. Although not required by the RTT, they can provide a means for increased performance through spatial diversity.
A3.2.6.4	*Unique antenna* : describe additional antenna systems which are either required or optional for the proposed system, e.g., beam shaping, leaky feeder. Include in the description the advantage or application of the antenna system.	q	A1.3.6	Both FDD and TDD operating modes of UTRA are able to use all the standard types of Base Station antennas. This includes those that provide omni-directional, sectored, fixed or variable patterns. Directive Antennas decrease the interference, leading to an increase in system capacity. Both FDD and TDD modes support Smart Antenna Systems (Beam Shaping). Signal-to-interference-plus-noise-ratio (SINR) can be improved significantly by incorporating various smart antenna concepts on the uplink as well as the downlink. These SINR gains may be exploited: * To increase the capacity (mainly in urban areas), e.g. by reducing the interference. *To increase the coverage (mainly in rural areas), e.g., by increasing the cell size (range extension) or by improving the edge coverage. *To increase the link quality. *To decrease the delay spread. *To reduce the transmission powers, or a combination thereof.
A3.2.7	BS frequency synchronization/time alignment requirements. Does the proposed RTT require base transmitter and/or receiver station synchronization or base-to-base bit time alignment? If so, specify the long term (1 year) frequency stability requirements, and also the required bit-to-bit time alignment. Describe the	Q and q	A1.4.1 A1.4.3	<u>FDD mode</u> BS-BS synchronization is not required. <u>TDD mode</u> Time slot synchronization is recommended to decrease the effort for neighbour cell interference suppression. Frame synchronization is recommended to speed up listening of neighbour cell beacon information.

	means of achieving this.			
A3.2.8	The number of users per RF carrier/frequency channel that the proposed RTT can support affects overall cost – especially as bearer traffic requirements increase or geographic traffic density varies widely with time.	Q	A1.2.17	<u>FDD mode</u> There are a maximum of 256 orthogonal downlink channels available, some of which must be allocated for downlink common transport channels. This leaves approximately 250 orthogonal channels for user traffic, such as voice. Normally, the cell capacity is interference limited, i.e. the actual number of voice channels is lower than this number (exact number of voice channels depends on operational conditions). Uplink is never limited by the number of orthogonal code channels, as the orthogonal code tree used is user-specific in the uplink. In some cases, e.g. for the case when adaptive antennas are used, the number of voice channels per cell can be increased above 250 by applying multiple non-orthogonal code sets on the downlink. The simulation results are: Indoor A: 85 Pedestrian A: 154 (2 code sets are considered) Vehicular A: 85
	Specify the maximum number of user channels that can be supported while still meeting ITU-T Recommendation G.726 performance requirements for voice traffic.			<u>TDD mode</u> There are a maximum of 128 orthogonal downlink channels available, some of which are allocated for downlink common transport channels. This leaves approximately 120 orthogonal channels for user traffic. The reason why the maximum number of channels in TDD is only 50% of that in FDD is the UL/DL sharing of one 5 MHz carrier in TDD.
A3.2.9	Base site implementation/installation requirements (not applicable to satellite)	q	A1.4.17	The base station configuration can be modular.
	BS size, mounting, antenna type and height can vary greatly as a function of cell size, RTT design and application environment. Discuss its positive or negative impact on system complexity and cost.			Thus the number of user supported can be increased modularly if desired, similar to introducing new TX/RX units to a GSM base station with the difference being that RF hardware is not effected as only single RX/TX per base station is required. WCDMA Base Stations are anticipated as possibly smaller than the current GSM Base Stations.
A3.2.10	Handover complexity	Q	A1.2.24	
	Consistent with handover quality objectives defined in criterion § 6.1.3, describe how user handover is implemented for both voice and data services and	and q	A1.4.6.1	<u>FDD mode</u> The handover scheme is based on a mobile assisted soft/softer handover mechanism and hard handover. The mobile station (MS) monitors the pilot signal levels received from neighbouring base stations and reports to the network those pilots crossing or above a given set of dynamic thresholds. Based on this information the network orders the MS to add or remove pilots from its *Active Set*. The

	its overall impact on infrastructure cost and complexity.			*Active Set* is defined as the set of base stations for which user signal is simultaneously demodulated and coherently combined. The same user information modulated by the appropriate base station code is sent from multiple base stations. Coherent combining of the different signals from different sectored antennas, from different base stations, or from the same antenna but on different multiple path components is performed in the MS by the usage of Rake receivers. Base stations with which the mobile station is in soft handover, process the signal transmitted by a mobile station. The received signal from different sectors of a base station (cell) can be combined in the base station, and the received signal from different base stations (cells) can be combined at the radio network controller. Soft handover results in increased coverage range on the uplink. This soft handover mechanism results in truly seamless handover without any disruption of service. The obtained spatial diversity reduces the frame error rate in the handover regions and allows for improved performance in difficult radio environment. Furthermore, the RTT supports various types of hard handover, e.g. inter-frequency handover. The measurements to detect other available carriers are made possible through the use of measurement slots. <u>TDD mode</u> TDD provides two different handover mechanisms depending on the service type: connection oriented or packet services. For connection oriented services, the basic HO scheme is a mobile assisted, network evaluated and decided hard handover using backward signalling. Appropriate measures are provided to accelerate the HO procedure, e.g. in case of a corner effect. Furthermore, the proposed RTT does not prevent the introduction of soft handover, which is for further study. For packet services, the basic HO scheme is a mobile evaluated and decided hard handover with background network control using forward signalling (cell reselection). Potential advantages: • Seamless HO for connection oriented, loss-less HO for packet bearer services • Mobile assisted network evaluated and decided handover scheme is most appropriate for RT services since it allows for both high flexibility in HO-algorithm design and implementation, e.g. to meet operator specific requirements in various deployment scenarios and system stability at high capacity. • Mobile evaluated and decided handover with network background control is most appropriate for packet services since it allows for both resource savings on the air interface by decentralised decision making and system integrity at high capacity by network co-ordinating measures.

Comments

The overall complexity is kept to a reasonable level, and it is possible to improve the system performance by considering additional features.

Summary

The RTT requires broadband linear amplifiers and receivers at the Base Station.

Power Control is available in both downlink and uplink directions. Dynamic range is 30 dB in downlink and 80 dB in uplink, with a variable step size between 0.25 and 3 dB, depending on the operating environment and mode. Power control rate varies between 100 and 1600 control messages per second. The RTT incorporates different types of diversity: • Time • Multipath • Macro (soft/softer handover) Different antenna systems may be available at the Base Station: • Distributed • Remote • Smart The handover scheme for FDD (connection services) is based on a mobile assisted soft/softer handover mechanism and hard handover. For connection oriented services in TDD mode, the basic handover scheme is a mobile assisted, network evaluated and decided hard handover using backward signalling.				
A3.3	Quality			
A3.3.1	Transparent reconnect procedure for dropped calls	q	A1.4.14	
	Dropped calls can result from shadowing and rapid signal loss. Air interfaces utilizing a transparent reconnect procedure – that is, the same as that employed for hand-off – mitigate against dropped calls whereas RTTs requiring a reconnect procedure significantly different from that used for hand-off do not.			The RTT supports the transparent reconnect procedure for handling dropped calls. In this context, “dropped call” is taken as the re-establishement of a call after radio resources have been released, but network resources are not yet released. In the case of rapid fades which are not compensated by soft handover, the use of pliot bits allow the the terminal or BS to rapidly reacquire synchronization.
A3.3.2	Round trip delay, D1 (with vocoder (ms)) and D2 (without vocoder (ms)) (See Fig. 6).	Q	A1.3.7.1 A1.3.7.2	The round trip delay, including source coding, is implementation dependent.
	NOTE 1 – The delay of the codec should be that specified by ITU-T for the common generic voice bearer and if there are any proposals for optional codecs include the			However, the approximate delay is 13 ms, excluding speech coding delay. The speech coding delay is around 12 ms for speech framing and processing, assuming a 10 ms frame length speech codec. Other delays such as delays on the interface between the BTS and the transcoder are not included.

	information about those also. (For the satellite component, the satellite propagation delay is not included.)			
A3.3.3	Handover/ALT quality	Q	A1.2.24	
	Intra switch/controller handover directly affects voice service quality. Handover performance, minimum break duration, and average number of handovers are key issues.		A1.2.24.1 A1.2.24.2 A1.4.6.1	<u>FDD mode</u> Soft handover: No break duration (make before break). Hard handover: no loss for packet services due to ARQ. <u>TDD mode</u> For the basic scheme of hard HO, the break duration on HO execution is the time interval between suspension of transmission on the traffic and signalling channels of the serving cell, and the successful establishment of these on the new target cell. This time is mainly dependent on the access procedure to the target cell. Since cells are assumed to be synchronised on a frame basis, a synchronous handover is executed, i.e. the MS performs a HO access onto the transport channels of the new cell with known synchronisation resulting in a very short HO execution time. Impact of HO on service performance: RT services: since the break duration is very short, seamless HO is possible. NRT services: ARQ mechanism ensures loss-less HO.
A3.3.4	Handover quality for data There should be a quantitative evaluation of the effect on data performance of handover.	Q	A1.2.24 A1.2.24.1 A1.2.24.2 A1.4.6.1	See A.3.3
A3.3.5	Maximum user bit rate for data (bit/s) A higher user bit rate potentially provides higher data service quality (such as high quality video service) from the user's point of view.	Q	A1.3.3	Protocols are designed in such a way that user bit rate can reach at least up to 2048 kbps.
A3.3.6	Channel aggregation to achieve higher user bit There should also be a qualitative evaluation of the method used to aggregate channels to provide higher bit rate services.	q	A1.2.32	<u>FDD mode</u> Channel aggregation to achieve higher rates is normally not needed for the FDD mode, due to different bit rates of the physical channels (maximum physical channel bit rate is 2048 Kbps in DL and 1024 Kbps in UL). Channel aggregation (multi-code transmission) is supported and used for the highest user rates (up to 2 Mbps). <u>TDD mode</u> Channel aggregation (multi-code and multi-slot) is used. Different rate services are implemented by multi-code (assigning more than one code) and multi-slot (assigning more than one

				time slot) transmission.
A3.3.7	Voice quality Recommendation ITU-R M.1079 specifies that FPLMTS speech quality without errors should be equivalent to ITU-T Recommendation G.726 (32 kbit/s ADPCM) with desired performance at ITU-T Recommendation G.711 (64 kbit/s PCM). NOTE 1 – Voice quality equivalent to ITU-T Recommendation G.726 error free with no more than a 0.5 degradation in MOS in the presence of 3% frame erasures might be a requirement.	Q and q	A1.2.19 A1.3.8	Different voice coding schemes can be supported, since the RTT has a flexible bearer capability supporting different bit rate allocation and voice coding frame length (e.g. 10 ms and 20 ms). Voice coding schemes envisaged to be used are those used in the GSM system, e.g. EFR and AMR coding schemes. The AMR is to be finalised end'98. For this speech codec, no exact values are available yet. For clean speech a MOS value of 4.1 (+0.1 compared to G.726 and –0.1 compared to G.711) can be expected, depending on the source coding bit rate.
A3.3.8	System overload performance (not applicable to satellite) Evaluate the effect on system blocking and quality performance on both the primary and adjacent cells during an overload condition, at e.g. 125%, 150%, 175%, 200%. Also evaluate any other effects of an overload condition.	Q and q	A1.3.9.1	FDD mode Overload causes graceful degradation of system performance, e.g. by decreasing the speech codec bit rate or increasing the BER. An additional mechanism of handling overload conditions is *Cell Breathing.* Under overload conditions, the increased interference will lead to a smaller boundary. Users near the boundary area will be handled over the adjacent, hopefully less loaded cells. TDD mode Under overload conditions DCA can be used to increase the allocated resources to the overloaded cell at the cost of the capacity of the neighbouring cells.

Comments

The RTT is able to provide high quality services at rates up to 2 Mbps.

The RTT has been designed to operate with different type of vocoders. This approach ensures the long term evolution of the system, as more efficient and higher quality voice coding schemes become available.

Summary

In the FDD mode, system overload causes graceful performance degradation, that might be traded by a decrement of the speech codec bit rate or increment of the BER.

In the TDD mode, a Dynamic Channel Allocation mechanism can be used to increase the resources available to the overloaded cell at the expense of capacity loss in the neighbouring cells.

A3.4	Flexibility of radio technologies			
A3.4.1	Services aspects			
A3.4.1.1	Variable user bit rate capabilities Variable user bit rate applications can consist of the following: – adaptive signal coding as a function of RF signal quality; – adaptive voice coder rate as a function of traffic loading as long as ITU-T Recommendation G.726 performance is met; – variable data rate as a function of user application; – variable voice/data channel utilization as a function of traffic mix requirements. Some important aspects which should be investigated are as follows: – how is variable bit rate supported? – what are the limitations? Supporting technical information should be provided such as – the range of possible data rates, – the rate of changes (ms).	q and Q	A1.2.18 A1.2.18.1	<u>FDD mode</u> Variable user bit rates between 0 and 2.048 Mbit/s can be supported with 100 bit/s granularity, with adjustments possible on a frame by frame basis. For a given connection, a sub-set of these rates is chosen at call set-up. During the call, the rate can be varied between the rates within the sub-set on a frame by frame basis. The rates sub-set can also be changed during a call, e.g. responding to services addition or removal. <u>TDD mode</u> Variable user bit rates between 0 and 2.048 Mbit/s can be supported, with a high degree of flexibility, by adjusting the number of used codes and time slots as well as by adjusting the channel coding and burst types.
A3.4.1.2	Maximum tolerable Doppler shift, F_d (Hz) for which voice and data quality requirements are	q and Q	A1.3.1.4	Implementation dependent. A Doppler Shift for a vehicular speed of 500 Km/h is tolerated.

	met (terrestrial only) Supporting technical information: F_d			
A3.4.1.3	Doppler compensation method (satellite component only) What is the Doppler compensation method and residual Doppler shift after compensation?	Q and q	A1.3.2.2	Not applicable.
A3.4.1.4	How the maximum tolerable delay spread of the proposed technology impact the flexibility (e.g., ability to cope with very high mobile speed)?	q	A1.3.1.3 A1.2.14 A1.2.14.1 A1.2.14.2 A1.3.10	Implementation dependent Exact requirement is for further study, but at least 50000 ns should be tolerated. FDD Mode: A Rake Receiver is used to obtain a diversity gain from multipath. Phase reference in the form of pilot symbols is available in both tranmission directions. Variations in the delay spread profiles, in terms of amplitude and phase, can be tracked at least on a slot-by-slot basis (0.625 ms). Additional variations, such as appearance and disappearance of rays can be tracked at least on a frame-by-frame basis. TDD Mode: A Joint Detector coherently detects the data corresponding to different CDMA codes, coping with multipath propagation effects at both BS and MS. Intersymbol interference (ISI) is reduced in the detection process by means of the Joint Detection Equalizer. All variations in the delay spread profile, including amplitude, phase and appearance/disappearance of rays can be tracked on a slot-by-slot basis (0.625 ms).
A3.4.1.5	Maximum user information bit rate, R_u (kbit/s) How flexibly services can be offered to customers ? What is the limitation in number of users for each particular service? (e.g. no more than two simultaneous 2 Mbit/s users)	Q and q	A1.3.3 A1.3.1.5.2 A1.2.31 A1.2.32	Protocols are designed in such a way that user bit rate at least up to 2048 kbps. <u>Flexibility of Service</u> See A.3.4.1.1.
A3.4.1.6	Multiple vocoder rate capability – bit rate variability, – delay variability,	Q and q	A1.2.19 A1.2.19.1 A1.2.7 A1.2.12	Different voice coding schemes can be supported since the supported RTT has a flexible bearer capability supporting different bit rate allocation and voice coding frame length (e.g. 10 ms and 20 ms). Voice coding schemes envisaged those used in the GSM system, e.g. EFR and AMR coding schemes. The AMR is to be finalised end’98.

	– error protection variability.			
A3.4.1.7	Multimedia capabilities The proponents should describe how multimedia services are handled. The following items should be evaluated: – possible limitations (in data rates, number of bearers), – ability to allocate extra bearers during of the communication, – constraints for handover.	Q and q	A1.2.21 A1.2.20 A1.3.1.5.2 A1.2.18 A1.2.24 A1.2.30 A1.2.30.1	Parallel services can be provided. The different services can have independent bit rate, bit-error rate, delay, etc., and can have different transfer modes (packet/circuit-switched). All service classes can be supported with the proposed RTT. The pooling of resource units bearer services at the radio interface with various data rates can be achieved. Further, by variation of the spreading factor, power, coding rate and interleaving depth, various BER and delay requirements can be met. For each service class, dedicated bearer services at the radio interface are defined. The bearer services at the radio interface are separated into low delay data (LDD), long constrained delay (LCD) and unconstrained delay data (UDD) bearer services. The LDD bearer is characterised by stringent delay (and stringent delay variation) requirements. In contrary, the LCD bearer is characterised by less stringent delay (and delay variation) requirements but more stringent BER requirements. Both LDD and LCD bearers can have a constant or variable bit rate. Finally, the UDD bearer is characterised by unconstrained delay requirements. The following mapping may be used: Class A: LDD Class B: LDD-VBR Class C: LCD Class D: UDD
A3.4.2	Planning			
A3.4.2.1	Spectrum related matters			
A3.4.2.1.1	Flexibility in the use of the frequency band The proponents should provide the necessary information related to this topic (e.g., allocation of sub-carriers with no constraints, handling of asymmetric services, usage of non-paired band).	q	A1.2.1 A1.2.2 A1.2.2.1 A1.2.3 A1.2.5.1	The minimum frequency band required to deploy the system is 2x5 MHz in FDD mode and 1x5 MHz in TDD mode.With these spectrum allocations, up to 2 Mbps user rate is possible. However, note that these are the minimum spectrum requirements. Larger spectrum allocation is recommended for more efficient operation. A larger spectrum allocation supporting two or more 5 MHz carriers would e.g. allow for more efficent trunking or multiple cell layers. Since it is possible to set uplink and downlink bearer service characteristics independently, asymmetric connections can be supported in both TDD and FDD modes. In the FDD mode, it is possible to assign more carriers to the downlink than uplink, or vice versa. In TDD mode, the ratio of uplink to downlink capacity of a carrier can be adjusted by changing the ratio of the number of uplink and downlink time slots within the frame. The basic chip rate of the RTT is 4.096 Mcps corresponding to a channel bandwidth of approximately 5 MHz. Additional chip rates 8.192 Mcps and 16.384 Mcps, corresponding to bandwidths of approximately 10 MHz and 20 MHz respectively, are also specificied for the FDD mode. These bandwidths are seen as future evolution of the RTT towards even higher user rates (>2 Mbps). The different bandwidths are not used to compensate for transmission medium impairments. The different bandwidths are transparent to the end user.
A3.4.2.1.2	Spectrum sharing capabilities The proponent should indicate how global spectrum allocation can be shared between operators in the same region.	q and Q	A1.2.26	For both FDD-mode and TDD-mode, sharing is always possible through frequency division. Furthermore, for the TDD-mode, sharing the same frequency with another TDD system including non-UMTS/IMT-2000 systems such as DECT and PHS is also possible due to the TDMA component. Both fixed time division and interference avoidance in the time domain using Dynamic Channel Allocation (DCA) can be used for this purpose. In addition, since ODMA relays do not own dedicated radio resource but share it in an asynchronous fashion with neighbouring nodes, they are tolerant to spectrum sharing.

	The following aspects may be detailed: – means for spectrum sharing between operators in the same region, – guardband between operators in case of fixed sharing.			
A3.4.2.1.3	Minimum frequency band necessary to operate the system in good conditions Supporting technical information: – impact of the frequency reuse pattern, – bandwidth necessary to carry high peak data rate.	Q and q	A1.2.1 A1.4.15 A1.2.5	FDD mode: 2×5 MHz TDD mode: 1×5 MHz With these spectrum allocations, up to 2 Mbps user rate is possible. However, note that these are the minimum spectrum requirements. Larger spectrum allocation is recommended for more efficient operation. A larger spectrum allocation supporting two or more 5 MHz carriers would e.g. allow for more efficent trunking or multiple cell layers.
A3.4.2.1.4	Band Plans and Frequency Duplexing The proponent should describe how their system will provide global service delivery in the different regional/national band plans and frequency duplexing arrangements for IMT-2000 systems.	Q and q	A1.2.2.5 A1.2.2.6	The bandwidth per duplex RF channel (MHz) measured at the 3 dB down points is: FDD mode: ≈ 8.2 MHz, (≈16.4 MHz and ≈ 32.8 MHz for higher chip rates which are not yet described) TDD mode: 4.1 MHz See also A.3.4.2.1.2.
A3.4.2.2	Radio resource planning			
A3.4.2.2.1	Allocation of radio resources The proponents and evaluators should focus on the requirements and constraints imposed by the proposed technology. More particularly, the following aspects should be considered: – what are the methods used to make the allocation and planning of radio resources	q	A1.2.25 A1.2.27 A1.4.15	<u>Frequency Planning</u> Due to frequency re-use = 1, no frequency planning is required. Different sectorizations are possible, for example, 3 sectors/site or 6 sectors/site. The RTT does not use interleaved frequency allocation. <u>Dynamic Channel Allocation (DCA)</u> <u>FDD mode</u> For the FDD mode, DCA is not generally needed. <u>TDD mode</u>

	flexible? – what are the impacts on the network side (e.g. synchronization of BSs, signalling,)? – other aspects. Examples of functions or type of planning required which may be supported by the proposed technology: – DCA, – frequency hopping, – code planning, – time planning, – nterleaved frequency planning. NOTE 1 – The use of the second adjacent channel instead of the adjacent channel at a neighbouring cluster cell is called "interleaved frequency planning". In some cases, no particular functions are necessary (e.g. frequency reuse = 1).			For the TDD mode, slow DCA (allocation of time slots to cells) and fast DCA (allocation of a channel to a certain call) can be distinguished. Additionally, for the TDD mode, the allocation of a slot in a cell to uplink or downlink traffic is managed by the slow DCA, too. The interference on different slots in the time frame may be different. Therefore, a DCA algorithm that allocates the least interfered slots to ongoing calls with high QoS requirements results in a considerable gain in quality and/or capacity. The capability to vary the ratio of slots allocated in the uplink and in the downlink allows an optimal adaptation to the traffic asymmetry. Since synchronised base stations are used, advanced combinations of fast and slow DCA can be implemented in order to allocate the maximum amount of resources to the cell with the momentarily highest amount of traffic.
A3.4.2.2.2	Adaptability to adapt to different and/or time varying conditions (e.g., propagation, traffic) How the proposed technology cope with varying propagation and/or traffic conditions? Examples of adaptive functions which may be supported by the proposed technology: – DCA, – link adaptation, – fast power control, – adaptation to large delay spreads.	q	A1.3.10 A1.2.27 A1.2.22 A1.2.14	See § A3.4.2.2.1 for description of DCA. Three types of power control systems are employed. One is the 'fast closed loop power control' (FDD mode only) which counteracts fading on a slot basis (0.625 ms): it is based on measurements of SIR. The second one is the 'open loop power control': it is only used for initial power setting. The third one is 'outer loop power control': it is based on BER and FER measurements. It has the role of changing the target C/I, when the situation of the mobile is changing, or for power control planning. It is done on a longer period basis. The use of fast power control significantly improves the link-performance (BER as a function of E_b/N_0) especially in the case of slow-moving mobile stations. For fast moving mobile stations (>100 km/h), there is less performance improvement due to fast power control. Multipath is compensated by the following systems: <u>FDD mode</u> A RAKE receiver or any other suitable receiver structures can coherently combine multiple paths and give diversity gains (the detailed receiver structure is implementation dependent). Phase reference in the form of pilot symbols is available on both transmission directions. <u>TDD mode</u> Typically, joint detection is used to coherently detect the data corresponding to different CDMA codes and copes with

	Some adaptivity aspects may be inherent to the RTT.			multipath propagation effects at the MS as well as the BS (the detailed receiver structure is implementation dependent). Phase reference in the form of a pilot sequence is available in both transmission directions.
A3.4.2.3	Mixed cell architecture (not applicable to satellite component)			
A3.4.2.3.1	Frequency management between different layers What kind of planning is required to manage frequencies between the different layers? e.g. – fixed separation, – dynamic separation, – possibility to use the same frequencies between different layers. Possible supporting technical information: – guard band.	q and Q	A1.2.28 A1.4.15	Mixed cell architectures are supported. Single cell reuse is anticipated, although different sectorizations are possible. Adjacent Channel Protection (ACP) allows deployement of 2 or 3 carriers within 10 or 15 MHz bands.
A3.4.2.3.2	User adaptation to the environment What are the constraints to the management of users between the different cell layers? e.g. – constraints for handover between different layers, – adaptation to the cell layers depending on services, mobile speed, mobile power.	q	A1.2.28 A1.3.10	Handover between RF carriers is provided by the RTT with minimal traffic disruption. Parameters which may vary per cell include: • Power step size. • Soft Handover thresholds. • Admission control thresholds. <u>FDD mode</u> Intra and Inter-frequency Handover are supported. Intra-frequency Handover is implemented by combining signals at a Rake receiver. The network orders the Mobile Station to add or remove links from its active set. Inter-frequency Handover may be implemented at the Mobile Station by means of two possible alternatives: • Slotted Mode (downlink). • Dual Receiver. <u>TDD mode</u> The TDD mode also supports Intra-and Inter-frequency Handover.

				Frame synchronization between Base Stations optimize interfrequency measurements.
A3.4.2.4	Fixed-wireless access			
A3.4.2.4.1	The proponents should indicate how well its technology is suited for operation in the fixed wireless access environment. Areas which would need evaluation include (not applicable to satellite component): – ability to deploy small BSs easily, – use of repeaters, – use of large cells, – ability to support fixed and mobile users within a cell, – network and signalling simplification.	q	A1.1.3 A1.3.5 A1.4.17 A1.4.7 A1.4.7.1	The RTT can be used for FWA applications. There are no differences in the radio transmission technology parameters with respect to FWA than what is being used for the cellular applications. The flexibility of the RTT allows for an optimisation of the transmission and receiver chains according to the specific deployment scenario such as cellular or FWA. The RTT is designed to be future proof taking advantage of extended range technologies such as adaptive antennas and antenna diversity in the downlink but also interference cancellation techniques. It is also possible to include those kinds of techniques later, if necessary, without requiring any frequency reconfiguration nor does it preclude the use of user equipment not supporting those techniques like antenna diversity in the downlink. Repeaters can be used. In addition, ODMA supports data transfer via a network of intermediate relaying nodes (dedicated fixed relays or relaying enable mobiles).
A3.4.2.4.2	Possible use of adaptive antennas (how well suited is the technology) (not applicable to satellite component) Is RTT suited to introduce adaptive antennas? Explain the reason if it is.	q	A1.3.6	Both FDD and TDD have been specially designed to support adaptive antennas. FDD mode The FDD mode incorporates connection-dedicated pilot bits in both uplink and downlink directions. TDD mode The TDD mode incorporates connection dedicated midamble sequences in both uplink and downlink . Moreover, the reciprocity between both directions facilitates an efficent implementation of Smart Antennas.
A3.4.2.4.3	Existing system migration capability	q	A1.4.16	The detailed parameters of the RTT have been chosen in order to facilitate the implementation of dual-mode UMTS/GSM terminals. Some specific aspects that have been considered in order to ease 2G/3G migration are: • Support of GSM/UMTS handover. • Ability to reuse existing cell strucutres and sites. • Support of signalling to evolved GSM networks. • Support of GSM services. • Support of the UIM concept. • Support of GSM authentication concept.

Comments

The RTT has been specially designed to support Hierarchical Cell Strucutures (HCS). This approach may be used to co-locate microcells intended to low-speed, high data rate Mobile Stations, with macro-cells mainly intended for vehicular-based (high speed) Mobile Stations.

As the RTT has special provisions for the use of Smart Antennas, very high capacity efficiency may be obtained in Fixed Wireless Access (FWA) applications.

Summary

User bit rates may reach up to 2 Mbps, with 100 bps granularity. The Bit rate may be changed in each 10 msec frame.

Due to Frequency Re-use=1, no frequency planning is required.

The TDD mode includes slow and fast Dynamic Channel Allocation.

Minimum spectrum requirements are 2*5 MHz for FDD and 1*5 MHz for TDD. Better performance is obtained in larger bands.

Hierarchical Cell Structures are possible.

Fixed Wireless Access applications are possible without special adaptation of the radio interface.

A3.5	Implication on network interface			
A3.5.1	Examine the synchronization requirements with respect to the network interfaces. *Best case* : no special accommodation necessary to provide synchronization. *Worst case* : special accommodation for synchronization is required, e.g. additional equipment at BS or special consideration for facilities.	q	A1.4.3	On a transmission level, the base stations are synchronised to the transmission network. The different base stations involved in soft handover (macro diversity) are synchronised per active mobile station connection. An accuracy of +/- 2 msec is acceptable for the fixed transmission between base stations-to-network (radio network controller (RNC)) and can be obtained through prioritised Iub (RNC-BS) and Iur (RNC-RNC) signalling.

Comments

The RTT is designed in order to interoperate with different types of Core Networks.

FDD Synchronization

Since no special accommodation is necessary to provide synchronization, a non-dedicated synchronization network is used for BS clock reference generation. The synchronization network is superimposed to a PDH, SDH or SONET traffic network.

No BS-BS synchronization is required.

TDD Synchronization

As in the FDD mode, a non-dedicated synchronization network is used for BS clock reference generation. The synchronization network is superimposed to a PDH, SDH or SONET traffic network. BS-BS synchronization is strongly recommended. Timeslot synchronization decreases the effort for neighbour cell interference suppression. Frame synchronization is recommended to speed up the listening of beacon information transmitted by neighbour cells. Note that TDD synchronization requirements refer to frame level, and not to chip level, as is required by other RTTs. This allows the use of less rigorous synchronization networks.				
Summary The Call Handover fucntion is shared between the RTT and the Core Network, according to the following distribution: • Radio Environment Survey (UE, UTRA). • Handover decision (UE, UTRA) • Macro-diversity control (UTRA) • Handover control (UTRA) • Handover Execution (UTRA, CN) • Handover Completion (UTRA, CN) • Serving Radio Network System (SRNS) relocation (UTRAN, CN) • Inter-system Handover (UE, UTRAN, CN)				
A3.6	Handportable performance optimization capability			
A3.6.1	Isolation between transmitter and receiver Isolation between transmitter and receiver has an impact on the size and weight of the handportable.	Q	A1.2.2 A1.2.2.1 A1.2.2.2	<u>FDD mode</u> A duplexer is required. Required transmitter/receiver isolation is 50 dB. <u>TDD mode</u> No duplexer is required.
A3.6.2	Average terminal power output P_0 (mW) Lower power gives longer battery life and greater operating time.	Q	A1.2.16.1.2	<u>FDD mode:</u> Activity is 100 % if a mobile operates a dedicated channel. For packet transmission on the common channels, smaller TX active cycles possible. Average power outputs depend on the environment. Simulations where based in the following values: • Indoor: 2.5 mW (4 dBm). • Pedestrian: 25 mW (14 dBm). • Vehicular: 250 mW (24 dBm). <u>TDD mode:</u> • 1 code (peak/average ratio 3.2 dB): Min. 14.8 dBm (1 timeslot), Max. 26.5 dBm (15 timeslots). • 8 codes (peak/average ratio 8.7dB): Min. 9.2 dBm (1 timeslot used), Max. 21 dBm (15 timeslots used). Calculation:

				Time average power =30dBm-peak/average ratio+10*log10(used timeslots/frame/16).
A3.6.3	System round trip delay impacts the amount of acoustical isolation required between handportable microphone and speaker components and, as such, the physical size and mechanical design of the subscriber unit. NOTE 1 – The delay of the codec should be that specified by ITU-T for the common generic voice bearer and if there are any proposals for optional codecs include the information about those also. (For the satellite component, the satellite propagation delay is not included.)	Q and q	A1.3.7 A1.3.7.1 A1.3.7.2 A1.3.7.3	The round trip delay including source coding is implementation dependent. However, the approximate delay is 13 ms excluding the speech codec delay. In addition, the speech coding delay should be added which is around 12 ms for speech framing and processing, assuming a 10 ms frame length speech codec. Other delays such as delays on the interface between the BTS and the transcoder are not included.
A3.6.4	Peak transmission power	Q	A1.2.16.1.1	Typically 30 dBm.
A3.6.5	Power control characteristics Does the proposed RTT utilize transmitter power control? If so, is it used in both forward and reverse links? State the power control range, step size (dB) and required accuracy, number of possible step sizes and number of power controls per second, which are concerned with mobile station technology complexity.			Three types of power control strategies are employed. One is the ‘fast closed loop power control’ (FDD mode only) which counteracts fading on a slot basis (0.625 ms). It is based on measurements on SIR. The second one is ‘open loop power control’. It is used only for the initial power setting. The third one is the ‘outer loop power control’. It is based on BER and FER measurements. It has the role to change the target C/I, when the situation of the mobile is changing or for power control planning. It is done on a longer period basis. The use of fast power control significantly improves the link-performance (BER as a function of E_b/N_0) especially in the case of slow-moving mobile stations. For fast moving mobile stations (>100 km/h), there is less performance improvement due to fast power control.
A3.6.5.1	Power control dynamic range Larger power control dynamic range gives longer battery life and greater operating time.	Q	A1.2.22 A1.2.22.3 A1.2.22.4	Uplink: 80 dB Downlink: 30 dB
A3.6.5.2	Power control step size, accuracy and speed	Q	A1.2.22 A1.2.22.1 A1.2.22.2 A1.2.22.5	<u>FDD mode</u> Uplink: Variable on a cell basis in the range 0.25-1.5 dB. Downlink: Variable on a cell basis in the range 0.25-1.5 dB.

				Speed: 1600 control cycles per second. <u>TDD mode:</u> The power control step size is variable, ranging from 0.5 to 3 dB. Speed: 100 to 800 control cycles per second, depending on the exact UL/DL time slot allocation.
A3.6.6	Linear transmitter requirements	q	A1.4.10	For 2 equal power signals being separated by 200 kHz leading to an output level of 21 dBm each, the resulting intermodulation spectrum shall not exceed (relative to peak spectrum): • -38 dB at 200 kHz to -90 dB at 800 kHz offset from higher/lower frequency signal (linear decrease) • <=-95 dB at 1 MHz from higher/lower frequency signal and above
A3.6.7	Linear receiver requirements (not applicable to satellite)	q	A1.4.11	The 3rd order intercept point will be specified between -10 dBm and -5 dBm.
A3.6.8	Dynamic range of receiver The lower the dynamic range requirement, the lower the complexity and ease of design implementation.	Q	A1.4.12	80 dB for Automatic Gain Control.
A3.6.9	Diversity schemes Diversity has an impact on handportable complexity and size. If utilized describe the type of diversity and address the following two attributes.	Q and q	A1.2.23 A1.2.23.1 A1.2.23.2	Diversity in the Mobile Station is possible but not mandatory.
A3.6.10	The number of antennas	Q	A1.2.23.1	The Mobile Station requires a single antenna. Implementations using more than one antenna are possible.
A3.6.11	The number of receivers	Q	A1.2.23.1	The Mobile Station requires a single receiver. Implementations including more than one receiver will provide additional advantages, for example, in the application of FDD mode inter-frequency handover.
A3.6.12	Frequency stability Tight frequency stability requirements contribute to handportable complexity.	Q	A1.4.1.2	3 ppm (unlocked) 0.1 ppm (locked)
A3.6.13	The ratio of “off (sleep)” time to “on” time	Q	A1.2.29 A1.2.29.1	In dedicated mode, i.e. during calls and in circuit oriented operation, the transmitter/receiver is continuously “on” (FDD) or transmits/receives only in allocated timeslots (TDD). Variable rate transmission is utilised whenever possible to reduce needed transmitted power. Employing power control methods also reduces the transmitted power levels to a minimum. With packet traffic, depending on the packet-access mode, the receiver and transmitter may only be used periodically, i.e. being switched-off until data is available for transmission, or until the base station indicates to the mobile station that is to be received.

				In idle mode, the mobile station uses a sleep mode that allows most of the circuits to be turned-off during the periods when the mobile station is not receiving. The mobile station is only awakens for short periods to listen to e.g. the paging channel or the broadcast channel.
A3.6.14	Frequency generator step size, switched speed and frequency range Tight step size, switch speed and wide frequency range contribute to handportable complexity. Conversely, they increase RTT flexibility.	Q	A1.4.5	- Step size: 200 kHz - Switched speed: 250 µs - Frequency range: 60 MHz
A3.6.15	Digital signal processing requirements Digital signal processing can be a significant proportion of the hardware for some radio interface proposals. It can contribute to the cost, size, weight and power consumption of the BS and influence secondary factors such as heat management and reliability. Any digital circuitry associated with the network interfaces should not be included. However any special requirements for interfacing with these functions should be included. This section of the evaluation should analyse the detailed description of the digital signal processing requirements, including performance characteristics, architecture and algorithms, in order to estimate the impact on complexity of the BSs. At a minimum the evaluation should review the signal	Q and q	A1.4.13	For a detailed description of the signal processing mechanisms, refer to the UTRA RTT System Description document. Signal Processing Estimates FDD mode 8 kbps – 2048 kbps: 5 – 86 million real multiplications. DSP and correlators included, taken the word length requirements relative to the DSP operation into account. The estimation is valid for both UL and DL. TDD mode It depends on the implementation, e.g. for a 110 kbps service, around 15-20 MIPS is needed. The convolutional encoding/decoding is not included in these figures, as it is the same regardless of the multiple access for the same data rate(s).

	processing estimates (MOPS, memory requirements, gate counts) required for demodulation, equalization, channel coding, error correction, diversity processing (including Rake receivers), adaptive antenna array processing, modulation, A-D and D-A converters and multiplexing as well as some IF and baseband filtering. For new technologies, there may be additional or alternative requirements (such as FFTs). Although specific implementations are likely to vary, good sample descriptions should allow the relative cost, complexity and power consumption to be compared for the candidate RTTs, as well as the size and the weight of the circuitry. The descriptions should allow the evaluators to verify the signal processing requirement metrics, such as MOPS, memory and gate count, provided by the RTT proponent.			

Comments

The proposed RTT can be implemented in handportable terminals without posing any particular technical constraint.

Summary

The RTT incorporates three schemes of Power Control:

- To counteract fast fading on a slot basis (0.625 msec).
- For initial power setting.
- To take into account the impact on BER and FER.

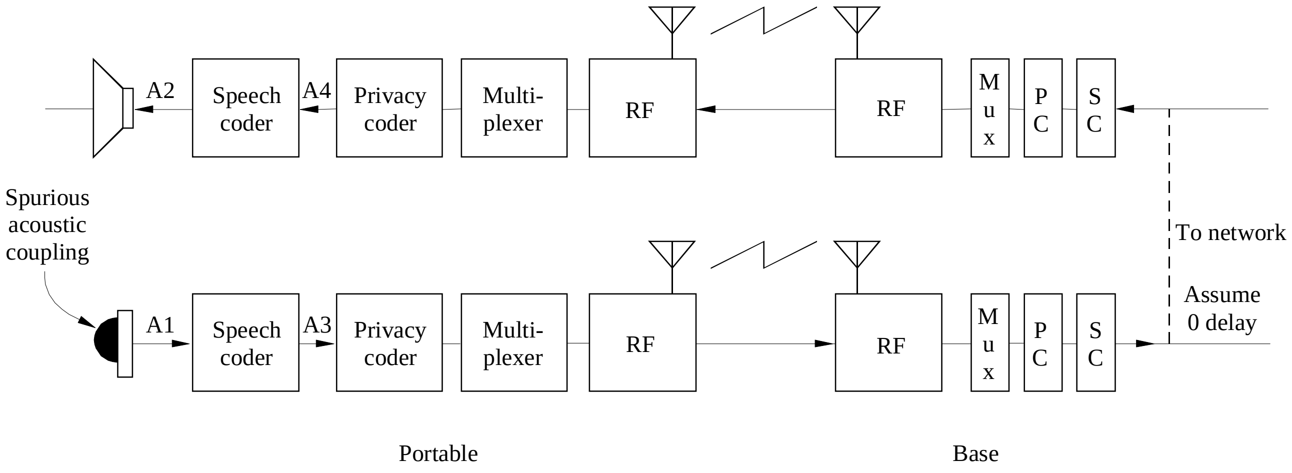
A single receiver is required at the Mobile Station for both FDD and TDD modes.

A single antenna is required at the Mobile Station.				
A3.7	Coverage/power efficiency			
A3.7.1	Terrestrial Coverage efficiency: – the coverage efficiency is considered for the lowest traffic loadings; – the base site coverage efficiency can be quantitatively determined by addressing coverage limitation and/or by calculating the maximum coverage range for the lowest traffic loading.			
A3.7.1.1	Base site coverage efficiency The number of base sites required to provide coverage at system start-up and ongoing traffic growth significantly impacts cost. From § 1.3.2 of Annex 2, determine the coverage efficiency, *C* (km^2/base sites), for the lowest traffic loadings. Proponent has to indicate the background of the calculation and also to indicate the maximum coverage range.	Q	A1.3.1.7 A1.3.1.7.1 A1.3.1.7.2 A1.3.4	All the following results are expressed in km^2/cell, DL/UL. **FDD MODE** **Speech** Indoors: 2.6 / 3.3 Pedestrian: 2.7 / 3.3 Vehicular: 35.5 / 22.6 **LCD** Indoors 2048: 0.2 / 0.1 Pedestrian 384: 0.9 / 0.7 Vehicular 144: 18.9 / 8.3 **UDD** Indoors 2048: 0.40 / 0.32 Pedestrian 384: 0.88 / 0.80 Vehicular 144: 27.43 / 12.99 **TDD MODE** **Speech** Indoors: 2.9 / 2.6 Pedestrian: 2.9 / 2.1 Vehicular: 28.7 / 8.9

				<u>LCD</u> Indoors 2048: 0.1/ 0.1 Pedestrian 384: 0.8 / 0.5 Vehicular 144: 53.4 / 12.2 **<u>UDD</u>** The proposed Type II Hybrid ARQ scheme allows for retransmission in case of unsuccessful data detection. Therefore, no fixed E_b/N_o or C/I values required to achieve the QoS at the cell border can be defined. However, due to ARQ retransmissions, the range of UDD services is larger than the range of the corresponding LCD service. **NOTE** These results are based on the following assumptions: • Vehicular Environment: BS with 3-sector antenna. Gain : 17 dBi. • MS antenna gain: 2 dBi. • TX Powers (DL/UL) (dBm) Indoor Office A: 13/10 Outdoor to Indoor: 23/20 Pedestrian: 23/20 Vehicular A: 30/24
A3.7.1.2	Method to increase the coverage efficiency Proponent describes the technique adopted to increase the coverage efficiency and drawbacks. Remote antenna systems can be used to economically extend vehicular coverage to low traffic density areas. RTT link budget, propagation delay system noise and diversity strategies can be impacted by their use. Distributed antenna designs – similar to remote antenna systems – interconnect multiple antennas to a single radio port via	q	A1.3.5 A1.3.6	Both FDD and TDD operating modes of UTRA are able to use all the standard types of Base Station antennas. This includes those that provide omni-directional, sectored, fixed or variable patterns. Directive Antennas decrease the interference, leading to an increase in system capacity. Both FDD and TDD mode support remote, distributed, and smart antenna systems. Signal-to-interference-plus-noise-ratio (SINR) can be significantly improved by incorporating various smart antenna concepts in the uplink as well as in the downlink. These SINR gains may be exploited: • to increase the capacity (mainly in urban areas), e.g. by reducing the interference. • to increase the coverage (mainly in rural areas), e.g., by increasing the cell size (range extension) or by improving the edge coverage. • to increase the link quality. • to decrease the delay spread. • to reduce the transmission power, or a combination thereof. Repeaters can be used. In addition, ODMA supports data transfer via a network of intermediate relaying nodes (dedicated fixed relays or relaying enable mobiles).

	broadband lines. However, their application is not necessary limited to providing coverage, but can also be used to economically provide continuous building coverage for pedestrian applications. System synchronization, delay spread, and noise performance can be impacted by their use.			
A3.7.2	Satellite Normalized power efficiency Supported information bit rate per required carrier power-to-noise density ratio for the given channel performance under the given interference conditions for voice Supported information bit rate per required carrier power-to-noise density ratio for the given channel performance under the given interference conditions for voice plus data mixed traffic.	Q	A1.3.2.4 A1.3.2.4.1 A1.3.2.4.2	Not applicable.
Comments ODMA may significantly increase the coverage for data communication services.				
Summary				

FIGURE 6



D1: delay between A1 and A2
D2: delay between A3 and A4
Mux: multiplexer
PC: privacy coder
SC: speech coder

1225-06

Attachment 2

Updated System description

1. INTRODUCTION

This attachment contains a revised version of the system description of the ETSI UMTS terrestrial radio access (UTRA) RTT candidate submission. The UTRA network is currently being developed in ETSI SMG2 and this document reflects the status as of September 1998. Thus, any modification to this RTT can be made as a result of that process.

The description contains a definition and abbreviation section as shown in Section 2. Section 3 describes the general architecture of the radio access network. In Section 4, Layer 2 and 3 of the radio protocol are described, i.e. from the radio resource management sub-layer to the MAC sub-layer as defined in ITU-R recommendation M.1035. Finally the Physical layer of the radio interface protocol is described in Sections 5 and 6 for the FDD mode and TDD mode respectively. Handover is discussed in Section 7.

2. DEFINITIONS, ABBREVIATIONS AND SYMBOLS

2.1 Definitions

Active Set

Set of radio links simultaneously involved in a specific communication service between an MS and a UTRAN.

Cell

Geographical area served from one *UTRAN Access Point*. A cell is defined by a cell identity broadcast from the *UTRAN Access Point*.

Coded Composite Transport Channel (CCTrCH)

A data stream resulting from encoding and multiplexing of one or several *transport channels*.

Iu

The interconnection point (interface) between the RNS and the Core Network. It is also considered as a reference point.

Iub

Interface between the RNC and the Node B.

Iur

Interface between two RNSs.

Logical Channel

A logical channel is a radio bearer, or part of it, dedicated for exclusive use of a specific communication process. Different types of logical channel are defined according to the type of information transferred on the radio interface.

Node B

A logical node responsible for radio transmission / reception in one or more cells to/from the UE. Terminates the Iub interface towards the RNC.

Physical Channel

In FDD mode, a physical channel is defined by code, frequency and, in the uplink, relative phase (I/Q). In TDD mode, code, frequency, and time-slot define a physical channel.

Physical channel data stream

In the uplink, a data stream that is transmitted on one *physical channel*.

In the downlink, a data stream that is transmitted on one *physical channel* in each cell of the *active set.*

Radio access bearer

The service that the access stratum provides to the non-access stratum for transfer of user data between MS and CN.

Radio Access Network Application Part

Radio Network Signalling over the Iu.

Radio Network Subsystem Application Part

Radio Network Signalling over the Iur.

Radio frame

A radio frame is a numbered time interval of 10 ms duration used for data transmission on the radio physical channel. A radio frame is divided into 16 slots of 0.625 ms duration. The unit of data that is mapped to a radio frame (10 ms time interval) may also be referred to as radio frame.

Radio link

A set of (radio) *physical channels* that link an MS to a *UTRAN access point.*

Radio link addition

A [soft handover] procedure whereby a branch through a new [sector of a cell] is added in case some of the already existing branches were using [sectors] of the same cell.

Radio link removal

A [soft handover] procedure whereby a branch through a new [sector of a cell] is removed in case some of the remaining existing branches use [sectors of] that cell.

Radio Network Controller

This equipment in the RNS is in charge of controlling the use and the integrity of the radio resources.

Radio Network Subsystem

Either a full network or only the access part of a UMTS network offering the allocation and the release of specific radio resources to establish means of connection in between an UE and the UTRAN.
A Radio Network Subsystem is responsible for the resources and transmission/reception in a set of cells.

Serving RNS

A role an RNS can take with respect to a specific connection between an UE and UTRAN. There is one Serving RNS for each UE that has a connection to UTRAN. The Serving RNS is in charge of the radio connection between a UE and the UTRAN. The Serving RNS terminates the Iu for this UE.

Drift RNS

The role an RNS can take with respect to a specific connection between an UE and UTRAN. An RNS that supports the Serving RNS with radio resources when the connection between the UTRAN and the UE need to use cell(s) controlled by this RNS is referred to as Drift RNS

RRC connection

A point-to-point bi-directional connection between RRC peer entities on the UE and the UTRAN sides, respectively. An UE has either zero or one RRC connection.

Signalling connection

An assured-mode link between the user equipment and the core network to transfer higher layer information between peer entities in the non-access stratum**.**

Signalling link

Provides an assured-mode link layer to transfer the MS-UTRAN signalling messages as well as MS - Core

Network signalling messages (using the *signalling connection*).

Transport channel

The channels that are offered by the physical layer to Layer 2 for data transport between peer L1 entities are denoted as Transport Channels.

Different types of transport channels are defined by how and with which characteristics data is transferred on the physical layer, e.g. whether using dedicated or common physical channels are employed.

Transport Format

A combination of encoding, interleaving, bit rate and mapping onto *physical channels*.

Transport Format Indicator (TFI)

A label for a specific *Transport Format* within a *Transport Format Set*.

Transport Format Set

A set of Transport *Formats*. For example, a variable rate DCH has a Transport Format Set (one Transport Format for each rate), whereas a fixed rate DCH has a single Transport Format.

UTRAN access point

The UTRAN-side end point of a *radio link*. A UTRAN access point is a *cell*.

User Equipment

A Mobile Equipment with one or several UMTS Subscriber Identity Module(s).

2.2 Abbreviations

For the purposes of this specification the following abbreviations apply.

ARQ	Automatic Repeat Request
AAL	Application Adaptation Layer
ATM	Asynchronous Transfer Mode
BCCH	Broadcast Control Channel
BER	Bit Error Ratio
BLER	Block Error Ratio
BS	Base Station
BSS	Base Station System
BPSK	Binary Phase Shift Keying
CA	Capacity Allocation
CAA	Capacity Allocation Acknowledgement
CBR	Constant Bit Rate
C-	Control-
CC	Call Control
CCCH	Common Control Channel
CCPCH	Common Control Physical Channel
CCTrCH	Coded Composite Transport Channel
CD	Capacity Deallocation
CDA	Capacity Deallocation Acknowledgement
CDMA	Code Division Multiple Access
CN	Core Network
CTDMA	Code Time Division Multiple Access
CRC	Cyclic Redundancy Check
DCA	Dynamic Channel Allocation
DCH	Dedicated Channel
DCCH	Dedicated Control Channel
DC-SAP	Dedicated Connection Service Access Point

DL	Downlink
DPCH	Dedicated Physical Channel
DPCCH	Dedicated Physical Control Channel
DPDCH	Dedicated Physical Data Channel
DRNS	Drift RNS
DRX	Discontinuous Reception
DTX	Discontinuous Transmission
DS-CDMA	Direct-Sequence Code Division Multiple Access
FACH	Forward Access Channel
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FER	Frame Error Ratio
HCS	Hierarchical Cellular Structures
HO	Handover
GMSK	Gaussian Minimum Shift Keying
GSM	Global System for Mobile Communication
ITU	International Telecommunication Union
JD	Joint Detection
kbps	kilo-bits per second
L1	Layer 1 (physical layer)
L2	Layer 2 (data link layer)
L3	Layer 3 (network layer)
LAC	Link Access Control
LLC	Logical Link Layer
MA	Multiple Access
MAC	Medium Access Control
MAHO	Mobile Assisted Handover
Mcps	Mega Chip Per Second
ME	Mobile Equipment
MM	Mobility Management
MO	Mobile Originated
MOHO	Mobile Originated Handover
MS	Mobile Station
MT	Mobile Terminated
MUI	Mobile User Identifier
NRT	Non-Real Time
ODMA	Opportunity Driven Multiple Access
OVSF	Orthogonal Variable Spreading Factor (codes)
PC	Power Control
PCH	Paging Channel
PDU	Protocol Data Unit
PHY	Physical layer
PhyCH	Physical Channel
QoS	Quality of Service
QPSK	Quaternary Phase Shift Keying
PG	Processing Gain
PI	Paging Indication
PRACH	Physical Random Access Channel
PUF	Power Up Function
RACH	Random Access Channel
RANAP	Radio Access Network Application Part
RF	Radio Frequency
RLC	Radio Link Control
RLCP	Radio Link Control Protocol
RNC	Radio Network Controller
RNS	Radio Network Subsystem
RNSAP	Radio Network Subsystem Application Part
RR	Radio Resource

RRC	Radio Resource Control
RRM	Radio Resource Management
RT	Real Time
RU	Resource Unit
RX	Receive
SAP	Service Access Point
SCH	Synchronisation Channel
SDCCH	Stand-alone Dedicated Control Channel
SDU	Service Data Unit
SF	Spreading Factor
SIR	Signal-to-Interference Ratio
SMS	Short message Service
SP	Switching Point
SRNS	Serving RNS
TCH	Traffic Channel
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TFI	Transport Format Indicator
TPC	Transmit Power Control
TX	Transmit
U-	User-
UE	User Equipment
UL	Uplink
UMTS	Universal Mobile Telecommunications System
USIM	UMTS Subscriber Identity Module
UTRA	UMTS Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
VA	Voice Activity
VBR	Variable Bit Rate

3. THE RADIO ACCESS NETWORK ARCHITECTURE

3.1 General architecture

Figure 1 shows the assumed UMTS architecture as outlined in ETSI/SMG. The figure shows the architecture in terms of its entities User Equipment (UE), UMTS Terrestrial Radio Access Network (UTRAN) and Core Network (CN). The respective reference points Uu (Radio Interface) and Iu (CN-UTRAN interface) are shown. The figure illustrates furthermore the high-level functional grouping into the Access Stratum and the Non-Access Stratum.

The Access Stratum offers services through the following Service Access Points (SAP) to the Non-Access Stratum:

- General Control (GC) SAPs,
- Notification (Nt) SAPs and
- Dedicated Control (DC) SAPs

The SAPs are marked with circles in Figure 1. It is assumed that GC, Nt, and DC SAPs are provided by the Radio Resource Control (RRC) layer[1] and that the protocols in the I_u-interface will transport the primitives of the SAPs over the I_u-interface (since the RRC protocol is terminated in the UTRAN).

[1] The radio interface protocol layers such as the RRC layer are described in Section 0.

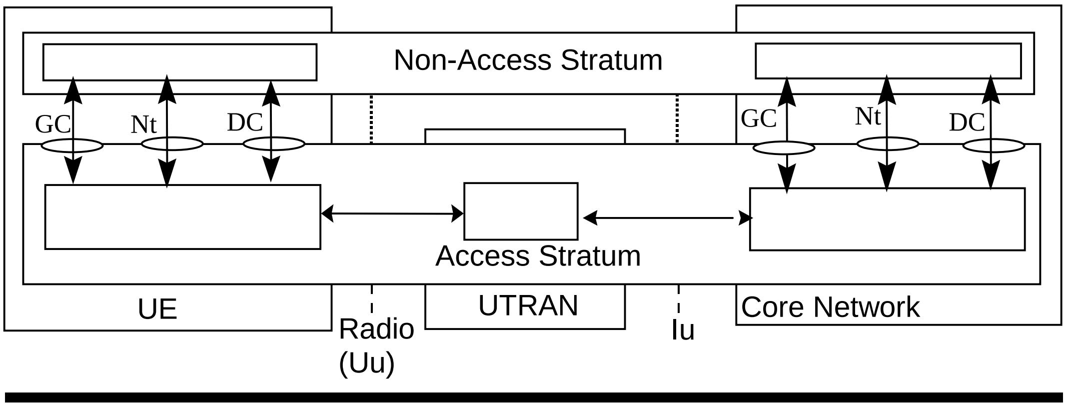


Figure 1. Assumed UMTS Architecture

Figure 2 shows a simplified UMTS architecture with the external reference points and interfaces to the UTRAN. (The terminal can be named both as Mobile Station (MS), User Equipment (UE) or Mobile Equipment (ME). The term MS is used as a generic term for the terminal part.) The figure forms the basis for the architecture description in the following subsections.

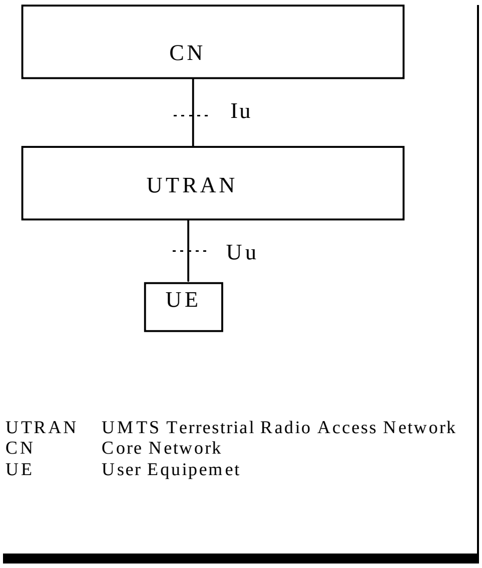


Figure 2. UMTS Architecture

3.2 Basic principles

Some basic principles agreed for the definition of the architecture are:

- Logical separation of signalling and data transport networks
- Macro diversity is fully handled in the UTRAN
- UTRAN and CN functions are fully separated from transport function. Addressing scheme used in UTRAN and CN shall not be tied to the addressing schemes of Transport functions. The fact that some UTRAN or CN function resides in the same equipment, as some transport functions does not make the transport functions part of the UTRAN or the CN.

3.2.1 Mobility handling

Radio access specific procedures should be handled within UTRAN only. This means that all cell level mobility should be handled within UTRAN. Also the cell structure of the radio network should not necessarily be known outside the UTRAN.

When a dedicated connection exists to the UE, the UTRAN shall handle the radio interface mobility of the UE. This includes procedures such as soft handover.

When a dedicated connection does not exist to the UE, no UE information in UTRAN is needed. In this case, the mobility is handled directly between UE and CN outside access stratum (e.g. by means of registration procedures). When paging the UE, the CN indicates a 'geographical area' that is translated within UTRAN to the actual cells that shall be paged. A 'geographical area' shall be identified in a cell-structure independent way. One possibility is the use of 'Location Area identities'.

During the lifetime of the dedicated connection, the registrations to the CN are suppressed by the UE. When a dedicated connection is released, the UE performs a new registration to the CN, if needed.

Thus the UTRAN does not contain any permanent 'location registers' for the UE, but only temporary contexts for the duration of the dedicated connection. This context may typically contain location information (e.g. current cell(s) of the UE) and information about allocated radio resources and related connection references.

3.3 UTRAN logical architecture

The UTRAN consists of a set of Radio Network Subsystems connected to the Core Network through the Iu.

A RNS consists of a Radio Network Controller and one or more Node B nodes. Node B is connected to the RNC through the Iub interface.

The RNC is responsible for the handover decisions that require signalling to the UE.

The RNC comprises a combining/splitting function to support macro diversity between different Node B.

However, a Node B can comprise an optional combining/splitting function to support macro diversity inside a Node B.

Inside the UTRAN, the RNCs of the Radio Network Subsystems can be interconnected together through the Iur. Iu and Iur are logical interfaces. Iur can be conveyed over physical direct connection between RNCs or via any suitable transport network.

The UTRAN architecture is shown in Figure 3.

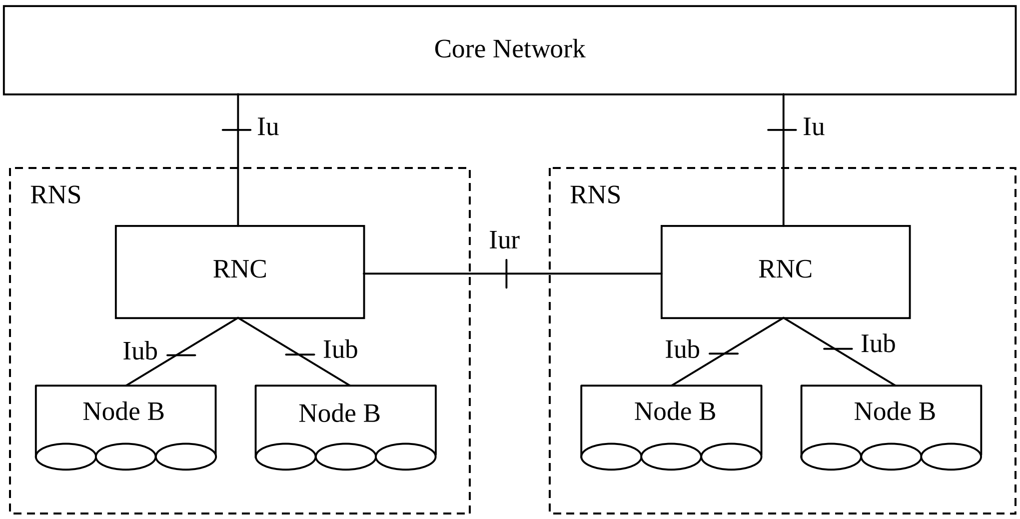


Figure 3. UTRAN Architecture

Each RNS is responsible for the resources of its set of cells.

For each connection between User Equipment and the UTRAN, one RNS is the Serving RNS. When required, Drift RNSs support the Serving RNS by providing radio resources as shown in Figure 4. The role of an RNS (Serving or Drift) is on a per connection basis between a UE and the UTRAN.

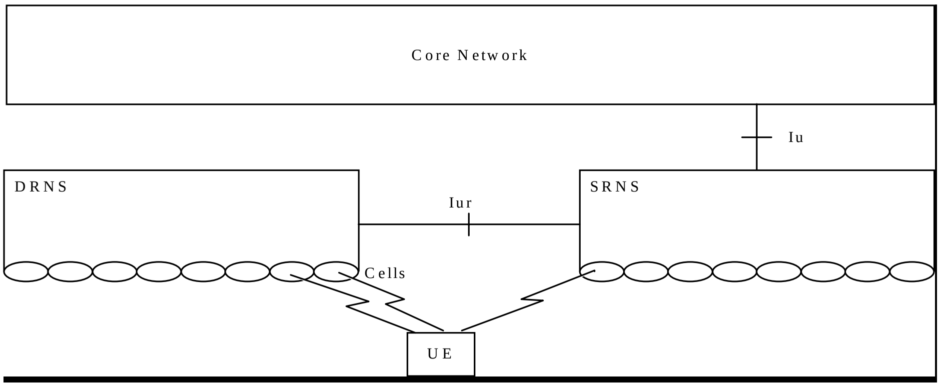


Figure 4. Serving and Drift RNS

3.4 Function descriptions

The functions that will be performed in UTRAN is grouped into 4 main groups:

- Functions related to overall system access control
  - System information broadcasting
- Functions related to radio channel ciphering
  - Radio channel ciphering
  - Radio channel deciphering
- Functions related to handover
  - Radio environment survey
  - Handover decision
  - Macro-diversity control
  - Handover Control
  - Handover execution
  - Handover completion
  - SRNS Relocation
  - Inter-System handover
- Functions related to radio resource management and control
  - Radio bearer connection set-up and release (Radio Bearer Control)
  - Reservation and release of physical radio channels
  - Allocation and de-allocation of physical radio channels
  - Packet data transfer over radio function
  - RF power control
  - RF power setting
  - Radio channel coding
  - Radio channel decoding
  - Channel coding control
  - Initial (random) access detection and handling

The different functions are described in the following subsections.

3.4.1 Functions description

3.4.1.1 Functions related to overall system access control

System access is the means by which a user is connected to the network in order to use services and/or facilities. User system access may be initiated from either the mobile side, e.g. a mobile originated call, or the network side, e.g. a mobile terminated call.

3.4.1.1.1 System information broadcasting

This function provides the mobile station with the information that is needed to camp on a cell and to set up a connection in idle mode and to perform handover or route packets in communication mode. The tasks may include:

- access rights
- frequency bands used
- configuration of transport channels, PCH, FACH and RACH channel structure of the cell etc
- network and cell identities
- information for location registration purposes
- UE idle mode cell selection and cell re-selection criteria
- UE transmission power control information
- UE access and admission control information

Because of its close relation to the basic radio transmission and the radio channel structure, the basic control and synchronisation of this function should be located in UTRAN.

3.4.1.2 Functions related to radio channel ciphering

3.4.1.2.1 Radio channel ciphering

This function is a pure computation function whereby the radio transmitted data can be protected against an non-authorised third party. Ciphering may be based on the usage of a session-dependent key, derived through signalling and/or session dependent information.

This function is located in the UE and in the UTRAN.

3.4.1.2.2 Radio channel deciphering

This function is a pure computation function that is used to restore the original information from the ciphered information. The deciphering function is the complement function of the ciphering function, based on the same ciphering key.

This function is located in the UE and in the UTRAN.

3.4.1.3 Functions related to handover

3.4.1.3.1 Radio environment survey

This function performs measurements on radio channels (current and surrounding cells) and translates these measurements into radio channel quality estimates. Measurements may include:

1. received signal strengths (current and surrounding cells),
2. estimated bit error ratios, (current and surrounding cells),
3. estimation of propagation environments (e.g. high-speed, low-speed, satellite, etc.),
4. transmission range (e.g. through timing information),
5. doppler shift,
6. synchronisation status,
7. received interference level.

In order for these measurements and the subsequent analysis to be meaningful, some association between the measurements and the channels to which they relate should be made in the analysis. Such association may include the use of identifiers for the network, the base station, the cell (base station sector) and/or the radio channel.

This function is located in the UE and in the UTRAN.

3.4.1.3.2 Handover decision

This function consists of gathering estimates of the quality of the radio channels (including estimates from surrounding cells) from the measuring entities and to assess the overall quality of service of the call. The overall quality of service is compared with requested limits and with estimates from surrounding cells. Depending on the outcome of this comparison, the *macro-diversity control function* or the *handover control function* may be activated.

This function may also include functionality to assess traffic loading distribution among radio cells and to decide on handing over traffic between cells for traffic reasons.

The location of this function is depending on the handover principle chosen.

- if network only initiated handover, this function is located in the RNC;
- if mobile only initiated handover, this function is located in the UE;
- if both the mobile and the network can initiate handover, this function will be located in both the RNC and the UE.

3.4.1.3.3 Macro-diversity control

Upon request of the *Handover Decision function*, this function control the duplication/ replication of information streams to receive/ transmit the same information through multiple physical channels (possibly in different cells) from/ towards a single mobile terminal.

This function also controls the combining of information streams generated by a single source (diversity link), but conveyed via several parallel physical channels (diversity sub-links). Macro diversity control should interact with channel coding control in order to reduce the bit error ratio when combining the different information streams. This function controls macro-diversity execution which is located at the two endpoints of the connection element on which macro-diversity is applied (diversity link), that is at the access point and also at the mobile termination.

In some cases, depending on physical network configuration, there may be several entities which combine the different information streams, e.g. one entity combines information streams on radio signal basis, another combines information streams on wire-line signal basis.

This function is typically located in the UTRAN.

3.4.1.3.4 Handover control

In the case of switched handover, this function is responsible for the overall control of the handover execution process. It initiates the handover execution process in the entities required and receives indications regarding the results.

Due to the close relationship with the radio access and the Handover Decision function, this function should be located in the UTRAN.

3.4.1.3.5 Handover execution

This function is in control of the actual handing over of the communication path. It comprises two sub-processes: *handover resource reservation* and *handover path switching.* The *handover resource reservation* process will reserve and activate the new radio and wire-line resources that are required for the handover. When the new resources are successfully reserved and activated, the *handover path switching* process will perform the final switching from the old to the new resources, including any intermediate path combination required, e.g. handover branch addition and handover branch deletion in the soft handover case.

This function is located in the UTRAN for UTRAN internal path switching and in the CN for CN path switching.

3.4.1.3.6 Handover completion

This function will free up any resources that are no longer needed. A re-routing of the call may also be triggered in order to optimise the new connection.

This function is located both in the UTRAN and in the CN.

3.4.1.3.7 SRNS relocation

The SRNS Relocation function co-ordinates the activities when the SRNS role is to be taken over by another RNS. SRNS relocation implies that the Iu interface connection point is moved to the new RNS.

This function is located in the RNC and the CN.

3.4.1.3.8 Inter-system handover

The Inter-system handover function enables handover to and from e.g. GSM BSS.

This function is located in the UTRAN, the UE and the CN.

3.4.1.4 Functions related to radio resource management and control

Radio resource management is concerned with the allocation and maintenance of radio communication resources. UMTS radio resources must be shared between circuit mode (voice and data) services and other modes of service (e.g. packet data transfer mode and connectionless services).

3.4.1.4.1 Radio bearer connection set-up and release (Radio Bearer Control)

This function is responsible for the control of connection element set-up and release in the radio access sub network. The purpose of this function is

1. to participate in the processing of the end-to-end connection set-up and release,
2. and to manage and maintain the element of the end-to-end connection, which is located in the radio access sub network.

In the former case, this function will be activated by request from other functional entities at call set-up/release. In the latter case, i.e. when the end-to-end connection has already been established, this function may also be invoked to cater for in-call service modification or at handover execution. This function interacts with the *reservation and release of physical (radio) channels* function.

This function is located both in the UE and in the RNC.

3.4.1.4.2 Reservation and release of physical radio channels

This function consists of translating the connection element set-up or release requests into physical radio channel requests, reserving or releasing the corresponding physical radio channels and acknowledging this reservation/ release to the requesting entity.

This function may also perform physical channel reservation and release in the case of a handover. Moreover, the amount of radio resource required may change during a call, due to service requests from the user or macro-diversity requests. Therefore, this function must also be capable of dynamically assigning physical channels during a call.

Note: This function may or may not be identical to the function reservation and release of physical radio channels. The distinction between the two functions is required e.g. to take into account sharing a physical radio channel by multiple users in a packet data transfer mode.

This function is located in the UTRAN.

3.4.1.4.3 Allocation and de-allocation of physical radio channels

This function is responsible, once physical radio channels have been reserved, for actual physical radio channel usage, allocating or de-allocating the corresponding physical radio channels for data transfer. Acknowledging this allocation/de-allocation to the requesting entity is for further study.

Note: This function may or may not be identical to the function reservation and release of physical radio channels. The distinction between the two functions is required e.g. to take into account sharing a physical radio channel by multiple users in a packet data transfer mode.

This function is located in the UTRAN.

3.4.1.4.4 Packet data transfer over radio function

This function provides packet data transfer capability across the UMTS radio interface. This function includes procedures which:

1. provide packet access control over radio channels,
2. provide packet multiplexing over common physical radio channels,
3. provide packet discrimination within the mobile terminal,
4. provide error detection and correction,
5. provide flow control procedures.

This function is located in both the UE and in the UTRAN.

3.4.1.4.5 RF power control

In order to minimise the level of interference (and thereby maximise the re-use of radio spectrum), it is important that the radio transmission power is not higher than what is required for the requested service quality. Based on assessments of radio channel quality, this function controls the level of the transmitted power from the mobile station as well as the base station.

This function is located in both the UE and in the UTRAN.

3.4.1.4.6 RF power setting

This function adjusts the output power of a radio transmitter according to control information from the *RF power control function*. The function forms an inherent part of any power control scheme, whether closed or open loop.

This function is located in both the UE and in the UTRAN.

3.4.1.4.7 Radio channel coding

This function introduces redundancy into the source data flow, increasing its rate by adding information calculated from the source data, in order to allow the detection or correction of signal errors introduced by the transmission medium. The channel coding algorithm(s) used and the amount of redundancy introduced may be different for the different types of transport channels and different types of data.

This function is located in both the UE and in the UTRAN.

3.4.1.4.8 Radio channel decoding

This function tries to reconstruct the source information using the redundancy added by the channel coding function to detect or correct possible errors in the received data flow. The channel decoding function may also employ a priori error likelihood information generated by the demodulation function to increase the efficiency of the decoding operation. The channel decoding function is the complement function to the channel coding function.

This function is located in both the UE and in the UTRAN.

3.4.1.4.9 Channel coding control

This function generates control information required by the channel coding/ decoding execution functions. This may include channel coding scheme, code rate, etc.

This function is located in both the UE and in the UTRAN.

3.4.1.4.10 Initial (random) access detection and handling

This function will have the ability to detect an initial access attempt from a mobile station and will respond appropriately. The handling of the initial access may include procedures for a possible resolution of colliding attempts, etc. The successful result will be the request for allocation of appropriate resources for the requesting mobile station.

This function is located in the UTRAN.

3.5 Description of UTRAN interfaces

The Uu (radio) interface is described in Sections Radio Interface ArcHitecture, Layer 1 description (FDD mode) and Layer 1 description (TDD modE).

3.5.1 Iu interface, assumptions

From a UTRAN perspective, maximising the commonality of the various protocols that flow on the Iu interface is desirable. This means at the minimum that :

- A common set of radio access bearer services will be offered by UTRAN to the Core Network nodes, regardless of their type.

There will be a common functional split between UTRAN and the Core Network nodes, regardless of their type.

3.5.1.1 Streamlining functions

3.5.1.1.1 Access network triggered streamlining

One Access Network triggered function needed over the Iu interface is the function for SRNS Relocation. SRNS Relocation needs support from the Core Network to be executed. Figure 5 shows the principle of streamlining.

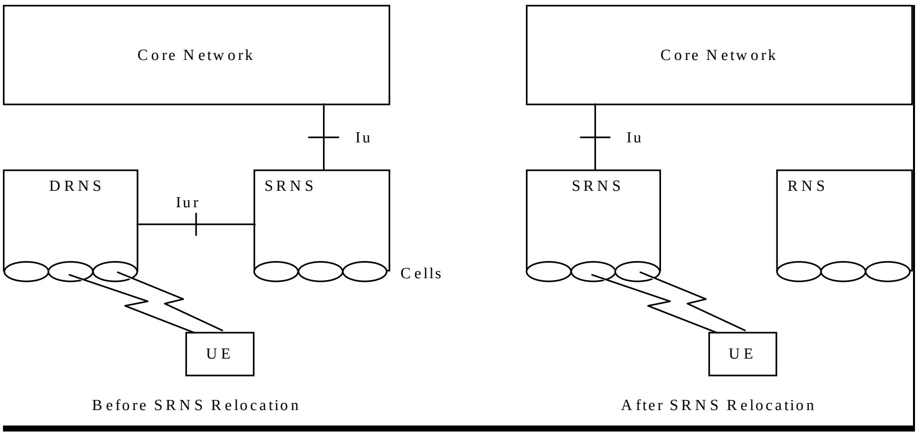


Figure 5. Serving RNS Relocation

[FDD — For the cases where handover can be performed independently from SRNS Relocation, the algorithm for triggering the SRNS relocation is not specified.]

[FDD — The specification of Iur Interface shall allow the support of soft handover throughout the UTRAN of PLMN without performing SRNS relocation.]

3.5.2 Iu interface protocol

The Radio Network signalling over Iu consists of the Radio Access Network Application Part (RANAP). The RANAP consists of mechanisms to handle all procedures between the CN and UTRAN. It is also capable of conveying messages transparently between the CN and the UE without interpretation or processing by the UTRAN.

Over the Iu interface the RANAP protocol is, e.g. used for:

- Facilitate a set of general UTRAN procedures from the Core Network such as paging -notification as defined by the notification SAP.
- Separate each User Equipment (UE) on the protocol level for mobile specific signalling management as defined by the dedicated SAP.
- Transfer of transparent non-access signalling as defined in the dedicated SAP.
- Request of various types of UTRAN Radio Access Bearers through the dedicated SAP.
- Perform the streamlining function.

The Access Stratum provides the Radio Access Bearers

Various transmission possibilities exist to convey the bearers over the Iu to the Core Network. It is therefore proposed to separate the Data Transport Resource and traffic handling from the RANAP (Figure 6). This resource and traffic handling is controlled by the Transport Signalling. A Signalling Bearer carries the

Transport Signalling over the Iu interface.

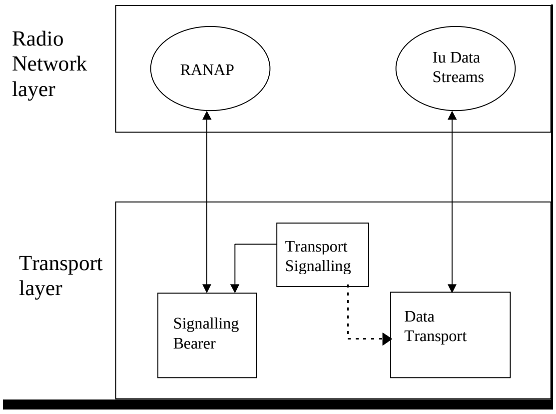


Figure 6. Separation of RANAP and transport over Iu

The RANAP is terminated in the SRNS.

3.5.3 Description of UTRAN internal interfaces

3.5.3.1 Iur interface

The Iur interface connects a SRNS and a DRNS.

This interface should be open.

The information exchanged across the Iur is categorised as below:

- One or more Iur Data stream which comprises
  - Radio frames
  - Simple, commonly agreed Quality estimate
  - Synchronisation information
- Signalling
  - Addition of Cells in the DRNS which may lead or not to the addition of an new Iur Data stream
  - Removal of Cells in the DRNS
  - Modify Radio bearer characteristics

Note: This list of procedures is not the full list over Iur interface. From a logical stand point, the Iur interface is a point to point interface between the SRNS and all the DRNS, i.e. there is no deeper hierarchy of RNSs than the SRNS and DRNS. However, this point to point logical interface should be feasible even in the absence of a physical direct connection between the two RNSs.

3.5.3.1.1 Functional split over Iur interface

Note: This is only an initial list.

3.5.3.1.1.1 Macro-diversity combining/splitting

DRNS may perform macro-diversity combining/splitting of data streams communicated via its cells. SRNS performs macro-diversity combining/splitting of Iur data streams received from/sent to DRNS(s), and data streams communicated via its own cells.

The internal DRNS handling of the macro-diversity combining/splitting of radio frames is controlled by the DRNS.

3.5.3.1.1.2 Control of macro-diversity combining/splitting topology

When requesting the addition of a new cell for an UE-UTRAN connection, the RNC of the SRNS (i.e. SNRC) can explicitly request to the RNC of the DRNS (i.e. the DRNC) a new Iur data stream. In this case the macro-diversity combining and splitting function within the DRNS is not used for that cell. Otherwise, the DRNS takes the decision whether macro-diversity combining and splitting function is used inside the DRNS for that cell i.e. whether a new Iur data stream shall be added or not.

3.5.3.1.1.3 Handling of DRNS hardware resources

Allocation and control of DRNS hardware resources, used for Iur data streams and radio interface transmission/reception in DRNS, is performed by DRNS.

3.5.3.1.1.4 Allocation of downlink channelisation codes

Allocation of downlink channelisation codes of cells belonging to DRNS is performed in DRNS.

Note that this does not imply that the signalling of the code allocation to the UE must be done from the DRNS.

3.5.3.1.2 Iur interface protocol

The signalling information across Iur interface as identified in section 0 is called Radio Network Subsystem Application Part (RNSAP).

The RNSAP is terminated in the SRNS and in the DRNS.

As already stated in Section 0 a clear separation shall exist between the Radio Network Layer and the Transport Layer. It is therefore proposed to separate the Data Transport resource and traffic handling from the RNSAP (Figure 7). This resource and traffic handling is controlled by the Transport Signalling. A Signalling Bearer carries the Transport Signalling over the Iur interface.

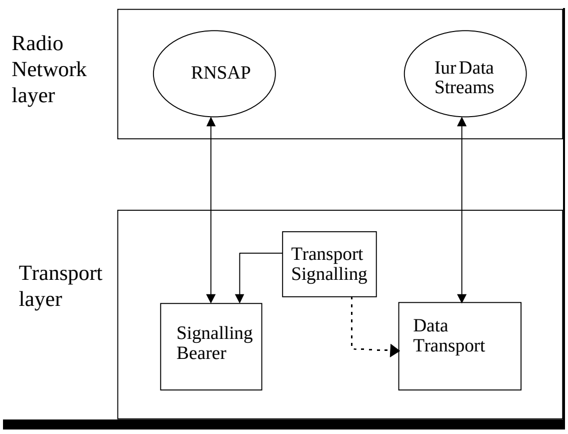


Figure 7. Separation of RNSAP and transport over Iur

3.5.3.2 Iub interface

Note: This description is applicable if the Iub interface will be standardised as an open interface

The Iub interface connects a RNC and a Node B.

The information transferred over the Iub reference point can be categorised as follows:

1. Radio application related signalling
   The Iub interface allows RNC and Node B to negotiate about radio resources, for example to add and delete cells controlled by the Node B to support communication of the dedicated connection between UE and SRNS.

2. Radio frame data blocks
   The Iub interface provides means for transport of uplink and downlink radio frame data blocks between

RNC and Node B. This transport can use pre-defined transmission links or switched connections.

3. Quality estimations of uplink radio frames and synchronisation data
The macro-diversity combining function of the RNC uses Node B quality estimations of the uplink radio frame data blocks. There is also a need for accurate time synchronisation between the soft handover branches.

The information in category 3 is tightly coupled to the radio frame data blocks in category 2. Therefore, category 2 and 3 information is multiplexed on the same underlying transport mechanism (e.g. switched connection), and is together referred to as an Iub data stream.

The Iub data stream shall follow the same specification as the Iur data stream.

Over the Iub interface between the RNC and one Node B, one or more Iub data streams are established, each corresponding to one or more cells belonging to the Node B.

3.5.3.2.1 Iub General Principles

The following principles shall be respected when defining the Iub interface :

1. The functional division between RNC and Node B shall have as few options as possible.
2. Complex functionality shall as far as possible be avoided over Iub. This is important so that the Iub specification is ready on time. Advanced optimisation solutions may be added in later versions of the standard.
3. The Iub functional split shall take into account the probability of frequent switching between different channel types.
4. Iub should be based on a logical model of Node B.
5. Node B controls a number of cells and can be ordered to add/remove radio links in those cells.
6. Neither the physical structure nor any internal protocols of node B should be standardised and are thus not limiting factors, e.g. when introducing future technology.
7. Operation and Maintenance of Node B hardware and software resources is not part of the Iub standardisation.

3.5.3.2.2 Functional split over Iub

Note: This is only an initial list.

3.5.3.2.2.1 Traffic management

3.5.3.2.2.1.1 Management of dedicated resources

These functions are related to the activation of logical resources (e.g. Uu radio ports, Iub ports), and the connection of these various resources together.

Some freedom may be left to Node B on some functions like allocation of codes or soft combining within Node B, since soft combining has merits for being executed as close as possible to the radio (both in terms of transmission cost and efficiency).

3.5.3.2.2.1.2 Management of common radio channels

The common channels need to be controlled from the RNC. This is typically the control of the RACH channel, the information that is broadcast on the Broadcast control channel, and the control and request for sending information on the paging channels.

3.5.3.2.2.1.3 Control of traffic flows

Congestion on the Iub interface will need to be covered for asynchronous flows (i.e. those which may flow via AAL5). This concerns in particular the flow from radio channels where retransmission takes place in Node B and where soft handover is not applied.

3.5.3.2.2.2 Macro-diversity combining of radio frame data blocks

Node B may perform macro-diversity combining/splitting of data streams communicated via its cells. RNC performs macro-diversity combining/splitting of Iub data streams received from/sent to several Node B(s).

3.5.3.2.2.3 Control of macro-diversity combining/splitting topology

When requesting the addition of a new cell for a UE to UTRAN connection, the RNC can explicitly request to the Node B a new Iub data stream, in which case the macro-diversity combining and splitting function within the Node B is not used for that cell. Otherwise, the Node B takes the decision whether macro-diversity combining and splitting function is used inside the Node B for that cell i.e. whether a new Iub data stream shall be added or not.

The Node B controls the internal Node B handling of the macro-diversity combining/splitting.

3.5.3.2.2.4 Soft handover decision

To support mobility of the UE to UTRAN connection between cells, UTRAN uses measurement reports from the MS and detectors at the cells. [The mechanisms for this are FFS.]

The RNC takes the decision to add or delete cells from the connection.

3.5.3.2.2.5 Handling of Node B hardware resources

Mapping of Node B logical resources onto Node B hardware resources, used for Iub data streams and radio interface transmission/reception, is performed by Node B.

3.5.3.2.2.6 Allocation of downlink channelisation codes

Allocation of downlink channelisation codes of cells belonging to Node B is performed in Node B.

Note that this does not imply that the signalling of the code allocation to the UE must be done from Node B.

3.5.3.2.3 Iub interface protocol

As already stated in Section Basic principles, there shall exist a clear separation between the radio network layer and the transport layer. Therefore the radio Network signalling and Iub data streams are separated from the data transport resource and traffic handling as show in Figure 8. This resource and traffic handling is controlled by the Transport Signalling. A Signalling Bearer carries the Transport Signalling over the Iub interface.

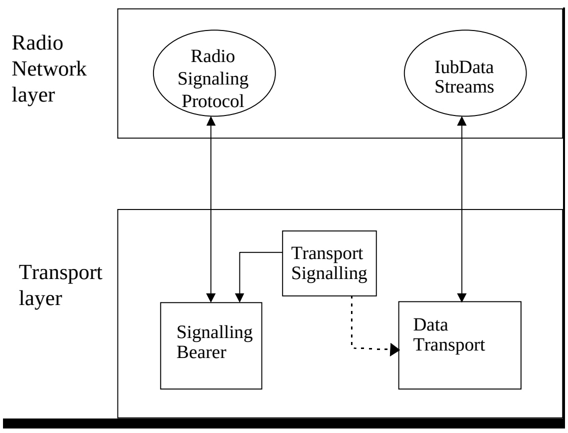


Figure 8. Separation of Radio Network protocols and transport over Iub

3.6 UTRAN internal bearers

For all open interfaces, one mandatory set of protocols must be specified. However, a clear separation between the Radio Network functions and the Transport functions should allow this Transport layer to be exchanged to another one with minimum impact on the Radio Network functions.

3.6.1 User data bearers

ATM and AAL type 2 (ITU-T recommendations I.363.2 and I.366.1) is used as the standard transport layer for Soft Handover data stream across the Iur interface.

Transport Network Control Plane is a functional plane in the interface protocol structure that is used for the transport bearer management. The actual signalling protocol that is in use within the Transport Network Control Plane depends on the underlying transport layer technology. The intention is not to specify a new UTRAN specific Application Part for the Transport Network Control Plane but to use signalling protocols standardised in other groups (if needed) for the applied transport layer technology.

3.6.2 Signalling bearers

Note: These requirements are initial requirements. Other requirements may be added later on.

3.6.2.1 Signalling bearer requirements for Iu interface

Over the Iu interface the RANAP protocol requires:

- A connectionless transport of RANAP messages to facilitate e.g. paging.
- A connection oriented transport of RANAP messages e.g. to facilitate messages belonging to a specific User equipment (UE) during a call.
- A reliable connection to make the RANAP simpler.
- Support of signalling inactivity testing of a specific UE connection.

3.6.2.2 Signalling bearer requirements for Iur interface

There exist at least two major types of soft handover over the Iur interface:

1. The case when a new physical transmission (Iur data stream) is set up over the Iur interface to provide an additional cell.
2. The case when the existing transmission (Iur data stream) is used over the Iur interface when an additional cell is added in the DRNS. In this case the DRNS must be able to identify the UE in order to perform the adding of the cell. Consequently a UE context must exist in the DRNS.

Over the Iur interface the RNSAP protocol requires:

- A connection oriented transport of RNSAP messages, i.e. one signalling bearer connection for each DRNS for a particular UE.
- A reliable connection to make the RNSAP simpler.
- Support of signalling inactivity testing of a specific UE connection.

3.6.2.3 Addressing of RNSs over the Iur interface

- For an RRC connection using a dedicated channel, the Iur standard shall allow the addition / deletion of cells belonging to any RNS within the PLMN.
- The specification of the Iur interface shall allow the SRNC (i.e. the RNC of the SRNS) to address any other RNC in the PLMN for establishing a signalling bearer over Iur.
- The specification of the Iur interface shall allow the SRNC (i.e. the RNC of the SRNS) to address any other RNC within the PLMN for establishing user data bearers for Iur data streams.

Note : Connectionless RNSAP over Iur is for further study.

4. RADIO INTERFACE ARCHITECTURE

4.1 Radio interface protocol architecture

4.1.1 Overall protocol structure

The radio interface is layered into three protocol layers:

- the physical layer (L1),
- the data link layer (L2),
- the network layer (L3).

Layer 1 is described in Sections Layer 1 description (FDD mode) and Layer 1 description (TDD modE).

Layer 2 is split into two sub-layers, Link Access Control (LAC) and Medium Access Control (MAC).

Layer 3 and LAC are divided into Control (C-) and User (U-) planes.

In the C-plane, Layer 3 is partitioned into sub-layers where the lowest sub-layer, denoted as Radio Resource Control (RRC), interfaces with layer 2. The higher layer signalling such as Mobility Management (MM) and Call Control (CC) are assumed to belong to the non-access stratum, see Figure 1. The MM and CC layers will not be described here. On the general level, the protocol architecture is similar to the current ITU-R protocol architecture, ITU-R M.1035.

Figure 9 shows the radio interface protocol architecture. Each block in Figure 9 represents an instance of the respective protocol. In the U-plane, the shaded LAC protocol may belong to the non-access stratum. Service Access Points (SAP) for peer-to-peer communication are marked with circles at the interface between sub-layers. The SAP to the physical layer provides the transport channels. In the C-plane, the interface between RRC and higher L3 sublayers (CC, MM) is defined by the General Control (GC), Notification (Nt) and Dedicated Control (DC) SAPs.

The SAPs to the MAC sublayer provides the logical channels (cf. Section Logical channels). The SAP to the physical layer provides the transport channels (cf. Section Transport channels).

Also shown in the figure are connections between RRC and MAC as well as RRC and L1 providing inter-layer services. It is for further study whether or not separate service access points need to be defined for these services (it may be merged with other SAP(s), or regarded as services provided through Layer Management).

The RLC sublayer provides ARQ functionality closely coupled with the radio transmission technique used. The requirement for a second ARQ layer is under discussion. From an ARQ point of view, the main purpose of this protocol would be to provide for the re-transmission of packets lost when the RLC can not provide retransmission, for instance during some handover scenarios. Given the possibility that a repetition protocol may be available in the core network this raised the question as to whether there is a requirement for a second level of ARQ within the UTRAN.

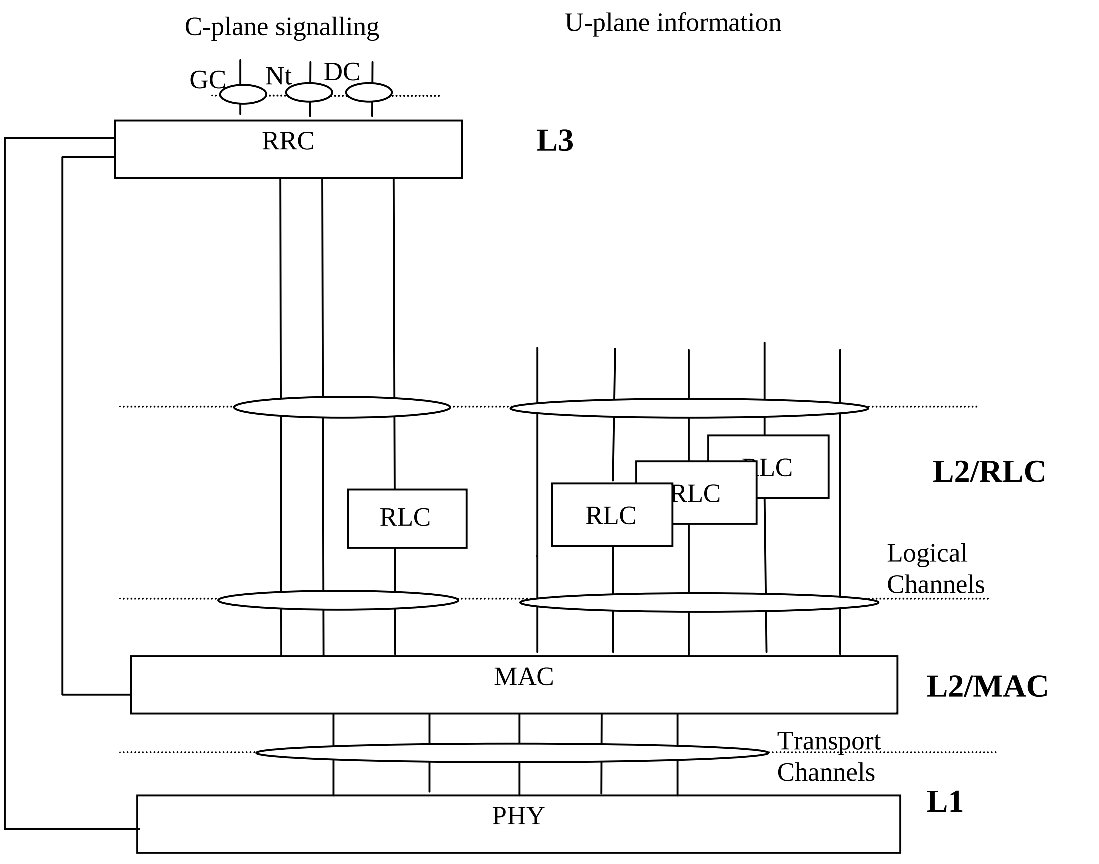


Figure 9.   Radio Interface protocol architecture (Service Access Points marked by circles)

4.1.2 Layer 1 Services and functions

4.1.2.1 L1 Services

The physical layer offers information transfer services to MAC and higher layers. The physical layer transport services are described by *how* and with what characteristics data are transferred over the radio interface. An adequate term for this is ‘Transport Channel’[2].

4.1.2.1.1 Transport channels

A general classification of transport channels is into two groups:

- common channels (where there is a need for in-band identification of the MSs when particular MSs are addressed) and
- dedicated channels (where the MSs are identified by the physical channel, i.e. code and frequency for FDD and code, timeslot and frequency for TDD)

Common transport channel types are:

1. Random Access Channel(s) (RACH) characterised by:
    - existence in uplink only,
    - collision risk,
    - open loop power control,
    - limited data field, and
    - requirement for in-band identification of the MSs.
2. Forward Access Channel(s) (FACH) characterised by:
    - existence in downlink only,
    - possibility to use beam-forming,
    - possibility to use slow power control,
    - lack of fast power control and
    - requirement for in-band identification of MSs.
3. Downlink Shared Channel(s) (DSCH) characterised by:
    - existence in downlink only,
    - possibility to use beamforming,
    - possibility to use slow power control,
    - possibility of either to be used as a stand-alone channel, or to be associated with dedicated channel(s) (DCH)
    - possibility to use fast power control, when associated with dedicated channel(s)
    - requirement for explicit identification of UEs.
4. Broadcast Control Channel (BCCH) characterised by:
    - existence in downlink only,
    - low fixed bit rate and
    - requirement to be broadcast in the entire coverage area of the cell.

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[2] This should be clearly separated from the classification of *what* is transported, which relates to the concept of logical channels. Thus DCH is used to denote that the physical layer offers the same type of service for both control and traffic.

Note that the SCH transport channel is defined for the TDD mode only. In the FDD mode, a synchronisation channel is defined as a physical channel. This channel however should not be confused with the SCH transport channel defined above.

5. Paging Channel (PCH) characterised by:

- existence in downlink only,
- possibility for sleep mode procedures and
- requirement to be broadcast in the entire coverage area of the cell.

The only type of dedicated transport channel is the:

1. Dedicated Channel (DCH) characterised by:

- possibility to use beam-forming,
- possibility to change rate fast (each 10ms),
- fast power control and
- inherent addressing of MSs.

To each transport channel, there is an associated Transport Format (for transport channels with a fixed or slow changing rate) or an associated Transport Format Set (for transport channels with fast changing rate). A Transport Format is defined as a combination of encoding, interleaving, bit rate and mapping onto physical channels. A Transport Format Set is a set of Transport Formats. E.g., a variable rate DCH has a Transport Format Set (one Transport Format for each rate), whereas a fixed rate DCH has a single Transport Format.

4.1.2.1.2 Model of physical layer of the MS

4.1.2.1.2.1 Uplink

Figure 10 shows a model of the MS's physical layer in the uplink for both FDD and TDD.

The model shows that one or several DCHs can be processed and multiplexed together by the same coding and multiplexing unit. The single output data stream from the coding and multiplexing unit is denoted *Coded Composite Transport Channel (CCTrCH)*.

The data stream of the CCTrCH is fed to a data demultiplexer/splitter unit that splits the CCTrCH's data stream onto one or several *Physical Channel Data Streams*.

The current configuration of the coding and multiplexing unit (transport format) is either signalled to, or optionally blindly detected by, the network for each 10 ms frame. If the configuration is signalled, the Transport Format Indicator (TFI) bits represent it. Note that the TFI signalling only consists of pointing out the current transport formats within the already configured transport format sets. In the uplink there is only one TFI representing the current transport formats on all DCHs of one CCTrCH simultaneously. In FDD the physical channel data stream carrying the TFI is mapped onto the physical channel carrying the power control bits and the pilot. The random access transport channel (RACH) is the only common type transport channel in the uplink. RACHs are always mapped one-to-one onto physical channels, i.e. there is no physical layer multiplexing of RACH. The MAC layer handles Service multiplexing.

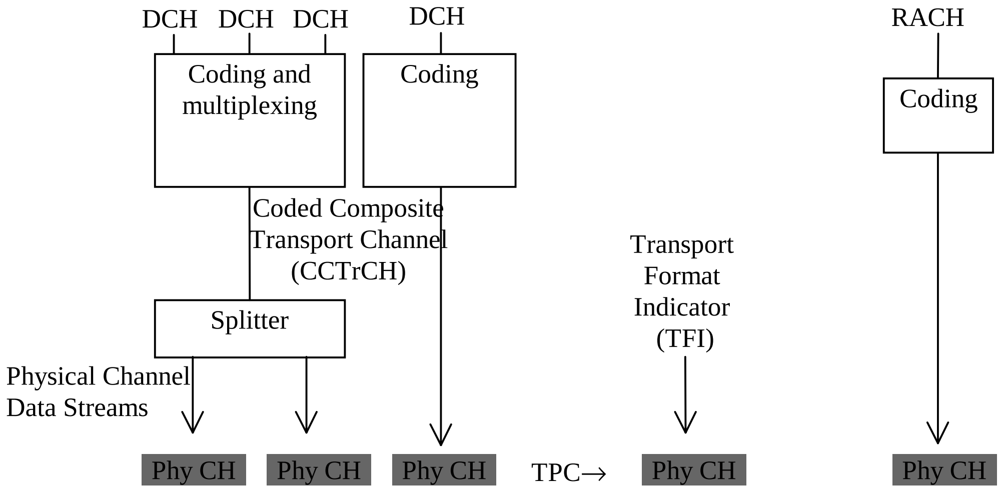

Figure 10. Model of the MS's physical layer – uplink

4.1.2.1.2.2 Downlink

Figure 11 and Figure 12 show the model of the MS's physical layer for the downlink in the FDD and TDD mode respectively.

For the DCHs the model is quite similar as the uplink model. The mapping between DCHs and physical channel data streams works in the same way as for the uplink. Note however, that the number of DCHs, the coding and multiplexing etc. may be different in uplink and downlink.

In the FDD mode differences are mainly due to the soft and softer handover. Further, the pilot, TPC bits and TFIs are time multiplexed onto the same physical channel(s) as the DCHs. Further, the definition of physical channel data is somewhat different from the uplink.

Note that it is logically one and the same physical data stream in the active set of cells, even though physically there is one stream for each cell. The same processing and multiplexing is done in each cell. The only difference between the cells is the actual codes, and these codes of course correspond to the same spreading factor.

The physical channels carrying the same physical data stream are combined in the MS receiver, excluding the pilot, and in some cases the TPC bits and transport format indicators (TFIs). TPC bits received on certain physical channels may be combined, e.g. physical channels from cells belonging to the same site (softer handover), provided that UTRAN has informed the MS that the TPC information on these channels is identical. The TFIs may also be combined provided that all physical data streams are identical in the associated cells. Figure 11 shows the case where one of the physical data streams is only transmitted in one of the cells, while two other physical data streams are transmitted in three cells, i.e. there are two different active sets for the MS. This would be the situation if e.g. a certain type of service should not employ soft handover whereas other simultaneous services should. Since the number of DCHs and thereby the combinations of transport format sets now will be different between different cells, the TFIs will also differ. In this example the TFIs transmitted from Cell 2 and Cell 3 will be exactly identical and may therefore be combined by the MS. However, the TFI from Cell 1 will be different. If different active sets between physical data streams are allowed, UTRAN must inform the MS of what TFIs are identical. Note that physical channel data streams that are related to the same CCTrCH are always transmitted in the same set of cells. There are three types of common transport channels in the downlink, namely BCCH, FACH and PCH. Downlink common transport channels are mapped one-to-one onto separate physical channels. The MAC layer handles Service multiplexing.

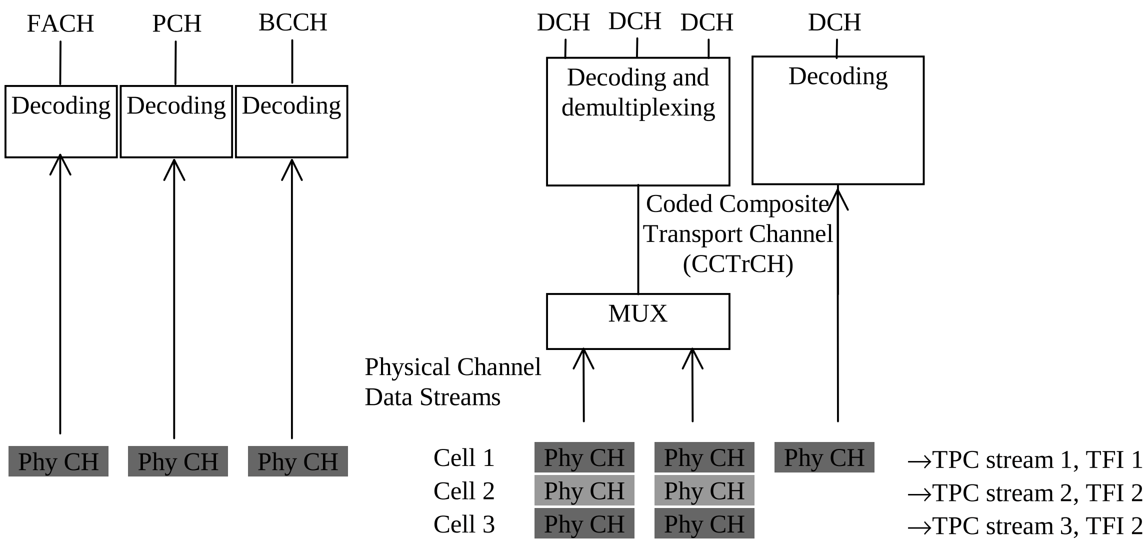

Figure 11. Model of the MS's physical layer – downlink FDD mode

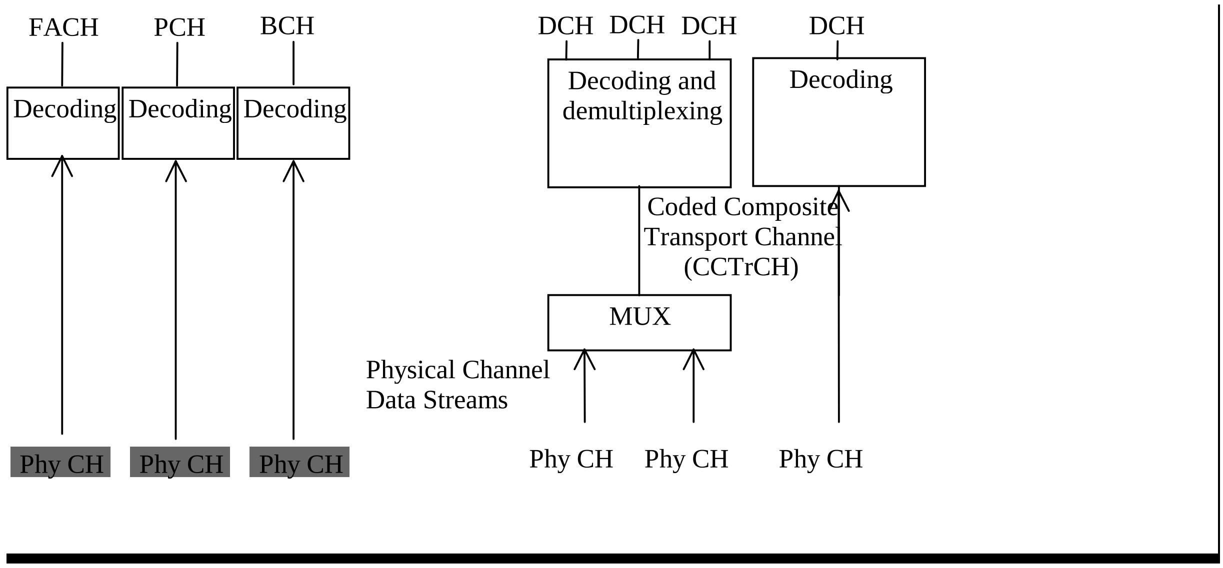

Figure 12: Model of the MS's physical layer – downlink TDD mode

4.1.2.2 L1 Functions

The physical layer performs the following main functions:

- FEC encoding/decoding of transport channels
- Measurements
- Macro diversity distribution/combining and soft handover execution
- Multiplexing/de-multiplexing of transport channels and of coded composite transport channels
- Mapping of coded composite transport channels on physical channels
- Modulation and spreading/demodulation and de-spreading of physical channels
- Frequency and time (chip, bit, slot, frame) synchronisation
- Closed-loop power control
- Power weighting and combining of physical channels

- RF processing

4.1.3 Layer 2 Services and Functions

4.1.3.1 MAC sub-layer

4.1.3.1.1 MAC services to upper layers

- **Data transfer**. This service provides unacknowledged transfer of MAC SDUs between peer MAC entities.

- **Reallocation of radio resources and MAC parameters**. This service performs on request of RRC execution of radio resource reallocation and change of MAC parameters, i.e. reconfiguration of MAC functions such as change of identity of UE, change of transport format (combination) sets, change of transport channel type. In TDD mode, in addition, the MAC can handle resource allocation autonomously.

The following potential services are regarded as further study items:

- **Reporting of measurements**. Local measurements such as traffic volume, quality indication, MAC status indication, [other MAC measurements tbd.], are reported to RRC.
- **Allocation/deallocation of radio resources**. Indication to RRC that allocation/deallocation of a MAC bearer is required. In TDD mode, the MAC can alternatively perform resource allocation autonomously.

4.1.3.1.2 Logical channels

The MAC layer provides data transfer services to upper layers on logical channels. A set of logical channel types is defined for different kinds of data transfer services as offered by MAC. Each logical channel type is defined by what type of information is transferred.

A general classification of logical channels is into two groups:

- Control Channels (for the transfer of control plane information)
- Traffic Channels (for the transfer of user plane information)

The configuration of logical channel types is depicted in Figure 13.

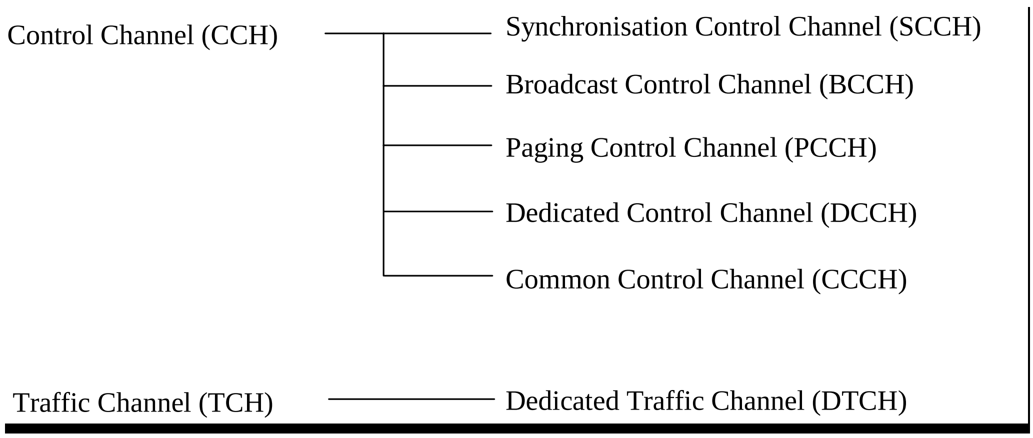

Figure 13: Logical channel structure

4.1.3.1.2.1 Control Channels

Control channels are used for transfer of control plane information only.

Synchronisation Control Channel (SCCH)

A downlink channel for broadcasting synchronisation information (cell ID, optional information) in case of TDD operation.

Broadcast Control Channel (BCCH)

A downlink channel for broadcasting system control information.

Paging Control Channel (PCCH)

A downlink channel that transfers paging information. This channel is used when the network does not know the location cell of the UE.

Common Control Channel (CCCH)

Bi-directional channel for transmitting control information between network and UEs. This channel is commonly used by the UEs having no RRC connection with the network.

Dedicated Control Channel (DCCH)

A point-to-point bi-directional channel that transmits dedicated control information between a UE and the network. This channel is established through RRC connection set-up procedure.

4.1.3.1.2.2 Traffic Channels

Traffic channels are used for the transfer of user plane information only.

Dedicated Traffic Channel (DTCH)

A Dedicated Traffic Channel (DTCH) is a point-to-point channel, dedicated to one UE, for the transfer of user information. A DTCH can exist in both uplink and downlink.

4.1.3.1.3 Mapping between logical channels and transport channels

The following connections between logical channels and transport channels exist:

- SCCH is connected to SCH
- BCCH is connected to BCH
- PCCH is connected to PCH
- CCCH is connected to RACH and FACH
- DCCH and DTCH can be connected to either RACH and FACH, to RACH and DSCH, to DCH and DSCH, or to a DCH

The mappings as seen from the UE and UTRAN sides are shown in Figure 14 and Figure 15 respectively.

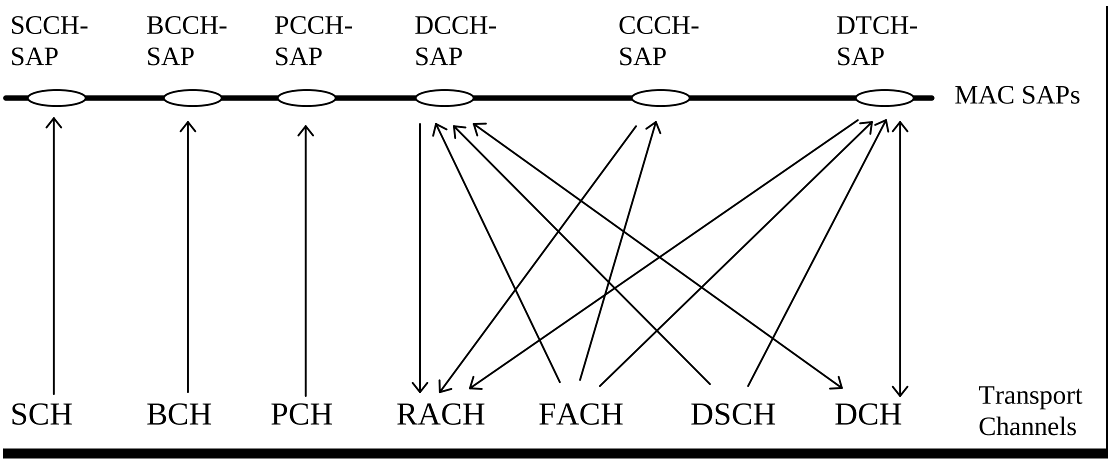

Figure 14: Logical channels mapped onto transport channels, seen from the UE side

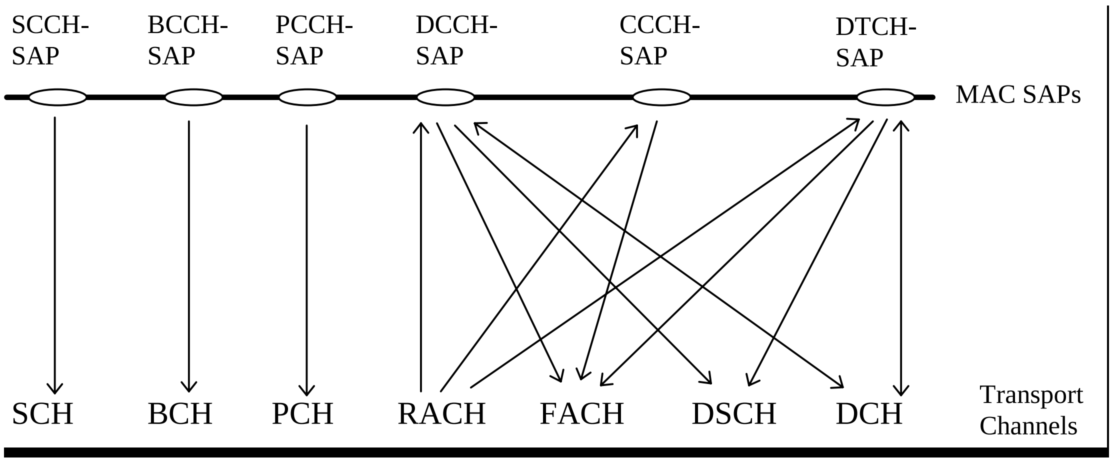

Figure 15: Logical channels mapped onto transport channels, seen from the UTRAN side

4.1.3.1.4 MAC functions

The functions of MAC include:

- **Selection of appropriate Transport Format for each Transport Channel depending on instantaneous source rate**. Given the Transport Format Combination Set assigned by L3, MAC selects the appropriate transport format within an assigned transport format set for each active transport channel depending on source rate**.** The control of transport formats ensures efficient use of transport channels.
- **Priority handling between data flows of one user**. When selecting between the Transport Format Combinations in the given Transport Format Combination Set, priorities of the data flows to be mapped onto the corresponding Transport Channels can be taken into account. Priorities are e.g. given by attributes of radio access bearer services and RLC buffer status. The priority handling is achieved by selecting a Transport Format Combination for which high priority data is mapped onto L1 with a “high bit rate” Transport Format, at the same time letting lower priority data be mapped with a “low bit rate” (could be zero bit rate) Transport Format.
- **Priority handling between users by means of dynamic scheduling**. In order to utilise the spectrum resources efficiently for bursty traffic (packet), a dynamic scheduling function may be applied. Priority handling on common channels is realised by MAC. Regarding dedicated channels in the downlink the traffic can be scheduled by co-ordinating the selection of Transport Format Combinations for different users appropriately, taking the maximum allowed interference level and radio access bearer priorities (or corresponding) of different users into account. In the uplink the same functionality can be achieved, but then requiring signalling for fast resource requests and allocations of Transport Formats or Transport Format Combinations. By allocations is meant pointing out Transport Formats or Transport Format Combinations in the already assigned Transport Format Combination Sets. Whether or not this dynamic scheduling on dedicated channels belongs to MAC or RRC is regarded as further study item.

Note that in the TDD mode the data to be transported are represented in terms of sets of resource units.

- **Scheduling of broadcast, paging and notification messages**. This function provides mechanisms for efficient transfer of broadcast, paging and notification messages by means of appropriate scheduling and repetition of the messages.
- **Identification of MSs on common transport channels**. When a particular MS is addressed on a common downlink channel, or when an MS is using the RACH, there is a need for inband identification of the MS. Since the MAC layer handles the access to, and multiplexing onto, the transport channels, the identification functionality is naturally also placed in MAC.

- **Multiplexing/demultiplexing of higher layer PDUs into/from transport blocks delivered to/from the physical layer on common transport channels**. MAC should support service multiplexing for common transport channels, since the physical layer does not support multiplexing of these channels.

- **Routing of higher layer signalling**. This function performs the mapping of higher layer signalling messages to the appropriate transport channel. This function is required in TDD mode, where the MAC performs resource allocation autonomously.
- **Maintenance of a MAC signalling connection between peer MAC entities**. This function supports unacknowledged transfer of MAC-internal messages between peer MAC entities. A MAC signalling connection is required in the TDD mode. Whether or not it is also needed in the FDD mode is ffs.

The following potential functions are regarded as further study items:

- **Multiplexing/demultiplexing of higher layer PDUs into/from transport block sets delivered to/from the physical layer on dedicated transport channels**. MAC should optionally support service multiplexing for dedicated transport channels. This function is needed for the case where the physical layer cannot offer sufficiently many DCHs or transport formats for each of these. In this case the identification of multiplexing is contained in the MAC protocol control information.
- **Dynamic Transport Channel type switching.** Execution of the switching between common and dedicated transport channels based on a switching decision derived by RRC.
- **Traffic volume monitoring**. Measurement of traffic volume and reporting to RRC.
- **Monitoring the links of the assigned resources**. This function provides means for monitoring link quality in TDD mode (may also be defined as L1 function).
- **Support of open loop power control**. Handling of parameters required for open loop power control, e.g. power and interference levels to be broadcast and computation of initial power levels to be used on physical channels.
- **Processing of messages received at common control channels**. This function is applied in TDD mode to support a data transfer on common control channels to support MAC operation.

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4.1.3.2 RLC sublayer

4.1.3.2.1 Services provided to the upper layer

- **L2 connection establishment/release**. This service performs establishment/release of L2 connections.
- **Transparent data transfer**. This service transmits L2 SDUs without adding any protocol information (“null” mode). This means that essentially the user of L2 accesses directly the SAP at the layer below.
- **Unacknowledged data transfer**. This service transmits L2 SDUs without guaranteeing delivery to the peer entity.
- **Acknowledged data transfer**. This service transmits L2 SDUs and guarantees delivery to the peer entity. In case L2 is unable to deliver the data correctly, the user of L2 at the transmitting side is notified. For this service, both in-sequence and out-of-sequence delivery are supported. In many cases a higher layer protocol can restore the order of its PDUs. As long as the out-of-sequence properties of the lower layer are known and controlled (i.e. the higher layer protocol will not immediately request retransmission of a missing PDU) allowing out-of-sequence delivery can save memory space in the receiving RLC.
- **QoS setting.** The retransmission protocol shall be configurable by layer 3 to provide different levels of QoS.

The length of an L2 SDU is variable up to the maximum length specified for each data transfer service [ffs.]. There is only a single L2 connection per Radio Access Bearer.

4.1.3.2.2 RLC functions

- **Connection Control.** This function performs establishment, release, and maintenance of a L2 connection.
- **Segmentation and reassembly.** This function performs segmentation/reassembly of variable-length L2 SDUs into/from smaller L2 PDUs. The size oft the smallest retransmission unit shall be determined by the

smallest possible bit rate. The L2 PDU size is adjustable to the actual set of transport formats. This function is applied in acknowledged and unacknowledged data transfer.

- **Transfer of user data.** This function is used for conveyance of data between users of L2 services. L2 supports acknowledged, unacknowledged and transparent data transfer.
- **Error correction.** This function provides error correction by retransmission (e.g. Selective Repeat, Go Back N, or a Stop-and-Wait ARQ) in acknowledged data transfer mode.
- **In-sequence delivery of L2 SDUs to higher layers**. This function preserves the order of L2 SDUs that were submitted for transfer by L2 using the acknowledged data transfer service. If this function is not used, out-of-sequence delivery is provided.
- **Duplicate Detection.** This function detects duplicated received L2 PDUs and ensures that the resultant L2 SDU is delivered only once to the upper layer.
- **Flow control**. This function allows an L2 receiver to control the rate at which the peer L2 transmitting entity may send information.
- **Protocol error detection and recovery**. This function detects and recovers from errors in the operation of the L2 protocol.

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· The following potential function(s) are regarded as further study items:

- **Suspend/resume function**. Suspension and resumption of data transfer as in e.g. LAPDm (cf. the standard GSM 04.05).
- **Quick Repeat**. This function provides mechanisms to transmit a L2 PDU several times as part of unacknowledged data transfer service.
- **Keep Alive**. This function ensures that the two peer L2 entities remain in a relationship even in the case of a prolonged absence of data transfer.
- **FCS error detection and handling**. This function provides mechanisms for the detection and handling of corrupted L2 PDU’s through FCS (Frame check sequence). A corrupted acknowledged L2 PDU is retransmitted. A corrupted unacknowledged L2 PDU is discarded. Note that Layer 1 already provides FCS error detection. Whether or not additional error detection functionality is required on RLC level is ffs.
- **Ciphering.** This function prevents unauthorised acquisition of data.

4.1.4 Layer 3 - RRC services and functions

4.1.4.1 RRC services

4.1.4.1.1 General control

The General Control (GC) SAP provides an information broadcast service. This service broadcasts information to all UEs in a certain geographical area. The basic requirements from such service are:

- It should be possible to broadcast non-access stratum information in a certain geographical area.
- The information is transferred on an unassured mode link. Unassured mode means that the delivery of the broadcast information can not be guaranteed (typically no retransmission scheme is used). It seems reasonable to use an unassured mode link since the information is broadcast to a lot of UEs and since broadcast information often is repeated periodically.
- It should be possible to do repeated transmissions of the broadcast information (the non-access stratum controls how it is repeated).
- The point where the UE received the broadcast information should be included, when the access stratum delivers broadcast information to the non-access stratum.

4.1.4.1.2 Notification

The Notification (Nt) SAP provides paging and notification broadcast services. The paging service sends information to a specific UE(s). The information is broadcast in a certain geographical area but addressed to a specific UE(s). The basic requirements from such service are:

- It should be possible to broadcast paging information to a number of UEs in a certain geographical area.
- The information is transferred on an unacknowledged mode link. It is assumed that the protocol entities in non-access stratum handle any kind of retransmission of paging information.

The notification broadcast service broadcasts information to all UEs in a certain geographical. The basic requirements from this service are typically the same as for the information broadcast service of the GC SAP:

- It should be possible to broadcast notification information in a certain geographical area.
- The information is transferred on an unacknowledged mode link.

4.1.4.1.3 Dedicated control

The Dedicated Control (DC) SAP provides services for establishment/release of a connection and transfer of messages using this connection. It should also be possible to transfer a message during the establishment phase. The basic requirements from the establishment/release services are:

- It should be possible to establish connections (both point and group connections).
- It should be possible to transfer an initial message during the connection establishment phase. This message transfer has the same requirements as the information transfer service.
- It should be possible to release connections.

The information transfer service sends a message using the earlier established connection. It is possible to specify the quality of service requirements for each message. A finite number of quality of service classes will be specified, but currently no class has been specified. In order to get an idea of the basic requirements, the CC and MM protocols in GSM are used as a reference. A GSM based core network is chosen since it is one main option for UMTS. Considering the existing GSM specification of CC and MM the basic requirements from the information transfer service are (these are services provided by RR and the data link layer in GSM):

- Acknowledged mode link for transfer of messages
  This assured mode link guarantees that the CC and MM messages are transferred to the corresponding side. Assured mode means that the delivery of the paging information can be guaranteed (some kind of retransmission scheme is used). A connection between two DC SAPs using an assured mode link has already been introduced and is called *signalling connection*. This link should also guarantee that no messages are lost or duplicated during handover.
- Preserved message order
  The order of the transferred messages is preserved.
- Priority handling
  If SMS messages should be transported through the control plane it should be possible to give higher priority to signalling messages.

The CC and MM protocols also expect other services, which can not be supported by the current primitives of the (DC) SAP, e.g. indication of radio link failure.

4.1.4.2 RRC functions

The Radio Resource Control (RRC) layer handles the control plane signalling of Layer 3 between the MSs and URAN. The RRC performs the following functions:

- **Broadcast of information provided by the non-access stratum (Core Network).** The RRC layer performs system information broadcasting from the network to all UEs. The system information is normally repeated on a regular basis. This function supports broadcast of higher layer (above RRC) information. Typically this information is not cell specific.
- **Broadcast of information related to the access stratum.** The RRC layer performs system information broadcasting from the network to all UEs. This function supports broadcast of typically cell-specific information.
- **Establishment, maintenance and release of an RRC connection between the UE and UTRAN.** The

establishment of an RRC connection is initiated by a request from higher layers at the UE side to establish the first Signalling Connection for the UE. The establishment of an RRC connection includes an optional cell re-selection, an admission control, and a layer 2 signalling link establishment. The release of an RRC connection can be initiated by a request from higher layers to release the last Signalling Connection for the UE or by the RRC layer itself in case of RRC connection failure. The RRC layer detects loss of RRC connection and releases resources assigned for the RRC connection in case of connection failure.

- **Establishment, reconfiguration and release of Radio Access Bearers.** The RRC layer can, on request from higher layers, perform the establishment, reconfiguration and release of radio access bearers in the user plane. A number of radio access bearers can be established to an UE at the same time. At establishment and reconfiguration, the RRC layer performs admission control and selects parameters describing the radio access bearer processing in layer 2 and layer 1, based on information from higher layers.
- **Assignment, reconfiguration and release of radio resources for the RRC connection.** The RRC layer handles the assignment of radio resources (e.g. codes) needed for the RRC connection including needs from both the control and user plane. The RRC layer may reconfigure radio resources during an established RRC connection. This function includes co-ordination of the radio resource allocation between multiple radio bearers related to the same RRC connection.
- **RRC connection mobility functions.** The RRC layer performs evaluation, decision and execution related to RRC connection mobility during an established RRC connection, such as handover, cell re-selection and cell/paging area update procedures, based on e.g. measurements done by the UE**.**
- **Arbitration of the radio resource allocation between the cells**. This function shall ensure optimal performance of the overall UTRAN capacity.
- **Control of requested QoS**. This function shall ensure that the QoS requested for the radio access bearers can be met. This includes the allocation of a sufficient number of radio resources.
- **UE measurement reporting and control of the reporting.** The measurements performed by the UE are controlled by the RRC layer, in terms of what to measure, when to measure and how to report. The RRC layer also performs the reporting of the measurements from the UE to the network.
- **Outer loop power control.** The RRC layer controls setting of the target of the closed loop power control.
- **Control of ciphering.** The RRC layer provides procedures for setting of ciphering (on/off) between the UE and UTRAN.

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· The following functions are regarded as further study items:

- **Initial cell selection and re-selection in idle mode.** Idle mode is a further study item.
- **Paging/notification.** The RRC layer can broadcast paging information from the network to selected UEs. Higher layers on the network side can request paging and notification. The RRC layer can also initiate paging during an established RRC connection.
- **Contention resolution**. Further study item.
- **Congestion control.** Further study item.

4.1.5 Interactions between MAC and RRC in the C plane

The RRC protocol controls and signals the allocation of radio resources to the UE. RRC allows MAC to arbitrate between users and radio access bearers within the radio resource allocation. The RRC uses the measurements done by the lower layers to determine which radio resources that are available. Therefore it is a need for a measurement report from the UE RRC to the UTRAN RRC. Figure 16 illustrates the principle. Whether there is a need for explicit MAC signalling for the arbitration between users and radio access bearers is FFS.

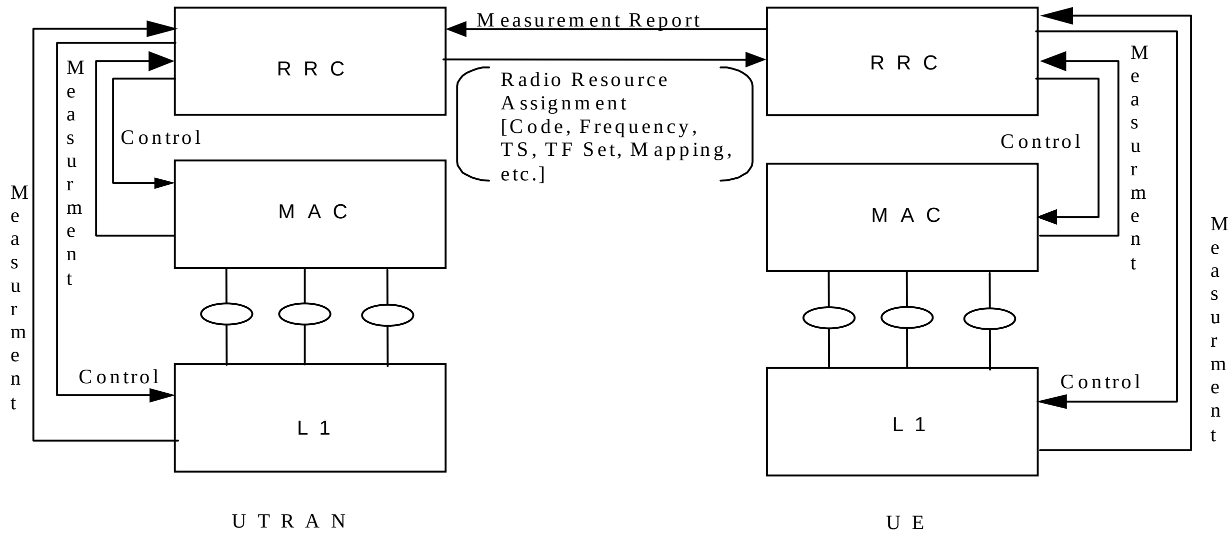


Figure 16. Interactions between MAC and RRC

5. LAYER 1 DESCRIPTION (FDD MODE)

5.1 Transport channels and physical channels (FDD)

5.1.1 Transport channels

Transport channels are the services offered by Layer 1 to the higher layers, cf. Section Layer 1 Services and functions.

5.1.1.1 Dedicated transport channel

5.1.1.1.1 DCH - Dedicated Channel

The Dedicated Channel (DCH) is a downlink or uplink transport channel that is used to carry user or control information between the network and a mobile station. The DCH thus corresponds to the three channels Dedicated Traffic Channel (DTCH), Stand-alone Dedicated Control Channel (SDCCH), and Associated Control Channel (ACCH) defined within ITU-R M.1035. The DCH is transmitted over the entire cell or over only a part of the cell using lobe-forming antennas.

5.1.1.2 Common transport channels

5.1.1.2.1 BCCH - Broadcast Control Channel

The Broadcast Control Channel (BCCH) is a downlink transport channel that is used to broadcast system- and cell-specific information. The BCCH is always transmitted over the entire cell.

5.1.1.2.2 FACH - Forward Access Channel

The Forward Access Channel (FACH) is a downlink transport channel that is used to carry control information to a mobile station when the system knows the location cell of the mobile station. The FACH may also carry short user packets. The FACH is transmitted over the entire cell or over only a part of the cell using lobe-forming antennas.

5.1.1.2.3 PCH - Paging Channel

The Paging Channel (PCH) is a downlink transport channel that is used to carry control information to a mobile station when the system does not know the location cell of the mobile station. The PCH is always transmitted over the entire cell.

5.1.1.2.4 RACH - Random Access Channel

The Random Access Channel (RACH) is an uplink transport channel that is used to carry control information from a mobile station. The RACH may also carry short user packets. The RACH is always received from the entire cell.

5.1.2 Physical channels

5.1.2.1 The physical resource

The basic physical resource is the code/frequency plane. In addition, on the uplink, different information streams may be transmitted on the I and Q branch. Consequently, a physical channel corresponds to a specific carrier frequency, code, and, on the uplink, relative phase (0 or $\pi/2$).

5.1.2.2 Uplink physical channels

5.1.2.2.1 Dedicated uplink physical channels

There are two types of uplink dedicated physical channels, the uplink Dedicated Physical Data Channel (uplink DPDCH) and the uplink Dedicated Physical Control Channel (uplink DPCCH).

The uplink DPDCH is used to carry dedicated data generated at Layer 2 and above, i.e. the dedicated transport channel (DCH). There may be zero, one, or several uplink DPDCHs on each Layer 1 connection.

The uplink DPCCH is used to carry control information generated at Layer 1. The Layer 1 control information consists of known pilot bits to support channel estimation for coherent detection, transmit power-control (TPC) commands, and an optional transport-format indicator (TFI). The transport-format indicator informs the receiver about the instantaneous parameters of the different transport channels multiplexed on the uplink DPDCH, see further Section 3. There is one and only one uplink DPCCH on each Layer 1 connection.

Frame structure

Figure 17 shows the frame structure of the uplink dedicated physical channels. Each frame of length 10 ms is split into 16 slots, each of length T_{slot} = 0.625 ms, corresponding to one power-control period. A super frame corresponds to 72 consecutive frames, i.e. the super-frame length is 720 ms.

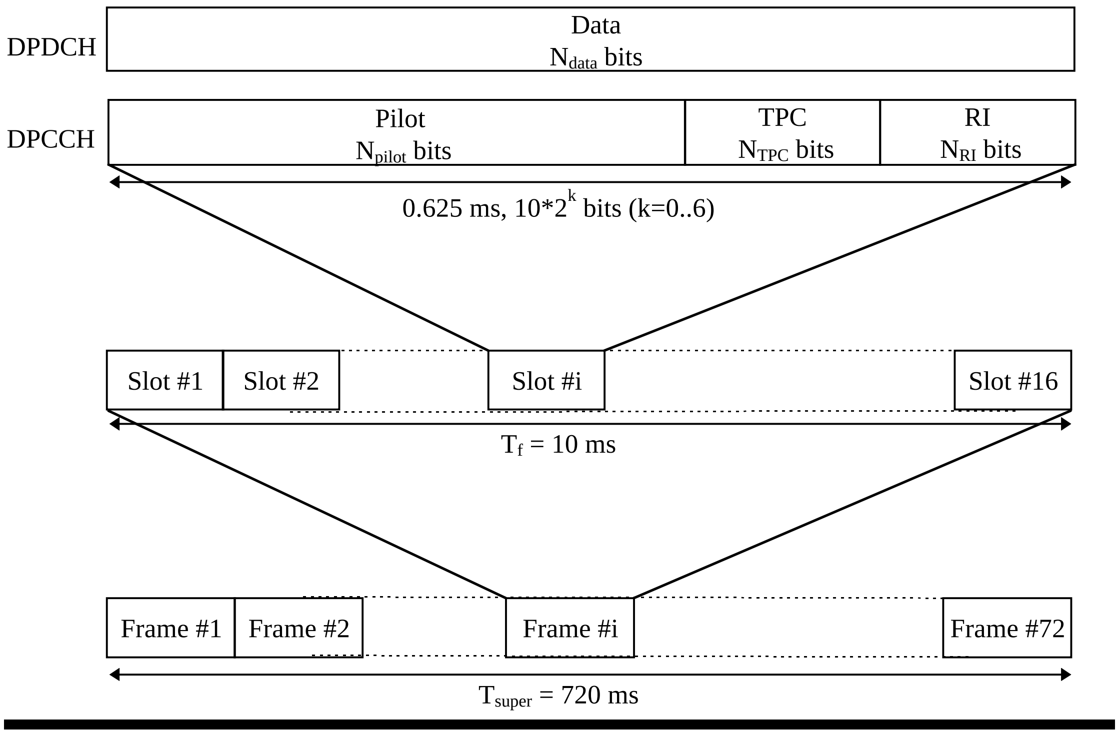


Figure 17. Frame structure for uplink DPDCH/DPCCH

The parameter k in Figure 17 determines the number of bits per uplink DPDCH/DPCCH slot. It is related to the spreading factor SF of the physical channel as SF = $256/2^k$. The spreading factor may thus range from 256 down to 4. Note that an uplink DPDCH and uplink DPCCH on the same Layer 1 connection generally are of different rates, i.e. have different spreading factors and different values of k.

The exact number of bits of the different uplink DPCCH fields in Figure 17 (N_{pilot}, N_{TPC}, and N_{TFI}) is yet to be determined.

Multi-code operation is possible for the uplink dedicated physical channels. When multi-code transmission is used, several parallel DPDCH are transmitted using different channelisation codes, see Section Channelisation codes. However, there is only one DPCCH per connection.

5.1.2.2.2 Common uplink physical channels

5.1.2.2.2.1 Physical Random Access Channel

The Physical Random Access Channel (PRACH) is used to carry the RACH. It is based on a Slotted ALOHA approach, i.e. a mobile station can start the transmission of the PRACH at a number of well-defined time-offsets, relative to the frame boundary of the received BCCH of the current cell. The different time offsets are denoted *access slots* and are spaced 1.25 ms apart as illustrated in Figure 18. Information on what access slots are available in the current cell is broadcast on the BCCH.

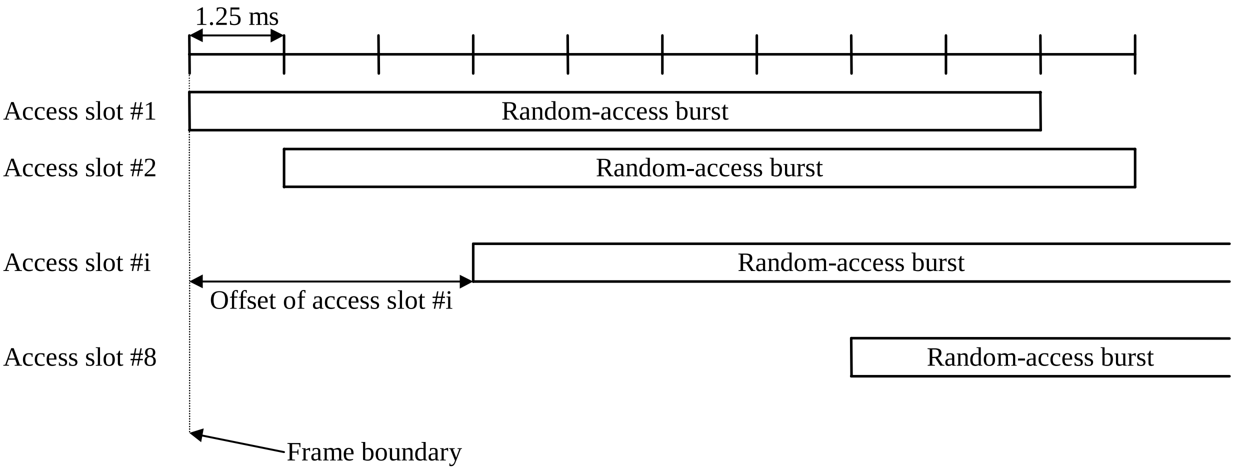


Figure 18. Access slots.

The structure of the random access burst of Figure 18 is shown in Figure 19. The random access burst consists of two parts, a *preamble* part of length 1 ms and a *message* part of length 10 ms. Between the preamble part and the message part there is an idle time period of length 0.25 ms (preliminary value). The idle time period allows for detection of the preamble part and subsequent on-line processing of the message part.

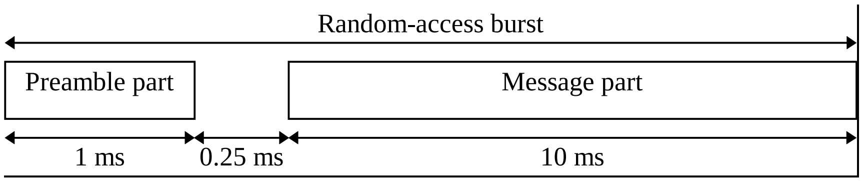


Figure 19. Structure of the Random Access burst.

Preamble part

The preamble part of the random-access burst consists of a *signature* of length 16 complex symbols (±1±j). Each preamble symbol is spread with a 256 chip real Orthogonal Gold code. There are a total of 16 different signatures, based on the Orthogonal Gold code set of length 16 (see Section Preamble spreading code for more details).

Message part

The message part of the random-access burst has the same structure as the uplink dedicated physical channel. It consists of a data part, corresponding to the uplink DPDCH, and a Layer 1 control part, corresponding to the uplink DPCCH, see Figure 20. The data and control parts are transmitted in parallel. The data part carries the random access request or small user packets. The spreading factor of the data part is limited to SF∈{256, 128, 64, 32} corresponding to channel bit rates of 16, 32, 64, and 128 kbps respectively. The control part carries pilot bits and rate information, using a spreading factor of 256. The rate information indicates which channelisation code (or rather the spreading factor of the channelisation code) is used on the data part, see further Section Random access codes.

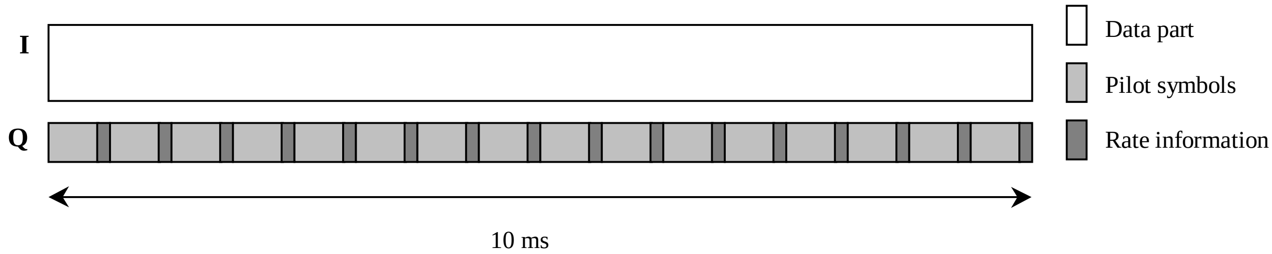


Figure 20. The message part of the random access burst.

Figure 21 shows the structure of the data part of the Random-Access burst. It consists of the following fields (the values in brackets are preliminary values):

- Mobile station identification (MS ID) [16 bits]. The MS ID is chosen at random by the mobile station at the time of each Random-Access attempt.
- Required Service [3 bits]. This field informs the base station what type of service is required (short packet transmission, dedicated-channel set-up, etc.)
- An optional user packet
- A CRC to detect errors in the data part of the Random-Access burst [8 bits].

Figure 21. Structure of Random-Access burst data part.

5.1.2.3 Downlink physical channels

<u>5.1.2.3.1 Dedicated physical channels</u>

There is only one type of downlink dedicated physical channel, the Downlink Dedicated Physical Channel (downlink DPCH).

Within one downlink DPCH, dedicated data generated at Layer 2 and above, i.e. the dedicated transport channel (DCH), is transmitted in time-multiplex with control information generated at Layer 1 (known pilot bits, TPC commands, and an optional TFI). The downlink DPCH can thus be seen as a time multiplex of a downlink DPDCH and a downlink DPCCH, compare Section Dedicated uplink physical channels.

Frame structure

Figure 22 shows the frame structure of the downlink DPCH. Each frame of length 10 ms is split into 16 slots, each of length T_{slot} = 0.625 ms, corresponding to one power-control period. A super frame corresponds to 72 consecutive frames, i.e. the super-frame length is 720 ms.

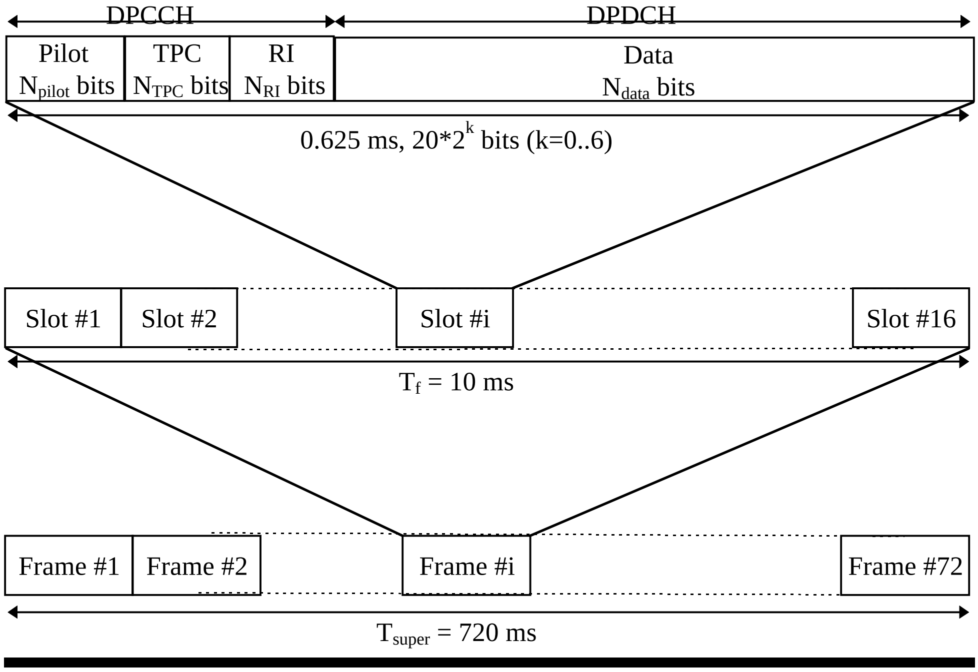


Figure 22. Frame structure for downlink DPCH.

The parameter k in Figure 22 determines the total number of bits per downlink DPCH slot. It is related to the spreading factor SF of the physical channel as SF = $256/2^k$. The spreading factor may thus range from 256 down to 4.

The exact number of bits of the different downlink DPCH fields in Figure 22 (N_{pilot}, N_{TPC}, N_{TFI}, and N_{data}) is yet to be determined. The overhead due to the DPCCH transmission has to be negotiated at the connection set-up and can be re-negotiated during the communication, in order to match particular propagation conditions.

The DPCCH fields are spread using the same channelisation code used for the DPDCH field. A channelisation code for the highest bit rate to be served during the connection (for a given DPCH) should be assigned (with spreading factor SF_1).

Note that connection-dedicated pilot bits are transmitted also for the downlink in order to support the use of downlink adaptive antennas.

When the total bit rate to be transmitted on one downlink connection exceeds the maximum bit rate for a downlink physical channel, multi-code transmission is employed, i.e. several parallel downlink DPCHs are transmitted for one connection using the same spreading factor. In this case, the Layer 1 control information is put on only the first downlink DPCH. The additional downlink DPCHs belonging to the connection do not transmit any data during the corresponding time period, see Figure 23.

Multiple codes may also transmitted in order to transmit different transport channels on different codes (code multiplex). In that case, the different parallel codes may have different spreading factors and the Layer 1 control information is transmitted on each code independently.

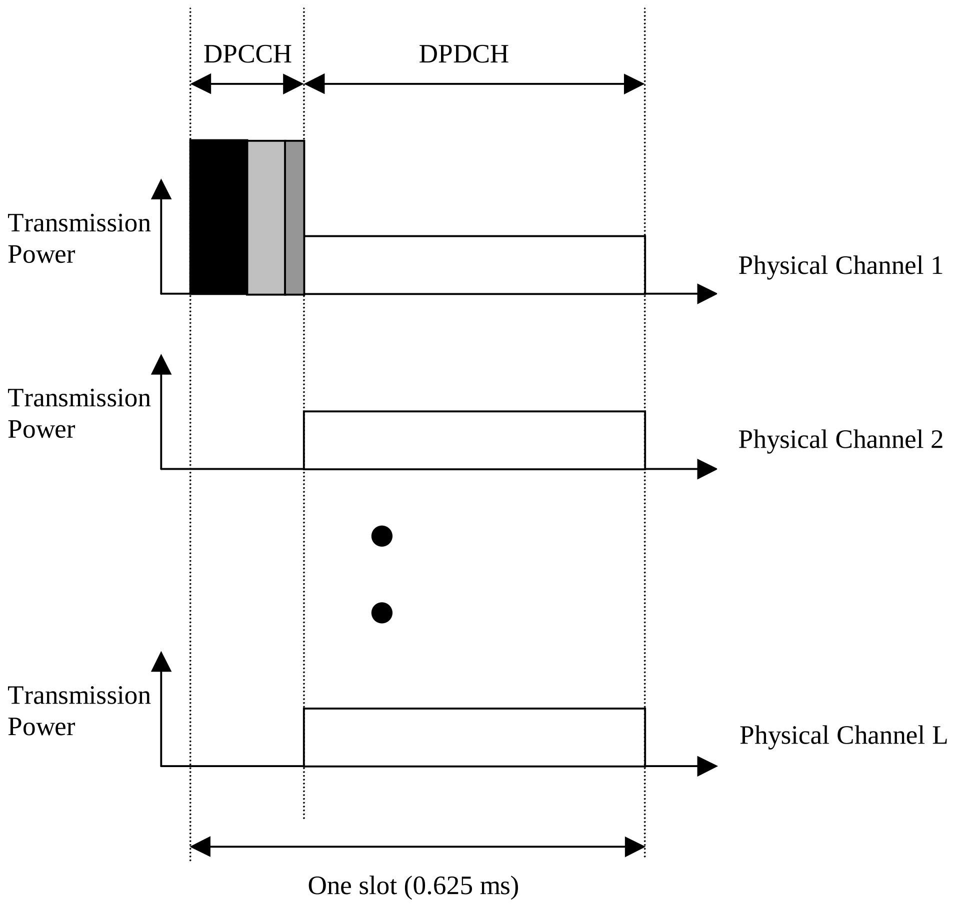


Figure 23. Downlink slot format in case of multi-code transmission.

5.1.2.3.2 Common physical channels

5.1.2.3.2.1 Primary Common Control Physical Channel (CCPCH)

The Primary CCPCH is a fixed rate (32 kbps, SF=256) downlink physical channels used to carry the BCCH.

Figure 24 shows the frame structure of the Primary CCPCH. The frame structure differs from the downlink DPCH in that no TPC commands or TFI is transmitted. The only Layer 1 control information is the common pilot bits needed for coherent detection.

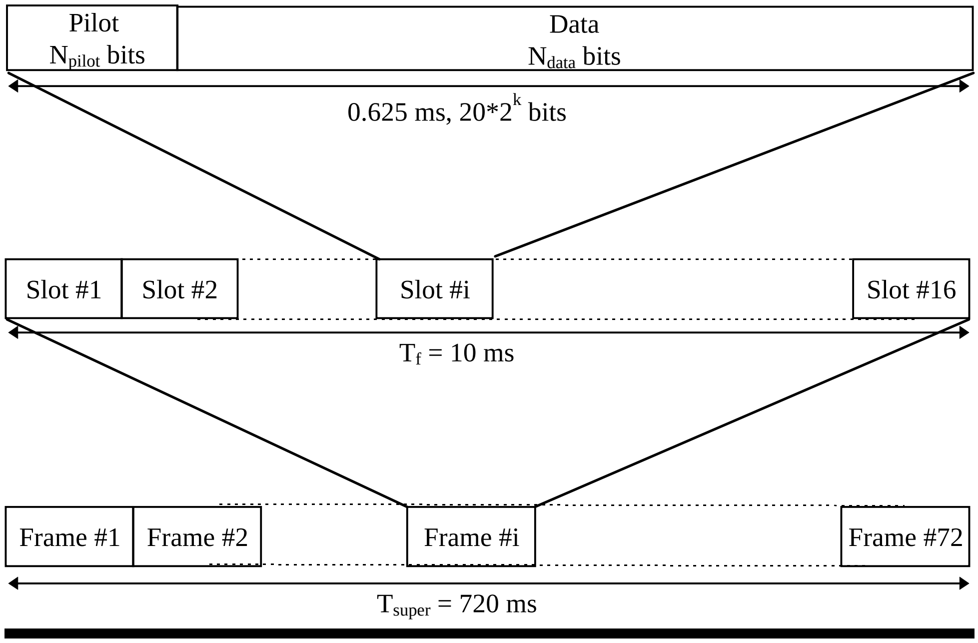


Figure 24. Frame structure for Primary Common Control Physical Channel.

5.1.2.3.2.2 Secondary Common Control Physical Channel

The secondary CCPCH is used to carry the FACH and PCH. It is of constant rate. However, in contrast to the Primary CCPCH, the rate may be different for different secondary CCPCH within one cell and between cells, in order to be able to allocate different amount of FACH and PCH capacity to a cell. The rate and spreading factor of each secondary CCPCH is broadcast on the BCCH. The set of possible rates is the same as for the downlink DPCH, see Section Dedicated physical channels.

The frame structure of the Secondary CCPCH is shown in Figure 25.

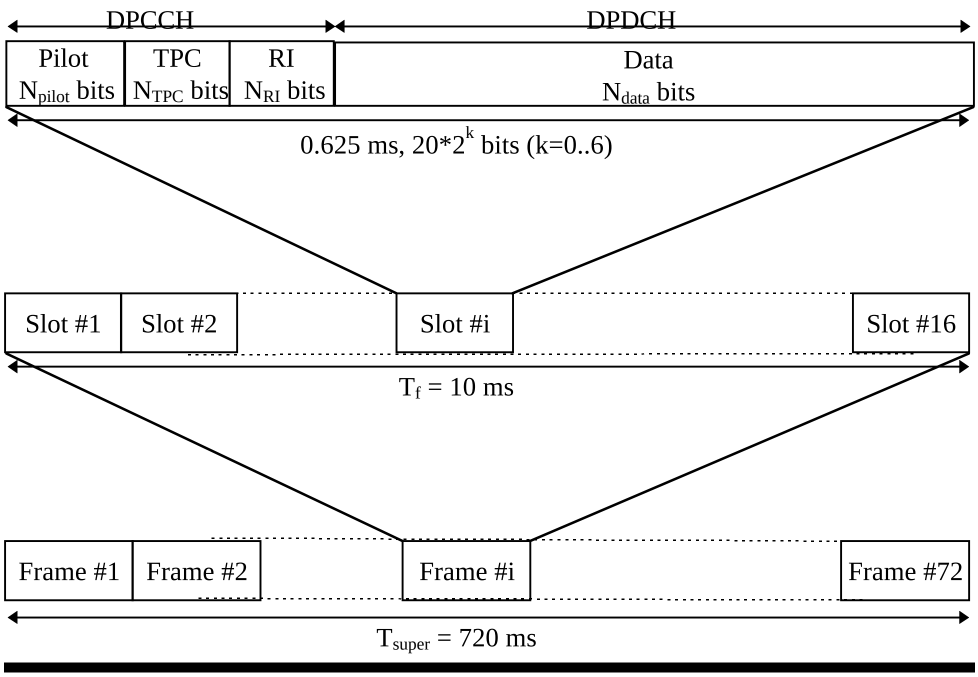
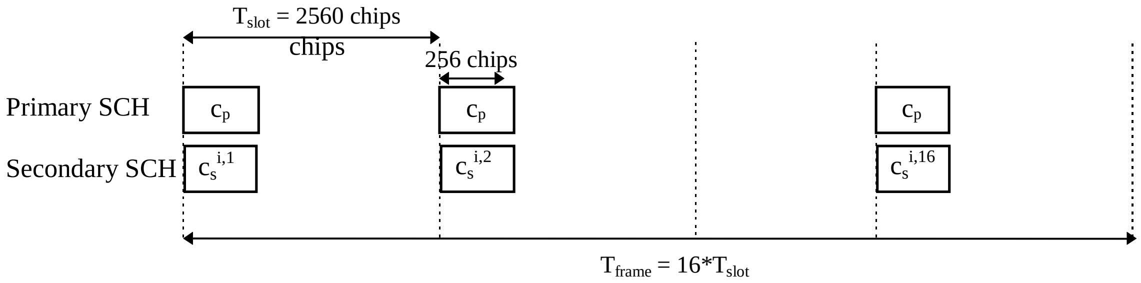


Figure 25. Frame structure for Secondary Common Control Physical Channel.

The FACH and PCH are mapped to separate Secondary CCPCHs. The main difference between a CCPCH and a downlink dedicated physical channel is that a CCPCH is not power controlled. The main difference between the Primary and Secondary CCPCH is that the Primary CCPCH has a fixed predefined rate while the Secondary CCPCH has a constant rate that may be different for different cells, depending on the capacity needed for FACH and PCH. Furthermore, a Primary CCPCH is continuously transmitted over the entire cell while a Secondary CCPCH is only transmitted when there is data available and may be transmitted in a narrow lobe in the same way as a dedicated physical channel (only valid for a Secondary CCPCH carrying the FACH).

5.1.2.3.2.3 Synchronisation Channel

The Synchronisation Channel (SCH) is a downlink signal used for cell search. The SCH consists of two sub channels, the Primary and Secondary SCH. Figure 26 illustrates the structure of the SCH:



c_p : Primary Synchronization Code
$c_s^{i,k}$: One of 17 possible Secondary Synchronization Codes
($c_s^{i,1}$, $c_s^{i,2}$, ..., $c_s^{i,16}$) encode cell specific long scrambling code group i

Figure 26. Structure of Synchronisation Channel (SCH).

The Primary SCH consists of an *unmodulated* orthogonal Gold code of length 256 chips, the Primary Synchronisation Code, transmitted once every slot. The Primary Synchronisation Code is the same for every base station (cell of Node B) in the system and is transmitted time-aligned with the BCCH slot boundary as illustrated in Figure 26.

The Secondary SCH consists of repeatedly transmitting a length 16 sequence of *unmodulated* Orthogonal Gold codes of length 256 chips, the Secondary Synchronisation Codes, transmitted in parallel with the Primary Synchronisation channel. Each Secondary Synchronisation code is chosen from a set of 17 different Orthogonal Gold codes of length 256. This sequence on the Secondary SCH indicates which of the 32 different code groups (see Section Scrambling code) the base station downlink scrambling code belongs. 32 sequences are used to encode the 32 different code groups each containing 16 scrambling codes. The 32 sequences are constructed such that their cyclic-shifts are unique, i.e., a non-zero cyclic shift less than 16 of any of the 32 sequences is not equivalent to some cyclic shift of any other of the 32 sequences. Also, a non-zero cyclic shift less than 16 of any of the sequences are not equivalent to itself with any other cyclic shift less than 16. This property is used to uniquely determine both the long code group and the frame timing in the second step of acquisition (see Section Initial cell search). The following sequences are used to encode the 32 different code groups each containing 16 scrambling codes (note that c_i indicates the i'th Secondary Short code of the 17 Orthogonal Gold codes).

$(c_1\ c_1\ c_2\ c_{11}\ c_6\ c_3\ c_{15}\ c_7\ c_8\ c_8\ c_7\ c_{15}\ c_3\ c_6\ c_{11}\ c_2)$
$(c_1\ c_2\ c_9\ c_3\ c_{10}\ c_{11}\ c_{13}\ c_{13}\ c_{11}\ c_{10}\ c_3\ c_9\ c_2\ c_1\ c_{16}\ c_{16})$
$(c_1\ c_3\ c_{16}\ c_{12}\ c_{14}\ c_2\ c_{11}\ c_2\ c_{14}\ c_{12}\ c_{16}\ c_3\ c_1\ c_{13}\ c_4\ c_{13})$
$(c_1\ c_4\ c_6\ c_4\ c_1\ c_{10}\ c_9\ c_8\ c_{17}\ c_{14}\ c_{12}\ c_{14}\ c_{17}\ c_8\ c_9\ c_{10})$
$(c_1\ c_5\ c_{13}\ c_{13}\ c_5\ c_1\ c_7\ c_{14}\ c_3\ c_{16}\ c_8\ c_8\ c_{16}\ c_3\ c_{14}\ c_7)$
$(c_1\ c_6\ c_3\ c_5\ c_9\ c_9\ c_5\ c_3\ c_6\ c_1\ c_4\ c_2\ c_{15}\ c_{15}\ c_2\ c_4)$
$(c_1\ c_7\ c_{10}\ c_{14}\ c_{13}\ c_{17}\ c_3\ c_9\ c_9\ c_3\ c_{17}\ c_{13}\ c_{14}\ c_{10}\ c_7\ c_1)$
$(c_1\ c_8\ c_{17}\ c_6\ c_{17}\ c_8\ c_1\ c_{15}\ c_{12}\ c_5\ c_{13}\ c_7\ c_{13}\ c_5\ c_{12}\ c_{15})$
$(c_1\ c_9\ c_7\ c_{15}\ c_4\ c_{16}\ c_{16}\ c_4\ c_{15}\ c_7\ c_9\ c_1\ c_{12}\ c_{17}\ c_{17}\ c_{12})$
$(c_1\ c_{10}\ c_{14}\ c_7\ c_8\ c_7\ c_{14}\ c_{10}\ c_1\ c_9\ c_5\ c_{12}\ c_{11}\ c_{12}\ c_5\ c_9)$
$(c_1\ c_{11}\ c_4\ c_{16}\ c_{12}\ c_{15}\ c_{12}\ c_{16}\ c_4\ c_{11}\ c_1\ c_6\ c_{10}\ c_7\ c_{10}\ c_6)$
$(c_1\ c_{12}\ c_{11}\ c_8\ c_{16}\ c_6\ c_{10}\ c_5\ c_7\ c_{13}\ c_{14}\ c_{17}\ c_9\ c_2\ c_{15}\ c_3)$
$(c_1\ c_{13}\ c_1\ c_{17}\ c_3\ c_{14}\ c_8\ c_{11}\ c_{10}\ c_{15}\ c_{10}\ c_{11}\ c_8\ c_{14}\ c_3\ c_{17})$
$(c_1\ c_{14}\ c_8\ c_9\ c_7\ c_5\ c_6\ c_{17}\ c_{13}\ c_{17}\ c_6\ c_5\ c_7\ c_9\ c_8\ c_{14})$
$(c_1\ c_{15}\ c_{15}\ c_1\ c_{11}\ c_{13}\ c_4\ c_6\ c_{16}\ c_2\ c_2\ c_{16}\ c_6\ c_4\ c_{13}\ c_{11})$
$(c_1\ c_{16}\ c_5\ c_{10}\ c_{15}\ c_4\ c_2\ c_{12}\ c_2\ c_4\ c_{15}\ c_{10}\ c_5\ c_{16}\ c_1\ c_8)$
$(c_1\ c_{17}\ c_{12}\ c_2\ c_2\ c_{12}\ c_{17}\ c_1\ c_5\ c_6\ c_{11}\ c_4\ c_4\ c_{11}\ c_6\ c_5)$
$(c_2\ c_8\ c_{11}\ c_{15}\ c_{14}\ c_1\ c_4\ c_{10}\ c_{10}\ c_4\ c_1\ c_{14}\ c_{15}\ c_{11}\ c_8\ c_2)$

$(c_2\ c_9\ c_1\ c_7\ c_1\ c_9\ c_2\ c_{16}\ c_{13}\ c_6\ c_{14}\ c_8\ c_{14}\ c_6\ c_{13}\ c_{16})$
$(c_2\ c_{10}\ c_8\ c_{16}\ c_5\ c_{17}\ c_{17}\ c_5\ c_{16}\ c_8\ c_{10}\ c_2\ c_{13}\ c_1\ c_1\ c_{13})$
$(c_2\ c_{11}\ c_{15}\ c_8\ c_9\ c_8\ c_{15}\ c_{11}\ c_2\ c_{10}\ c_6\ c_{13}\ c_{12}\ c_{13}\ c_6\ c_{10})$
$(c_2\ c_{12}\ c_5\ c_{17}\ c_{13}\ c_{16}\ c_{13}\ c_{17}\ c_5\ c_{12}\ c_2\ c_7\ c_{11}\ c_8\ c_{11}\ c_7)$
$(c_2\ c_{13}\ c_{12}\ c_9\ c_{17}\ c_7\ c_{11}\ c_6\ c_8\ c_{14}\ c_{15}\ c_1\ c_{10}\ c_3\ c_{16}\ c_4)$
$(c_2\ c_{14}\ c_2\ c_1\ c_4\ c_{15}\ c_9\ c_{12}\ c_{11}\ c_{16}\ c_{11}\ c_{12}\ c_9\ c_{15}\ c_4\ c_1)$
$(c_2\ c_{15}\ c_9\ c_{10}\ c_8\ c_6\ c_7\ c_1\ c_{14}\ c_1\ c_7\ c_6\ c_8\ c_{10}\ c_9\ c_{15})$
$(c_2\ c_{16}\ c_{16}\ c_2\ c_{12}\ c_{14}\ c_5\ c_7\ c_{17}\ c_3\ c_3\ c_{17}\ c_7\ c_5\ c_{14}\ c_{12})$
$(c_2\ c_{17}\ c_6\ c_{11}\ c_{16}\ c_5\ c_3\ c_{13}\ c_3\ c_5\ c_{16}\ c_{11}\ c_6\ c_{17}\ c_2\ c_9)$
$(c_2\ c_1\ c_{13}\ c_3\ c_3\ c_{13}\ c_1\ c_2\ c_6\ c_7\ c_{12}\ c_5\ c_5\ c_{12}\ c_7\ c_6)$
$(c_2\ c_2\ c_3\ c_{12}\ c_7\ c_4\ c_{16}\ c_8\ c_9\ c_9\ c_8\ c_{16}\ c_4\ c_7\ c_{12}\ c_3)$
$(c_2\ c_3\ c_{10}\ c_4\ c_{11}\ c_{12}\ c_{14}\ c_{14}\ c_{12}\ c_{11}\ c_4\ c_{10}\ c_3\ c_2\ c_{17}\ c_{17})$
$(c_2\ c_4\ c_{17}\ c_{13}\ c_{15}\ c_3\ c_{12}\ c_3\ c_{15}\ c_{13}\ c_{17}\ c_4\ c_2\ c_{14}\ c_5\ c_{14})$
$(c_2\ c_5\ c_7\ c_5\ c_2\ c_{11}\ c_{10}\ c_9\ c_1\ c_{15}\ c_{13}\ c_{15}\ c_1\ c_9\ c_{10}\ c_{11})$

The use of the SCH for cell search is described in detail in Section Cell search.

5.1.3 Mapping of Transport Channels to Physical Channels

Figure 27 summarises the mapping of transport channels to physical channels.

Transport Channels	**Physical Channels**
BCCH	Primary Common Control Physical Channel (Primary CCPCH)
FACH	Secondary Common Control Physical Channel (Secondary CCPCH)
PCH	
RACH	Physical Random Access Channel (PRACH)
DCH	Dedicated Physical Data Channel (DPDCH)
	Synchronisation Channel (SCH)

Figure 27. Transport-channel to physical-channel mapping.

The DCHs are coded and multiplexed as described in Section Multiplexing, channel coding and interleaving (FDD), and the resulting data stream is mapped sequentially (first-in-first-mapped) directly to the physical channel(s). The mapping of BCCH and FACH is equally straightforward, where the data stream after coding and interleaving is mapped sequentially to the Primary and Secondary CCPCH respectively. Also for the RACH, the coded and interleaved bits are sequentially mapped to the physical channel, in this case the message part of the random access burst on the PRACH. The mapping of the PCH to the Secondary CCPCH is slightly more complicated to allow for an efficient sleep mode, and is described below.

Mapping Method of PCH to Secondary Common Control Physical Channel

The mapping method is shown in Figure 28.

The PCH is divided into several groups in one super-frame, and layer 3 information is transmitted in each group.

Each group of PCH shall have information amount worth of 4 slots, and consists of a total of 6 information parts: 2 Paging Indication (PI) parts - for indicating whether there are MS-terminated calls or not, and 4 Mobile User Identifier (MUI) parts - for indicating Identity of the paged mobile user.

In each group, PI parts are transmitted ahead of MUI parts.

In all groups, 6 information parts are allocated with a certain pattern in the range of 24 slots. By shifting each pattern by 4 slots, multiple 288 groups of PCH can be allocated on one Secondary Common Control Physical Channel.

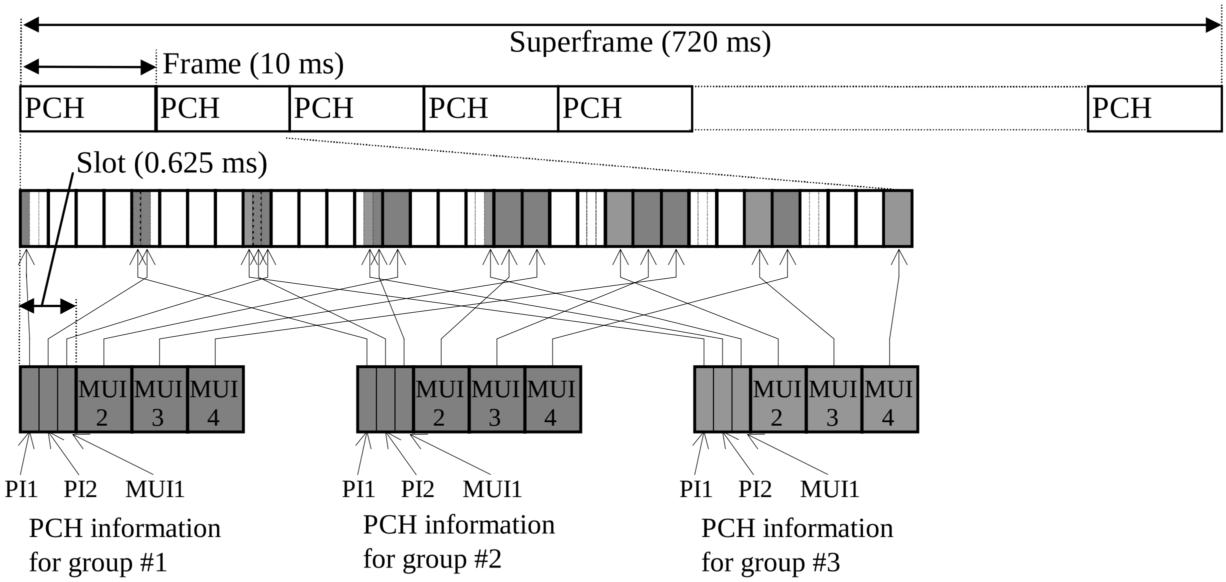


Figure 28. PCH mapping method.

5.1.4 Timing relationship between physical channels

In general, a BTS (Node B) covers *N* cells, where $N \geq 1$. Each BTS has a Reference System Frame Number (SFN), which counts from *0* to *M-1* in Radio Frame (10 ms) intervals. *M* is a multiple of the super-frame (72), and is TBD. The purpose of the Reference SFN is to make sure that the correct frames are combined at soft handover. Each cell has a Cell SFN, which is broadcast on the BCCH.

Figure 29 shows the proposed physical channel timing parameters in a soft handover situation including two BTSs, BTS1 and BTS2. The timing parameters in Figure 29 refer to frame-timing.

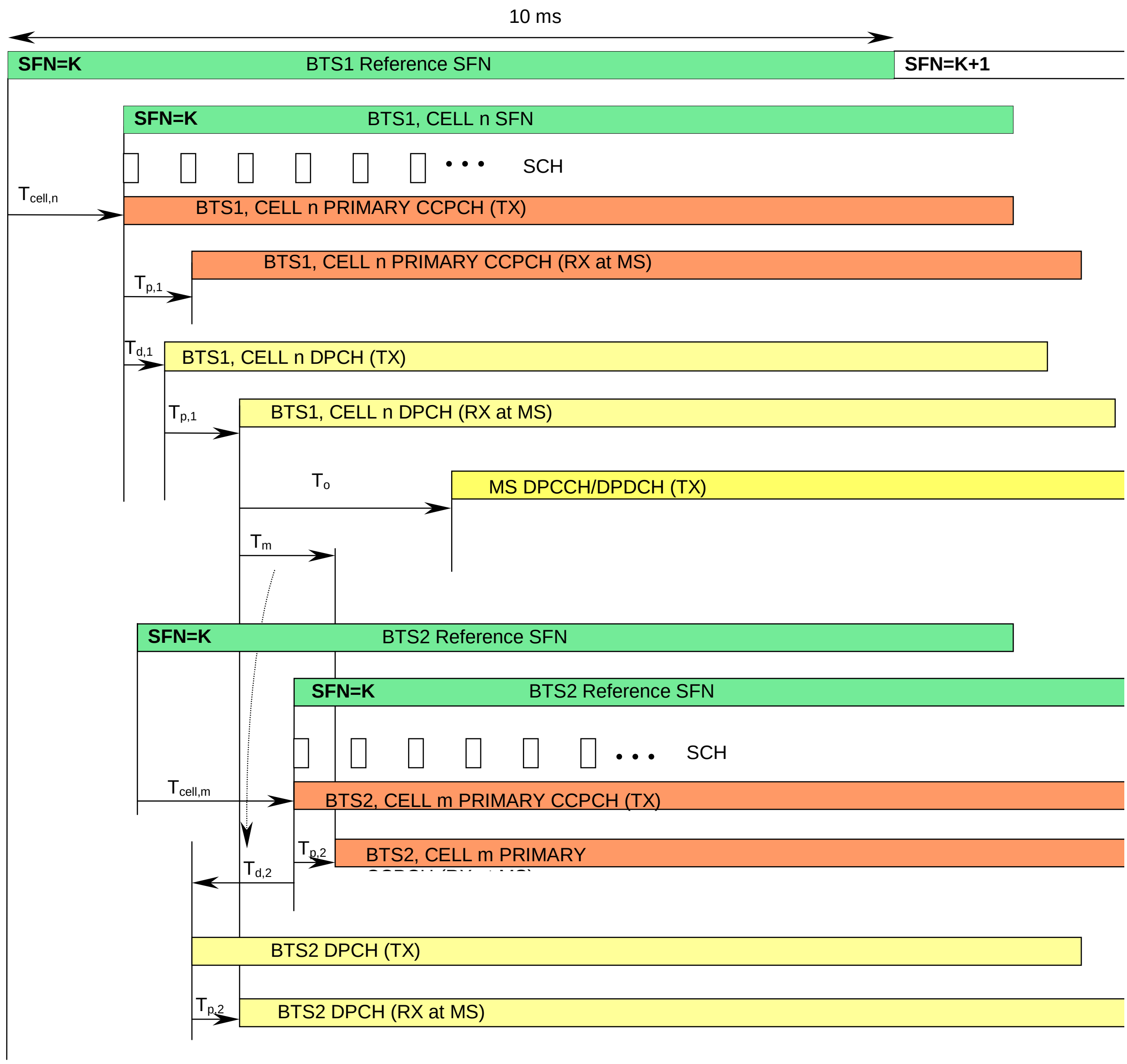


Figure 29: Physical channel timing relations.

The parameters in Figure 29 are explained below:

- **T_p**: Propagation delay between BTS and MS.
- **T_{cell}**: This timing offset is used for the frame timing of SCH, Primary CCPCH and the starting phase of all downlink scrambling codes in a cell. The main purpose is to avoid having overlapping SCHs in different cells belonging to the same BTS. The resolution, which affects the number of possible sectors in a BTS, is TBD and depends on the maximum expected time-dispersion. The range is one slot. T_{cell} is also the reference frame timing for the PRACH.
- **T_d**: This timing offset is used for the frame timing of DPCHs and Secondary CCPCHs. It can be individually set up for each DPCH and Secondary CCPCH. The T_d values for the latter may be broadcast on the BCCH, or known a-priori. The purpose of T_d is:
  - In an originating/terminating cell, to distribute discontinuous transmission periods in time, and also to distribute BTS-RNC transmission traffic in time.
  - At soft handover, to synchronise downlink DPCHs to the same MS, in order to minimise the buffering requirements at the MS.
  The resolution is 256 chips in order to maintain downlink orthogonality and the range is TBD.
- **T_o**: This constant timing offset is used to set up the transmission frame timing of an uplink DPCCH/DPDCH in the MS. The uplink DPCCH/DPDCH transmission frame timing should be set to T_o

seconds after the frame timing of the earliest received path of the downlink DPCH. T_o should be chosen to minimise the closed loop PC delay in as large cell-radii as possible. The value is TBD.
The starting phase of the uplink scrambling code is synchronised with the uplink DPCCH/DPDCH frame timing.

- **T_m**: This value is measured by the MS and reported to the RNC prior to soft handover. The RNC can then notify this value to the target BTS, which then knows how to set T_d to achieve proper reception and transmission frame timing of the dedicated physical channel.

Note that since the MS reports the value T_m as the time-difference between the received Primary CCPCH frame-timing from the target BTS and the earliest received existing DPCH path, the propagation delay to the target BTS is already compensated for in the setting of T_d at the target BTS. The DPCH signal from the target BTS will reach the MS at the same time as the earliest received existing DPCH path. The only remaining error, besides frequency-drift and MS mobility related errors, is due to a (known) rounding error at the target BTS in order to maintain downlink orthogonality.

5.2 Multiplexing, channel coding and interleaving (FDD)

5.2.1 Transport-channel coding/multiplexing

Figure 30 illustrates the overall concept of transport-channel coding and multiplexing. The following steps can be identified:

- Channel coding, including optional transport-channel multiplexing
- Static rate matching
- Inter-frame interleaving
- Transport-channel multiplexing
- Dynamic rate matching
- Intra-frame interleaving

The different steps are described in detail below

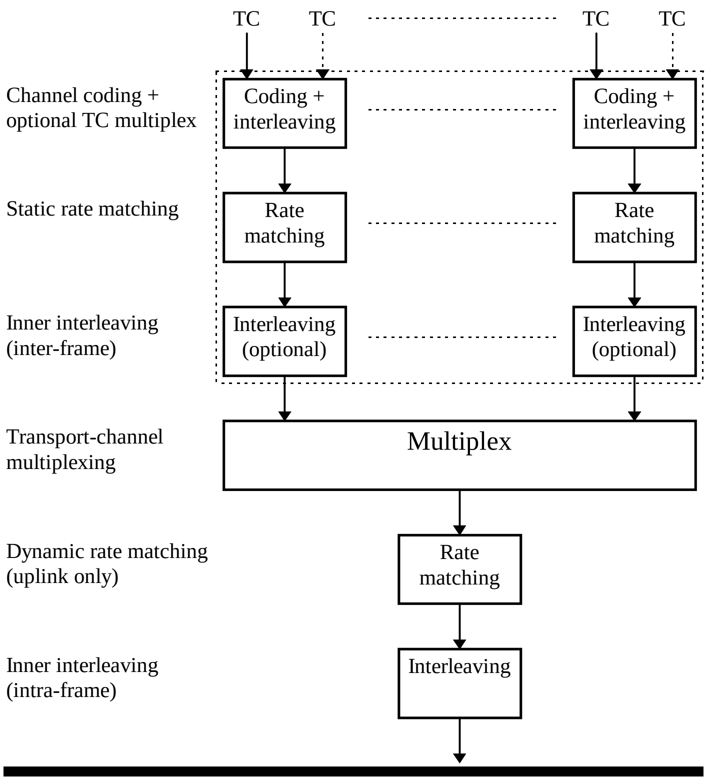


Figure 30. Coding and multiplexing of transport channels.

Note that although the coding, static rate matching, and inter-frame interleaving is done in parallel chains for different transport channels, some co-ordination in the parameter setting may be needed when adding, removing, or modifying transport channels (indicated by the dashed box in Figure 30).

The output after the inner interleaving is typically mapped to one DPDCH. Only for the very highest bit rates the output is split onto several DPDCHs, i.e. multi-code transmission.

Primarily, transport channels are coded and multiplexed as described above, i.e. into one data stream mapped on one or several physical channels. However, an alternative way of multiplexing services is to use code multiplexing, which corresponds to having several parallel multiplexing chains as in Figure 30, resulting in several data stream, each mapped to one or several physical channels.

5.2.1.1 Channel coding

Channel coding is done on a per-transport-channel basis, i.e. before transport-channel multiplexing.

The following options are available for the transport-channel specific coding, see also Figure 31:

- Convolutional coding
- Outer Reed-Solomon coding + Outer interleaving + Convolutional coding
- Turbo coding
- Service-specific coding, e.g. unequal error protection for some types of speech codecs.

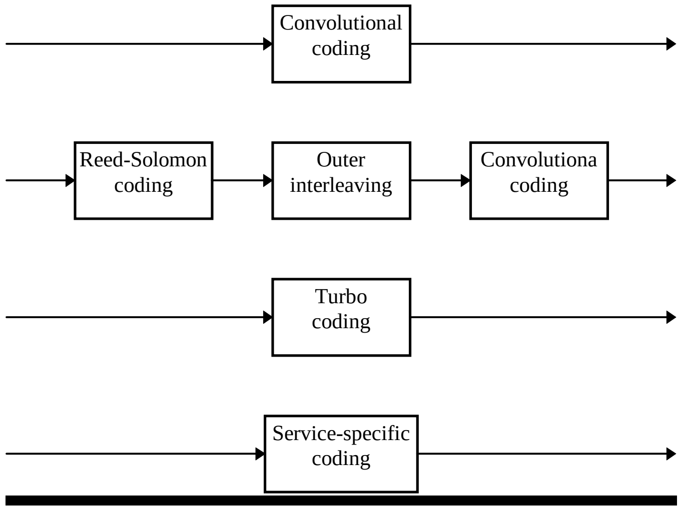


Figure 31. Channel coding in UTRA/FDD.

5.2.1.1.1 Convolutional coding

Convolutional coding is typically applied for services that require a BER in the order of 10^{-3}. Convolutional coding is also, in concatenation with RS coding + outer interleaving, applied to services that require a BER in the order of 10^{-6}, see also Section Outer Reed-Solomon coding and outer interleaving.

Table 1 lists the possible parameters for the convolutional coding.

Table 1. Generator polynomials for the convolutional codes.

Rate	Constraint length	Generator polynomial 1	Generator polynomial 2	Generator polynomial 3
1/3	9	557	663	711
½	9	561	753	N/A

Typically, rate-1/3 convolutional coding is applied to dedicated transport channels (DCHs) in normal (non-slotted) mode while rate ½ convolutional coding is applied to DCHs in slotted mode, see Section Coding for slotted mode.

5.2.1.1.2 Outer Reed-Solomon coding and outer interleaving

Reed-Solomon coding + outer interleaving, is, in concatenation with inner convolutional coding, typically applied to transport channels that require a BER in the order of 10^{-6}.

The RS-coding is of approximate rate 4/5 using the 256-ary alphabet.

The outer interleaving is symbol-based block interleaver with interleaver width equal to the block length of the RS code. The interleaver span is variable and can be 10, 20, 40, or 80 ms [longer spans e.g. 160 ms are ffs].

5.2.1.1.3 Turbo coding

The use of Turbo coding for high data rate (above 32 kbps), high quality services, is currently being investigated within ETSI. Turbo codes of rate 1/3 and ½ (for the highest data rates), have been proposed to replace the concatenation of convolutional and Reed-Solomon codes.

The block diagram for the basic Turbo Encoder is shown in Figure 32.

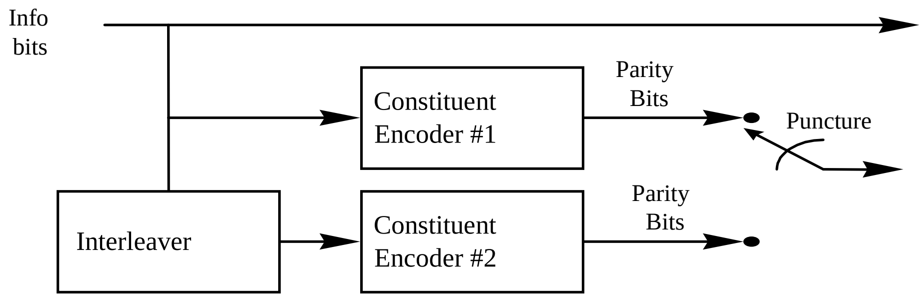


Figure 32. Block diagram of a Turbo code encoder.

5.2.1.1.4 Service specific coding

The service-specific-coding option allows for additional flexibility of the UTRA Layer 1 by allowing for additional coding schemes, in addition to the standard coding schemes listed above. One example is the use of unequal-error-protection coding schemes for certain speech-codecs.

5.2.1.2 Inner inter-frame interleaving

Inner inter-frame bit interleaving is carried out on a per-transport-channel basis on those transport-channels that can allow for and require interleaving over more than one radio frame (10 ms). The span of the inner inter-frame interleaving is variable and can be 20, 40 or 80 ms [longer spans, e.g. 160 ms, is ffs].

5.2.1.3 Rate matching

Two types of rate matching is carried out:

- Static rate matching carried out on a slow basis, typically every time a transport channel is added or removed from the connection.
- Dynamic rate matching carried out on a frame-by-frame (10 ms) basis

5.2.1.3.1 Static rate matching

Static rate matching is used for two different reasons:

- to adjust the coded transport channel bit rate to a level where minimum transmission quality requirements of each transport channel is fulfilled with the smallest differences in channel bit energy
- to adjust the coded transport channel bit rate so that the maximum total bit rate after transport channel multiplexing is matched to the channel bit rate of the uplink and downlink dedicated physical channel

The static rate matching is based on code puncturing and unequal repetition.

Note that, although static rate matching is carried out prior to transport-channel multiplexing, the rate matching must be co-ordinated between the different transport channels.

5.2.1.3.2 Dynamic rate matching

Dynamic rate matching is carried out after the multiplexing of the parallel coded transport channels and is used to match the total instantaneous rate of the multiplexed transport channels to the channel bit rate of the uplink DPDCH. Dynamic rate matching uses unequal repetition and is only applied to the uplink. On the downlink, discontinuous transmission (DTX) is used when the total instantaneous rate of the multiplexed transport channels does not match the channel bit rate.

5.2.1.3.3 Rate matching algorithm

Let's denote:

$S_N = \{N_1, N_2, \ldots, N_L\}$ = ordered set (in ascending order from left to right) of allowed number of bits per block

N_C = number of bits per matching block

$S_0 = \{d_1, d_2, \ldots, d_{N_C}\}$ = set of N_C data bits

P = maximum amount of puncturing allowed (tentatively 0.2, for further study)

The rate-matching rule is as follows:

- find N_i and N_{i+1} so that $N_i \le N_C < N_{i+1}$
- j = 0
- $z = \left\lfloor \frac{N_{i+1}}{N_C} \right\rfloor$
- $if(z > 1 \& N_C \ne N_i)$
  - ↔ repeat every bit from set S_j z times
  - ↔ $N_C = N_C z$
- $if(\frac{N_C - N_i}{N_C} < P)$
  - ↔ $x = N_C$
  - ↔ $y = N_C - N_i$
  - ↔ $S_j = \{d_1, d_2, \ldots, d_{N_C}\}$
  - ↔ do while $y > 1$
    - ∂ $z = \left\lceil \frac{x}{y} \right\rceil$, $k = \left\lfloor \frac{x}{z} \right\rfloor$
    - ∂ $x = x-k$
    - ∂ $y = y-k$
    - ∂ puncture every z'th bit from set S_j
    - ∂ form new set S_{j+1} from not punctured bits of set S_j
    - ∂ $j = j+1$
  - ↔ end do
  - ↔ if y == 1
    - ∂ puncture last bit from set S_j
- *else*
  - ↔ $x = N_C$
  - ↔ $y = N_{i+1} - N_C$
  - ↔ $S_j = \{d_1, d_2, \ldots, d_{N_C}\}$
  - ↔ do while $y > 1$
    - ∂ $z = \left\lceil \frac{x}{y} \right\rceil$, $k = \left\lfloor \frac{x}{z} \right\rfloor$
    - ∂ $x = x-k$
    - ∂ $y = y-k$
    - ∂ repeat every z'th bit from set S_j
    - ∂ form new set S_{j+1} from not repeated bits of set S_j
    - ∂ $j = j+1$
  - ↔ end do
  - ↔ if y == 1

∂ repeat first bit from set S_j

5.2.1.4 Transport-channel multiplexing

The coded transport channels are serially multiplexed within one radio frame. The output after the multiplexer (before the inner interleaving) will thus be according to Figure 33.

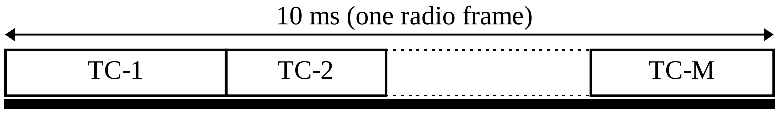


Figure 33. Transport channel multiplexing.

As an option, transport channels may be multiplexed within the channel-coding unit, typically after outer RS coding but before outer interleaving.

5.2.1.5 Inner intra-frame interleaving

Inner intra-frame interleaving over one radio frame (10 ms) is applied to the multiplexed set of transport channels.

5.2.2 Automatic Repeat Request (ARQ)

The details of the UTRA ARQ schemes are not yet specified. Therefore, the impact on Layer 1, e.g. if soft combining of retransmitted packets is to take place, is not yet fully specified.

5.2.3 Coding for layer 1 control

5.2.3.1 Transport-format-indicator coding

The TFI bits are encoded using bi-orthogonal (32, 6) block code. The coding procedure is as shown in Figure 34.

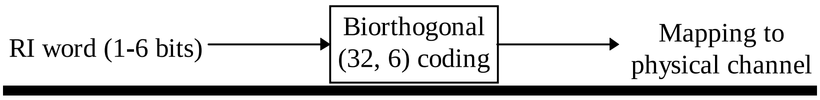


Figure 34. Channel coding of TFI bits.

The length of the TFI code word is 32 bits. Thus there are 2 bits of (encoded) TFI in every slot of the radio frame. The code words of the bi-orthogonal block code are from the level 6 of the code three of OVSF codes defined in chapter 4.3.2.1. The code words, $C_6(i)$, i = 0, ..., 31, form an orthogonal set, $S_{C_6} = \{C_6(0), C_6(1), \ldots, C_6(31)\}$, of 32 functions. By taking the binary complements of the code words of S_{C_6}, another set, $\bar{S}_{C_6} = \{\bar{C}_6(0), \bar{C}_6(1), \ldots, \bar{C}_6(31)\}$ is formed. These two sets are mutually bi-orthogonal yielding total of 64 different code words.

Mapping of the TFI bits to the code words is done as shown in the Figure 35.

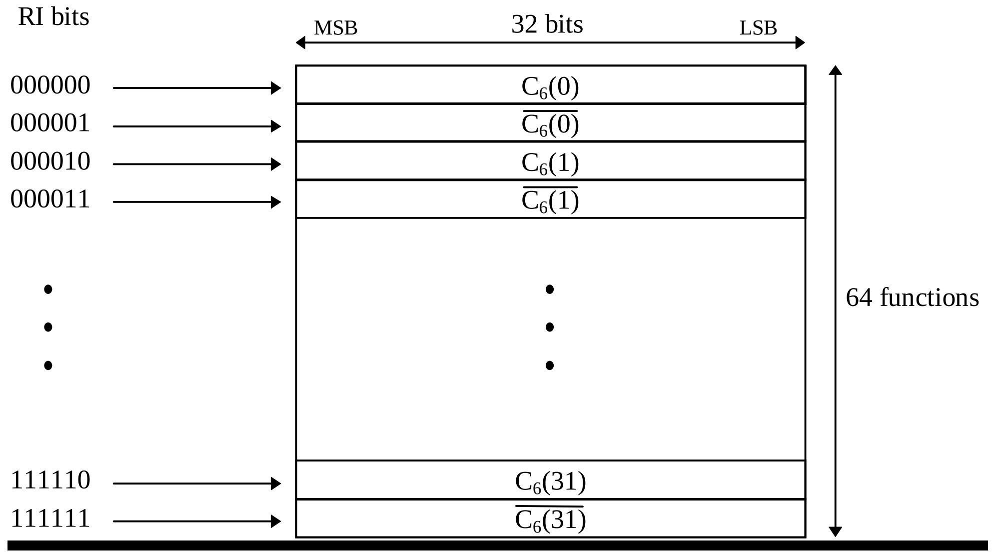


Figure 35. Mapping of TFI bits to bi-orthogonal code words.

Bits of the TFI code words are time multiplexed to DPCCH as shown in the Figure 36. Within a slot the more significant bit is transmitted before the less significant bit.

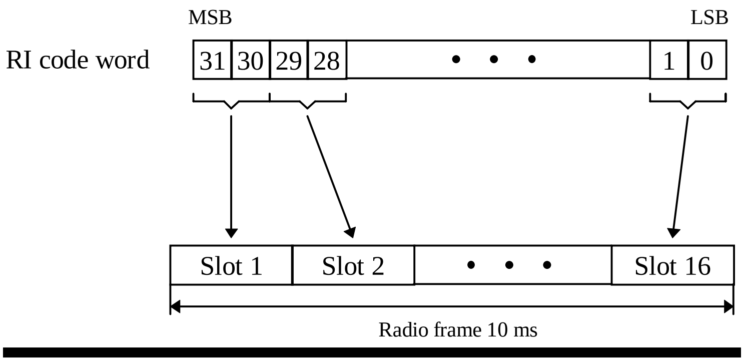


Figure 36. Time multiplexing of the bits of a TFI code word to radio frame.

5.2.4 Coding for slotted mode

With slotted downlink transmission, it is possible for a single-receiver mobile station to carry out measurements on other frequencies without affecting the ordinary data flow. The principle of slotted downlink transmission is illustrated in Figure 37. When in slotted mode, the information normally transmitted during 10 ms frame is compressed in time. This can be achieved by:

- code puncturing, for lower compression factors,
- changing the FEC rate, for higher compression factors.

Note that the idle slot is created without any loss of data as the number of information bits per frame is kept constant, while the processing gain is reduced by increasing the coding rate. As illustrated in Figure 37, the instantaneous transmit power is increased in the slotted frame in order to keep the quality (BER, FER, etc.) unaffected by the reduced processing gain.

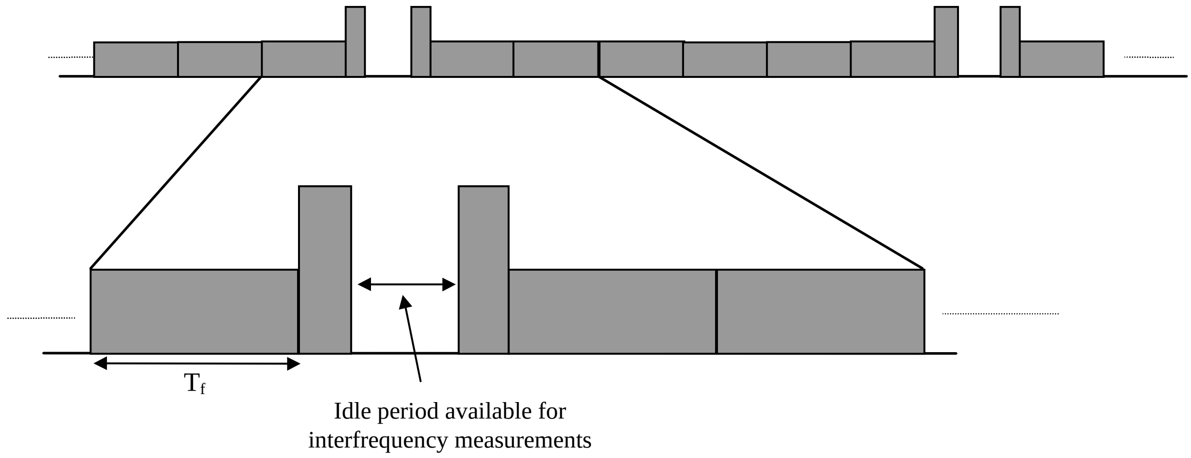


Figure 37. Downlink slotted transmission.

Although Figure 37 shows slotted transmission with a mid-frame idle-period, there are in general three types of possible slotted transmission mechanisms, as illustrated in Figure 38.

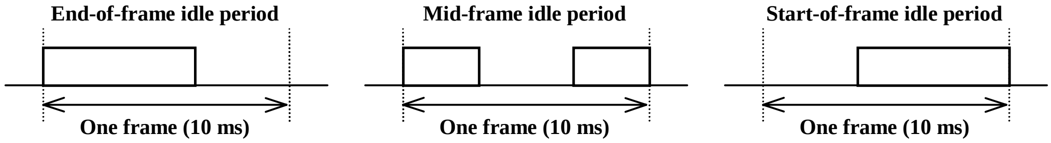


Figure 38. Possible idle period positions.

The default position is the mid-frame idle period. The start-of-frame and end-of-frame idle are supported in order to be able to create an even longer double-frame idle period, as illustrated in Figure 39.

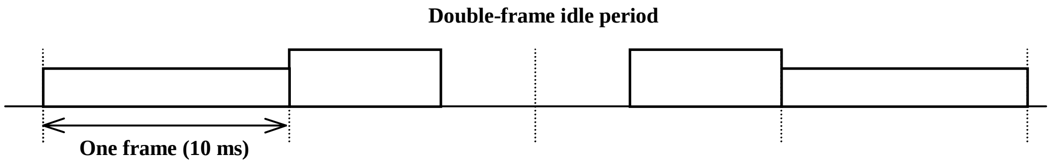


Figure 39. Double-frame idle period.

When in slotted mode, slotted frames can occur periodically, as illustrated in Figure 37, or requested on demand. The rate of and type of slotted frames is variable and depends on the environment and the measurement requirements.

For UTRA-to-GSM inter-frequency handover considerations, see Section UTRA - GSM handover.

For services that allows for a larger delay, e.g. data services with interleaving over several frames, multiple frames can be compressed together in order to create a short measurement slot. As an example, for a 2 Mbps service, with interleaving of 5 frames (50 ms), a 5 ms idle slot can be created by puncturing only 10% of 5 frames, as illustrated in Figure 40.

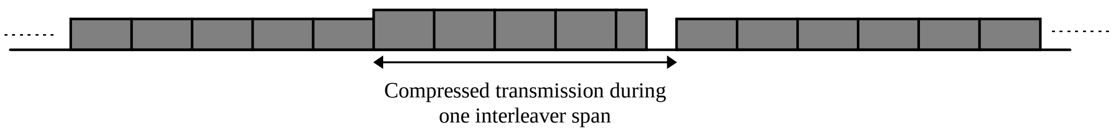


Figure 40. Multi-frame compressed mode for long-delay services.

5.3 Spreading and modulation (FDD)

5.3.1 Uplink spreading and modulation

5.3.1.1 Spreading

Uplink Dedicated Physical Channels (uplink DPDCH/DPCCH)

Figure 41 illustrates the spreading and modulation for the case of a single uplink DPDCH. Data modulation is dual-channel QPSK, where the uplink DPDCH and DPCCH are mapped to the I and Q branch respectively. The I and Q branch are then spread to the chip rate with two different channelisation codes c_D/c_C and subsequently complex scrambled by a mobile-station specific complex scrambling code c_{scramb}

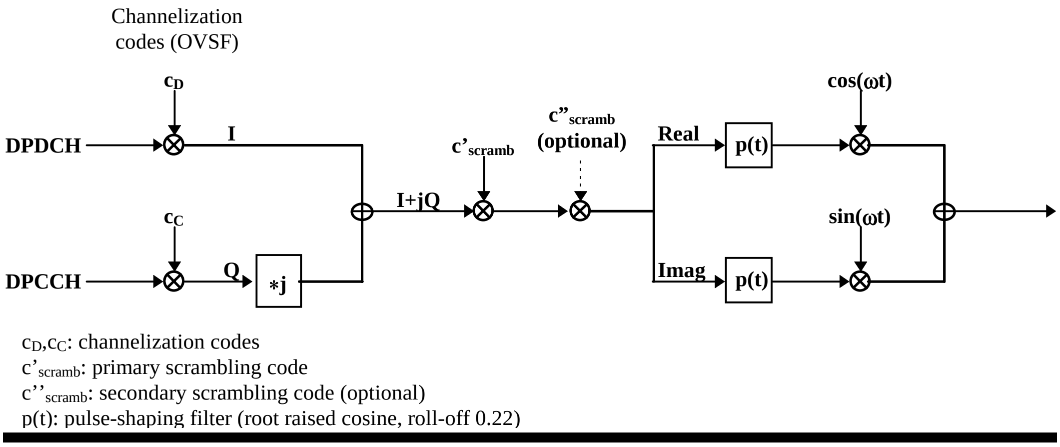


Figure 41. Spreading/modulation for uplink DPDCH/DPCCH.

For multi-code transmission, each additional uplink DPDCH may be transmitted on either the I or the Q branch. For each branch, each additional uplink DPDCH should be assigned its own channelisation code. Uplink DPDCHs on different branches may share a common channelisation code.

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The spreading and modulation of the message part of the Random-Access burst is basically the same as for the uplink dedicated physical channels, see Figure 41, where the uplink DPDCH and uplink DPCCH are replaced by the data part and the control part respectively. The scrambling code for the message part is chosen based on the base-station-specific preamble code.

5.3.1.2 Code generation and allocation

5.3.1.2.1 Channelisation codes

The channelisation codes of Figure 41 are Orthogonal Variable Spreading Factor (OVSF) codes that preserve the orthogonality between a user's different physical channels. The OVSF codes can be defined using the code tree of Figure 42.

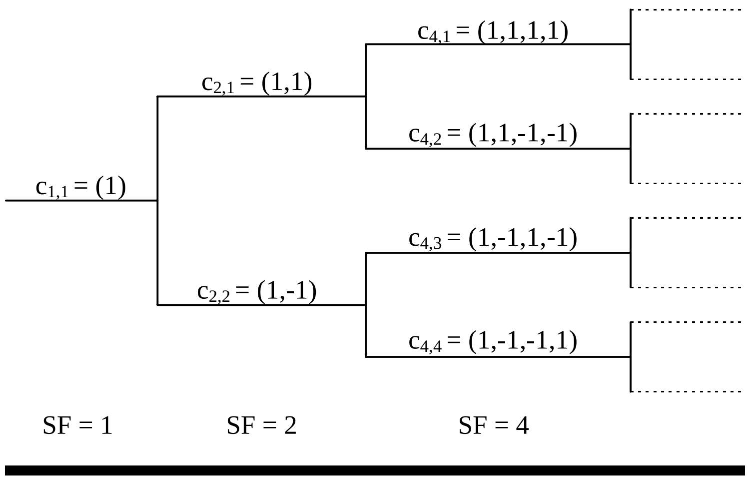


Figure 42. Code-tree for generation of Orthogonal Variable Spreading Factor (OVSF) codes.

Each level in the code tree defines channelisation codes of length SF, corresponding to a spreading factor of SF in Figure 42. All codes within the code tree cannot be used simultaneously by one mobile station. A mobile station can use a code if and only if the same mobile station uses no other code on the path from the specific code to the root of the tree or in the sub-tree below the specific code. This means that the number of available channelisation codes is not fixed but depends on the rate and spreading factor of each physical channel.

Each connection is allocated at least one uplink channelisation code, to be used for the uplink DPCCH. In most cases, at least one additional uplink channelisation code is allocated for an uplink DPDCH. Further uplink channelisation codes may be allocated if more than one uplink DPDCH is required. All channelisation codes used for the DPDCHs must be orthogonal to the code used for the DPCCH.

As different mobile stations use different uplink scrambling codes, the uplink channelisation codes may be allocated with no co-ordination between different connections. The uplink channelisation codes are therefore always allocated in a pre-defined order. The mobile-station and network only need to agree on the number and length (spreading factor) of the uplink channelisation codes. The exact codes to be used are then implicitly given. In order to obtain the benefits associated with the scrambling code design at the mobile station transmitter (see Section Scrambling codes

Either short or long scrambling codes should be used on uplink. The short scrambling code is typically used in cells where the base station is equipped with an advanced receiver, such as a multi-user detector or interference canceller. With the short scrambling code the cross-correlation properties between different physical channels and users does not vary in time in the same way as when a long code is used. This means that the cross-correlation matrices used in the advanced receiver do not have to be updated as often as for the long scrambling code case, thereby reducing the complexity of the receiver implementation. In cells where there is no gain in implementation complexity using the short scrambling code, the long code is used instead due to its better interference averaging properties.

Both short and long scrambling codes are formed as follows:

$$Cscramb = c_1(w_0 + jc_2'w_1)$$

where w_0 and w_1 are chip rate sequences defined as repetitions of:

w_0 = {1 1symbol 125 \f "Symbol" \s 10

w_1 = {1 -1symbol 125 \f "Symbol" \s 10

and where c_1 is a real chip rate code, and c_2' is a decimated version of the real chip rate code c_2. The preferred decimation factor is 2, however other decimation factors should be possible in future evolutions of the RTT if proved desirable.

With a decimation factor N=2, c_2' is given as:

$$c_2'(2k-1) = c_2'(2k) = c_2(2k-1), \quad k=1,2,3\ldots.$$

These scrambling codes are designed such that at N-1 out of N consecutive chip times they produce +/-90° rotations of the IQ multiplexed data and control channels. At the remaining 1 out of N chip times, they produce 0, +/-90 or 180° rotations . This limits the transitions of the complex baseband signal that is inputted to the root raised cosine pulse shaping filter. This in turn reduces the peak to average ratio of the signal at the filter output, allowing a more efficient power amplifier implementation. To guarantee these desirable properties, restrictions on the choice of uplink OVSF codes are also required.

The constituent codes c_1 and c_2 are formed differently for the short and long scrambling codes as described in Sections Short scrambling code and Long scrambling code.

5.3.1.2.2.1 Short scrambling code

The short scrambling codes are formed as described in Section **Error! Not a valid bookmark self-reference.**, where c_1 and c_2 are two different codes from the extended Very Large Kasami set of length 256.

The network decides the uplink short scrambling code. The mobile station is informed about what short scrambling code to use in the downlink Access Grant message that is the base-station response to an uplink Random Access Request.

The short scrambling code may, in rare cases, be changed during the duration of a connection.

5.3.1.2.2.2 Long scrambling code

The long uplink scrambling code is typically used in cells without multi-user detection in the base station. The mobile station is informed if a long scrambling code should be used in the Access Grant Message following a Random-Access request and in the handover message.

What long scrambling code to use is directly given by the short scrambling code. No explicit allocation of the long scrambling code is thus needed.

The long scrambling codes are formed as described in Section **Error! Not a valid bookmark self-reference.**, where c_1 and c_2 are constructed as the position wise modulo 2 sum of 40960 chip segments of two binary *m*-sequences generated by means of two generator polynomials of degree 41. Let *x*, and *y* be the two *m*-sequences respectively. The *x* sequence is constructed using the primitive (over GF(2)) polynomial $1+X^3+X^{41}$. The *y* sequence is constructed using the polynomial $1+X^{20}+X^{41}$. The resulting sequences thus constitute segments of a set of Gold sequences.

The scrambling code for the quadrature component is a 1024-chip shifted version of the in-phase scrambling code.

The uplink scrambling code word has a period of one radio frame of 10 ms.

Let $n_{40} \ldots n_0$ be the binary representation of the scrambling code number *n* (decimal) with n_0 being the least significant bit. The *x* sequence depends on the chosen scrambling code number *n* and is denoted x_n, in the sequel. Furthermore, let $x_n(i)$ and $y(i)$ denote the *i:*th symbol of the sequence x_n and *y*, respectively

The *m*-sequences x_n and *y* are constructed as:

Initial conditions:

$x_n(0)=n_0$, $x_n(1)= n_1$, $\ldots =x_n(39)= n_{39}$, $x_n(40)= n_{40}$

$y(0)=y(1)= \ldots =y(39)= y(40)=1$

Recursive definition of subsequent symbols:

$x_n(i+41) =x_n(i+3) + x_n(i)$ modulo 2, $i=0,\ldots, 2^{41}-43$,

$y(i+41) = y(i+20)+y(i)$ modulo 2, $i=0,\ldots, 2^{41}-43$.

The definition of the *n*:th scrambling code word for the in phase and quadrature components follows as (the left most index correspond to the chip scrambled first in each radio frame):

$C^{I}_{long,n} = < x_n(0)+y(0), x_n(1)+y(1), \ldots, x_n(40959)+y(40959) >$,

$C^{Q}_{long,n} = < x_n(1024)+y(1024), x_n(1025)+y(1025), \ldots, x_n(41983) + y(41983) >$,

again all sums being modulo 2 additions.

Now, the complex long scrambling code $C_{long,n}$ is defined by:

$$C_{long,n} = (C^{I}_{long,n} + jC^{Q}_{long,n}) =$$

$$= < ((x_n(0)+y(0)) + j(x_n(1024)+y(1024))), \ldots,$$

$$((x_n(40959)+y(40959)) + j(x_n(41983) + y(41983))) >$$

The code generator must be able to generate the sequence shifted arbitrarily from the initial state.

5.3.1.2.3 Random access codes

5.3.1.2.3.1 Preamble spreading code

The spreading code for the preamble part is cell specific and is broadcast by the base station. More than one preamble code can be used in a base station if the traffic load is high. The preamble codes must be code planned, since two neighbouring cells should not use the same preamble code.

The code used is a real-valued 256 chip Orthogonal Gold code. All 256 codes are used in the system. The preamble codes are generated in the same way as the codes used for the downlink synchronisation channel and are defined in Section Synchronisation codes.

5.3.1.2.3.2 Preamble signature

The preamble part carries one of 16 different orthogonal complex signatures of length 16, $<P_0, P_1, \ldots, P_{15}>$. The signatures are based on a set of Orthogonal Gold codes of length 16 and are specified in Table 2. The base station broadcasts which signatures that are allowed in a cell.

Table 2. Preamble signatures. A = 1+j.

	Preamble symbols															
Signature	P_0	P_A	P_2	P_3	P_4	P_5	P_6	P_7	P_8	P_9	P_{10}	P_{11}	P_{12}	P_{13}	P_{14}	P_{15}
1	A	A	A	-A	-A	-A	A	-A	-A	A	A	-A	A	-A	A	A
2	-A	A	-A	-A	A	A	A	-A	A	A	A	-A	-A	A	-A	A
3	A	-A	A	A	A	-A	A	A	-A	A	A	A	-A	A	-A	A
4	-A	A	-A	A	-A	-A	-A	-A	-A	A	-A	A	-A	A	A	A
5	A	-A	-A	-A	-A	A	A	-A	-A	-A	-A	A	-A	-A	-A	A
6	-A	-A	A	-A	A	-A	A	-A	A	-A	-A	A	A	A	A	A
7	-A	A	A	A	-A	-A	A	A	A	-A	-A	-A	-A	-A	-A	A
8	A	A	-A	-A	-A	-A	-A	A	A	-A	A	A	A	A	-A	A
9	A	-A	A	-A	-A	A	-A	A	A	A	-A	-A	-A	A	A	A
10	-A	A	A	-A	A	A	-A	A	-A	-A	A	A	-A	-A	A	A
11	A	A	A	A	A	A	-A	-A	A	A	-A	A	A	-A	-A	A
12	A	A	-A	A	A	A	A	A	-A	-A	-A	-A	A	A	A	A
13	A	-A	-A	A	A	-A	-A	-A	A	-A	A	-A	-A	-A	A	A
14	-A	-A	-A	A	-A	A	A	A	A	A	A	A	A	-A	A	A
15	-A	-A	-A	-A	A	-A	-A	A	-A	A	-A	-A	A	-A	-A	A

16	-A	-A	A	A	-A	A	-A	-A	-A	-A	A	-A	A	A	-A	A

5.3.1.2.3.3 Channelisation codes for the message part

The signature in the preamble specifies one of the 16 nodes in the code-tree that corresponds to channelisation codes of length 16, as shown in Figure 43. The sub-tree below the specified node is used for spreading of the message part. The control (Q-branch) is spread with the channelisation code of spreading factor 256 in the lowest branch of the sub-tree. The data part (I-branch) can use any of the channelisation codes from spreading factor 32 to 256 in the upper-most branch of the sub-tree. However, the system may restrict the set of codes (spreading factors) actually allowed in the cell, through the use of a BCCH message.

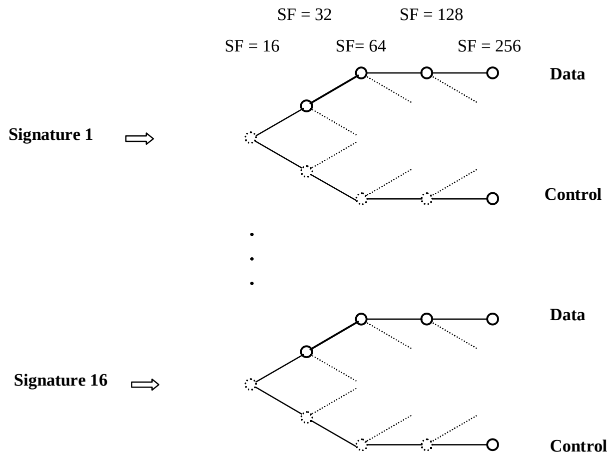


Figure 43. Channelisation codes for the random access message part.

Since the control part is always spread with a known channelisation code of length 256, the base station can detect it. The rate information field of the control part informs the base station about the spreading factor used on the data part. With knowledge of the sub-tree (obtained from the preamble signature) and the spreading factor (obtained from the rate information), the base station knows which channelisation code is used for the data part.

This structure allows for simultaneous detection of multiple random access messages arriving in the same access slot, as long as different signatures are used.

5.3.1.2.3.4 Scrambling code for the message part

In addition to spreading, the message part is also subject to scrambling with a 10 ms complex code. The scrambling code is cell-specific and has a one-to-one correspondence to the spreading code used for the preamble part. Note that although the scrambling code is the same for every access slot, there is no scrambling-code collision problem between different access slots due to the 1.25 ms time shifts between the access slots.

The scrambling codes used are from the same set of codes as is used for the other dedicated uplink channels. The first 256 codes are used for the random access channel. The generation of these codes is explained in Section Scrambling code.

5.3.1.3 Modulation

5.3.1.3.1 Modulating chip rate

The modulating chip rate is 4.096 Mcps. This basic chip rate can be extended to 8.192 or 16.384 Mcps.

5.3.1.3.2 Pulse shaping

The pulse-shaping filters are root-raised cosine (RRC) with roll-off α=0.22 in the frequency domain.

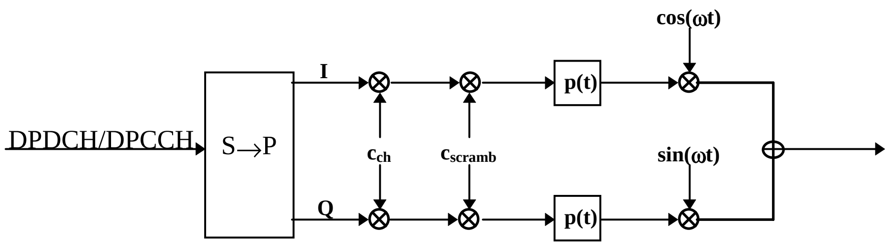
5.3.1.3.3 Modulation

QPSK modulation is used. Note that phase restrictions are introduced by the scrambling code design.

5.3.2 Downlink spreading and modulation

5.3.2.1 Spreading

Figure 44 illustrates the spreading and modulation for the downlink DPCH. Data modulation is QPSK where each pair of two bits are serial-to-parallel converted and mapped to the I and Q branch respectively. The I and Q branch are then spread to the chip rate with the same channelisation code c_{ch} (real spreading) and subsequently scrambled by the same cell specific scrambling code c_{scramb} (real scrambling).



c_{ch}: channelization code
c_{scramb}: scrambling code
p(t): pulse-shaping filter (root raised cosine, roll-off 0.22)

Figure 44. Spreading/modulation for downlink dedicated physical channels.

The different physical channels use different channelisation codes, while the scrambling code is the same for all physical channels in one cell.

The multiplexing of the SCH with the other downlink physical channels (DPCH and CCPCH) is illustrated in Figure 45. The figure illustrates that the SCH is only transmitted intermittently (one codeword per slot) and also that the SCH is multiplexed *after* long code scrambling of the DPCH and CCPCH. Consequently, the SCH is *non-orthogonal* to the other downlink physical channels.

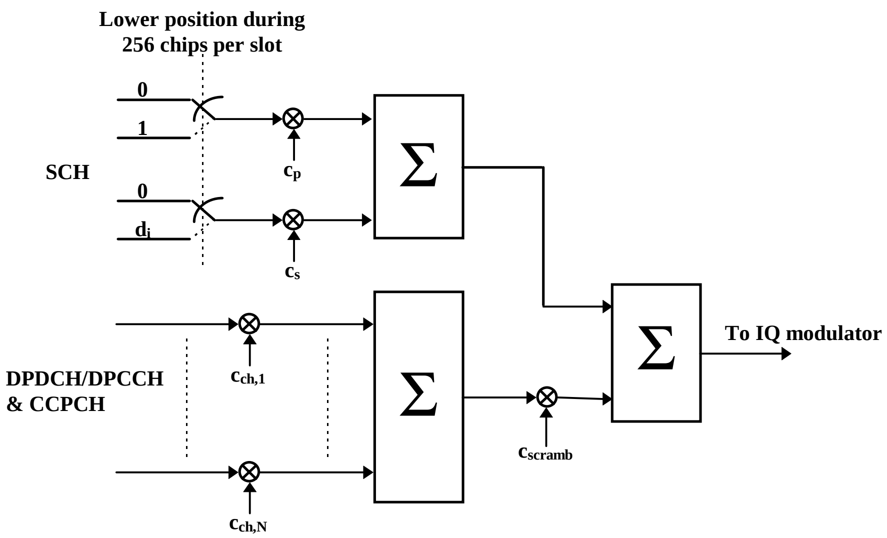


Figure 45. Multiplexing of SCH.

5.3.2.2 Code generation and allocation

5.3.2.2.1 Channelisation codes

The channelisation codes of Figure 44 are the same codes used in the uplink, namely Orthogonal Variable Spreading Factor (OVSF) codes that preserve the orthogonality between downlink channels of different rates and spreading factors. The same restriction on code allocation applies as for the uplink, but for a cell and not a mobile station as in the uplink. Hence, in the downlink a code can be used in a cell if and only if no other code on the path from the specific code to the root of the tree or in the sub-tree below the specific code is used in the same cell.

The channelisation code for the BCCH is a predefined code, which is the same for all cells within the system.

The channelisation code(s) used for the Secondary Common Control Physical Channel is broadcast on the BCCH.

The channelisation codes for the downlink dedicated physical channels are decided by the network. The mobile station is informed about what downlink channelisation codes to receive in the downlink Access Grant message that is the base-station response to an uplink Random Access request. The set of channelisation codes may be changed during the duration of a connection, typically as a result of a change of service or an inter-cell handover. A change of downlink channelisation codes is negotiated over a DCH.

5.3.2.2.2 Scrambling code

The total number of available scrambling codes is 512, divided into 32 code groups with 16 codes in each group. The grouping of the downlink codes is done in order to facilitate a fast cell search, see Section Cell search. The downlink scrambling code is assigned to the cell (sector) at the initial deployment. The mobile station learns about the downlink scrambling code during the cell search process, see Section Cell search.

The scrambling code sequences are constructed as the position wise modulo 2 sum of 40960 chip segments of two binary *m*-sequences generated by means of two generator polynomials of degree 18. Let x, and y be the two sequences respectively. The x sequence is constructed using the primitive (over GF(2)) polynomial $1+X^7+X^{18}$. The y sequence is constructed using the polynomial $1+X^5+X^7+ X^{10}+X^{18}$. The resulting sequences thus constitute segments of a set of Gold sequences.

The scrambling codes are repeated for every 10 ms radio frame.

Let $n_{17} \dots n_0$ be the binary representation of the scrambling code number n (decimal) with n_0 being the least significant bit. The x sequence depends on the chosen scrambling code number n and is denoted x_n, in the sequel. Furthermore, let $x_n(i)$ and $y(i)$ denote the i:th symbol of the sequence x_n and y, respectively

The *m*-sequences x_n and y are constructed as:

Initial conditions:

$x_n(0)=n_0$, $x_n(1)= n_1$, $\dots =x_n(16)= n_{16}$, $x_n(17)= n_{17}$

$y(0)=y(1)= \dots =y(16)= y(17)=1$

Recursive definition of subsequent symbols:

$x_n(i+18) =x_n(i+7) + x_n(i)$ *modulo 2,* $i=0,\dots,2^{18}-20$,

$y(i+18) = y(i+10)+y(i+7)+y(i+5)+y(i)$ *modulo 2,* $i=0,\dots, 2^{18}-20$.

All sums of symbols are taken modulo 2.

The definition of the n:th scrambling code word follows as (the left most index correspond to the chip scrambled first in each radio frame):

$C_{scramb,n} = < x_n(0)+y(0), x_n(1)+y(1), \dots,x_n(40959)+y(40959) >$,

again all symbol sums being modulo 2 additions.

The index n runs from 0 to 511 giving 512 distinct 40960 chip segments of a corresponding Gold code sequence. The leftmost chip in $C_{scramb,n}$ corresponds to the first chip in a 10 ms radio frame and the rightmost to the last.

The sign of the I- and Q-branch component is changed if and only if the corresponding chip in $C_{scramb,n}$ equals ‘1’.

The code generator must be able to generate the sequence shifted arbitrarily from the initial state.

5.3.2.2.3 Synchronisation codes

The Primary and Secondary code words, C_p and $\{C_1,\dots,C_{17}\}$ respectively, consist of pair wise mutually orthogonal Gold codes of length 256. The Primary SCH is furthermore chosen to have good aperiodic auto correlation properties. The code sequences are constructed with the help of two binary *m*-sequences of length 255, x, and y, respectively. The x sequence is constructed using the polynomial $1+X^2+X^3+X^4+X^8$. The y sequence is constructed using the polynomial $1+X^3+X^5+X^6+X^8$.

Before we define the Primary and Secondary code words, we define the set of orthogonal Gold codes.

Let $n_7 \ldots n_0$ be the binary representation of the scrambling code number *n* (decimal) with n_0 being the least significant bit. The *x* sequence depends on the chosen code number *n* and is denoted x_n in the sequel. Furthermore, let $x_n(i)$ and $y(i)$ denote the *i:*th symbol of the sequence x_n and *y*, respectively

The *m*-sequences x_n and *y* are constructed as:

Initial conditions:

$x_n(0)=n_0$, $x_n(1)= n_1$, … $=x_n(6)= n_6$, $x_n(7)= n_7$

$y(0)=y(1)= \ldots =y(6)= y(7)=1$

Recursive definition of subsequent symbols:

$x_n(i+8) =x_n(i+4) + x_n(i+3) + x_n(i+2) + x_n(i)$ *modulo 2, i=0,…,* 246,

$y(i+8) = y(i+6)+ y(i+5)+ y(i+3)+y(i)$ *modulo 2, i=0,…,* 246.

The definition of the *n*:th SCH code word follows (the left most index correspond to the chip transmitted first in each slot):

$C_{SCH,n} = < 0, x_n(0)+y(0), x_n(1)+y(1), \ldots,x_n(254)+y(254) >$,

All sums of symbols are taken modulo 2.

Note that the code words always start with a constant ‘0’. symbol.

Before modulation and transmission these binary code words are converted to real valued sequences by the transformation ‘0’ -> ‘+1’, ‘1’ -> ‘-1’.

The Primary and Secondary code words are defined in terms of $C_{SCH,n}$ and the definition of C_p and $\{C_1,\ldots,C_{17}\}$ now follows as:

$C_p= C_{SCH, 0}$

and

$C_i = C_{SCH, i}$, i=1,…,17

5.3.2.3 Modulation

5.3.2.3.1 Modulating chip rate

The modulating chip rate is 4.096 Mcps. This basic chip rate can be extended to 8.192 or 16.384 Mcps.

5.3.2.3.2 Pulse shaping

The pulse-shaping filters are root raised cosine (RRC) with roll-off α=0.22 in the frequency domain.

5.3.2.3.3 Modulation

QPSK modulation is used.

5.4 Radio transmission and reception (FDD)

5.4.1 General

The information presented in this section is based on a chip rate of 4.096 Mcps. Appropriate adjustments should be made for higher chip rate options.

5.4.2 Frequency bands and channel arrangement

5.4.2.1 Proposed frequency bands for operation

UTRA/FDD is designed to operate in the following paired band:

Table 3. Proposed frequency band for UTRA/FDD

1920 – 1980 MHz Mobile station transmit Base station receive	2110 – 2170 MHz Mobile station receive Base station transmit

Deployment in other frequency bands is not precluded.

5.4.2.2 Carrier spacing

The nominal channel spacing is 5 MHz, but this can be adjusted to optimise performance in particular deployment scenarios. The channel raster is 200 kHz, which means that the carrier frequency must be a multiple of 200 kHz.

5.4.2.3 TX – RX frequency separation

The minimum transmit to receive separation is 130 MHz when operating in the paired band defined in Table 3. If used in other frequency bands like the American PCS band the minimum separation would be 80 MHz.

5.4.2.4 Variable duplex distance

UTRA/FDD should support a variable duplex distance, i.e. $D_{duplexer} = F_{down} – F_{up}$ is not necessary a constant but is, in general, allowed to vary within certain limits. The specific limits for the duplex distance applicable for different frequency bands and terminal classes are yet to be determined.

5.4.3 Service classes

5.4.3.1 Terminal service classes

A number of different service classes will be used to define the data rate and code allocation for a UTRA/FDD terminal. Possible types of service class profiles are 144 kbps, 384 kbps and 2048 kbps.

5.4.4 Transmitter characteristics

The output power is given in terms of power level at the antenna connector of the equipment. For equipment with integral antenna only, a reference antenna with a gain of 0 dBi is assumed.

5.4.4.1 Mobile station output power

The mobile station output power profile would be used to define a range of terminal output powers for use in different system scenarios. The power class would be based on the mobile station's peak power for example 30 dBm. For mobile station using directive antennas for transmission, a class dependent limit will be placed on the maximum EIRP (Equivalent Isotropic Radiated Power).

5.4.4.2 Base station output power

The base station output power profile would be used to cater for different system scenarios. The power class would be based on the peak power specified for the base stations.

5.4.4.3 Output power dynamics

The transmitter uses fast closed-loop Carrier/Interference based power control and slow quality-based power control on both the uplink and downlink.

Table 4. Output power dynamics for UL and DL

	Uplink (UL)	Downlink (DL)
Power control steps	Variable 0.25-1.5 dB	Variable 0.25-1.5 dB
Minimum transmit power	-50 dBm	[ ] dBm
Power control cycles per second	1.6 kHz	1.6 kHz
Power control dynamic	80 dB	30 dB

5.4.4.4 Output RF spectrum emissions

5.4.4.4.1 Out of band emissions

The assumed spectrum mask has been derived from simulations on a real wide band amplifier as shown in

Figure 46 below. These emission levels will be dependent on the power class and code allocation of the mobile and base station.

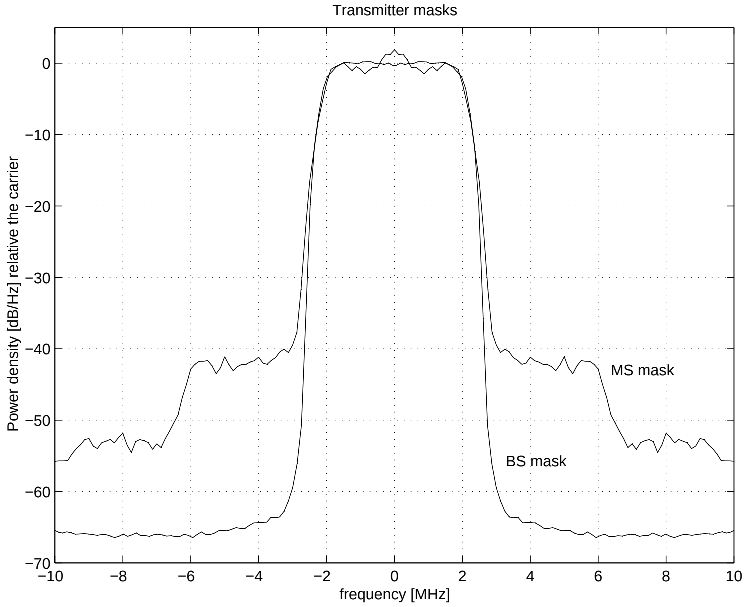


Figure 46. Assumed spectrum masks.

5.4.4.4.2 Spurious emissions

The limits for spurious emissions at frequencies greater than ± 250% of the necessary bandwidth would be based on the applicable tables from ITU-R Recommendation SM.329. Further guidance would be taken from the ERC recommendation that is currently under progress.

5.4.4.5 Adjacent channel protection (ACP)

Adjacent channel protection (ACP) is the ratio of the transmitted power and the power measured after a receiver filter in the adjacent channel.

The ACP envisaged for 5 MHz channel spacing is in the order of 35 dB to 40 dB. The possibility is being considered of dynamically relaxing the ACP requirements for mobile stations under conditions when this would not lead to significant interference (with respect to other systems or UMTS operators). This would be carried out under network control, primarily to facilitate reduction in MS power consumption.

5.4.4.6 Occupied bandwidth

The channel bandwidth is less than 5 MHz based on a chip rate of 4.096 Mcps.

5.4.4.7 Frequency stability

The frequency stability for the mobile and base station is indicated in Table 5.

Table 5. Mobile and base station frequency stability.

Mobile station	Base station
3 PPM (unlocked), 0.1 PPM (locked)	0.05 PPM

5.4.5 Receiver characteristics

A Rake receiver or any other suitable receiver structure using coherent reception in both channel impulse response estimation, and code tracking procedures is assumed.

5.4.5.1 Diversity characteristics

Three forms of diversity are available in UTRA / FDD:

Table 6. Diversity characteristics for UTRA/FDD.

Time diversity	Channel coding and interleaving in both uplink and down link.
Multi-path diversity	Rake receiver or other suitable receiver structure with maximum combining. Additional processing elements can increase the delay-spread performance due to increased capture of signal energy.
Space diversity	Antenna diversity with maximum ratio combing in the base station and optionally in the mobile stations. Possibility for downlink transmit diversity in the base station.

5.4.5.2 Reference sensitivity level

The reference sensitivity for the following services; 8 kbps, 144 kbps, 384 kbps and 2048 kbps are specified in the link budget template for a number of test environments and multi-path channel classes.

5.4.5.3 BER noise floor level

The BER noise floor level for voice services is significantly less than 10^{-3} BER. The BER noise floor level for data services is significantly less than 10^{-6} BER.

5.4.5.4 Maximum tolerable delay spread

To maintain the voice and data service quality requirements the UTRA/FDD concept allows for a time dispersion spread suitable for the various propagation models specified in UMTS 30.03 (which contains the models defined in ITU-R recommendation M.1225).

5.4.5.5 Maximum tolerable Doppler spread

The maximum tolerable Doppler spread is 1000 Hz, which at a 2 GHz carrier frequency corresponds to a maximum velocity of about 500 km/hr. Parameters determining system performance are not necessarily optimised for this value of Doppler spread.

5.5 Physical layer procedures (FDD)

5.5.1 Power control

5.5.1.1 Uplink power control

5.5.1.1.1 Closed loop power control

The uplink closed loop power control adjusts the mobile station transmit power in order to keep the received uplink Signal-to-Interference Ratio (SIR) at a given SIR target.

The base station should estimate the received uplink DPCCH power after RAKE combining of the connection to be power controlled. Simultaneously, the base station should estimate the total uplink received interference in the current frequency band and generate a SIR estimate SIR_{est}. The base station then generates TPC commands according to the following rule:

$SIR_{est} > SIR_{target,UL} \rightarrow$ TPC command = “down”
$SIR_{est} < SIR_{target,UL} \rightarrow$ TPC command = “up”

Upon the reception of a TPC command, the mobile station should adjust the transmit power of both the uplink DPCCH and the uplink DPDCH in the given direction with a step of Δ_{TPC} dB. The step size Δ_{TPC} is a parameter that may differ between different cells, in the region 0.25 – 1.5 dB.

In case of receiver diversity (e.g., space diversity) or softer handover at the base station, the TPC command should be generated after diversity combining.

In case of soft handover, the following procedure is considered:

- in the base stations a quality measurement is performed on the received signals; in case the quality measurement indicates a value below a given threshold, an increase command is sent to the mobile, otherwise a decrease command is transmitted; all the base stations in the active set send power control commands to the mobile;

- the mobile compares the commands received from different base stations and increases its power only if all the commands indicate an increase value (this means that all the receivers are below the threshold); in case one command indicates a decrease step (that is, at least one receiver is operating in good conditions), the mobile reduces its power; in case more than one decrease commands are received by the mobile, the mobile station should adjust the power with the largest step in the "down" direction ordered by the TPC commands received from each base station in the active set.
- the quality threshold for the base stations in the active set should be adjusted by the outer loop power control (to be implemented in the network node were soft handover combining is performed).

5.5.1.1.2 Outer loop (SIR target adjustment)

The outer loop adjusts the SIR target used by the closed-loop power control. The SIR target is independently adjusted for each connection based on the estimated quality of the connection. In addition, the power offset between the uplink DPDCH and uplink DPCCH may be adjusted. How the quality estimate is derived and how it affects the SIR target is decided by the radio-resource management, i.e. it is not a physical-layer issue.

5.5.1.1.3 Open-loop power control

Open-loop power control is used to adjust the transmit power of the physical Random-Access channel. Before the transmission of a Random-Access burst, the mobile station should measure the received power of the downlink Primary CCPCH over a sufficiently long time to remove effects of the non-reciprocal multi-path fading. From the power estimate and knowledge of the Primary CCPCH transmit power (broadcast on the BCCH) the downlink path-loss including shadow fading can be found. From this path loss estimate and knowledge of the uplink interference level and the required received SIR, the transmit power of the physical Random-Access channel can be determined. The uplink interference level as well as the required received SIR are broadcast on the BCCH.

5.5.1.2 Downlink power control

5.5.1.2.1 Closed loop power control

The downlink closed loop power control adjusts the base station transmit power in order to keep the received downlink SIR at a given SIR target.

The mobile station should estimate the received downlink DPCH power after RAKE combining of the connection to be power controlled. Simultaneously, the mobile station should estimate the total downlink received interference in the current frequency band. The mobile station then generates TPC commands according to the following rule:

$SIR_{est} > SIR_{target,DL} \rightarrow$ TPC command = "down"
$SIR_{est} < SIR_{target,DL} \rightarrow$ TPC command = "up"

Upon the reception of a TPC command, the base station should adjust the transmit power in the given direction with a step of Δ_{TPC} dB. The step size Δ_{TPC} is a parameter that may differ between different cells, in the range 0.25 – 1.5 dB.

In case of receiver diversity (e.g., space diversity) at the mobile station, the TPC command should be generated after diversity combining.

5.5.1.2.2 Outer loop (SIR target adjustment)

The outer loop adjusts the SIR target used by the closed-loop power control. The SIR target is independently adjusted for each connection based on the estimated quality of the connection. In addition, the power offset between the downlink DPDCH and DPCCH may be adjusted. How the quality estimate is derived and how it affects the SIR target is decided by the radio-resource management, i.e. it is not a physical-layer issue.

5.5.2 Cell search

5.5.2.1 Initial cell search

During the initial cell search, the mobile station searches for the base station to which it has the lowest path loss. It then determines the downlink scrambling code and frame synchronisation of that base station. The initial cell search uses the synchronisation channel (SCH), shown in Figure 47 below (repeated from Section Synchronisation Channel).

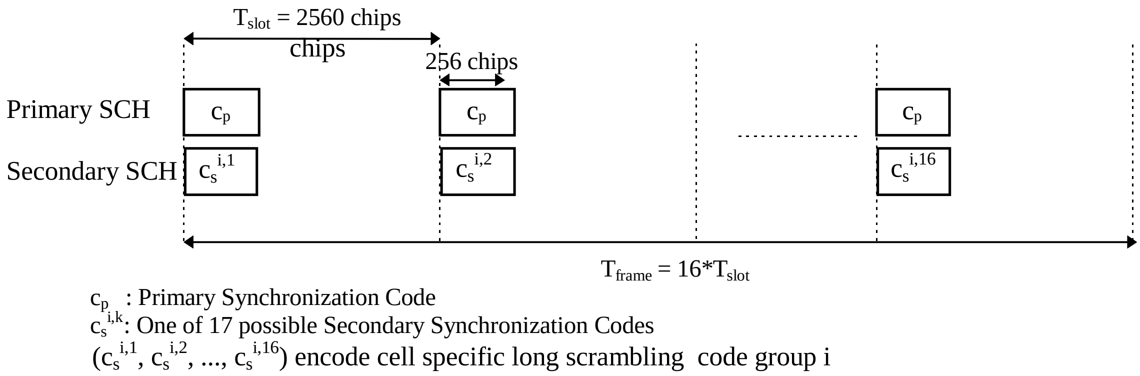


Figure 47. Structure of synchronisation channel (SCH).

This initial cell search is carried out in three steps:

Step 1: Slot synchronisation

During the first step of the initial cell search procedure the mobile station uses the primary SCH to acquire slot synchronisation to the strongest base station. This is done with a single matched filter (or any similar device) matched to the primary synchronisation code c_p which is common to all base stations. The output of the matched filter will have peaks for each ray of each base station within range of the mobile station, see Figure 48. Detecting the position of the strongest peak gives the timing of the strongest base station modulo the slot length. For better reliability, the matched-filter output should be non-coherently accumulated over a number of slots.

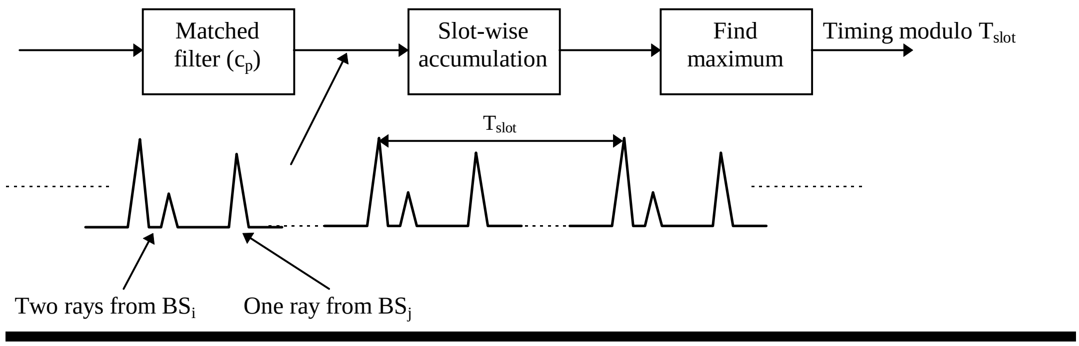


Figure 48. Matched-filter search for primary synchronisation code to slot synchronisation (timing modulo the slot length).

Step 2: Frame synchronisation and code-group identification

During the second step of the initial cell search procedure, the mobile station uses the secondary SCH to find frame synchronisation and identify the code group of the base station found in the first step. This is done by correlating the received signal at the positions of the Secondary Synchronisation Code with all possible (16) Secondary Synchronisation Codes. Note that the position of the Secondary Synchronisation Code is known after the first step. The outputs of all the 17 correlators for 16 consecutive secondary SCH locations are used to form the decision variables. The decision variables are obtained by *non-coherently* summing the correlator outputs corresponding to each 16 length sequence out of the 32 possible sequences and its 16 cyclic shifts giving a total of 512 decision variables. Note that the cyclic shifts of the sequences are unique (see Section Synchronisation Channel). Thus, by identifying the sequence/shift pair that gives the maximum correlation value, the code group as well as the frame synchronisation is determined.

Step 3: Scrambling-code identification

During the third and last step of the initial cell-search procedure, the mobile station determines the exact scrambling code used by the found base station. The scrambling code is identified through symbol-by-symbol correlation over the Primary CCPCH with all scrambling codes within the code group identified in the second step. Note that, from step 2, the frame boundary and consequently the start of the scrambling code is known. Correlation must be carried out symbol-wise, due to the unknown data of the primary CCPCH. Also, in order to reduce the probability of wrong/false acquisition, due to combat background noise/interference, averaging the correlator outputs over a sequence of symbols (diversity) might be required before using the outputs to determine the exact scrambling code.

After the scrambling code has been identified, the Primary CCPCH can be detected, super-frame synchronisation can be acquired and the system- and cell specific BCCH information can be read.

5.5.2.2 Idle mode cell search

When in idle mode, the mobile station continuously searches for new base stations on the current and other carrier frequencies. The cell search is done in basically the same way as the initial cell search. The main difference compared to the initial cell search is that an idle mobile station has received a priority list from the network. This priority list describes in which order the downlink scrambling codes should be searched for and does thus significantly reduce the time and effort needed for the scrambling-code search (step 3). Also the complexity in the second step may be reduced if the priority list only includes scrambling codes belonging to a subset of the total set of code groups. The priority list is continuously updated to reflect the changing neighbourhood of a moving mobile station.

5.5.2.3 Active mode cell search

When in active mode, the mobile station continuously searches for new base stations on the current carrier frequency. This cell search is carried out in basically the same way as the idle mode cell search. The mobile station may also search for new base stations on other carrier frequencies using the slotted mode, see Section Coding for slotted mode.

5.5.3 Random access

The procedure of a random access request is:

1. The mobile station acquires synchronisation to a base station
2. The mobile station reads the BCCH to get information about:
    - 2.1 The preamble spreading code(s) /message scrambling code(s) used in the cell
    - 2.2 The available signatures
    - 2.3 The available access slots
    - 2.4 The available spreading factors for the message part
    - 2.5 The interference level at the base station
    - 2.6 The primary CCPCH transmit power level
3. The mobile station selects a preamble spreading code/message scrambling code
4. The mobile station selects a spreading factor for the message part.
5. The mobile station estimates the downlink path loss (by using information about the transmitted and received power level of the primary CCPCH), and determines the required uplink transmit power (by using information about the interference level at the base station).
6. The mobile station randomly selects an access slot and signature from the available access slots and signatures.
7. The mobile station transmits its random access burst.
8. The mobile station waits for an acknowledgement from the base station. If no acknowledgement is received within a predefined time-out period, the mobile station starts again from step 5.

A typical implementation of the base-station random-access receiver for a given preamble code and preamble sequence is illustrated in Figure 49. The received signal is fed to a matched filter, matched to the preamble code. The output of the matched filter is then correlated with the preamble sequence. The output of the preamble correlator will have peaks corresponding to the timing of any received Random-Access burst using the specific preamble code and preamble sequence. The estimated timing can then be used in an ordinary RAKE combiner for the reception of the data part of the Random-Access burst.

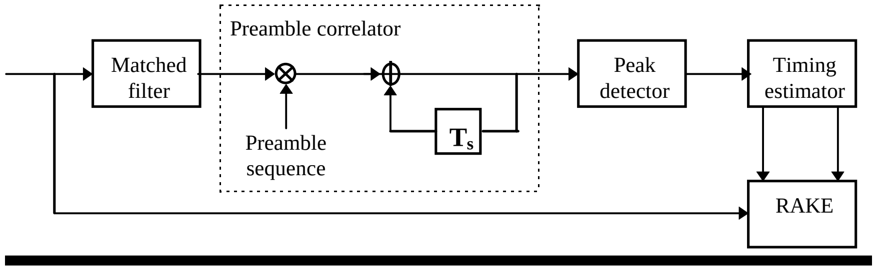


Figure 49. Base-station Random-Access receiver.

Upon reception of the Random-Access burst, the base station responds with an Access Grant message on the FACH. In case the Random Access request is for a dedicated channel (circuit-switched or packet) and the request is granted, the Access Grant message includes a pointer to the dedicated physical channel(s) to use. As soon as the mobile station has moved to the dedicated channel, closed-loop power control is activated.

5.5.4 Idle mode tasks

5.5.4.1 Paging control

5.5.4.1.1 Base Station operation

Every mobile station belongs to one group. When a paging message should be sent to a mobile, the paging message is transmitted on the PCH in the MUI-parts belonging to the terminating mobile's group. The paging message includes the mobile station identification number of the mobile station for which the paging message was intended. When a MUI is transmitted, the corresponding PI1 and PI2 fields are also transmitted.

The exact behaviour of the base station is described as:

For the PCH of the group which does not have terminating information:

- The BS shall transmit the two PI parts (PI1 and PI2) in the PCH as “all 0”.
- The MUI part shall not be transmitted.

For the PCH of the group which have terminating information:

- The BS shall transmit the two PI parts (PI1 and PI2) in the PCH as “all 1”.
- The MUI part shall be transmitted within the same PCH.

5.5.4.1.2 Mobile Station operation

The idea behind the detection of paging messages is to open the receiver to detect one of or both the paging indicators (PI1 and PI2), and if they indicate a paging message for the group the mobile belongs to, the actual paging information part (MUI) is received. When the MUI part is received, the existence of a paging message for the mobile is determined from the information included in the MUI part.

The mobile station operation for detection of paging information in group *n* is shown in Figure 50. $PI1_n$, $PI2_n$, and MUI_n are the PCH components belong to group *n*.

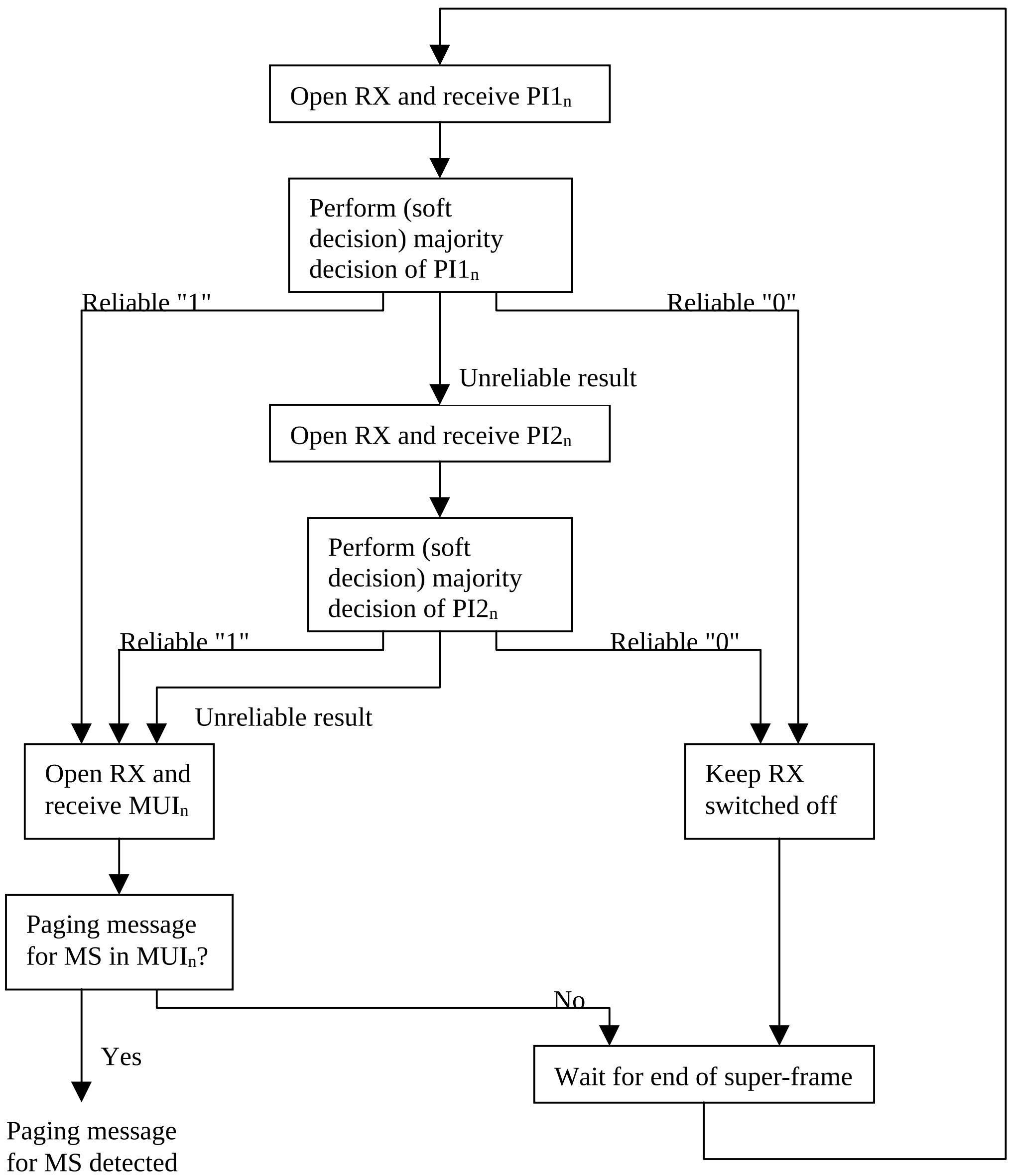


Figure 50. Detection of paging messages.

5.6 Additional features and options (FDD)

5.6.1 Adaptive antennas

Adaptive antennas are recognised as a way to enhance capacity and coverage of the system. Solutions employing adaptive antennas are already supported in the UTRA/FDD concept through the use of connection-dedicated pilot bits on both uplink and downlink.

5.6.2 Multi-user detection

UTRA/FDD is designed to work without requiring joint detection of multiple user signals. However, the potential capacity gains of such receivers in a UTRA/FDD system have been recognised and taken into account in the design of the concept. In the uplink the possibility to use only short codes facilitates more advanced receiver structures with reasonable complexity.

5.6.3 Downlink transmit diversity

Transmitter diversity in the downlink provides a means to significantly improve capacity and coverage of

UTRA/FDD, without the requirement for a second receiver chain in the mobile station that receiver diversity would entail. However, a typical transmit diversity technique, such as delay transmit diversity, has two main drawbacks: self-interference at locations with good SINR; and the requirement for additional Rake fingers in the mobile receiver. In order to overcome these drawbacks, diversity schemes have been proposed for UTRA/FDD, that maintain the orthogonality between diverse downlink transmit antennas, whilst offering significant advantages in the downlink performance. Simulation results for the proposed techniques have shown a gain of up to 7 dB (compared with the non-diversity case) for slow speed mobiles in a single path fading environment. In the proposed schemes, the orthogonality between antennas, is maintained using either code, or time division.

5.6.3.1 Code division transmit diversity

5.6.3.1.1 Orthogonal Transmit Diversity

Orthogonal Transmit Diversity (OTD) utilises code division transmission diversity. The implementation of OTD is as follows. Coded bits are split into two data streams and transmitted via two separate antennas. Different orthogonal channelisation codes are used per antenna for spreading. This maintains the orthogonality between the two output streams, and hence self-interference is eliminated in flat fading. Note that by splitting the coded data into two separate data streams, the effective number of channelisation codes per user is the same as the case without OTD.

The above structure is highly flexible, it may be easily extended to more antennas (4, 8, etc.)

OTD may be an optional feature that can be turned on only if needed. In addition, it is possible to support a mixture of mobiles with and without OTD capability.

The additional required processing at the mobile station is small. Figure 51 illustrates Rake finger processing with OTD. It is important to note that the Pilot signal is also split and transmitted on both antennas, which allows coherent detection of the signals received from both antennas. The data is processed using a Rake finger with parallel processing capability. Both transmitted signal streams are received simultaneously at the same delay (for a given multipath ray), hence no additional buffering and skewing of data is necessary. This significantly reduces the hardware complexity/cost associated with OTD implementation.

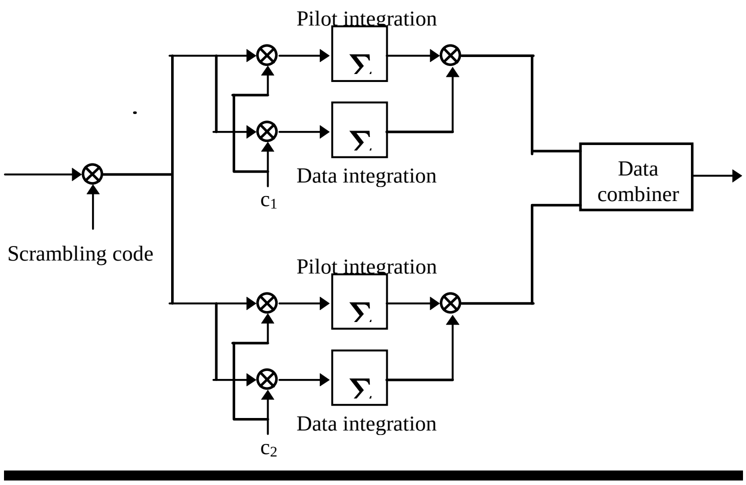


Figure 51. Rake finger processing with OTD.

In the base station transmitter, the base-band processing (i.e. data splitting and separate spreaders) required for OTD already exists with multi-code transmission in the downlink. From the OTD viewpoint, it is advantageous to employ multi-code transmission for all data rates, and it is also recommended to match the number of codes assigned to the user with the number of transmit antennas.

5.6.3.2 Time division transmit diversity

Two schemes have been put forward utilising time division transmission diversity for downlink UTRA/FDD mode operation. The basic Base Station Transmitter block diagram for Time Transmission Diversity is shown in Figure 52. In time division transmission diversity the signal is switched between antennas in one of two ways. Either, the signal is switched according to a pattern decided by the base station, or it is switched depending on signalling received from the mobile station.

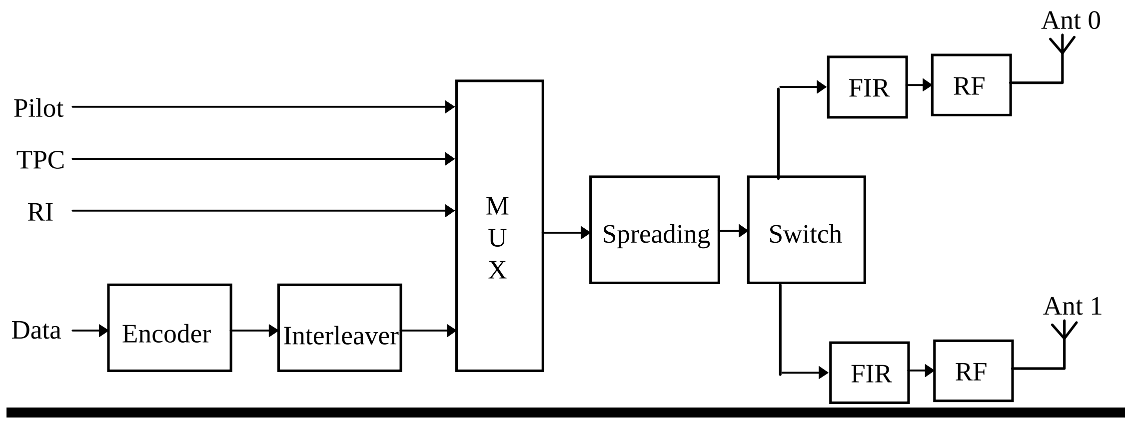


Figure 52. Base station transmitter block diagram for time division transmission diversity.

5.6.3.2.1 Time Switched Transmission Diversity

Time switched transmission diversity (TSTD) is implemented using the block diagram exactly as shown in Figure 52. TSTD does not assume any change to the UTRA/FDD physical layer channel structure other than switching at the filter input. There is no change to the channel coding, rate matching, interleaving and spreading within the UTRA/FDD physical layer description.

TSTD is used for the transmission of downlink Dedicated Physical Channels (DPCHs). All other downlink channels, i.e. the Common Control Physical Channels (CCPCHs) and the Synchronisation Channel (SCH), are transmitted from a single antenna, without diversity. TSTD is implemented by transmitting consecutive slots of the downlink DPCHs through two separate antennas. After scrambling, the spread time slots are switched consecutively to each antenna (i.e. the baseband signal is switched before modulation is applied, between transmitter antennas, at a rate of once every 0.625 ms).

The BCCH informs all mobile stations of the corresponding base station's capability for TSTD. The DPDCH and the DPCCH in the same slot for a given mobile station are then transmitted from one of the antennas. The next slot of the DPCH is transmitted from the other antenna. The DPCHs of other users operating in TSTD mode, may have different switching patterns in order to reduce the peak transmit power and peak to average power ratio in each power amplifier.

The spread time slots are transmitted to each antenna sequentially as shown in Figure 53.

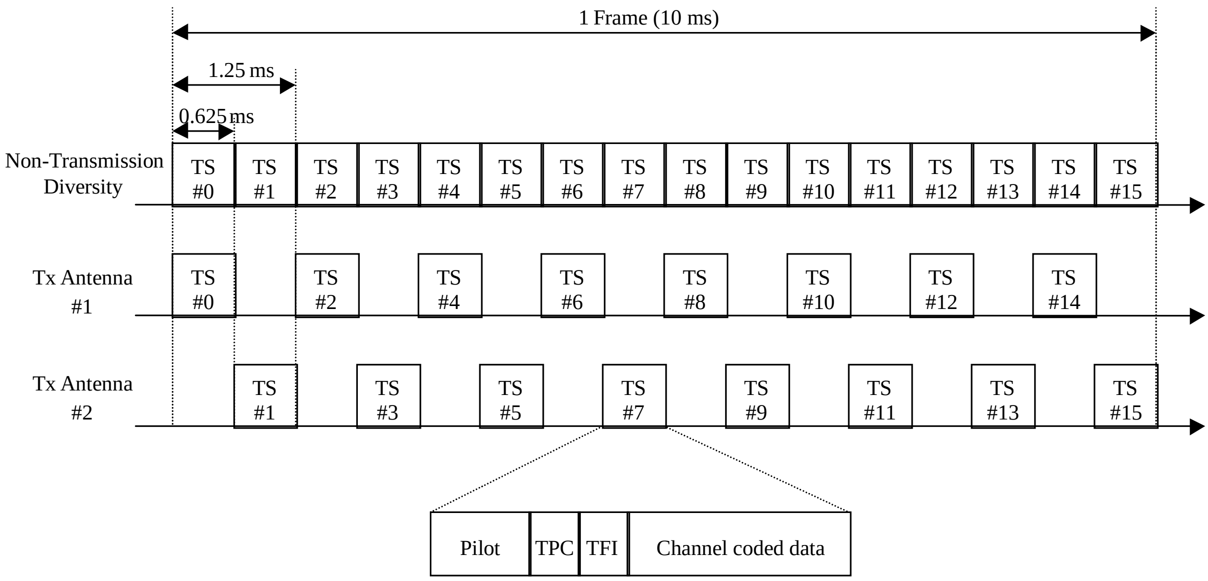


Figure 53. Switching pattern for Dedicated Physical Channels in TSTD.

5.6.3.2.2 Selection Transmit Diversity

Selection Transmit Diversity (STD) with fast closed loop control may be used to provide transmit diversity. For STD, the structure of the Base Station Transmitter is as shown in Figure 54. The implementation of STD is as follows. In the case of no soft handover, the base station antenna is dynamically selected, based on a fast transmit antenna selection (AS) control signal, transmitted by the mobile station (similar to fast PC loop). The value of the AS bit is determined, based on measurements on the antenna specific Primary CCPCH channel. The control loop speed is 400 Hz (note: the exact AS control loop speed is for further study). In order to guarantee that the mobile station is decoding the right downlink signal, the pilot symbols of the antennas are selected to be orthogonal with each other.

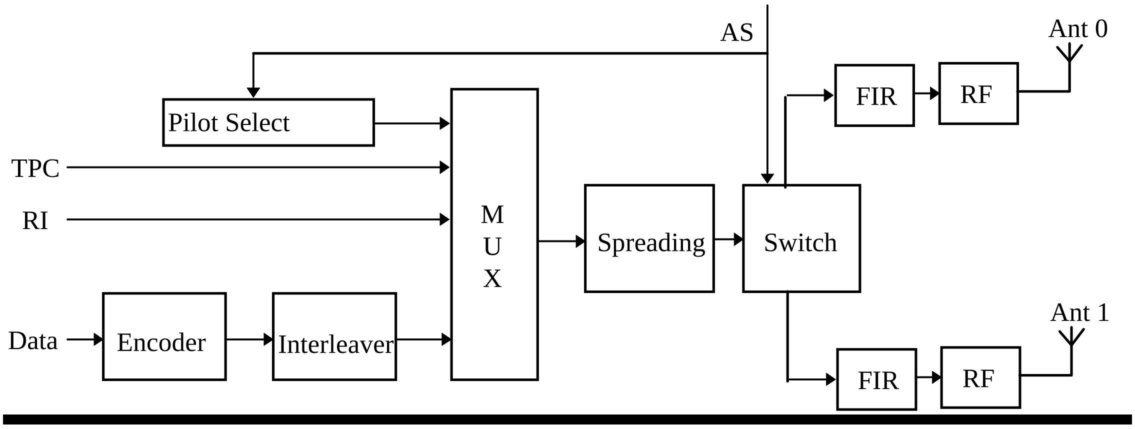


Figure 54. Selective Transmit Diversity: Base station transmitter block diagram.

5.6.4 Locationing function support

The wideband nature of the UTRA/FDD facilitates the high resolution in position location as the resolution achievable is directly proportional to the channel symbol rate, in this case chip rate. The duration of one chip corresponds to approximately 73 meters in propagation distance and if the delay estimation operates on the accuracy of samples/chip then the achievable maximum accuracy is approximately 18 meters with the 4.096 Mcps chip rate. Naturally there are then other inaccuracies that will cause degradation to the positioning but 18 meters can be considered as kind of lower bound on the positioning performance. With higher sampling rate or chip rate the bound is then naturally even lower.

With the UTRA/FDD concept the position location has been discussed in several ETSI/SMG2 input documents. One example solution to use is the proposed power up function (PUF) which in the need for an MS to be heard by several base stations will increase the transmission power over short interval. Other aspects of the position mechanism are how the issue of actual measurement is done and whether that is based on loop around time or on Time Difference Of Arrival (TDOA) or other measures.

6. LAYER 1 DESCRIPTION (TDD MODE)

6.1 Transport channels and physical channels (TDD)

6.1.1 Transport channels

This chapter describes the transport channels that are required for data transfer. Transport channels are the services offered by Layer 1 to the higher layers. A general classification of transport channels is into two groups:

- dedicated channels

- common channels

6.1.1.1 Dedicated transport channels

The only type of dedicated transport channel is the:

1. Dedicated Channel (DCH) characterised by:
    - possibility to use beamforming,
    - possibility to change rate fast (each 10ms),
    - possibility to use enhanced power control and
    - inherent addressing of MSs.

6.1.1.2 Common transport channels

Common transport channels are:

1. Random Access Channel(s) (RACH) characterised by:
    - existence in uplink only,
    - collision risk,
    - open loop power control,
    - limited data field, and
    - requirement for in-band identification of the MSs.
2. Forward Access Channel(s) (FACH) characterised by:
    - existence in downlink only,
    - possibility to use beamforming,
    - possibility to use enhanced power control,
    - requirement for in-band identification of MSs.
3. Broadcast Control Channel (BCCH) characterised by:
    - existence in downlink only,
    - low fixed bit rate and
    - requirement to be broadcast in the entire coverage area of the cell.
4. Paging Channel (PCH) characterised by:
    - existence in downlink only,
    - possibility for sleep mode procedures and
    - requirement to be broadcast in the entire coverage area of the cell.
5. Synchronisation Channel (SCH) characterised by:
    - existence in TDD and downlink only,
    - low fixed bit rate and
    - requirement to be broadcast in the entire coverage area of the cell.

6.1.2 Physical channels

A physical channel is defined as the association of one code, one time slot and one frequency.

6.1.2.1 Frame structure

In the following sections, an overview about the frame, time slot and code structure is outlined.

6.1.2.1.1 Time slots

The TDMA frame has duration of 10 ms and is subdivided into 16 time slots (TS) of 625 µs duration each. A time slot corresponds to 2560 chips. The physical content of the time slots is the bursts of corresponding length as described in Section Burst types.

6.1.2.1.2 TDD frame

Each 10 ms frame consists of 16 time slots; each allocated to either the uplink or the downlink (Figure 55). With such flexibility, the TDD mode can be adapted to different environments and deployment scenarios. In any configuration at least one time slot has to be allocated for the downlink and at least one time slot has to be allocated for the uplink.

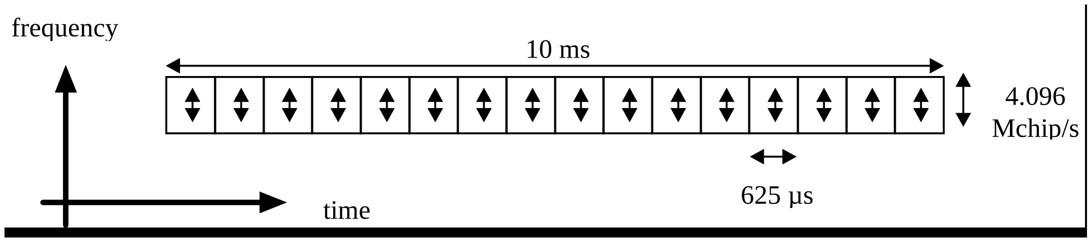


Figure 55. The TDD frame structure

Examples for multiple and single switching point configurations as well as for symmetric and asymmetric UL/DL allocations are given in Figure 56.

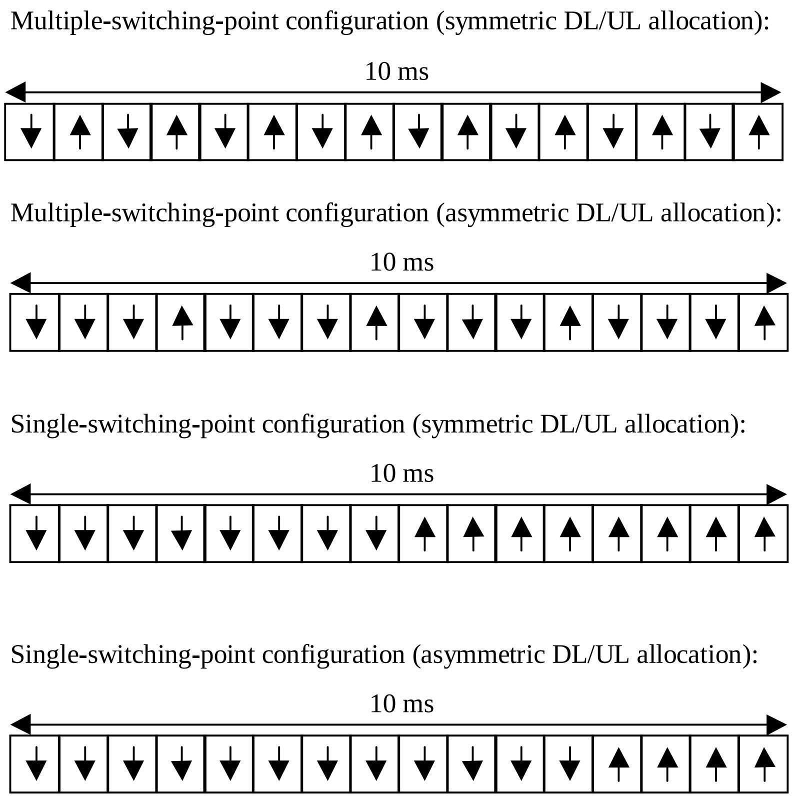


Figure 56. TDD frame structure examples

6.1.2.1.3 Spreading codes

Two options are being considered for the bursts that can be sent as described below. Both options allow a high degree of bit rate granularity and flexibility, thus allowing the implementation of the whole service range from low to high bit rates.

6.1.2.1.4 Multi-code transmission with fixed spreading

Within each time slot of length 625 µs, an additional separation of user signals by spreading codes is used. This means, that within one time slot of length 625 µs, more than one burst of corresponding length as described in Section 0 can be transmitted. These multiple bursts within the same time slot can be allocated to different users as well as partly or all to a single user. For the multiple bursts within the same time slot, different spreading codes are used to allow the distinction of the multiple bursts.

The bursts as described in Section 0 are designed in such a way, that up to 8 bursts could be transmitted within one time slot, if the bursts are allocated to different users in the uplink. In the downlink or if several bursts in the time slot are allocated to one single user in the uplink, even more than 8 bursts (e.g. 9 or 10) can be transmitted within one time slot.

6.1.2.1.5 Single code transmission with variable spreading

Within each time slot of 625 µs,

- a mobile always uses single code transmission by adapting the spreading factor as a function of the data rate. This limits the peak-to-average ratio of the modulated signal and consequently the stress imposed to the power amplifier resulting in an improved terminal autonomy. Several mobiles can be received in the same time slot by the base station, they are separated by their codes and the individual decoding can take profit of the joint detection.
- a base station should broadcast a single burst per mobile again by adapting the spreading as a function of the data rate. High rate data transmissions, requiring more than one timeslot per mobile, can be supported by terminals having the processing power for joint detection on a single slot: the required throughput occupies in a general way an integer number of slots plus a fraction of an extra slot. Single burst transmission should occur in the integer number of slots, while the extra slot can be occupied by a burst for the considered mobile plus extra bursts for other mobiles, joint detection is only needed for this last time slot in the considered mobile.

6.1.2.2 Burst types

As explained in the section Spreading codes, two options are being considered for the spreading.

6.1.2.2.1 Bursts for dedicated transport channels

Two types of bursts for dedicated transport channels are defined: The burst type 1 and the burst type 2. Both consist of two data symbol fields, a midamble and a guard period. The burst type 1 has a longer midamble of 512 chips than the burst type 2 with a midamble of 256 chips. Sample sets of midambles are given in sections Sample midamble code set for burst type 1 and Sample midamble code set for burst type 2.

Because of the longer midamble, the burst type 1 is suited for the uplink, where up to 8 different channel impulse responses have to be estimated. The burst type 2 can be used for the downlink and, if the bursts within a time slot are allocated to less than four users, also for the uplink.

Thus the burst type 1 can be used for

- uplink, independent of the number of active users in one time slot
- downlink, independent of the number of active users in one time slot

The burst type 2 can be used for

- uplink, if the bursts within a time slot are allocated to less than four users
- downlink, independent of the number of active users in one time slot

The data fields of the burst type 1 are 976 chips long, whereas the data fields length of the burst type 2 are 1104 chips. The corresponding number of symbols depends on the spreading factor, as indicated in Table 7 below. The guard period for the burst types 1and 2 is 96 chip periods long.

Table 7. Number of symbols per data field in bursts 1 and 2

Spreading factor (Q)	Number of symbols (N) per data field in Burst 1	Number of symbols (N) per data field in Burst 2
1	976	1104
2	488	552
4	244	276
8	122	138
16	61	69

The burst types 1 and 2 are shown in Figure 57 and Figure 58. The contents of the traffic burst fields are described in Table 8and Table 9.

Table 8. The contents of the burst type 1 fields

Chip number (CN)	Length of field In chips	Length of field in symbols	Length of field in µs	Contents of field
0-975	976	Cf. Table 7	238.3	Data symbols
976-1487	512	-	125.0	Midamble
1488-2463	976	Cf. Table 7	238.3	Data symbols
2464-2559	96	-	23.4	Guard period

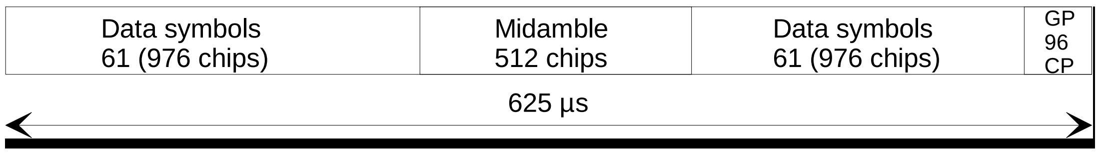


Figure 57. Burst structure of the burst type 1. GP denotes the guard period and CP the chip periods.

Table 9. The contents of the burst type 2 fields

Chip number (CN)	Length of field in chips	Length of field In symbols	Length of field in µs	Contents of field
0-1103	1104	Cf. Table 7	269.55	Data symbols
1104-1359	256	-	62.5	Midamble
1360-2463	1104	Cf. Table 7	269.55	Data symbols
2464-2559	96	-	23.4	Guard period

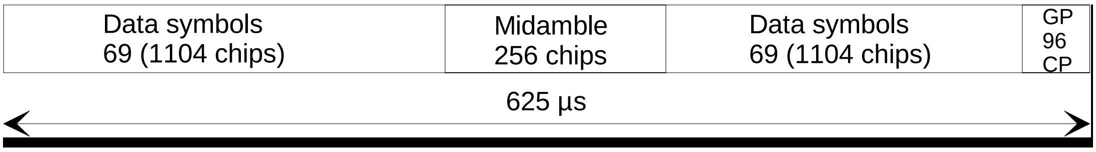


Figure 58. Burst structure of the burst type 2. GP denotes the guard period and CP the chip periods.

The two different bursts defined here are well-suited for the different applications mentioned above. It may be possible to further optimise the burst structure for specific applications, for instance for unlicensed operation.

6.1.2.3 Training sequences for spread bursts

As explained in the section Spreading codes, two options are being considered for the spreading. The training sequences presented here are common to both options.

Section Bursts contains a description of the spread speech/data bursts. These traffic bursts contain L_m midamble chips, which are also termed midamble elements. The L_m elements $\underline{m}_i^{(k)}$; i=1,...,L_m; k=1,...,K; of the midamble codes $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; of the K users are taken from the complex set

$$\underline{\mathrm{V}}_{\mathrm{m}} = \{1,\ \mathrm{j},\ -1,\ -\mathrm{j}\}. \qquad (0\text{-}1)$$

The elements $\underline{m}_i^{(k)}$ of the complex midamble codes $\underline{\mathbf{m}}^{(k)}$ fulfil the relation

$$\underline{m}_i^{(k)} = (\mathrm{j})^i \cdot m_i^{(k)} \qquad m_i^{(k)} \in \{1,\ -1\};\ i = 1,\ldots,L_m;\ k = 1,\ldots,K. \qquad (0\text{-}2)$$

Hence, the elements $\underline{m}_i^{(k)}$ of the complex midamble codes $\underline{\mathbf{m}}^{(k)}$ of the K users are alternating real and imaginary.

With W being the number of taps of the impulse response of the mobile radio channels, the L_m binary elements $m_i^{(k)};\ \mathrm{i} = 1,\ldots,L_{\mathrm{m}};\ k = 1,\ldots,K;$ of (6-2) for the complex midambles $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; of the K users are generated according to Steiner's method from a single periodic basic code

$$\mathbf{m} = \left(m_1, m_2, \ldots, m_{L_m+(K-1)W}\right)^{\mathrm{T}} \qquad m_i \in \{1,\ -1\};\ \ i = 1,\ldots,(L_m + (K-1)W). \qquad (0\text{-}3)$$

The elements $m_i;\ i = 1,\ldots,(L_m + (K-1)W),$ of (6-3) fulfil the relation

$$m_i = m_{i-P} \qquad \text{for the subset} \qquad i = (P+1),\ldots,(L_m + (K-1)W). \qquad (0\text{-}4)$$

The P elements $m_i;\ i = 1,\ldots,P,$ of one period of $\mathbf{m}$ according to (6-3) are contained in the vector

$$\mathbf{m}_{\mathrm{P}} = \left(m_1, m_2, \ldots, m_P\right)^{\mathrm{T}}. \qquad (0\text{-}5)$$

With $\mathbf{m}$ according to (6-3) the L_m binary elements $m_i^{(k)};\ \mathrm{i} = 1,\ldots,\mathrm{L}_{\mathrm{m}};\ k = 1,\ldots,K;$ of (6-2) for the midambles of the K users are generated based on Steiner's formula

$$m_i^{(k)} = m_{i+(K-k)W} \qquad i = 1,\ldots,L_m;\ k = 1,\ldots,K. \qquad (0\text{-}6)$$

In the following the term *'a midamble code set'* or *'a midamble code family'* denotes K specific midamble codes $\underline{\mathbf{m}}^{(k)}$; k=1,...,K. Different midamble code sets $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; are in the following specified based on different periods $\underline{\mathbf{m}}_{\mathrm{P}}$ according (6-5).

In adjacent cells of the cellular mobile radio system, different midamble codes sets $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; should be used to guarantee a proper channel estimation.

As mentioned above a single midamble code set $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; consisting of K midamble codes is based on a single period $\underline{\mathbf{m}}_{\mathrm{P}}$ according to (6-5).

In the following several periods $\underline{\mathbf{m}}_{\mathrm{P}}$ according (6-5) which should be used to generate different midamble code sets $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; will be listed in tables in a hexadecimal representation. As shown in Table 10 always 4 binary elements $\underline{m}_i$ are mapped on a single hexadecimal digit.

Table 10. Mapping of 4 binary elements $\underline{m}_i$ on a single hexadecimal digits

4 binary elements $\underline{m}_i$	**Mapped on hexadecimal digit**
-1 -1 -1 -1	0
-1 -1 -1 1	1
-1 -1 1 –1	2
-1 -1 1 1	3
-1 1 -1 –1	4
-1 1 -1 1	5
-1 1 1 –1	6
-1 1 1 1	7
1 -1 -1 –1	8
1 -1 -1 1	9
1 -1 1 –1	A
1 -1 1 1	B
1 1 -1 –1	C
1 1 -1 1	D
1 1 1 –1	E
1 1 1 1	F

The mean degradations [2, equation (38)] which serve as a quality information of the periods $\underline{\mathbf{m}}_{\mathrm{P}}$ according to (6-5) and hence of the specified midamble code sets $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; will be also given.

6.1.2.3.1 Sample midamble code set for burst type 1

In the case of burst type 1 (see section Bursts) the midamble has a length of L_{m}=512, which is corresponding to:

K=8; W=57; P=456

Table 11. Sample Periods $\underline{\mathbf{m}}_P$ *according (6-5) for case of burst type 1.*

Periods $\underline{\mathbf{m}}_P$ of length *P*=456	Degradation in dB
C482462CA7846266060D21688BA00B72E1EC84A3D5B7194C8DA39E21A3CE12BF512 C8AAB6A7079F73C0D3E4F40AC555A4BCC453F1DFE3F6C82	0.649471
56F3ACE0A65B96FC326A30B91665BD4380907C2B08DEC98C16A0B0339AEA855C3D 8BDD016E4C3E0F3DA5DF5C0891C851BA30A6C19ABE6C3ED4	0.695320
1D566C76440333CBF3CA2A405386068E19A2D6A53560CC50138B3A15BF7D9683F95F 66FF096431363E09A514D61099DD3EAD52903BF4A27D14	0.705751
9A0A349E49389CC184F7A3420D3FBE06B3A40BEE933D8E04E61FAA4A5214D918A1 ADD5BE25D833579FBCF17B422300D0CA1B419393F9722AA8	0.706513
B760E5694E49169C225A2FBCDACCCA8847F8486A6A351EB7D045BA2271B2A4CB90 0404C0D2BBA00F80F963861BD7DCE748F0F10AE6B785D0F0	0.707417
ECE93B83CE32E395405F7C889751970E84AFD632500B91E17C4E7846FE68D3C841013 5D3114D3281211214D1F5F1996A6B656259F11728AA52	0.708587
DE1B6F6219A0AD1A3EB5EEA02173D704C3340AAE7310B93A21BCF979BC7B6C081 7003AA300B1704BCE62524EC48C505977A1570F6C6BA1A2D8	0.711320

6.1.2.3.2 Sample midamble code set for burst type 2

In the case of burst type 2 (see section Bursts) the midamble has a length of L_m=256, which is corresponding to:

K=3; W=64; P=192

Table 12. Sample Periods $\underline{\mathbf{m}}_P$ *according (6-5) for case of burst type 2.*

Periods of length *P*=192	Degradation in dB
D4A124FE4D11BC14C258546A18C5DE0E3AA3F0617245DBFE	0.615566
48D76A687E21D22321C5201977F620D7A4CB5945F5693A1C	0.638404
9EEF5E79606DCAAB046769524691E09E816DC688ABC12030	0.663436
D2369A2B704878F55B58A300C853A2F62233E6207E39F944	0.677739
A26C7D9697B002714E9285D2AFC3AF1E233FC8C6C7486080	0.686287
8A615F5D7EE05668415E626482E90B11C95305E4707015B5	0.686660
5CC2D7409922FA463D2D14377EBCF0CC0E888426B06F0A82	0.688977
A68238D5BD37B2B4C48B466B9815087898409AFCB804FA0B	0.692613

6.1.2.3.3 Midamble transmit power

In the case of the downlink, 2*K* data blocks are transmitted in a burst simultaneously. Also in the uplink, if *K*' greater than one CDMA code are assigned to a single user, 2*K*' data blocks are transmitted in a burst simultaneously by this user. This is the so-called multi-code uplink situation. In the downlink and the multi-code uplink, the mean power used to transmit the midambles on the one hand and the 2*K* (or 2*K*') data blocks on the other hand shall be equal. This shall be achieved by multiplying the midamble codes $\underline{\mathbf{m}}^{(k)}$, *k*=1,...,*K*, with a proper real factor to achieve an attenuation or an amplification.

6.1.3 Mapping of transport channels to physical channels

This section describes the way in which transport channels are mapped onto physical channels as described in

Section Physical channels. A description of the multi-frame structure is given in Section Multi-frame structure.

6.1.4 Dedicated transport channels

A dedicated transport channel is mapped onto one or more sets of slots and codes within a frame. An interleaving period is associated with each allocation. The frame is subdivided into slots that are available for uplink and downlink information transfer. Each set of slots and codes over an interleaving period maps to a data unit and a data unit can correspond to one or more FEC code blocks and one or more RLC protocol data units dependent from the service being supported. The following diagram illustrates the mapping:

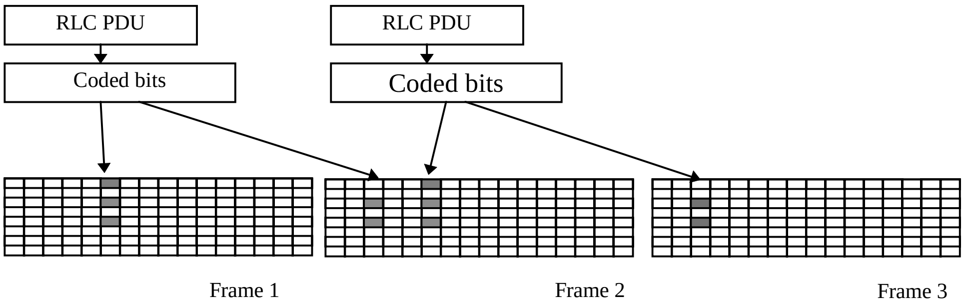


Figure 59. Mapping of PDU onto the physical bearer

For NRT packet data services an allocation is made only for a relatively short period of time. In general, for RT services an allocation is made for a certain time period and a release procedure is necessary to release the resource. For the efficient use of resources the slot/ code set allocated to a radio bearer may be changed from time to time and the resources allocated to a VBR service may increase or decrease along with the changes in the data rate. Traffic channels are power controlled, cf. Section Power Control.

6.1.5 Multi-frame structure

A strong requirement for the multi-frame structure comes from the realisation of low cost dual mode FDD-TDD terminals and from the GSM compatibility of the UTRA proposal. In this respect the super-frame and multi-frame structure for FDD and TDD mode have to be compatible and harmonised with GSM.

Thus in the proposed structure a multi-frame is composed by 24 frames each of length 10 ms. So the multi-frame period is 240 ms (twice the GSM TCH-F multi-frame).

All frames in the traffic channel multi-frames are used to carry both user data and dedicated signalling because:

- The use of in-band dedicated signalling or allocation of a dedicated signalling channel avoids the use of a signalling frame like SACCH frame in GSM.

  The most flexible method to distribute different user data blocks such as in-band signalling is under study.

- There is no need for an idle slot to read BCCH’s of adjacent cells as in GSM

  Adjacent cells in the TDD network are frame-synchronised.

- The bursty nature of TD-CDMA transmission and reception allows the MS in idle time slots to make measurements on GSM and FDD networks. This is valid also for high bit rate users (BCCH and RACH slots could also be used to this purpose)

The multi-frame length is therefore given by the common channel with the lowest bit rate in the present case the SCH, if its multi-frame structure is compatible with the GSM TCH-F multi-frame. This leads to a multi-frame length of 240 ms. Three TDD multi-frames match exactly into a FDD multi-frame ensuring the compatibility of both components.

6.2 Multiplexing, channel coding and interleaving (TDD)

6.2.1 General

This section describes the services multiplexing, channel coding/interleaving and rate matching.

In the UTRA-TDD mode, the total number of basic physical channels (a certain time slot one spreading code on a certain carrier frequency) per frame is given by the maximum number of time slots which is 16 and the maximum number of CDMA codes per time slot. This maximum number of codes is 8 in case the different codes within one time slot are allocated to different users in the uplink and is higher than 8 (e.g. 9 or 10) in the downlink or if several codes are allocated to one single user in the uplink.

The service classes given in the following represent only a selection of all possibilities that are conceivable.

Two types of traffic bursts are used. They are described in section Physical channels.

6.2.2 Multiplexing

In a same connection, multiple services could be treated with separate channel coding/interleaving and mapping to different basic physical channels (slot/code), see Figure 60. In this way QoS can be separately and independently controlled.

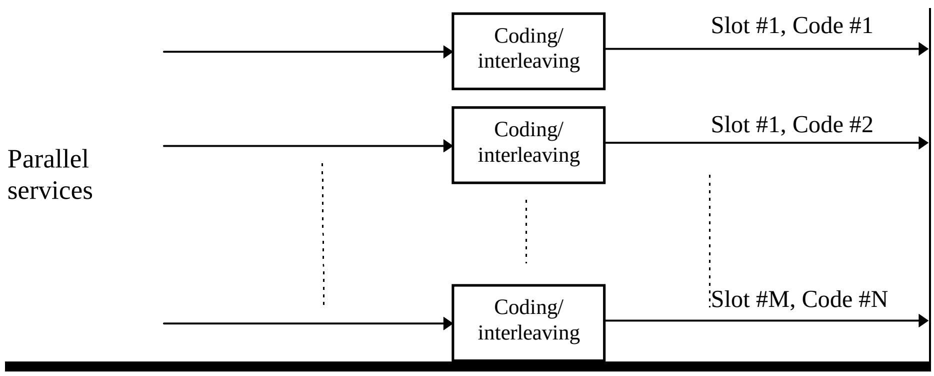


Figure 60. Service multiplexing (a)

A second alternative is time multiplexing at different points of the channel coding scheme, as shown in Figure 61.

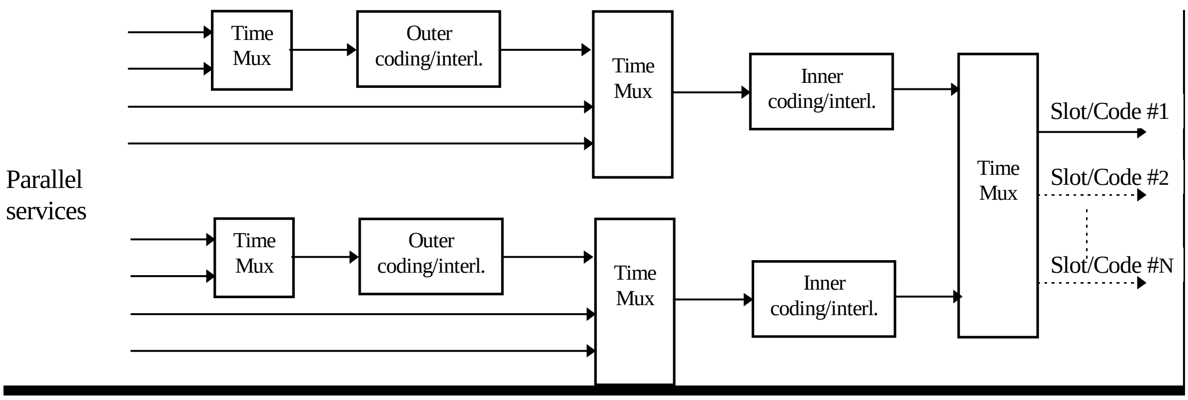


Figure 61. Service multiplexing (b)

After service multiplexing and channel coding, the multi-service data stream is mapped to one or, if the total rate exceeds the upper limit for single-code transmission, several resource units.

6.2.3 Channel coding and interleaving

In Real Time (RT) services a FEC coding is used, instead Non Real Time (NRT) services could be well managed with a proper combination of FEC and ARQ.

For the RT services two levels of QoS (10^{-3}, 10^{-6}) have been considered as examples in Figure 62.

Only convolutional coding is used in case of BER=10^{-3}, while a concatenated code scheme (Reed-Solomon, outer interleaving and convolutional coding) or Turbo codes could be used to achieve BER=10^{-6}.

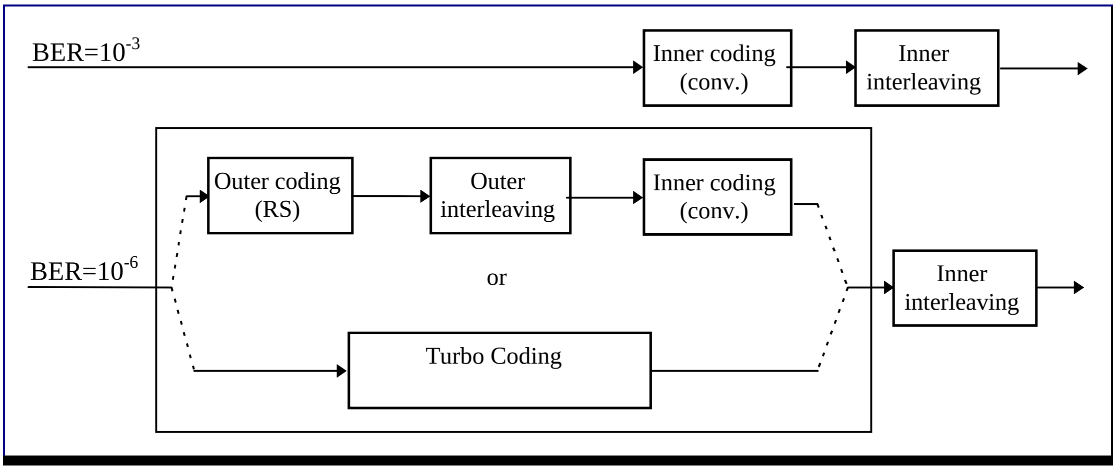


Figure 62. FEC coding

6.2.3.1 Inner coding/interleaving

The convolutional coding rates change according to the rates of different services. The convolutional coding rates from 1/4 to 1 have been chosen such that the complete system will be able to use as much as possible the same decoding structure.

After convolutional coding, interleaving is used. For LDD services, inter-frame interleaving over two 10 ms frames is applied. For LCD services an interleaving over 300 ms is applied.

6.2.3.2 Outer coding/interleaving

The outer RS coding, on GF(2^8) has different rate for different services. An outer interleaver to break the error burst at the output of the Viterbi decoder is needed in addition to an inner interleaver for breaking the error bursts due to fading.

6.2.4 Rate matching

To map the services on the air interface either puncturing or unequal repetition is used after channel coding.

This rate matching is performed considering both bursts:

- burst 1 (long midamble) used in uplink;
- burst 2 (short midamble) used in downlink as well as for uplink transmission in the case of multi-code transmission.

6.3 Spreading and modulation (TDD)

6.3.1 General

In this chapter, there has been made a separation between the data modulation and the spreading modulation. The data modulation is defined in Section Data modulation and the spreading modulation in Section Spreading modulation.

Table 13. Basic modulation parameters

Chip rate	Same as FDD basic chiprate: 4.096 Mchip/s
Carrier spacing	5.0 MHz
Data modulation	QPSK
Chip modulation	Same as FDD chip modulation: root raised cosine roll-off α = 0.22
Spreading characteristics	Orthogonal Q chips/symbol, where $Q = 2^p$, 0<= p <= 4

6.3.2 Data modulation

6.3.2.1 Symbol rate

The symbol rate and duration are indicated below.

Ts = Q. T_c, where $T_c = \frac{1}{chiprate}$ = 0.24414 µs, reflecting the dependence of the symbol time T_s upon the spreading factor Q.

6.3.2.2 Mapping of bits onto signal point constellation

A certain number K of CDMA codes can be assigned to either a single user or to different users who are simultaneously transmitting bursts in the same time slot and the same frequency. The maximum possible number of CDMA codes, which is smaller than or equal to 16, depends on the individual spreading factors, the actual interference situation and the service requirements. In Section Bursts examples of bodies of such spread bursts associated with a particular user are shown. Each user burst has 2 data carrying parts termed data blocks

$$\underline{\mathbf{d}}^{(k,i)} = (\underline{d}_1^{(k,i)}, \underline{d}_2^{(k,i)}, \ldots, \underline{d}_N^{(k,i)})^{\mathrm{T}} \qquad \mathrm{i} = 1,\ 2;\ \mathrm{k} = 1,\ldots,\mathrm{K}. \tag{0-7}$$

N_k is the number of symbols per data field for the user k. This number is linked to the spreading factor Q_k as described in Table 7.

Data block $\underline{\mathbf{d}}^{(k,1)}$ is transmitted before the midamble and data block $\underline{\mathbf{d}}^{(k,2)}$ after the midamble. Each of the *N* data symbols $\underline{d}_n^{(k,i)}$; i=1, 2; k=1,...,K; n=1,...,N; of (6-7) has the symbol duration $T_s^{(k)} = Q_k . T_c$ as already given.

The data modulation is QPSK; thus the data symbols $\underline{d}_n^{(k,i)}$ are generated from two interleaved and encoded data bits

$$b_{l,n}^{(k,i)} \in \{0,1\} \qquad \mathrm{l} = 1,2;\ \mathrm{n} = 1,\ldots,\mathrm{N};\ \mathrm{k} = 1,\ldots,\mathrm{K};\ \mathrm{i} = 1,\ 2 \tag{0-8}$$

using the equation

$$\mathrm{Re}\{\underline{d}_n^{(k,i)}\} = \frac{1}{\sqrt{2}}(2b_{1,n}^{(k,i)} - 1)$$

$$\mathrm{Im}\{\underline{d}_n^{(k,i)}\} = \frac{1}{\sqrt{2}}(2b_{2,n}^{(k,i)} - 1) \qquad \mathrm{n} = 1,\ldots,\mathrm{N};\ \mathrm{k} = 1,\ldots,\mathrm{K};\ \mathrm{i} = 1,\ 2. \tag{0-9}$$

Equation (6-9) corresponds to a QPSK modulation of the interleaved and encoded data bits $b_{l,n}^{(k,i)}$ of (6-8).

6.3.2.3 Pulse shape filtering

The pulse shape filtering is applied to each chip at the transmitter. In this context the term chip represents a single element $\underline{c}_q^{(k)}$ with k=1,...,K; q=1,...,Q_k; of a spreading code $\underline{\mathbf{c}}^{(k)}$; see also Section Spreading codes.

The impulse response of the above mentioned chip impulse filter $Cr_0(t)$ shall be a root raised cosine. The corresponding raised cosine impulse $C_0(t)$ is defined as

$$C_0(t) = \frac{\sin \pi \frac{t}{T_C}}{\pi \frac{t}{T_C}} \cdot \frac{\cos \alpha\pi \frac{t}{T_C}}{1 - 4\alpha^2 \frac{t^2}{T_C{}^2}} \tag{0-10}$$

The roll-off factor shall be $\alpha = 0.22$. T_C is the chip duration:

$$T_C = \frac{1}{\mathrm{Chiprate}} = 0.24414\mu\mathrm{s}$$

The impulse response $C_0(t)$ according to (6-10) and the energy density spectrum $\Phi_{C0}(f)$ of $C_0(t)$ are depicted in the figure below:

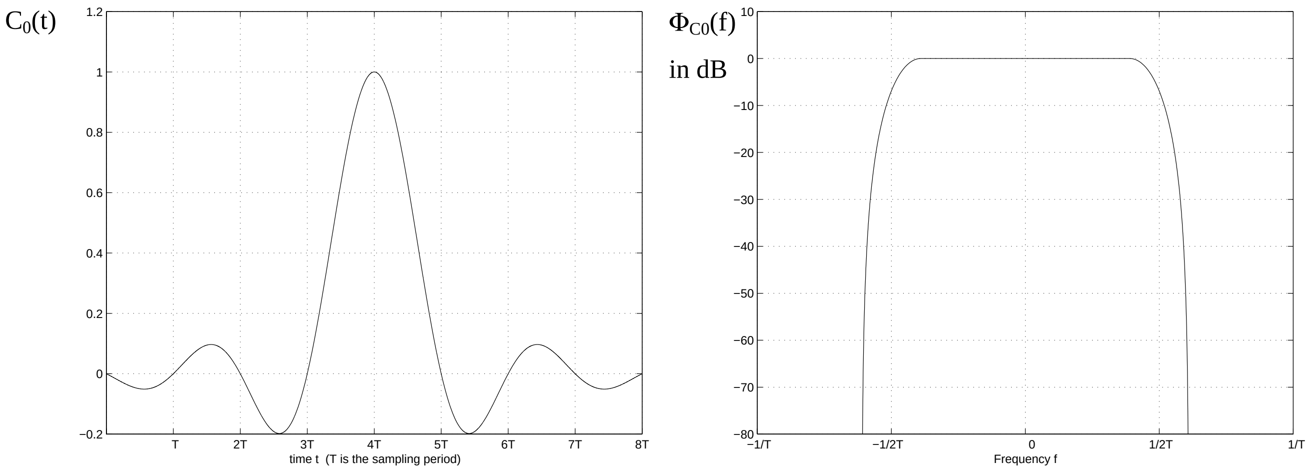


Figure 63. Basic impulse $C_0(t)$ and the corresponding energy density spectrum $\Phi_{C0}(f)$ of $C_0(t)$

6.3.3 Spreading modulation

6.3.3.1 Basic spreading parameters

Each data symbol $\underline{d}_n^{(k,i)}$ of (6-7) is spread with a spreading code $\underline{\mathbf{c}}^{(k)}$ of length. $Q_k \in \{1,2,4,8,16\}$. The resulting sequence is then scrambled by a sequence ν of length 16.

6.3.3.2 Spreading codes

The elements $\underline{c}_q^{(k)}$; k=1,...,K; q=1,...,Q_k; of the spreading codes $\underline{\mathbf{c}}^{(k)}=(\underline{c}_1^{(k)},\underline{c}_2^{(k)},...,\underline{c}_{Q_k}^{(k)})$; k=1,...,K; shall be taken from the complex set

$$\underline{V}_c = \{1,\ j,\ -1,\ -j\}. \tag{0-11}$$

In equation (6-11) the letter j denotes the imaginary unit. The spreading code $\underline{\mathbf{c}}^{(k)}$ = $\underline{\mathbf{c}}^{(k)}=(\underline{c}_1^{(k)},\underline{c}_2^{(k)},...,\underline{c}_{Q_k}^{(k)})$ is generated from the binary CDMA codes $\mathbf{a}^{(k)}$ of length Q_k shown in Figure 64allocated to the k^{th} user. . The relation between the elements $\underline{c}_q^{(k)}$ and $\underline{a}_q^{(k)}$ is given by:

$$\underline{c}_q^{(k)} = (\mathrm{j})^q \cdot a_q^{(k)} \qquad a_q^{(k)} \in \{1,\ -1\};\ q=1,\dots,Q;\ k=1,\dots,K. \tag{0-12}$$

Hence, the elements $\underline{c}_q^{(k)}$ of the CDMA codes $\underline{\mathbf{c}}^{(k)}$ are alternating real and imaginary. The $\mathbf{a}^{(k)}$ are Orthogonal Variable Spreading Factor (OVSF) codes, allowing mixing in the same timeslot channels with different spreading factors while preserving the orthogonality. The OVSF codes can be defined using the code tree of Figure 64.

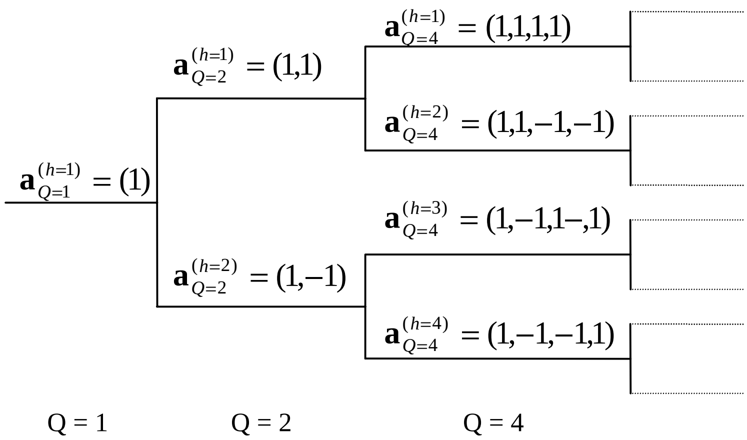


Figure 64. Code-tree for generation of Orthogonal Variable Spreading Factor (OVSF) codes.

Each level in the code tree defines spreading factors indicated by the value of Q in the figure. All codes within the code tree cannot be used simultaneously in a given timeslot. A code can be used in a timeslot if and only if no other code on the path from the specific code to the root of the tree or in the sub-tree below the specific code is used in this timeslot. This means that the number of available codes in a slot is not fixed but depends on the rate and spreading factor of each physical channel.

The spreading factor goes up to Q_{MAX}=16.

6.3.3.3 Scrambling codes

The spreading of data by a code $\mathbf{a}^{(k)}$ of length Q_k is followed by a cell specific scrambling sequence $\mathbf{v}$=(ν1, ν2, … ν_{QMAX}). The length matching is obtained by concatenating Q_{MAX}/Q_k spread words before the scrambling. The scheme is illustrated in Figure 65 below and is described in more detail in Section Spread and scrambled signal of data symbols and data blocks.

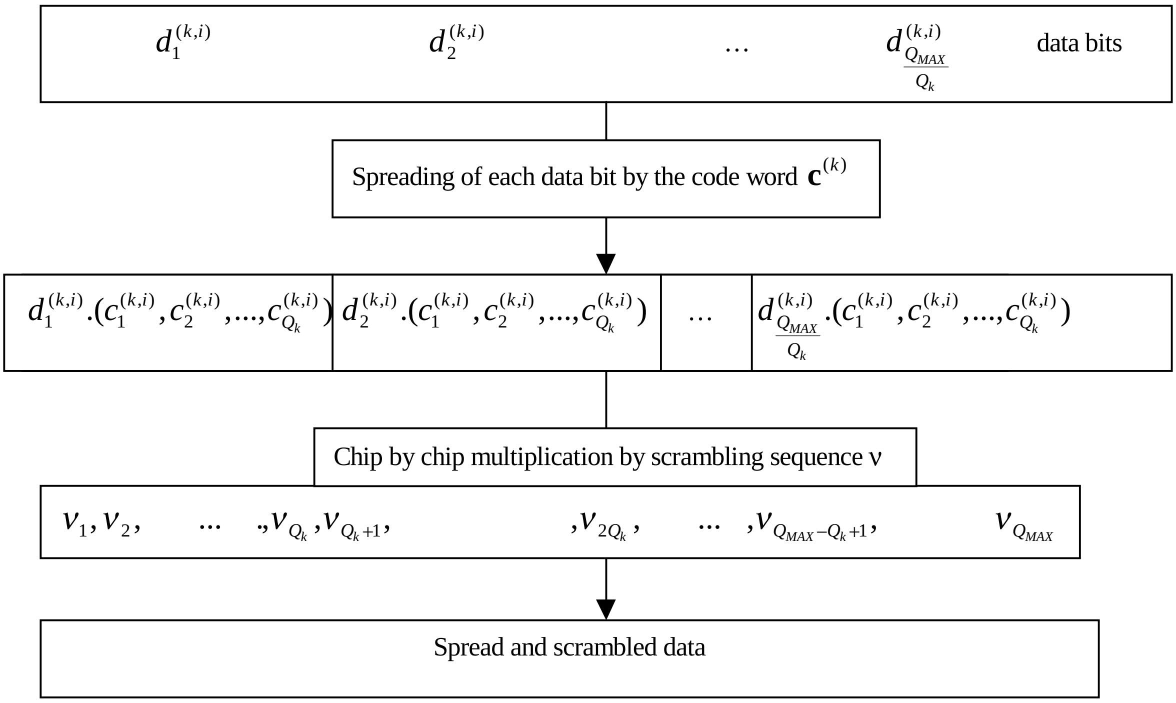


Figure 65. Spreading and subsequent scrambling of data bits.

6.3.3.4 Spread and scrambled signal of data symbols and data blocks

The combination of the spreading and cell specific scrambling codes can be seen as a user and cell specific spreading code $\mathbf{s}^{(k)} = \left(s_p^{(k)}\right)$ with $s_p^{(k)} = c^{(k)}_{1+[(p-1) \bmod Q_k]} .\text{í}_{1+[(p-1) \bmod Q_{MAX}]}$, k=1,…,K, p=1,…,$N_k Q_k$. The transmitted signal belonging to the data block $\underline{\mathbf{d}}^{(k,1)}$ of (6-7) transmitted before the midamble is

$$\underline{d}_n^{(k,1)}(t) = \sum_{n=1}^{N} \underline{d}_n^{(k,1)} \sum_{q=1}^{Q} \underline{c}_q^{(k)} \cdot Cr_o(t-(q-1)T_c - nT_c) \qquad \text{(0-13)}$$

and for the data block $\underline{\mathbf{d}}^{(k,2)}$ of (6-7) transmitted after the midamble

$$\underline{d}_n^{(k,2)}(t) = \sum_{n=1}^{N} \underline{d}_n^{(k,2)} \sum_{q=1}^{Q} \underline{c}_q^{(k)} \cdot Cr_0(t-(q-1)T_C - nT_C - NQT_c - L_m T_c). \qquad \text{(0-14)}$$

where L_m is the number of midamble chips.

6.4 Radio transmission and reception (TDD)

6.4.1 Frequency bands and channel arrangement

6.4.1.1 Proposed frequency bands for operation

UTRA/TDD is designed to operate in any frequency band that will accommodate at least one 4,096 Mcps carrier.

6.4.1.2 Carrier raster

The channel raster is 200 kHz.

6.4.1.3 Tx - Rx Frequency Separation

Tx and Rx are not separated in frequency.

6.4.2 Service Class

See relevant chapter for FDD mode

6.4.3 Transmitter characteristics

6.4.3.1 Output power

The mobile station and base station output power profiles would be used to define a range of output powers for the use in different system scenarios. The power class would be based on the peak power, e. g. 30 dBm for the terminals.

6.4.3.2 Output power dynamics

The transmitter uses fast closed-loop carrier/interference based power control and slow quality based power control on both the up- and downlink. The step size is variable and in the range 1.5 ...3 dB with 100-800 steps/s. The power control dynamic is 80 dB on the uplink and 30 dB on the downlink.

6.4.3.3 Output RF spectrum emissions, adjacent channel power, occupied bandwidth, frequency stability

See relevant Sections for FDD mode (Section Radio transmission and reception (FDD)).

6.4.4 Receiver characteristics

The receiver is typically a Joint Detection Receiver. Except for this the relevant chapters for the receiver characteristic of the FDD system apply also for the TDD system.

6.5 Physical layer procedures (TDD)

6.5.1 Synchronisation of the TDD base stations

It is required that BTS supporting the TDD mode, are operated in synchronised mode, so far the coverage area of the cells are overlapping, i.e. we have contiguous coverage for a certain area. The nature of the TDD operation requires BTS frame synchronisation, to achieve good spectral efficiency. The fact that MS and BTS are receiving and transmitting on the same frequency makes it desirable, that in the reuse cell the same TX / RX timing get used.

The lack of a frame synchronisation can cause, depending on the actual time slip, interference events that will effect several time slots.

By means of a frame synchronisation this effect should be minimised. However, it will be necessary for a cost efficient solution to allow some slip. The tolerance of the frame synchronisation shall be such, that the affected timeslots receive only minor performance degradation. I.e. only some of the symbols shall be corrupted by the frame slip, rather than a full slot. However synchronisation on a chip level is not required.

6.5.2 Channel Allocation

For the UTRA-TDD mode a physical channel is characterised by a combination of its carrier frequency, time slot, and spreading code as explained in the chapter on the physical channel structure

Channel allocation covers both:

- resource allocation to cells (slow DCA)
- resource allocation to bearer services (fast DCA)

6.5.2.1 Resource allocation to cells (slow DCA)

Channel allocation to cells follows the rules below:

- A reuse one cluster is used in the frequency domain. In terms of an interference-free DCA strategy a timeslot-to-cell assignment is performed, resulting in a time slot clustering. A reuse one cluster in frequency domain does not need frequency planning. If there is more than one carrier available for a single operator also other frequency reuse patters >1 are possible.

- Any specific time slot within the TDD frame is available either for uplink or downlink transmission. UL/DL resources allocation is thus able to adapt itself to time varying asymmetric traffic.
- In order to accommodate the traffic load in the various cells the assignment of the timeslots (both UL and DL) to the cells is dynamically (on a coarse time scale) rearranged (slow DCA) taking into account that strongly interfering cells use different timeslots. Thus resources allocated to adjacent cells may also overlap depending on the interference situation.

Due to idle periods between successive received and transmitted bursts, mobiles can provide the network with interference measurements in time slots different from the currently used one. The availability of such information enables the operator to implement the DCA algorithm suited to the network.

For instance, the prioritised assignment of time slots based on interference measurements results in a clustering in the time domain and in parallel takes into account the demands on locally different traffic loads within the network.

6.5.2.2 Resource allocation to bearer services (fast DCA)

Fast channel allocation refers to the allocation of one or multiple physical channels to any bearer service Resource units (RUs) are acquired (and released) according to a cell-related preference list derived from the slow DCA scheme.

The following principles hold for fast channel allocation:

1. The basic RU used for channel allocation is one code / time slot / (frequency).
2. Multi-rate services are achieved by pooling of resource units. This can be made both in the code domain (pooling of multiple codes within one time slot = **multi-code** operation) and time domain (pooling of multiple time slots within one frame = **multi-slot** operation). Additionally, any combination of both is possible.
3. Since the maximal number of codes per time slot in UL/DL depends on several physical circumstances like, channel characteristics, environments, etc. (see description of physical layer) and whether additional techniques to further enhance capacity are applied (for example smart antennas). The DCA algorithm has to be independent of this number. Additionally, time hopping can be used to average inter-cell interference in case of low-medium bit rate users.
4. Channel allocation differentiates between RT and NRT bearer services:
    - RT services: Channels remain allocated for the whole duration the bearer service is established. The allocated resources may change because of a channel reallocation procedure (e.g. VBR).
    - NRT services: Channels are allocated for the period of the transmission of a dedicated data packet only UDD channel allocation is performed using 'best effort strategy', i.e. resources available for NRT services are distributed to all admitted NRT services with pending transmission requests. The number of channels allocated for any NRT service is variable and depends at least on the number of current available resources and the number of NRT services attempting for packet transmission simultaneously. Additionally, prioritisation of admitted NRT services is possible.
5. Channel reallocation procedures (intra-cell handover) can be triggered for many reasons:
    - To cope with varying interference conditions.
    - In case of high rate RT services (i.e. services requiring multiple resource units) a 'channel reshuffling procedure' is required to prevent a fragmentation of the allocated codes over to many timeslots. This is achieved by freeing the least loaded timeslots (timeslots with minimum used codes) by performing a channel reallocation procedure.
    - When using smart antennas, channel reallocation is useful to keep spatially separated the different users in the same timeslot.

6.5.3 Power Control

Power control is applied for UTRA/TDD to limit the interference level within the system thus reducing the inter-cell interference level and to reduce the power consumption in the MS.

As mandatory power control scheme, a slow C-level based power control scheme (similar to GSM) is used both for up- and downlink. Power control is made individually for each resource unit (code) with the following characteristics:

Table 14. PC characteristics

	Uplink	**Downlink**
Dynamic range	80 dB	30 dB
Power control rate	variable; 1-800 cycles / second	variable; 1-800 cycles / second
Step size	1.5 … 3 dB	1.5 … 3 dB
Remarks	A cycle rate of 100 means that every frame the power level is controlled	within one timeslot the powers of all active codes are balanced to be within a range of 20 dB

- All codes within one timeslot allocated to the same bearer service use the same transmission power.
- For RT services, in UL and DL a closed loop power control is used
- For NRT services, both open loop power control and closed loop power control are used according to the MS state and the operators' needs
- The initial power value is based on the path-loss estimate to the serving BS
- In case of one user with simultaneous RT and NRT bearer service, the closed loop power control is used both for RT and NRT bearer service. However, depending on the current services different power levels are used.

Optional enhancements concerning power control for further study:

- Introduction of quality based power control

6.5.4 Cell Search

"Cell Search" is the procedure activated by the MS to find out suitable BS, which it can synchronise to.

Depending on the MS state, the cell search procedure can be performed in one of the following ways:

- Initial Mode Cell Search
- Idle Mode Cell Search
- Active Mode Cell Search

6.5.4.1 Initial Mode Cell Search

The Initial Mode Cell Search procedure is activated by the MS at the power on.

As soon as the MS has been powered on, it tries to find suitable BS to synchronise to. With "Suitable" means a BS broadcasting the identities of the system/network the MS has access rights to and whose reference power level is detected with the lowest path loss.

From the selected BS, the MS shall derive all the TDD-TDMA timings (i.e. chip, slot, frame configuration, multi-frame, super-frame synchronism), the frequency synchronisation and all the system information, which are required to access to the network services.

During the first step of the procedure, the MS scans, over an open time window of 10 ms, for the synchronisation pattern that a BS transmits on the Synchronisation Channel (SCH).

From the detection of the auto-correlation pulse, frequency; chip; time slot; frame and multi-frame synchronism and path loss measurement can be derived.

As a result of this first step, the MS has registered the TDMA timings from the strongest base station.

In a following step, the MS tries to detect from the Broadcast Control Channel (BCCH) of the locked BS all the other information (e.g. the switching point synchronism; the system identities; the RACH position etc.) which allows accessing the network services.

For this second step, the closed time window (i.e. a time window centred around the midamble field of the time slot) is used.

6.5.4.2 Idle Mode Cell Search

The Idle Mode Cell Search procedure is activated by the MS when it is already synchronised to a BS but has no physical channel allocated, that is when the MS is in the “Idle” state.

This procedure is activated when the locked BS is detected under a predetermined power threshold, which could also depend on the link quality.

In the Idle state, the MS has all the TDD-TDMA timings from the locked BS, but still monitors the radio environment in order to find out stronger BS.

In order to save power consumption, the MS can perform the radio monitoring periodically.

By receiving the SCH, BCCH of other BS, the MS learns the information for a possible new cell selection. Furthermore, the MS monitors all other time slots for interference measurements that are utilised by the BS later when the MS tries to get into the cell via the RACH mechanism.

When stronger BS is detected, the MS can lock to this new cell after checking access rights and aligning its TDD-TDMA timings (by the detection of the BCCH and SCH).

6.5.4.3 Active Mode Cell Search

The Active Mode Cell Search procedure is activated by the MS when it is already synchronised to a BS with which at least one physical channel is allocated, that is when the MS is in the “Active” State.

In the Active state, the MS periodically scans for the radio environment in order to keep updated the list of the strongest cells, in respect to the serving one, it can detects. The list, which contains the identity and the Received Signal Strength Intensity (RSSI) of each detected cell, is periodically forward to the serving BS (or it can be forwarded on demand) and can be used to perform inter-cell handovers of the allocated physical channels.

By receiving the SCH, BCCH of other BS, the MS learns about its radio environment. Furthermore, the MS monitors all time slots that are not occupied by the MS for interference measurements being utilised by the BS.

6.5.5 Random Access

The MS that needs to access the network services or needs more capacity shall transmit, to the selected BS, a random access burst on the Random Access Channel (RACH).

The RACH can be positioned in one or more time slots of the uplink part of the frame, as indicated by the Broadcast Control Channel (BCCH). The random access burst can be accommodated either in the first half or in the second half of the assigned time slot(s), so that the time slot capacity is doubled. A further improvement of the capacity and, as a consequence, a further reduction of collisions is achieved by allowing up to eight orthogonal codes per random access time slot.

The network can regulate the RACH use by allowing separate access groups of MS at a time.

Upon reception of a random access burst, the selected BS shall answer to the MS by sending an access grant message on the Forward Access Channel (FACH).

This message shall indicate the physical channels/time slots within the cell, which are assigned to the MS.

6.6 Additional features and options (TDD)

6.6.1 Joint detection

Joint detection of simultaneously active CDMA codes in the uplink as well as the downlink will already be performed in the introductory phase of the UTRA TDD mode. Therefore, this subject is treated in other sections of this system description.

6.6.2 Adaptive antennas

In the UTRA TDD-component, adaptive antennas are supported through the use of connection dedicated midamble sequences in both uplink and downlink (they are optional in the downlink). Moreover, the reciprocity between the uplink and the downlink channel facilitates an efficient implementation of smart antennas. Although the UTRA TDD component does not require the use of smart antennas, the resulting signal-to-interference-plus-noise-ratio (SINR) can significantly be improved by incorporating various smart antenna concepts at the base station on the uplink as well as the downlink.

These SINR gains may be exploited

- to increase the capacity,

e.g., by reducing the amount of interference suffered (BS receiver) and created (BS transmitter) in the system

- to increase the quality,
- to decrease the delay spread,
- to reduce the transmission powers,
- to reduce the electromagnetic pollution and user health hazards,
- to enhance spatial user location due to the estimation of the dominant directions of arrivals,

or a combination thereof. Three different smart antenna concepts, namely

- diversity antennas,
- sector antennas,
- and adaptive antenna arrays,

can be incorporated into the UTRA TDD mode.

6.6.3 Downlink transmit diversity

Downlink transmit diversity is supported by the UTRA TDD mode.

6.6.4 Locationing function support

The fact that the base stations in a local area are synchronised facilitates the implementation of mobile positioning algorithms in the UTRA TDD mode. Time delay or delay difference measurements to the base stations are obtained in a very efficient fashion. They are required as input for mobile positioning algorithm.

6.6.4.1 Relaying and ODMA

The UTRA TDD mode is a suitable platform for the support of relaying. Relaying is a widely used technique for radio packet data transmission both in commercial and military systems but it has so far not been widely used in cellular systems. Relaying has the potential among others

- to improve coverage and/or maximum user bit rates by reduced effective path loss, optimum link adaptation and link diversity and
- to increase capacity by lowering transmission powers and associated inter-cell interference.

The UTRA TDD design is sufficiently flexible to support both simple relaying and advanced relaying protocols such as Opportunity Driven Multiple Access (ODMA) with negligible increase to the MS complexity or cost.

ODMA supports packet data transfer between an origin and destination via a network of intermediate relay nodes (dedicated fixed relays or relaying enabled mobiles). TDD operation enables each node to receive other nodes‘ transmissions and build a connectivity table neighbours at each node exploiting path loss and delay information to. This table is subsequently used to route packets across a network in a dynamic manner without incurring a significant routing overhead.

6.6.4.2 Radio-Resource Organisation and Synchronisation

ODMA relaying requires MS to MS transmission allowing information to be sent from one mobile to another without passing via BS. Each MS can receive broadcast-signalling information over a large cell area. Reception of the broadcast information will allow frequency, chip and slot/frame synchronisation and determine connectivity/path loss to the BS. The BCCH will also indicate which physical channels are available for conventional use and which channels are reserved for MS-MS transmission. The MS-MS communications may use a different unpaired frequency channel to the one generating the BCCH. In fact it may be feasible for the broadcast cell to be FDD. The BS common channels will also be used for initial authentication and mobile location. An additional advantage of receiving a BCCH is that it avoids violating any RX before TX regulations that may apply to the mobile.

Figure 66 shows an example how conventional TDD and MS-MS can be incorporated in the same frame structure. The MS-MS resources are sub-divided into Calling and Traffic Channels. The Calling Channel is RACH like i.e. random access with collision risk and the Traffic Channels are used for MS-MS transfers after negotiation on the Calling Channel(s). Traffic Channels are preferably full time slots seized exclusively for one MS-MS communication. Multi-code transmission is used to achieve high throughput if necessary to avoid excessive delay supporting the ODMA operation. Use of higher order modulation would further assist even higher rate transmission.

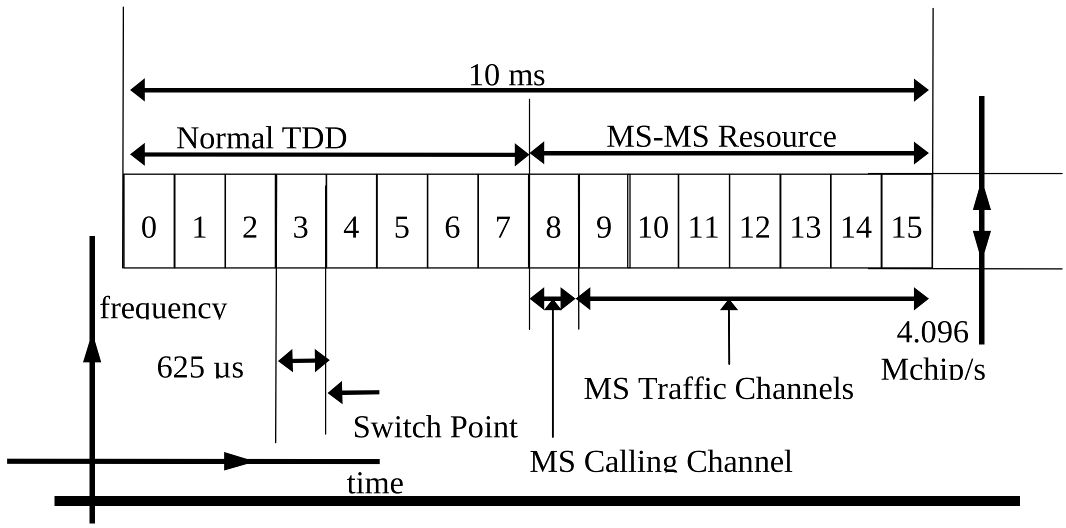


Figure 66. Example TDD Frame Structure Incorporating MS-MS Resources

An ODMA enabled MS will behave such that in MS-MS reserved slots it will be capable of receiving unless it has something to transmit and then it should be capable of transmitting in any of the MS-MS slots.

An MS can choose to just monitor the Calling Channel slot until it determines a need to also use Traffic Channels. This may be triggered by detecting its own address in a message field or by a requirement to source/sink data. To access a Calling Channel it is proposed that the MS transmits a type 1 or 2 traffic burst using a randomly chosen code. With joint detection the MS can simultaneously receive multiple signals with up to 30dB power difference and thus resolve collisions. It is assumed that MS-MS transmissions take place over micro-cellular range. Thus, the channel estimation can cope with the slight asynchronism between MS synchronised to the central BS.

The access to the traffic channels is based on a dynamic channel selection scheme based on interference measurements by the seizing MS. The seizure of channels by MS can be indicated in the calling channel burst structure as described below under addressing.

6.6.4.3 Idle-Mode Procedure

Each mobile requires some knowledge about other mobiles that it may communicate with and their relative connectivity. How it acquires this information could be implementation specific. For example an ODMA system would generate probe (RACH-like) signals to determine its neighbours and find end to end addresses. Probing is a mechanism used to indicate mobile activity in the ODMA network. When a mobile station is switched on for the first time it has no information about its surroundings. In this case the mobile will camp on one of the MS-MS CCH (after establishing synchronisation with the central BS) which are used by all mobile stations to receive and broadcast probes. With no ODMA system information stored in memory, the MS will begin a probing session, where the mobile initially camps on a CCH and periodically broadcasts a probe packet. The neighbour list will initially be empty. If another MS receives the broadcast packet it will register the probing MS as a neighbour and sends an addressed probe in response. The response probe is transmitted at random to avoid contention with other mobiles and typically one response is sent for every n broadcast probes received from a particular MS. The probe-response mechanism enables each MS to build a neighbour list that should contain at least 5 MS.

When using the probing approach initially there is no connectivity information and so the probe power must start low and ramp up until the required number of neighbouring MSs are determined. Additionally link adaptation mechanisms could be used for setting the local connectivity area to contain at least 5 other MS. Probe acknowledgements will appear on the Calling Channel (RACH-like). The acknowledgement will contain information to help refine the power control.

If the probing mechanism is allowed to occur at any time the MSs must RX continuously which may reduce battery life. To avoid this, a low duty cycle probing window, co-ordinated by BS, broadcast information can be used, i.e. the sleeping MSs wake up periodically to send and receive probes (e.g. every minute) and then go back to sleep. The window could be of the order of 0.5 seconds long. The BS has the capability to send a wakeup page to all the MSs via the BS's paging channel. A sleeping MS that is then paged awake will stay active whilst it can detect local ODMA transmissions. If it has not participated in such communication for a timeout period it will fall asleep. Similarly it may decide to sleep after a long period of activity.

An alternative approach, if feasible, would be for some central intelligence to determine where all the mobiles are located, their relative connectivity and somehow pass this information in an efficient manner to the MS.

Other MSs monitoring the probe/acks will determine connectivity between the nodes and themselves and refine their own knowledge for future communications.

6.6.4.4 Addressing

There are 2 types of addressing to be considered, Relay-Relay and End-End i.e. the former manages a particular relay hop and the latter identifies the origin and destination of the relayed transmission - within the cell. Note that all messages will have a BTS as one end of a transmission - and so a BTS should have a special generic address e.g. 0. It is assumed that each mobile has some unique end-to-end address e.g. MSISDN. The MSISDN should not be used to address MS-MS transmissions, as these fields are unencrypted (or use encryption common to the cell). When an MS registers onto the network it may be given a temporary identity (like a TMSI) which can be used for relaying purposes. For efficiency the size of this identity (or derived version for relaying) should be kept to a minimum.

The probe information mapped onto traffic burst and transmitted on a Calling Channel contains in the first half of the transmission a message independent header and enables a relay transmission to be identified. The header reveals source and destination addressing, link quality and power control parameters and which resources (Traffic Channels) to be used next. The second part of the burst is message type dependent consisting of message type, source and destination and flow information as message number, creation time, time to die and time elapsed.

6.6.4.5 Call Set-up

When a MO wishes to start a call it makes a conventional RACH access to the BS. A conventional authentication/call set-up will take place but during the negotiation of resource it will be decided to use ODMA mode. Firstly the BS will send a broadcast wakeup page to the MS relays. The BS will then ask the originating MS to send a message to it via ODMA relaying which it then acknowledges. The initial route for these messages will be based on knowledge acquired from the background probing. Alternatively, the BS could indicate the route to be used to the mobile. The transmissions will be monitored by relays not directly involved in the link. These relays then determine connectivity routes between the MO and BTS and are available to make further transmissions more optimum and reliable. Other mobiles will fall asleep using the page-awake rules. A similar procedure is used for MT calls.

6.7 System scenarios

6.7.1 Uncoordinated operation

A system requirement for uncoordinated residential operation is that systems can be bought and installed independently. The reference points for power control will be different for the different systems and their spatial separation can be arbitrarily small. Also, time synchronisation is very difficult to obtain leading to the loss of orthogonality in the code domain not only in the uplink but also on the downlink. For these reasons it is has been considered a requirement that time orthogonality is achieved between residential systems operating in close proximity.

The consequence is that contrary to public systems that are synchronised and seek to maximise the interleaving gain and hence performance and capacity, residential systems need to occupy as few slots as possible. In this way, the scope for interference avoidance increases and more systems can be accommodated.

The unsynchronised base stations, upon installation and in periodic intervals thereafter, measure interference on all slots and transmit the common control slot in the optimal position with regard to the slots in the frame used by other systems.

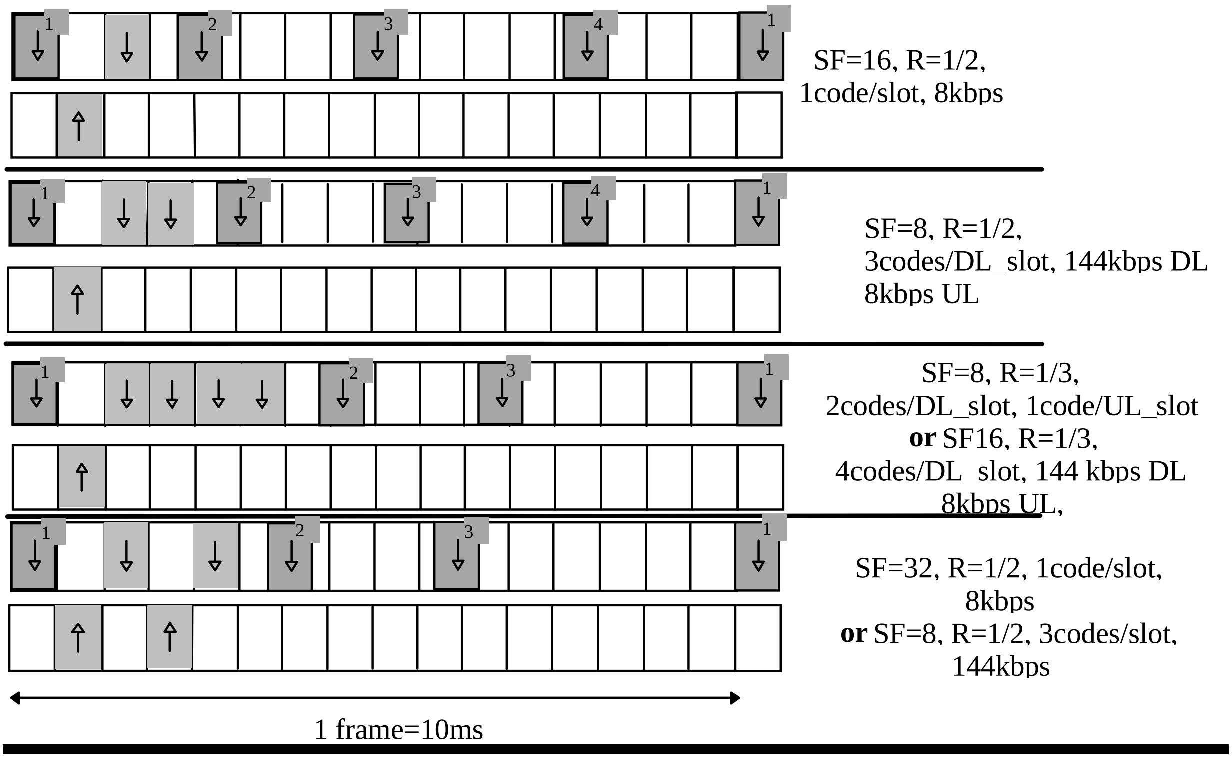


Figure 67. Frame structure and timing relation example for more than one system and several asymmetry patterns. The numbers in the grey boxes differentiate the common control channels of the different systems.

7. HANDOVER

7.1 Overview of handover types

The mobile station will support three types of handover:

- *Soft Handover*: A handover in which the mobile station communicates with a new base station without interrupting communications with the current serving base station. Soft handover can only be performed with base stations having identical frequency assignments.
- *UTRA to UTRA Hard Handover*: A handover in which the mobile is transitioned between disjoint

sets of base stations, either because those base stations are operating on a different frequency assignment or in a different mode, (UTRA/FDD to UTRA/TDD or UTRA/FDD to UTRA/FDD or UTRA/TDD to UTRA/FDD or UTRA/TDD to UTRA/TDD handover), or same frequency when soft handover is either not possible or not needed.

- *UTRA to GSM Hard Handover:* A handover in which the mobile is directed from a UTRA traffic channel to a GSM traffic channel.

The soft handover and UTRA internal hard handover are briefly outlined below.

7.2 Soft and softer handover

7.2.1 Soft handover

When in active mode, the mobile station continuously searches for new base stations on the current carrier frequency. During the search, the mobile station monitors the received signal level from neighbouring base stations, compares them to a set of thresholds, and reports them accordingly back to the base station. Based on this information the network orders the mobile station to add or remove base station links from its *active set*. The *active set* is defined as the set of base stations from which the same user information is sent, simultaneously demodulated and coherently combined, i.e. the set of base stations involved in the soft handover.

From the cell-search procedure, the mobile station knows the frame offset of the Primary CCPCH of potential soft-handover candidates relative to that of the source base station(s) (the base stations currently within the active set). When a soft handover is to take place, this offset together with the frame offset between the downlink DPCH and the Primary CCPCH of the source base station, is used to calculate the required frame offset between the downlink DPCH and the Primary CCPCH of the destination base station (the base station to be added to the active set). This offset is chosen so that the frame offset between the downlink DPCH of the source and destination base stations at the mobile-station receiver is minimised. Note that the offset between the downlink DPCH and Primary CCPCH can only be adjusted in steps of one downlink DPCH symbol in order to preserve downlink orthogonality.

7.2.2 Softer handover

Softer handover is the special case of a soft handover between sectors/cells belonging to the same base station site. Conceptually, a softer handover is initiated and executed in the same way as an ordinary soft handover. The main differences are on the implementation level within the network. For softer handover, it is e.g. more feasible to do uplink maximum-ratio combining instead of selection combining as the combining is done on the Node B level rather than on the RNC level.

7.3 UTRA to UTRA hard handover

UTRA to UTRA inter-frequency hard handover may typically occur in the following situations:

- Handover between cells to which different number of carriers have been allocated, e.g. due to different capacity requirements (hot-spot scenarios).
- Handover between cells of different overlapping orthogonal cell layers using different carrier frequencies.
- Handover between different UTRA operators/systems using different carrier frequency.

A key requirement for the support of seamless inter-frequency handover is the possibility for the mobile station to carry out cell search on a carrier frequency different from the current one, without affecting the ordinary data flow. UTRA/FDD and UTRA/TDD supports inter-frequency cell search in two different ways, a dual-receiver approach and a slotted-downlink-transmission approach (see Section Coding for slotted mode for details).

7.4 UTRA/FDD - UTRA/TDD handover

For terminals with both FDD and TDD capability the handover between the UTRA modes can be used. Both modes use the same 10 ms frame length and can perform measurements on each other. The UTRA FDD mode can use the slotted mode or other measurement ways described in Section Coding for slotted mode to perform measurements on the UTRA TDD mode. The UTRA FDD mode must search first the downlink activity part(s) in the 10 ms frame. As the UTRA TDD cells within the area are frame synchronised, the downlink/uplink timing obtained for a single TDD cell is also valid for other cells belonging to the same network in the same area.

For the UTRA TDD mode, measurement time can be obtained between the activity periods (between uplink/downlink transmission) to facilitate sufficient measurement frequency from UTRA FDD cells.

In the FDD mode, the mobile is continuously transmitting and receiving information. In order to perform a handover to the TDD mode, it should be able to make measurements on TDD carriers. However, the spectral separation between FDD carriers and TDD carriers may not be sufficient in some cases to be able to implement a filter to protect the TDD receiver making the measurements. Therefore, the mobile might need to interrupt FDD transmission in order to perform measurements in the TDD band. This can be implemented through a slotted mode in the uplink direction similar to the one defined for the downlink transmission.

For both modes it is expected that the UTRA base station is able to indicate the channel numbers used for the FDD and TDD cells in the area as well as the base station spreading/scrambling codes used. This does not cover the unlicensed TDD use where handovers are not likely to happen as the networks are not likely to be inter-connected.

7.4.1 UTRA - GSM handover

The handover between UTRA and GSM system offering world-wide coverage already today has been one of the main design criteria taken into account in the UTRA frame timing definition. The GSM compatible multi-frame structure, with the super-frame being multiple of 120 ms, allows similar timing for inter-system measurements as in the GSM system itself. The compatibility in timing is important, that when operating in UTRA mode, a multi-mode terminal is able to catch the desired information from the synchronisation bursts in the synchronisation frame on a GSM carrier with the aid of the frequency correction burst. This way the relative timing between a GSM and UTRA carriers is maintained similar to the timing between two asynchronous GSM carriers.

7.4.1.1 UTRA/FDD to GSM handover

UTRA/FDD-GSM dual mode terminals can be implemented without simultaneous use of two receiver chains. Although the frame length is different from GSM frame length, the GSM traffic channel and UTRA FDD channels use similar 120 ms multi-frame structure. Similar timing can be naturally done with UTRA TDD mode as well.

A UTRA terminal can do the measurements either by requesting the measurement intervals in a form of slotted mode where there are breaks in the downlink transmission or then it can perform the measurements independently with a suitable measurement pattern. Independent measurements do not use slotted mode, but use dual receiver approach, where the GSM receiver branch can operate independently of the UTRA FDD receiver branch.

For smooth inter-operation between the systems, information needs to be exchanged between the systems, in order to allow UTRA base station to notify the terminal of the existing GSM frequencies in the area. Further more integrated operation is needed for the actual handover where the current service is maintained, taking naturally into account the lower data rate capabilities in GSM when compared to UMTS maximum data rates reaching all the way to 2 Mbits/s.

Measurements of GSM using slotted mode

6 ms idle periods (similar to that of GSM) can be created by using double-frame idle periods, as described in Section Coding for slotted mode. Therefore, it is possible to capture the GSM FCCH and SCH in the same way as in GSM-to-GSM handover. The GSM Frequency Correction Channel (FCCH) and GSM Synchronisation Channel (SCH) use one slot out of the eight GSM slots in the indicated frames with the FCCH frame with one time slot for FCCH always preceding the SCH frame with one time slot for SCH. The principle is indicated in Figure 68.

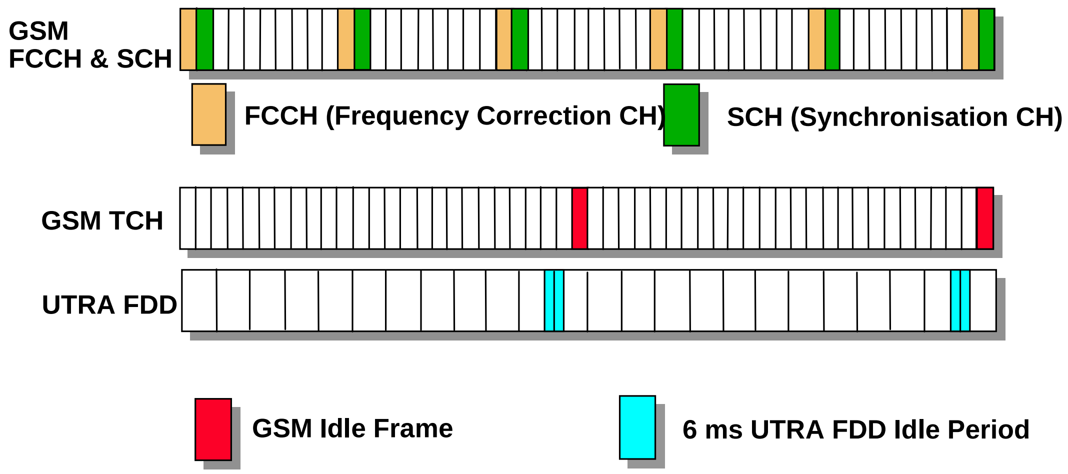


Figure 68. Example of GSM measurement timing relation between UTRA/FDD and GSM frame structures.

Alternatively, several shorter mid-frame idle periods (as described in Section Coding for slotted mode) with a certain spacing and every GSM super-frame, can be used to capture the GSM FCCH and SCH. For instance, two 3 ms idle periods every 120 ms, offset from each other by 30 ms, as illustrated in Figure 69.

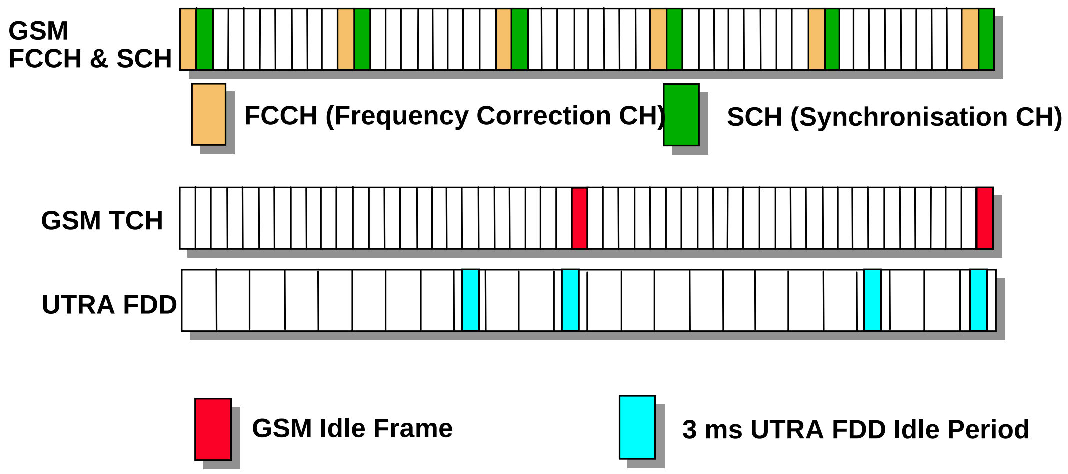


Figure 69. Another example of measurement timing relation between UTRA/FDD and GSM frame structures.

For the power measurements of GSM carriers, additional slotted frames will be used for single receiver FDD/GSM mobiles. Requirements concerning the number of power measurements per slotted frame are for further study.

), the OVSF codes should be selected from the top half of the code tree given in Section Channelisation codes.

5.3.1.2.2 Scrambling codes

Either short or long scrambling codes should be used on uplink. The short scrambling code is typically used in cells where the base station is equipped with an advanced receiver, such as a multi-user detector or interference canceller. With the short scrambling code the cross-correlation properties between different physical channels and users does not vary in time in the same way as when a long code is used. This means that the cross-correlation matrices used in the advanced receiver do not have to be updated as often as for the long scrambling code case, thereby reducing the complexity of the receiver implementation. In cells where there is no gain in implementation complexity using the short scrambling code, the long code is used instead due to its better interference averaging properties.

Both short and long scrambling codes are formed as follows:

$$Cscramb = c_1(w_0 + jc_2'w_1)$$

where w_0 and w_1 are chip rate sequences defined as repetitions of:

w_0 = {1 1symbol 125 \f "Symbol" \s 10

w_1 = {1 -1symbol 125 \f "Symbol" \s 10

and where c_1 is a real chip rate code, and c_2' is a decimated version of the real chip rate code c_2. The preferred decimation factor is 2, however other decimation factors should be possible in future evolutions of the RTT if proved desirable.

With a decimation factor N=2, c_2' is given as:

$$c_2'(2k-1) = c_2'(2k) = c_2(2k-1), \quad k=1,2,3\ldots.$$

These scrambling codes are designed such that at N-1 out of N consecutive chip times they produce +/-90° rotations of the IQ multiplexed data and control channels. At the remaining 1 out of N chip times, they produce 0, +/-90 or 180° rotations . This limits the transitions of the complex baseband signal that is inputted to the root raised cosine pulse shaping filter. This in turn reduces the peak to average ratio of the signal at the filter output, allowing a more efficient power amplifier implementation. To guarantee these desirable properties, restrictions on the choice of uplink OVSF codes are also required.

The constituent codes c_1 and c_2 are formed differently for the short and long scrambling codes as described in Sections Short scrambling code and Long scrambling code.

5.3.1.2.2.1 Short scrambling code

The short scrambling codes are formed as described in Section **Error! Not a valid bookmark self-reference.**, where c_1 and c_2 are two different codes from the extended Very Large Kasami set of length 256.

The network decides the uplink short scrambling code. The mobile station is informed about what short scrambling code to use in the downlink Access Grant message that is the base-station response to an uplink Random Access Request.

The short scrambling code may, in rare cases, be changed during the duration of a connection.

5.3.1.2.2.2 Long scrambling code

The long uplink scrambling code is typically used in cells without multi-user detection in the base station. The mobile station is informed if a long scrambling code should be used in the Access Grant Message following a Random-Access request and in the handover message.

What long scrambling code to use is directly given by the short scrambling code. No explicit allocation of the long scrambling code is thus needed.

The long scrambling codes are formed as described in Section **Error! Not a valid bookmark self-reference.**, where c_1 and c_2 are constructed as the position wise modulo 2 sum of 40960 chip segments of two binary *m*-sequences generated by means of two generator polynomials of degree 41. Let *x*, and *y* be the two *m*-sequences respectively. The *x* sequence is constructed using the primitive (over GF(2)) polynomial $1+X^3+X^{41}$. The *y* sequence is constructed using the polynomial $1+ X^{20}+X^{41}$. The resulting sequences thus constitute segments of a set of Gold sequences.

The scrambling code for the quadrature component is a 1024-chip shifted version of the in-phase scrambling code.

The uplink scrambling code word has a period of one radio frame of 10 ms.

Let $n_{40} \ldots n_0$ be the binary representation of the scrambling code number *n* (decimal) with n_0 being the least significant bit. The *x* sequence depends on the chosen scrambling code number *n* and is denoted x_n, in the sequel. Furthermore, let $x_n(i)$ and $y(i)$ denote the *i*:th symbol of the sequence x_n and *y*, respectively

The *m*-sequences x_n and y are constructed as:

Initial conditions:

$x_n(0)=n_0$, $x_n(1)= n_1$, … $=x_n(39)= n_{39}$, $x_n(40)= n_{40}$

$y(0)=y(1)=$ … $=y(39)= y(40)=1$

Recursive definition of subsequent symbols:

$x_n(i+41) =x_n(i+3) + x_n(i)$ *modulo 2*, $i=0,\ldots, 2^{41}-43$,

$y(i+41) = y(i+20)+y(i)$ *modulo 2*, $i=0,\ldots, 2^{41}-43$.

The definition of the *n*:th scrambling code word for the in phase and quadrature components follows as (the left most index correspond to the chip scrambled first in each radio frame):

$C^{I}_{long,n} = < x_n(0)+y(0), x_n(1)+y(1), \ldots, x_n(40959)+y(40959) >$,

$C^{Q}_{long,n} = < x_n(1024)+y(1024), x_n(1025)+y(1025), \ldots, x_n(41983) + y(41983) >$,

again all sums being modulo 2 additions.

Now, the complex long scrambling code $C_{long,n}$ is defined by:

$$C_{long,n} = (C^{I}_{long,n} + jC^{Q}_{long,n}) =$$

$$= < ((x_n(0)+y(0)) + j(x_n(1024)+y(1024))), \ldots,$$

$$((x_n(40959)+y(40959)) + j(x_n(41983) + y(41983))) >$$

The code generator must be able to generate the sequence shifted arbitrarily from the initial state.

5.3.1.2.3 Random access codes

5.3.1.2.3.1 Preamble spreading code

The spreading code for the preamble part is cell specific and is broadcast by the base station. More than one preamble code can be used in a base station if the traffic load is high. The preamble codes must be code planned, since two neighbouring cells should not use the same preamble code.

The code used is a real-valued 256 chip Orthogonal Gold code. All 256 codes are used in the system. The preamble codes are generated in the same way as the codes used for the downlink synchronisation channel and are defined in Section Synchronisation codes.

5.3.1.2.3.2 Preamble signature

The preamble part carries one of 16 different orthogonal complex signatures of length 16, $<P_0, P_1, ..., P_{15}>$. The signatures are based on a set of Orthogonal Gold codes of length 16 and are specified in Table 2. The base station broadcasts which signatures that are allowed in a cell.

Table 2. Preamble signatures. A = 1+j.

	Preamble symbols															
Signature	P_0	P_A	P_2	P_3	P_4	P_5	P_6	P_7	P_8	P_9	P_{10}	P_{11}	P_{12}	P_{13}	P_{14}	P_{15}
1	A	A	A	-A	-A	-A	A	-A	-A	A	A	-A	A	-A	A	A
2	-A	A	-A	-A	A	A	A	-A	A	A	A	-A	-A	A	-A	A
3	A	-A	A	A	A	-A	A	A	-A	A	A	A	-A	A	-A	A
4	-A	A	-A	A	-A	-A	-A	-A	-A	A	-A	A	-A	A	A	A
5	A	-A	-A	-A	-A	A	A	-A	-A	-A	-A	A	-A	-A	-A	A
6	-A	-A	A	-A	A	-A	A	-A	A	-A	-A	A	A	A	A	A
7	-A	A	A	A	-A	-A	A	A	A	-A	-A	-A	-A	-A	-A	A
8	A	A	-A	-A	-A	-A	-A	A	A	-A	A	A	A	A	-A	A

9	A	-A	A	-A	-A	A	-A	A	A	A	-A	-A	-A	A	A	A
10	-A	A	A	-A	A	A	-A	A	-A	-A	A	A	-A	-A	A	A
11	A	A	A	A	A	A	-A	-A	A	A	-A	A	A	-A	-A	A
12	A	A	-A	A	A	A	A	A	-A	-A	-A	-A	A	A	A	A
13	A	-A	-A	A	A	-A	-A	-A	A	-A	A	-A	-A	-A	A	A
14	-A	-A	-A	A	-A	A	A	A	A	A	A	A	A	-A	A	A
15	-A	-A	-A	-A	A	-A	-A	A	-A	A	-A	-A	A	-A	-A	A
16	-A	-A	A	A	-A	A	-A	-A	-A	-A	A	-A	A	A	-A	A

5.3.1.2.3.3 Channelisation codes for the message part

The signature in the preamble specifies one of the 16 nodes in the code-tree that corresponds to channelisation codes of length 16, as shown in Figure 43. The sub-tree below the specified node is used for spreading of the message part. The control (Q-branch) is spread with the channelisation code of spreading factor 256 in the lowest branch of the sub-tree. The data part (I-branch) can use any of the channelisation codes from spreading factor 32 to 256 in the upper-most branch of the sub-tree. However, the system may restrict the set of codes (spreading factors) actually allowed in the cell, through the use of a BCCH message.

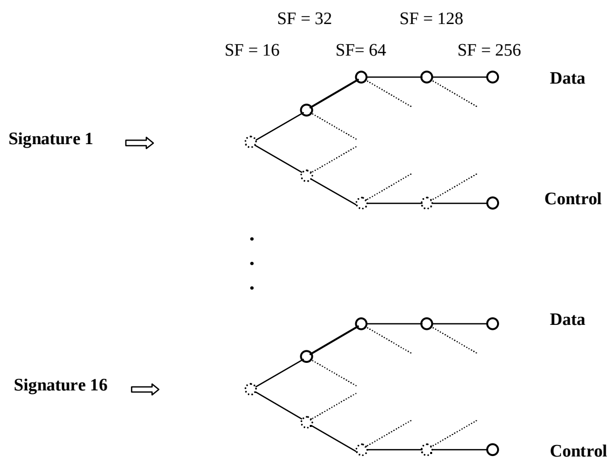


Figure 43. Channelisation codes for the random access message part.

Since the control part is always spread with a known channelisation code of length 256, the base station can detect it. The rate information field of the control part informs the base station about the spreading factor used on the data part. With knowledge of the sub-tree (obtained from the preamble signature) and the spreading factor (obtained from the rate information), the base station knows which channelisation code is used for the data part.

This structure allows for simultaneous detection of multiple random access messages arriving in the same access slot, as long as different signatures are used.

5.3.1.2.3.4 Scrambling code for the message part

In addition to spreading, the message part is also subject to scrambling with a 10 ms complex code. The scrambling code is cell-specific and has a one-to-one correspondence to the spreading code used for the preamble part. Note that although the scrambling code is the same for every access slot, there is no scrambling-code collision problem between different access slots due to the 1.25 ms time shifts between the access slots.

The scrambling codes used are from the same set of codes as is used for the other dedicated uplink channels. The first 256 codes are used for the random access channel. The generation of these codes is explained in Section Scrambling code.

5.3.1.3 Modulation

5.3.1.3.1 Modulating chip rate

The modulating chip rate is 4.096 Mcps. This basic chip rate can be extended to 8.192 or 16.384 Mcps.

5.3.1.3.2 Pulse shaping

The pulse-shaping filters are root-raised cosine (RRC) with roll-off α=0.22 in the frequency domain.

5.3.1.3.3 Modulation

QPSK modulation is used. Note that phase restrictions are introduced by the scrambling code design.

5.3.2 Downlink spreading and modulation

5.3.2.1 Spreading

Figure 44 illustrates the spreading and modulation for the downlink DPCH. Data modulation is QPSK where each pair of two bits are serial-to-parallel converted and mapped to the I and Q branch respectively. The I and Q branch are then spread to the chip rate with the same channelisation code c_{ch} (real spreading) and subsequently scrambled by the same cell specific scrambling code c_{scramb} (real scrambling).

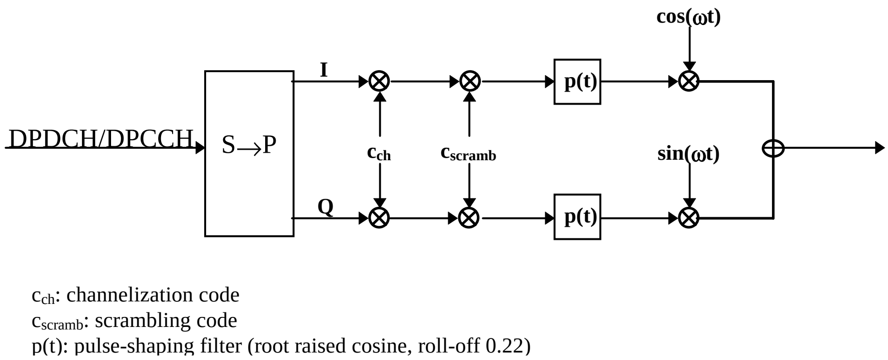


Figure 44. Spreading/modulation for downlink dedicated physical channels.

The different physical channels use different channelisation codes, while the scrambling code is the same for all physical channels in one cell.

The multiplexing of the SCH with the other downlink physical channels (DPCH and CCPCH) is illustrated in Figure 45. The figure illustrates that the SCH is only transmitted intermittently (one codeword per slot) and also that the SCH is multiplexed *after* long code scrambling of the DPCH and CCPCH. Consequently, the SCH is *non-orthogonal* to the other downlink physical channels.

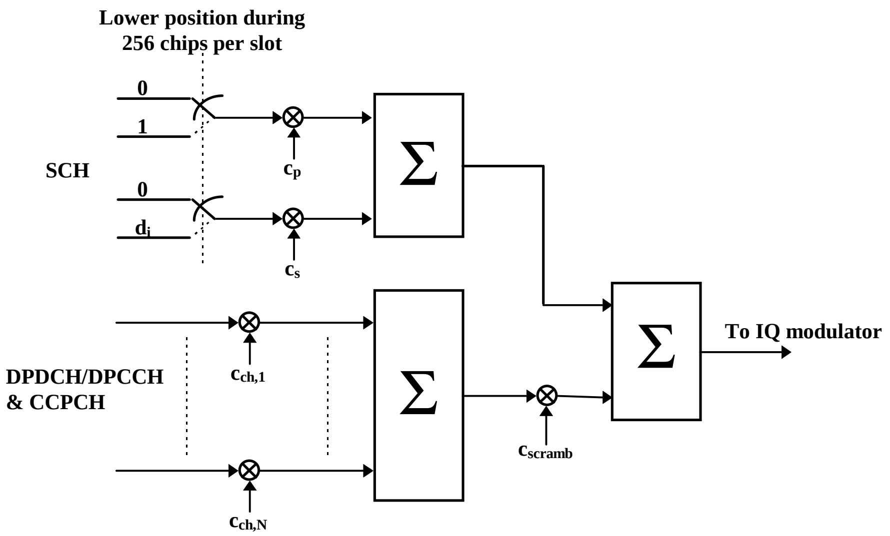


Figure 45. Multiplexing of SCH.

5.3.2.2 Code generation and allocation

5.3.2.2.1 Channelisation codes

The channelisation codes of Figure 44 are the same codes used in the uplink, namely Orthogonal Variable Spreading Factor (OVSF) codes that preserve the orthogonality between downlink channels of different rates and spreading factors. The same restriction on code allocation applies as for the uplink, but for a cell and not a mobile station as in the uplink. Hence, in the downlink a code can be used in a cell if and only if no other code on the path from the specific code to the root of the tree or in the sub-tree below the specific code is used in the same cell.

The channelisation code for the BCCH is a predefined code, which is the same for all cells within the system.

The channelisation code(s) used for the Secondary Common Control Physical Channel is broadcast on the BCCH.

The channelisation codes for the downlink dedicated physical channels are decided by the network. The mobile station is informed about what downlink channelisation codes to receive in the downlink Access Grant message that is the base-station response to an uplink Random Access request. The set of channelisation codes may be changed during the duration of a connection, typically as a result of a change of service or an inter-cell handover. A change of downlink channelisation codes is negotiated over a DCH.

5.3.2.2.2 Scrambling code

The total number of available scrambling codes is 512, divided into 32 code groups with 16 codes in each group. The grouping of the downlink codes is done in order to facilitate a fast cell search, see Section Cell search. The downlink scrambling code is assigned to the cell (sector) at the initial deployment. The mobile station learns about the downlink scrambling code during the cell search process, see Section Cell search.

The scrambling code sequences are constructed as the position wise modulo 2 sum of 40960 chip segments of two binary *m*-sequences generated by means of two generator polynomials of degree 18. Let x, and y be the two sequences respectively. The x sequence is constructed using the primitive (over GF(2)) polynomial $1+X^7+X^{18}$. The y sequence is constructed using the polynomial $1+X^5+X^7+ X^{10}+X^{18}$. The resulting sequences thus constitute segments of a set of Gold sequences.

The scrambling codes are repeated for every 10 ms radio frame.

Let $n_{17} \ldots n_0$ be the binary representation of the scrambling code number n (decimal) with n_0 being the least significant bit. The x sequence depends on the chosen scrambling code number n and is denoted x_n, in the sequel. Furthermore, let $x_n(i)$ and $y(i)$ denote the i:th symbol of the sequence x_n and y, respectively

The *m*-sequences x_n and y are constructed as:

Initial conditions:

$x_n(0)=n_0$, $x_n(1)= n_1$, $\ldots =x_n(16)= n_{16}$, $x_n(17)= n_{17}$

$y(0)=y(1)= \ldots =y(16)= y(17)=1$

Recursive definition of subsequent symbols:

$x_n(i+18) =x_n(i+7) + x_n(i)$ *modulo 2,* $i=0,\ldots,2^{18}-20$,

$y(i+18) = y(i+10)+y(i+7)+y(i+5)+y(i)$ *modulo 2,* $i=0,\ldots, 2^{18}-20$.

All sums of symbols are taken modulo 2.

The definition of the n:th scrambling code word follows as (the left most index correspond to the chip scrambled first in each radio frame):

$C_{scramb,n} = < x_n(0)+y(0), x_n(1)+y(1), \ldots,x_n(40959)+y(40959) >$,

again all symbol sums being modulo 2 additions.

The index n runs from 0 to 511 giving 512 distinct 40960 chip segments of a corresponding Gold code sequence. The leftmost chip in $C_{scramb,n}$ corresponds to the first chip in a 10 ms radio frame and the rightmost to the last.

The sign of the I- and Q-branch component is changed if and only if the corresponding chip in $C_{scramb,n}$ equals '1'.

The code generator must be able to generate the sequence shifted arbitrarily from the initial state.

5.3.2.2.3 Synchronisation codes

The Primary and Secondary code words, C_p and $\{C_1,\ldots,C_{17}\}$ respectively, consist of pair wise mutually orthogonal Gold codes of length 256. The Primary SCH is furthermore chosen to have good aperiodic auto correlation properties. The code sequences are constructed with the help of two binary *m*-sequences of length 255, *x*, and *y*, respectively. The *x* sequence is constructed using the polynomial $1+X^2+X^3+X^4+X^8$. The *y* sequence is constructed using the polynomial $1+X^3+X^5+X^6+X^8$.

Before we define the Primary and Secondary code words, we define the set of orthogonal Gold codes.

Let $n_7 \ldots n_0$ be the binary representation of the scrambling code number *n* (decimal) with n_0 being the least significant bit. The *x* sequence depends on the chosen code number *n* and is denoted x_n in the sequel. Furthermore, let $x_n(i)$ and $y(i)$ denote the *i*:th symbol of the sequence x_n and *y*, respectively

The *m*-sequences x_n and *y* are constructed as:

Initial conditions:

$x_n(0)=n_0$, $x_n(1)= n_1$, … $=x_n(6)= n_6$, $x_n(7)= n_7$

$y(0)=y(1)= \ldots =y(6)= y(7)=1$

Recursive definition of subsequent symbols:

$x_n(i+8) =x_n(i+4) + x_n(i+3) + x_n(i+2) + x_n(i)$ *modulo 2*, $i=0,\ldots, 246$,

$y(i+8) = y(i+6)+ y(i+5)+ y(i+3)+y(i)$ *modulo 2*, $i=0,\ldots, 246$.

The definition of the *n*:th SCH code word follows (the left most index correspond to the chip transmitted first in each slot):

$C_{SCH,n} = < 0, x_n(0)+y(0), x_n(1)+y(1), \ldots,x_n(254)+y(254) >$,

All sums of symbols are taken modulo 2.

Note that the code words always start with a constant '0'. symbol.

Before modulation and transmission these binary code words are converted to real valued sequences by the transformation '0' -> '+1', '1' -> '-1'.

The Primary and Secondary code words are defined in terms of $C_{SCH,n}$ and the definition of C_p and $\{C_1,\ldots,C_{17}\}$ now follows as:

$C_p= C_{SCH, 0}$

and

$C_i = C_{SCH, i}$, $i=1,\ldots,17$

5.3.2.3 Modulation

5.3.2.3.1 Modulating chip rate

The modulating chip rate is 4.096 Mcps. This basic chip rate can be extended to 8.192 or 16.384 Mcps.

5.3.2.3.2 Pulse shaping

The pulse-shaping filters are root raised cosine (RRC) with roll-off α=0.22 in the frequency domain.

5.3.2.3.3 Modulation

QPSK modulation is used.

5.4 Radio transmission and reception (FDD)

5.4.1 General

The information presented in this section is based on a chip rate of 4.096 Mcps. Appropriate adjustments should be made for higher chip rate options.

5.4.2 Frequency bands and channel arrangement

5.4.2.1 Proposed frequency bands for operation

UTRA/FDD is designed to operate in the following paired band:

Table 3. Proposed frequency band for UTRA/FDD

1920 – 1980 MHz Mobile station transmit Base station receive	2110 – 2170 MHz Mobile station receive Base station transmit

Deployment in other frequency bands is not precluded.

5.4.2.2 Carrier spacing

The nominal channel spacing is 5 MHz, but this can be adjusted to optimise performance in particular deployment scenarios. The channel raster is 200 kHz, which means that the carrier frequency must be a multiple of 200 kHz.

5.4.2.3 TX – RX frequency separation

The minimum transmit to receive separation is 130 MHz when operating in the paired band defined in Table 3. If used in other frequency bands like the American PCS band the minimum separation would be 80 MHz.

5.4.2.4 Variable duplex distance

UTRA/FDD should support a variable duplex distance, i.e. $D_{duplexer} = F_{down} – F_{up}$ is not necessary a constant but is, in general, allowed to vary within certain limits. The specific limits for the duplex distance applicable for different frequency bands and terminal classes are yet to be determined.

5.4.3 Service classes

5.4.3.1 Terminal service classes

A number of different service classes will be used to define the data rate and code allocation for a UTRA/FDD terminal. Possible types of service class profiles are 144 kbps, 384 kbps and 2048 kbps.

5.4.4 Transmitter characteristics

The output power is given in terms of power level at the antenna connector of the equipment. For equipment with integral antenna only, a reference antenna with a gain of 0 dBi is assumed.

5.4.4.1 Mobile station output power

The mobile station output power profile would be used to define a range of terminal output powers for use in different system scenarios. The power class would be based on the mobile station's peak power for example 30 dBm. For mobile station using directive antennas for transmission, a class dependent limit will be placed on the maximum EIRP (Equivalent Isotropic Radiated Power).

5.4.4.2 Base station output power

The base station output power profile would be used to cater for different system scenarios. The power class would be based on the peak power specified for the base stations.

5.4.4.3 Output power dynamics

The transmitter uses fast closed-loop Carrier/Interference based power control and slow quality-based power control on both the uplink and downlink.

Table 4. Output power dynamics for UL and DL

	Uplink (UL)	Downlink (DL)
Power control steps	Variable 0.25-1.5 dB	Variable 0.25-1.5 dB
Minimum transmit power	-50 dBm	[ ] dBm
Power control cycles per second	1.6 kHz	1.6 kHz
Power control dynamic	80 dB	30 dB

5.4.4.4 Output RF spectrum emissions

5.4.4.4.1 Out of band emissions

The assumed spectrum mask has been derived from simulations on a real wide band amplifier as shown in Figure 46 below. These emission levels will be dependent on the power class and code allocation of the mobile and base station.

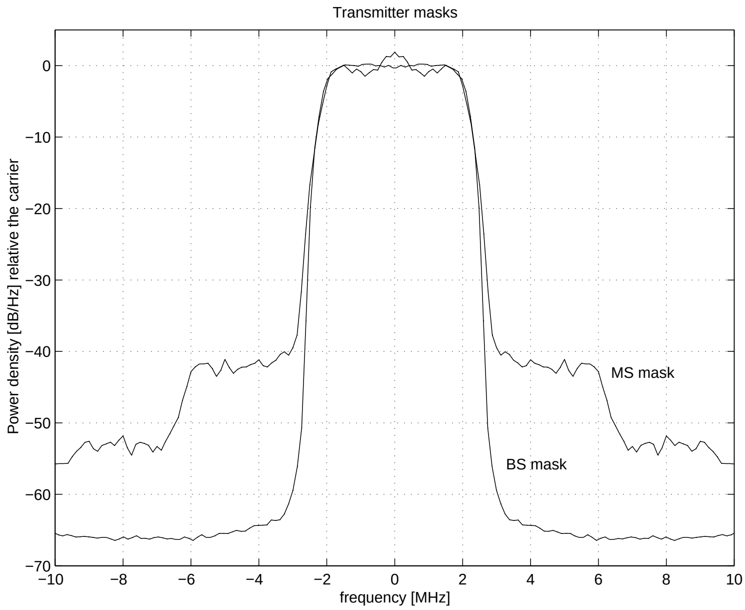


Figure 46. Assumed spectrum masks.

5.4.4.4.2 Spurious emissions

The limits for spurious emissions at frequencies greater than ± 250% of the necessary bandwidth would be based on the applicable tables from ITU-R Recommendation SM.329. Further guidance would be taken from the ERC recommendation that is currently under progress.

5.4.4.5 Adjacent channel protection (ACP)

Adjacent channel protection (ACP) is the ratio of the transmitted power and the power measured after a receiver filter in the adjacent channel.

The ACP envisaged for 5 MHz channel spacing is in the order of 35 dB to 40 dB. The possibility is being considered of dynamically relaxing the ACP requirements for mobile stations under conditions when this would not lead to significant interference (with respect to other systems or UMTS operators). This would be carried out under network control, primarily to facilitate reduction in MS power consumption.

5.4.4.6 Occupied bandwidth

The channel bandwidth is less than 5 MHz based on a chip rate of 4.096 Mcps.

5.4.4.7 Frequency stability

The frequency stability for the mobile and base station is indicated in Table 5.

Table 5. Mobile and base station frequency stability.

Mobile station	Base station
3 PPM (unlocked), 0.1 PPM (locked)	0.05 PPM

5.4.5 Receiver characteristics

A Rake receiver or any other suitable receiver structure using coherent reception in both channel impulse response estimation, and code tracking procedures is assumed.

5.4.5.1 Diversity characteristics

Three forms of diversity are available in UTRA / FDD:

Table 6. Diversity characteristics for UTRA/FDD.

Time diversity	Channel coding and interleaving in both uplink and down link.
Multi-path diversity	Rake receiver or other suitable receiver structure with maximum combining. Additional processing elements can increase the delay-spread performance due to increased capture of signal energy.
Space diversity	Antenna diversity with maximum ratio combing in the base station and optionally in the mobile stations. Possibility for downlink transmit diversity in the base station.

5.4.5.2 Reference sensitivity level

The reference sensitivity for the following services; 8 kbps, 144 kbps, 384 kbps and 2048 kbps are specified in the link budget template for a number of test environments and multi-path channel classes.

5.4.5.3 BER noise floor level

The BER noise floor level for voice services is significantly less than 10^{-3} BER. The BER noise floor level for data services is significantly less than 10^{-6} BER.

5.4.5.4 Maximum tolerable delay spread

To maintain the voice and data service quality requirements the UTRA/FDD concept allows for a time dispersion spread suitable for the various propagation models specified in UMTS 30.03 (which contains the models defined in ITU-R recommendation M.1225).

5.4.5.5 Maximum tolerable Doppler spread

The maximum tolerable Doppler spread is 1000 Hz, which at a 2 GHz carrier frequency corresponds to a maximum velocity of about 500 km/hr. Parameters determining system performance are not necessarily optimised for this value of Doppler spread.

5.5 Physical layer procedures (FDD)

5.5.1 Power control

5.5.1.1 Uplink power control

5.5.1.1.1 Closed loop power control

The uplink closed loop power control adjusts the mobile station transmit power in order to keep the received uplink Signal-to-Interference Ratio (SIR) at a given SIR target.

The base station should estimate the received uplink DPCCH power after RAKE combining of the connection to be power controlled. Simultaneously, the base station should estimate the total uplink received interference in the current frequency band and generate a SIR estimate SIR_{est}. The base station then generates TPC commands according to the following rule:

$SIR_{est} > SIR_{target,UL} \rightarrow$ TPC command = “down”
$SIR_{est} < SIR_{target,UL} \rightarrow$ TPC command = “up”

Upon the reception of a TPC command, the mobile station should adjust the transmit power of both the uplink DPCCH and the uplink DPDCH in the given direction with a step of Δ_{TPC} dB. The step size Δ_{TPC} is a parameter that may differ between different cells, in the region 0.25 – 1.5 dB.

In case of receiver diversity (e.g., space diversity) or softer handover at the base station, the TPC command should be generated after diversity combining.

In case of soft handover, the following procedure is considered:

- in the base stations a quality measurement is performed on the received signals; in case the quality measurement indicates a value below a given threshold, an increase command is sent to the mobile, otherwise a decrease command is transmitted; all the base stations in the active set send power control commands to the mobile;
- the mobile compares the commands received from different base stations and increases its power only if all the commands indicate an increase value (this means that all the receivers are below the threshold); in case one command indicates a decrease step (that is, at least one receiver is operating in good conditions), the mobile reduces its power; in case more than one decrease commands are received by the mobile, the mobile station should adjust the power with the largest step in the "down" direction ordered by the TPC commands received from each base station in the active set.
- the quality threshold for the base stations in the active set should be adjusted by the outer loop power control (to be implemented in the network node were soft handover combining is performed).

5.5.1.1.2 Outer loop (SIR target adjustment)

The outer loop adjusts the SIR target used by the closed-loop power control. The SIR target is independently adjusted for each connection based on the estimated quality of the connection. In addition, the power offset between the uplink DPDCH and uplink DPCCH may be adjusted. How the quality estimate is derived and how it affects the SIR target is decided by the radio-resource management, i.e. it is not a physical-layer issue.

5.5.1.1.3 Open-loop power control

Open-loop power control is used to adjust the transmit power of the physical Random-Access channel. Before the transmission of a Random-Access burst, the mobile station should measure the received power of the downlink Primary CCPCH over a sufficiently long time to remove effects of the non-reciprocal multi-path fading. From the power estimate and knowledge of the Primary CCPCH transmit power (broadcast on the BCCH) the downlink path-loss including shadow fading can be found. From this path loss estimate and knowledge of the uplink interference level and the required received SIR, the transmit power of the physical Random-Access channel can be determined. The uplink interference level as well as the required received SIR are broadcast on the BCCH.

5.5.1.2 Downlink power control

5.5.1.2.1 Closed loop power control

The downlink closed loop power control adjusts the base station transmit power in order to keep the received downlink SIR at a given SIR target.

The mobile station should estimate the received downlink DPCH power after RAKE combining of the connection to be power controlled. Simultaneously, the mobile station should estimate the total downlink received interference in the current frequency band. The mobile station then generates TPC commands according to the following rule:

$SIR_{est} > SIR_{target,DL} \rightarrow$ TPC command = "down"
$SIR_{est} < SIR_{target,DL} \rightarrow$ TPC command = "up"

Upon the reception of a TPC command, the base station should adjust the transmit power in the given direction with a step of Δ_{TPC} dB. The step size Δ_{TPC} is a parameter that may differ between different cells, in the range 0.25 – 1.5 dB.

In case of receiver diversity (e.g., space diversity) at the mobile station, the TPC command should be generated after diversity combining.

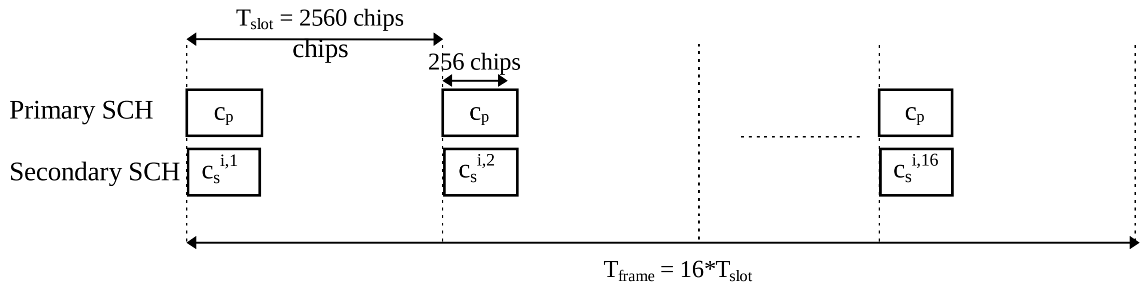
5.5.1.2.2 Outer loop (SIR target adjustment)

The outer loop adjusts the SIR target used by the closed-loop power control. The SIR target is independently adjusted for each connection based on the estimated quality of the connection. In addition, the power offset between the downlink DPDCH and DPCCH may be adjusted. How the quality estimate is derived and how it affects the SIR target is decided by the radio-resource management, i.e. it is not a physical-layer issue.

5.5.2 Cell search

5.5.2.1 Initial cell search

During the initial cell search, the mobile station searches for the base station to which it has the lowest path loss. It then determines the downlink scrambling code and frame synchronisation of that base station. The initial cell search uses the synchronisation channel (SCH), shown in Figure 47 below (repeated from Section Synchronisation Channel).



c_p : Primary Synchronization Code
$c_s^{i,k}$: One of 17 possible Secondary Synchronization Codes
($c_s^{i,1}$, $c_s^{i,2}$, ..., $c_s^{i,16}$) encode cell specific long scrambling code group i

Figure 47. Structure of synchronisation channel (SCH).

This initial cell search is carried out in three steps:

Step 1: Slot synchronisation

During the first step of the initial cell search procedure the mobile station uses the primary SCH to acquire slot synchronisation to the strongest base station. This is done with a single matched filter (or any similar device) matched to the primary synchronisation code c_p which is common to all base stations. The output of the matched filter will have peaks for each ray of each base station within range of the mobile station, see Figure 48. Detecting the position of the strongest peak gives the timing of the strongest base station modulo the slot length. For better reliability, the matched-filter output should be non-coherently accumulated over a number of slots.

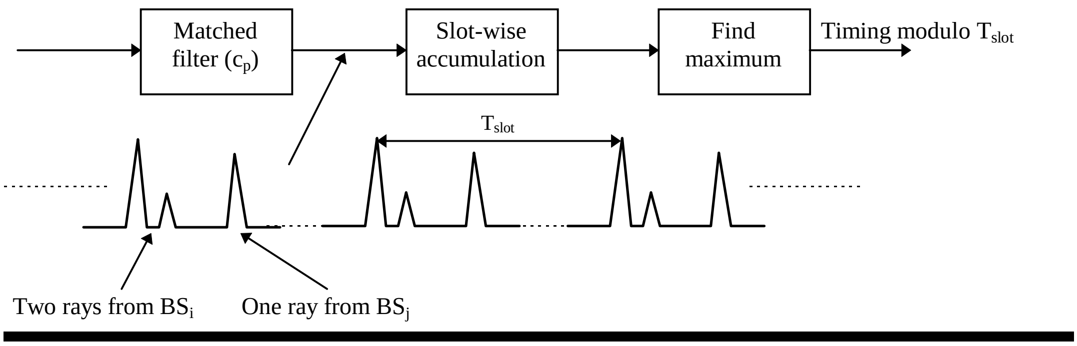


Figure 48. Matched-filter search for primary synchronisation code to slot synchronisation (timing modulo the slot length).

Step 2: Frame synchronisation and code-group identification

During the second step of the initial cell search procedure, the mobile station uses the secondary SCH to find frame synchronisation and identify the code group of the base station found in the first step. This is done by correlating the received signal at the positions of the Secondary Synchronisation Code with all possible (16) Secondary Synchronisation Codes. Note that the position of the Secondary Synchronisation Code is known after the first step. The outputs of all the 17 correlators for 16 consecutive secondary SCH locations are used to form the decision variables. The decision variables are obtained by *non-coherently* summing the correlator outputs corresponding to each 16 length sequence out of the 32 possible sequences and its 16 cyclic shifts giving a total of 512 decision variables. Note that the cyclic shifts of the sequences are unique (see Section Synchronisation Channel). Thus, by identifying the sequence/shift pair that gives the maximum correlation value, the code group as well as the frame synchronisation is determined.

Step 3: Scrambling-code identification

During the third and last step of the initial cell-search procedure, the mobile station determines the exact scrambling code used by the found base station. The scrambling code is identified through symbol-by-symbol correlation over the Primary CCPCH with all scrambling codes within the code group identified in the second step. Note that, from step 2, the frame boundary and consequently the start of the scrambling code is known. Correlation must be carried out symbol-wise, due to the unknown data of the primary CCPCH. Also, in order to reduce the probability of wrong/false acquisition, due to combat background noise/interference, averaging the correlator outputs over a sequence of symbols (diversity) might be required before using the outputs to determine the exact scrambling code.

After the scrambling code has been identified, the Primary CCPCH can be detected, super-frame synchronisation can be acquired and the system- and cell specific BCCH information can be read.

5.5.2.2 Idle mode cell search

When in idle mode, the mobile station continuously searches for new base stations on the current and other carrier frequencies. The cell search is done in basically the same way as the initial cell search. The main difference compared to the initial cell search is that an idle mobile station has received a priority list from the network. This priority list describes in which order the downlink scrambling codes should be searched for and does thus significantly reduce the time and effort needed for the scrambling-code search (step 3). Also the complexity in the second step may be reduced if the priority list only includes scrambling codes belonging to a subset of the total set of code groups. The priority list is continuously updated to reflect the changing neighbourhood of a moving mobile station.

5.5.2.3 Active mode cell search

When in active mode, the mobile station continuously searches for new base stations on the current carrier frequency. This cell search is carried out in basically the same way as the idle mode cell search. The mobile station may also search for new base stations on other carrier frequencies using the slotted mode, see Section Coding for slotted mode.

5.5.3 Random access

The procedure of a random access request is:

1. The mobile station acquires synchronisation to a base station
2. The mobile station reads the BCCH to get information about:
    - 2.1 The preamble spreading code(s) /message scrambling code(s) used in the cell
    - 2.2 The available signatures
    - 2.3 The available access slots
    - 2.4 The available spreading factors for the message part
    - 2.5 The interference level at the base station
    - 2.6 The primary CCPCH transmit power level
3. The mobile station selects a preamble spreading code/message scrambling code
4. The mobile station selects a spreading factor for the message part.
5. The mobile station estimates the downlink path loss (by using information about the transmitted and received power level of the primary CCPCH), and determines the required uplink transmit power (by using information about the interference level at the base station).
6. The mobile station randomly selects an access slot and signature from the available access slots and signatures.

7. The mobile station transmits its random access burst.

8. The mobile station waits for an acknowledgement from the base station. If no acknowledgement is received within a predefined time-out period, the mobile station starts again from step 5.

A typical implementation of the base-station random-access receiver for a given preamble code and preamble sequence is illustrated in Figure 49. The received signal is fed to a matched filter, matched to the preamble code. The output of the matched filter is then correlated with the preamble sequence. The output of the preamble correlator will have peaks corresponding to the timing of any received Random-Access burst using the specific preamble code and preamble sequence. The estimated timing can then be used in an ordinary RAKE combiner for the reception of the data part of the Random-Access burst.

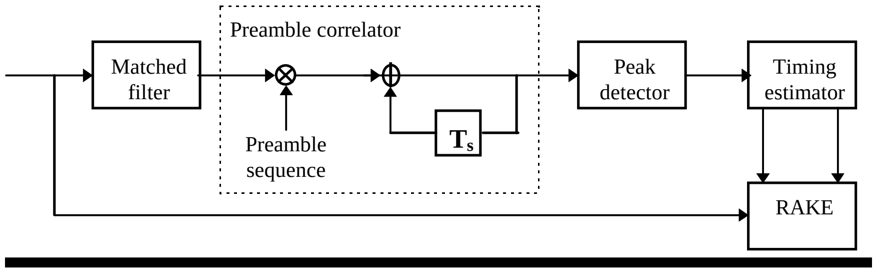


Figure 49. Base-station Random-Access receiver.

Upon reception of the Random-Access burst, the base station responds with an Access Grant message on the FACH. In case the Random Access request is for a dedicated channel (circuit-switched or packet) and the request is granted, the Access Grant message includes a pointer to the dedicated physical channel(s) to use. As soon as the mobile station has moved to the dedicated channel, closed-loop power control is activated.

5.5.4 Idle mode tasks

5.5.4.1 Paging control

5.5.4.1.1 Base Station operation

Every mobile station belongs to one group. When a paging message should be sent to a mobile, the paging message is transmitted on the PCH in the MUI-parts belonging to the terminating mobile's group. The paging message includes the mobile station identification number of the mobile station for which the paging message was intended. When a MUI is transmitted, the corresponding PI1 and PI2 fields are also transmitted.

The exact behaviour of the base station is described as:

For the PCH of the group which does not have terminating information:

- The BS shall transmit the two PI parts (PI1 and PI2) in the PCH as “all 0”.
- The MUI part shall not be transmitted.

For the PCH of the group which have terminating information:

- The BS shall transmit the two PI parts (PI1 and PI2) in the PCH as “all 1”.
- The MUI part shall be transmitted within the same PCH.

5.5.4.1.2 Mobile Station operation

The idea behind the detection of paging messages is to open the receiver to detect one of or both the paging indicators (PI1 and PI2), and if they indicate a paging message for the group the mobile belongs to, the actual paging information part (MUI) is received. When the MUI part is received, the existence of a paging message for the mobile is determined from the information included in the MUI part.

The mobile station operation for detection of paging information in group *n* is shown in Figure 50. $PI1_n$, $PI2_n$, and MUI_n are the PCH components belong to group *n*.

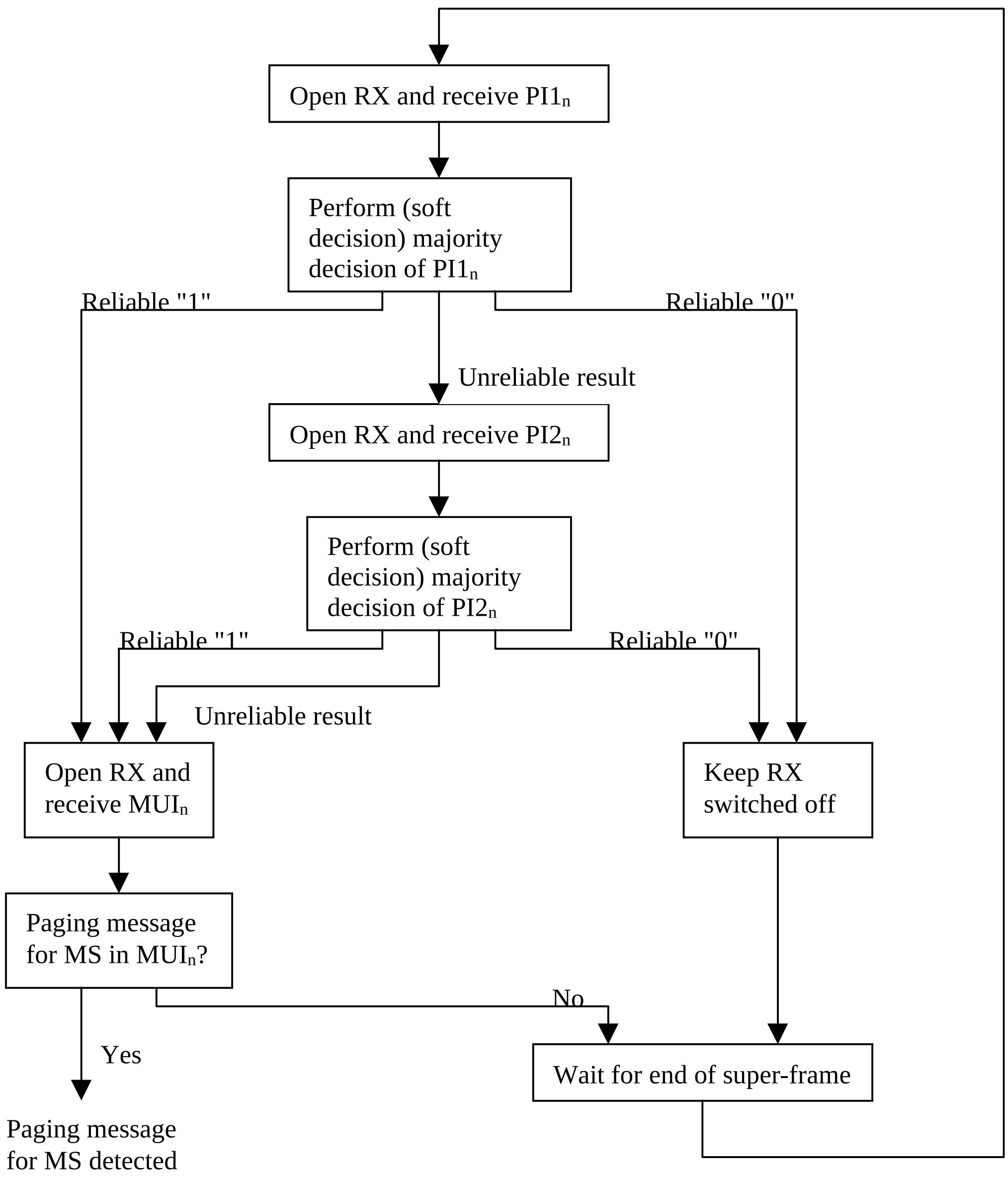


Figure 50. Detection of paging messages.

5.6 Additional features and options (FDD)

5.6.1 Adaptive antennas

Adaptive antennas are recognised as a way to enhance capacity and coverage of the system. Solutions employing adaptive antennas are already supported in the UTRA/FDD concept through the use of connection-dedicated pilot bits on both uplink and downlink.

5.6.2 Multi-user detection

UTRA/FDD is designed to work without requiring joint detection of multiple user signals. However, the potential capacity gains of such receivers in a UTRA/FDD system have been recognised and taken into account in the design of the concept. In the uplink the possibility to use only short codes facilitates more advanced receiver structures with reasonable complexity.

5.6.3 Downlink transmit diversity

Transmitter diversity in the downlink provides a means to significantly improve capacity and coverage of

UTRA/FDD, without the requirement for a second receiver chain in the mobile station that receiver diversity would entail. However, a typical transmit diversity technique, such as delay transmit diversity, has two main drawbacks: self-interference at locations with good SINR; and the requirement for additional Rake fingers in the mobile receiver. In order to overcome these drawbacks, diversity schemes have been proposed for UTRA/FDD, that maintain the orthogonality between diverse downlink transmit antennas, whilst offering significant advantages in the downlink performance. Simulation results for the proposed techniques have shown a gain of up to 7 dB (compared with the non-diversity case) for slow speed mobiles in a single path fading environment. In the proposed schemes, the orthogonality between antennas, is maintained using either code, or time division.

5.6.3.1 Code division transmit diversity

5.6.3.1.1 Orthogonal Transmit Diversity

Orthogonal Transmit Diversity (OTD) utilises code division transmission diversity. The implementation of OTD is as follows. Coded bits are split into two data streams and transmitted via two separate antennas. Different orthogonal channelisation codes are used per antenna for spreading. This maintains the orthogonality between the two output streams, and hence self-interference is eliminated in flat fading. Note that by splitting the coded data into two separate data streams, the effective number of channelisation codes per user is the same as the case without OTD.

The above structure is highly flexible, it may be easily extended to more antennas (4, 8, etc.)

OTD may be an optional feature that can be turned on only if needed. In addition, it is possible to support a mixture of mobiles with and without OTD capability.

The additional required processing at the mobile station is small. Figure 51 illustrates Rake finger processing with OTD. It is important to note that the Pilot signal is also split and transmitted on both antennas, which allows coherent detection of the signals received from both antennas. The data is processed using a Rake finger with parallel processing capability. Both transmitted signal streams are received simultaneously at the same delay (for a given multipath ray), hence no additional buffering and skewing of data is necessary. This significantly reduces the hardware complexity/cost associated with OTD implementation.

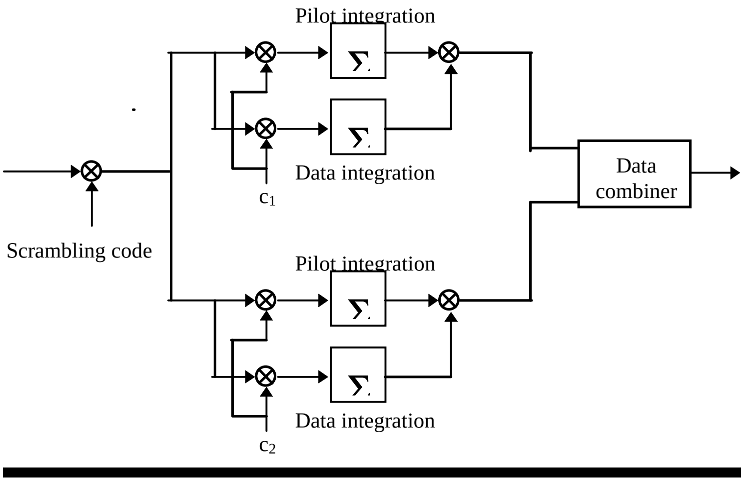


Figure 51. Rake finger processing with OTD.

In the base station transmitter, the base-band processing (i.e. data splitting and separate spreaders) required for OTD already exists with multi-code transmission in the downlink. From the OTD viewpoint, it is advantageous to employ multi-code transmission for all data rates, and it is also recommended to match the number of codes assigned to the user with the number of transmit antennas.

5.6.3.2 Time division transmit diversity

Two schemes have been put forward utilising time division transmission diversity for downlink UTRA/FDD mode operation. The basic Base Station Transmitter block diagram for Time Transmission Diversity is shown in Figure 52. In time division transmission diversity the signal is switched between antennas in one of two ways. Either, the signal is switched according to a pattern decided by the base station, or it is switched depending on signalling received from the mobile station.

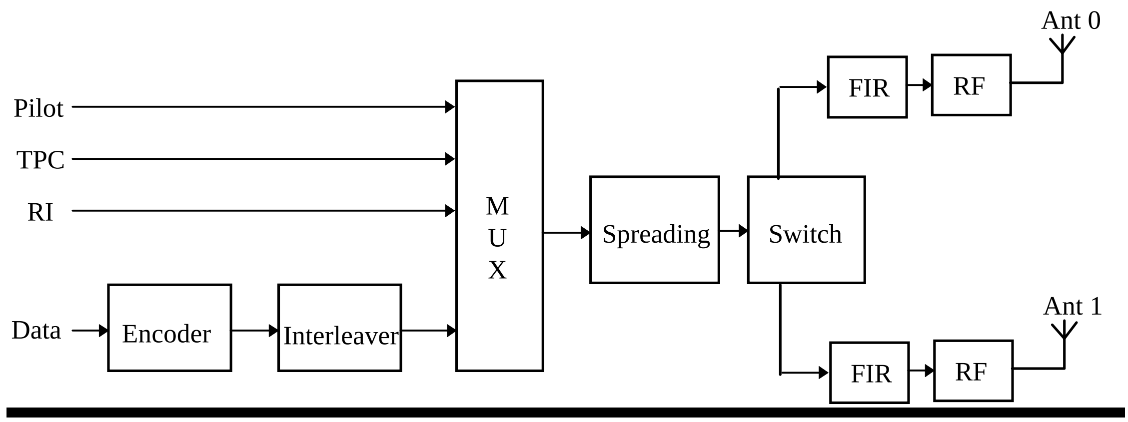


Figure 52. Base station transmitter block diagram for time division transmission diversity.

5.6.3.2.1 Time Switched Transmission Diversity

Time switched transmission diversity (TSTD) is implemented using the block diagram exactly as shown in Figure 52. TSTD does not assume any change to the UTRA/FDD physical layer channel structure other than switching at the filter input. There is no change to the channel coding, rate matching, interleaving and spreading within the UTRA/FDD physical layer description.

TSTD is used for the transmission of downlink Dedicated Physical Channels (DPCHs). All other downlink channels, i.e. the Common Control Physical Channels (CCPCHs) and the Synchronisation Channel (SCH), are transmitted from a single antenna, without diversity. TSTD is implemented by transmitting consecutive slots of the downlink DPCHs through two separate antennas. After scrambling, the spread time slots are switched consecutively to each antenna (i.e. the baseband signal is switched before modulation is applied, between transmitter antennas, at a rate of once every 0.625 ms).

The BCCH informs all mobile stations of the corresponding base station's capability for TSTD. The DPDCH and the DPCCH in the same slot for a given mobile station are then transmitted from one of the antennas. The next slot of the DPCH is transmitted from the other antenna. The DPCHs of other users operating in TSTD mode, may have different switching patterns in order to reduce the peak transmit power and peak to average power ratio in each power amplifier.

The spread time slots are transmitted to each antenna sequentially as shown in Figure 53.

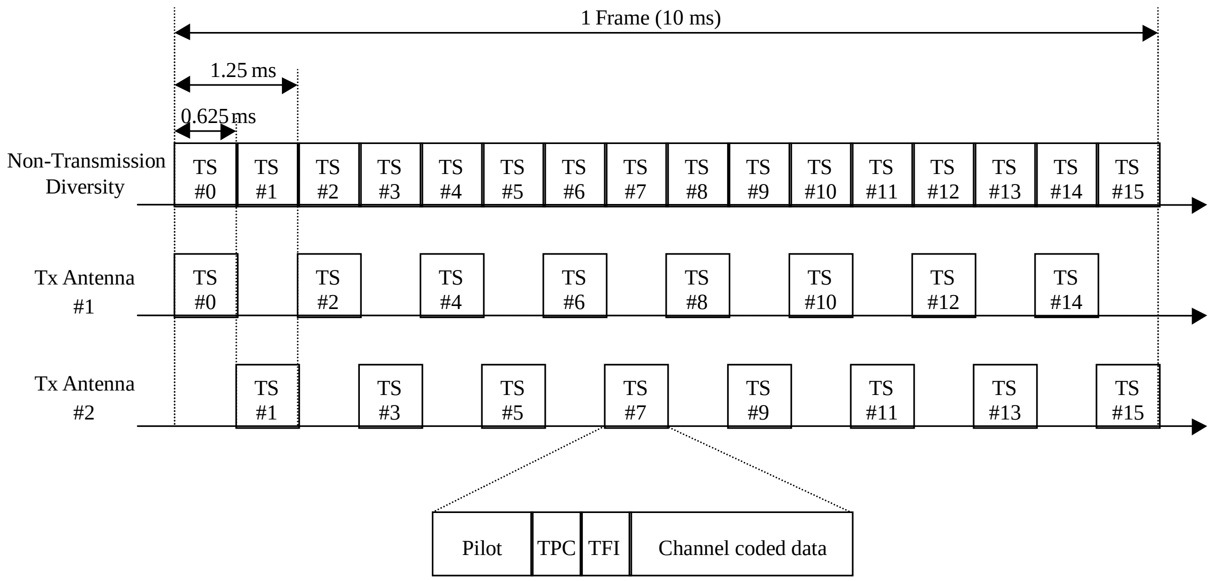


Figure 53. Switching pattern for Dedicated Physical Channels in TSTD.

5.6.3.2.2 Selection Transmit Diversity

Selection Transmit Diversity (STD) with fast closed loop control may be used to provide transmit diversity. For STD, the structure of the Base Station Transmitter is as shown in Figure 54. The implementation of STD is as follows. In the case of no soft handover, the base station antenna is dynamically selected, based on a fast transmit antenna selection (AS) control signal, transmitted by the mobile station (similar to fast PC loop). The value of the AS bit is determined, based on measurements on the antenna specific Primary CCPCH channel. The control loop speed is 400 Hz (note: the exact AS control loop speed is for further study). In order to guarantee that the mobile station is decoding the right downlink signal, the pilot symbols of the antennas are selected to be orthogonal with each other.

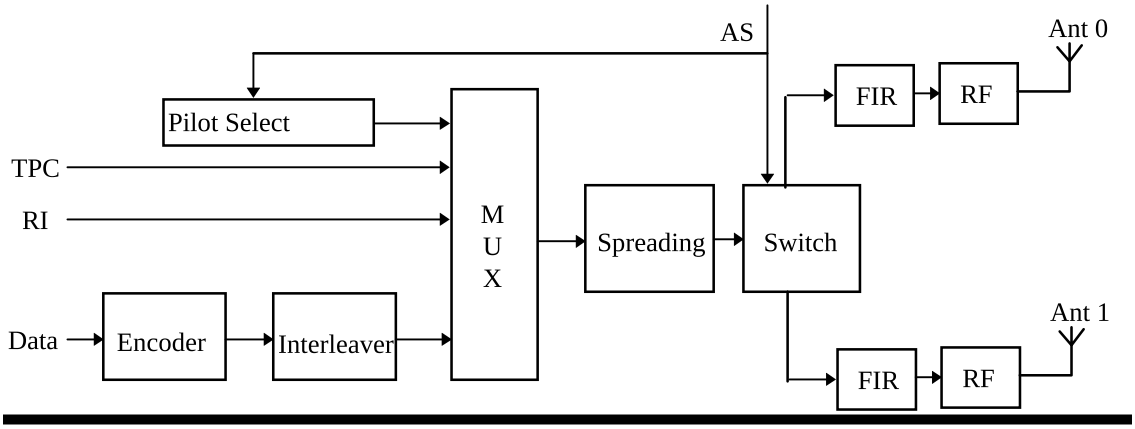


Figure 54. Selective Transmit Diversity: Base station transmitter block diagram.

5.6.4 Locationing function support

The wideband nature of the UTRA/FDD facilitates the high resolution in position location as the resolution achievable is directly proportional to the channel symbol rate, in this case chip rate. The duration of one chip corresponds to approximately 73 meters in propagation distance and if the delay estimation operates on the accuracy of samples/chip then the achievable maximum accuracy is approximately 18 meters with the 4.096 Mcps chip rate. Naturally there are then other inaccuracies that will cause degradation to the positioning but 18 meters can be considered as kind of lower bound on the positioning performance. With higher sampling rate or chip rate the bound is then naturally even lower.

With the UTRA/FDD concept the position location has been discussed in several ETSI/SMG2 input documents. One example solution to use is the proposed power up function (PUF) which in the need for an MS to be heard by several base stations will increase the transmission power over short interval. Other aspects of the position mechanism are how the issue of actual measurement is done and whether that is based on loop around time or on Time Difference Of Arrival (TDOA) or other measures.

6. LAYER 1 DESCRIPTION (TDD MODE)

6.1 Transport channels and physical channels (TDD)

6.1.1 Transport channels

This chapter describes the transport channels that are required for data transfer. Transport channels are the services offered by Layer 1 to the higher layers. A general classification of transport channels is into two groups:

- dedicated channels

- common channels

6.1.1.1 Dedicated transport channels

The only type of dedicated transport channel is the:

1. Dedicated Channel (DCH) characterised by:
    - possibility to use beamforming,
    - possibility to change rate fast (each 10ms),
    - possibility to use enhanced power control and
    - inherent addressing of MSs.

6.1.1.2 Common transport channels

Common transport channels are:

1. Random Access Channel(s) (RACH) characterised by:
    - existence in uplink only,
    - collision risk,
    - open loop power control,
    - limited data field, and
    - requirement for in-band identification of the MSs.
2. Forward Access Channel(s) (FACH) characterised by:
    - existence in downlink only,
    - possibility to use beamforming,
    - possibility to use enhanced power control,
    - requirement for in-band identification of MSs.
3. Broadcast Control Channel (BCCH) characterised by:
    - existence in downlink only,
    - low fixed bit rate and
    - requirement to be broadcast in the entire coverage area of the cell.
4. Paging Channel (PCH) characterised by:
    - existence in downlink only,
    - possibility for sleep mode procedures and
    - requirement to be broadcast in the entire coverage area of the cell.
5. Synchronisation Channel (SCH) characterised by:
    - existence in TDD and downlink only,
    - low fixed bit rate and
    - requirement to be broadcast in the entire coverage area of the cell.

6.1.2 Physical channels

A physical channel is defined as the association of one code, one time slot and one frequency.

6.1.2.1 Frame structure

In the following sections, an overview about the frame, time slot and code structure is outlined.

<u>6.1.2.1.1 Time slots</u>

The TDMA frame has duration of 10 ms and is subdivided into 16 time slots (TS) of 625 μs duration each. A time slot corresponds to 2560 chips. The physical content of the time slots is the bursts of corresponding length as described in Section Burst types.

6.1.2.1.2 TDD frame

Each 10 ms frame consists of 16 time slots; each allocated to either the uplink or the downlink (Figure 55). With such flexibility, the TDD mode can be adapted to different environments and deployment scenarios. In any configuration at least one time slot has to be allocated for the downlink and at least one time slot has to be allocated for the uplink.

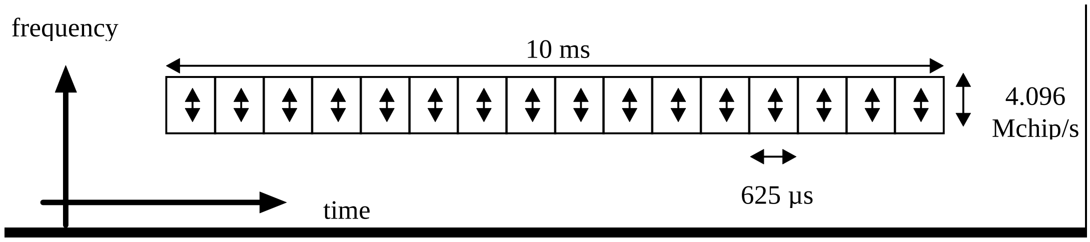


Figure 55. The TDD frame structure

Examples for multiple and single switching point configurations as well as for symmetric and asymmetric UL/DL allocations are given in Figure 56.

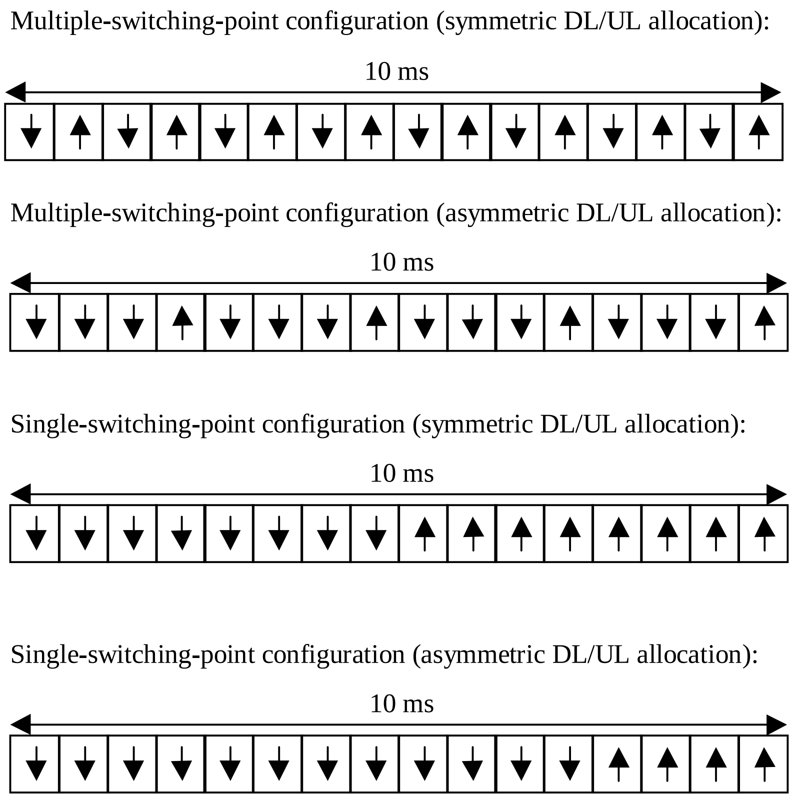


Figure 56. TDD frame structure examples

6.1.2.1.3 Spreading codes

Two options are being considered for the bursts that can be sent as described below. Both options allow a high degree of bit rate granularity and flexibility, thus allowing the implementation of the whole service range from low to high bit rates.

6.1.2.1.4 Multi-code transmission with fixed spreading

Within each time slot of length 625 µs, an additional separation of user signals by spreading codes is used. This means, that within one time slot of length 625 µs, more than one burst of corresponding length as described in Section 0 can be transmitted. These multiple bursts within the same time slot can be allocated to different users as well as partly or all to a single user. For the multiple bursts within the same time slot, different spreading codes are used to allow the distinction of the multiple bursts.

The bursts as described in Section 0 are designed in such a way, that up to 8 bursts could be transmitted within one time slot, if the bursts are allocated to different users in the uplink. In the downlink or if several bursts in the time slot are allocated to one single user in the uplink, even more than 8 bursts (e.g. 9 or 10) can be transmitted within one time slot.

6.1.2.1.5 Single code transmission with variable spreading

Within each time slot of 625 µs,

- a mobile always uses single code transmission by adapting the spreading factor as a function of the data rate. This limits the peak-to-average ratio of the modulated signal and consequently the stress imposed to the power amplifier resulting in an improved terminal autonomy. Several mobiles can be received in the same time slot by the base station, they are separated by their codes and the individual decoding can take profit of the joint detection.
- a base station should broadcast a single burst per mobile again by adapting the spreading as a function of the data rate. High rate data transmissions, requiring more than one timeslot per mobile, can be supported by terminals having the processing power for joint detection on a single slot: the required throughput occupies in a general way an integer number of slots plus a fraction of an extra slot. Single burst transmission should occur in the integer number of slots, while the extra slot can be occupied by a burst for the considered mobile plus extra bursts for other mobiles, joint detection is only needed for this last time slot in the considered mobile.

6.1.2.2 Burst types

As explained in the section Spreading codes, two options are being considered for the spreading.

6.1.2.2.1 Bursts for dedicated transport channels

Two types of bursts for dedicated transport channels are defined: The burst type 1 and the burst type 2. Both consist of two data symbol fields, a midamble and a guard period. The burst type 1 has a longer midamble of 512 chips than the burst type 2 with a midamble of 256 chips. Sample sets of midambles are given in sections Sample midamble code set for burst type 1 and Sample midamble code set for burst type 2.

Because of the longer midamble, the burst type 1 is suited for the uplink, where up to 8 different channel impulse responses have to be estimated. The burst type 2 can be used for the downlink and, if the bursts within a time slot are allocated to less than four users, also for the uplink.

Thus the burst type 1 can be used for

- uplink, independent of the number of active users in one time slot
- downlink, independent of the number of active users in one time slot

The burst type 2 can be used for

- uplink, if the bursts within a time slot are allocated to less than four users
- downlink, independent of the number of active users in one time slot

The data fields of the burst type 1 are 976 chips long, whereas the data fields length of the burst type 2 are 1104 chips. The corresponding number of symbols depends on the spreading factor, as indicated in Table 7 below. The guard period for the burst types 1and 2 is 96 chip periods long.

Table 7. Number of symbols per data field in bursts 1 and 2

Spreading factor (Q)	Number of symbols (N) per data field in Burst 1	Number of symbols (N) per data field in Burst 2
1	976	1104
2	488	552
4	244	276
8	122	138
16	61	69

The burst types 1 and 2 are shown in Figure 57 and Figure 58. The contents of the traffic burst fields are described in Table 8and Table 9.

Table 8. The contents of the burst type 1 fields

Chip number (CN)	Length of field In chips	Length of field in symbols	Length of field in µs	Contents of field
0-975	976	Cf. Table 7	238.3	Data symbols
976-1487	512	-	125.0	Midamble
1488-2463	976	Cf. Table 7	238.3	Data symbols
2464-2559	96	-	23.4	Guard period

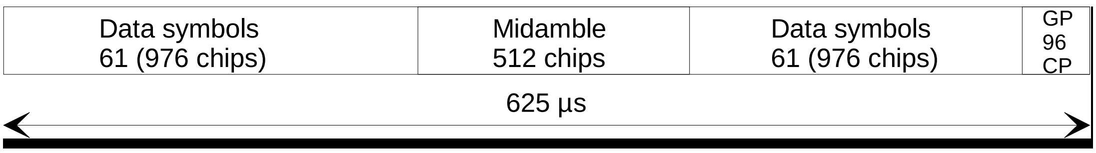


Figure 57. Burst structure of the burst type 1. GP denotes the guard period and CP the chip periods.

Table 9. The contents of the burst type 2 fields

Chip number (CN)	Length of field in chips	Length of field In symbols	Length of field in µs	Contents of field
0-1103	1104	Cf. Table 7	269.55	Data symbols
1104-1359	256	-	62.5	Midamble
1360-2463	1104	Cf. Table 7	269.55	Data symbols
2464-2559	96	-	23.4	Guard period

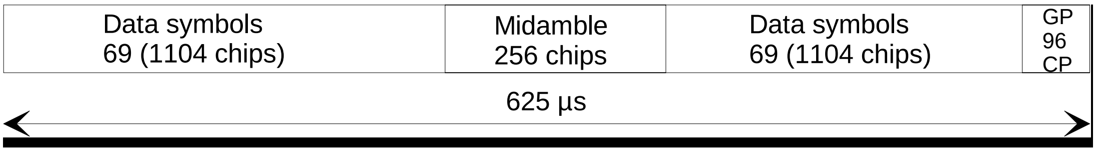


Figure 58. Burst structure of the burst type 2. GP denotes the guard period and CP the chip periods.

The two different bursts defined here are well-suited for the different applications mentioned above. It may be possible to further optimise the burst structure for specific applications, for instance for unlicensed operation.

6.1.2.3 Training sequences for spread bursts

As explained in the section Spreading codes, two options are being considered for the spreading. The training sequences presented here are common to both options.

Section Bursts contains a description of the spread speech/data bursts. These traffic bursts contain L_m midamble chips, which are also termed midamble elements. The L_m elements $\underline{m}_i^{(k)}$; i=1,...,L_m; k=1,...,K; of the midamble codes $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; of the K users are taken from the complex set

$$\underline{\mathrm{V}}_\mathrm{m} = \{1,\ \mathrm{j},\ -1,\ -\mathrm{j}\}. \tag{0-1}$$

The elements $\underline{m}_i^{(k)}$ of the complex midamble codes $\underline{\mathbf{m}}^{(k)}$ fulfil the relation

$$\underline{m}_i^{(k)} = (\mathrm{j})^i \cdot m_i^{(k)} \qquad m_i^{(k)} \in \{1,\ -1\};\ i = 1,\ldots,L_m;\ k = 1,\ldots,K. \tag{0-2}$$

Hence, the elements $\underline{m}_i^{(k)}$ of the complex midamble codes $\underline{\mathbf{m}}^{(k)}$ of the K users are alternating real and imaginary.

With W being the number of taps of the impulse response of the mobile radio channels, the L_m binary elements $m_i^{(k)};\ \mathrm{i} = 1,\ldots,L_\mathrm{m};\ k = 1,\ldots,K;$ of (6-2) for the complex midambles $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; of the K users are generated according to Steiner's method from a single periodic basic code

$$\mathbf{m} = \left(m_1, m_2, \ldots, m_{L_m+(K-1)W}\right)^\mathrm{T} \qquad m_i \in \{1,\ -1\};\ \ i = 1,\ldots,(L_m + (K-1)W). \tag{0-3}$$

The elements $m_i;\ i = 1,\ldots,(L_m + (K-1)W),$ of (6-3) fulfil the relation

$$m_i = m_{i-P} \qquad \text{for the subset} \qquad i = (P+1),\ldots,(L_m + (K-1)W). \tag{0-4}$$

The P elements $m_i;\ i = 1,\ldots,P,$ of one period of **m** according to (6-3) are contained in the vector

$$\mathbf{m}_\mathrm{P} = \left(m_1, m_2, \ldots, m_P\right)^\mathrm{T}. \tag{0-5}$$

With $\mathbf{m}$ according to (6-3) the L_m binary elements $m_i^{(k)};\ \mathrm{i} = 1,\ldots,\mathrm{L}_\mathrm{m};\ k = 1,\ldots,K;$ of (6-2) for the midambles of the K users are generated based on Steiner's formula

$$m_i^{(k)} = m_{i+(K-k)W} \qquad i = 1,\ldots,L_m;\ k = 1,\ldots,K. \tag{0-6}$$

In the following the term *'a midamble code set'* or *'a midamble code family'* denotes K specific midamble codes $\underline{\mathbf{m}}^{(k)}$; k=1,...,K. Different midamble code sets $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; are in the following specified based on different periods $\underline{\mathbf{m}}_{\mathrm{P}}$ according (6-5).

In adjacent cells of the cellular mobile radio system, different midamble codes sets $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; should be used to guarantee a proper channel estimation.

As mentioned above a single midamble code set $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; consisting of K midamble codes is based on a single period $\underline{\mathbf{m}}_{\mathrm{P}}$ according to (6-5).

In the following several periods $\underline{\mathbf{m}}_{\mathrm{P}}$ according (6-5) which should be used to generate different midamble code sets $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; will be listed in tables in a hexadecimal representation. As shown in Table 10 always 4 binary elements $\underline{m}_i$ are mapped on a single hexadecimal digit.

Table 10. Mapping of 4 binary elements $\underline{m}_i$ on a single hexadecimal digits

4 binary elements $\underline{m}_i$	**Mapped on hexadecimal digit**
-1 -1 -1 -1	0
-1 -1 -1 1	1
-1 -1 1 –1	2
-1 -1 1 1	3
-1 1 -1 –1	4
-1 1 -1 1	5
-1 1 1 –1	6
-1 1 1 1	7
1 -1 -1 –1	8
1 -1 -1 1	9
1 -1 1 –1	A
1 -1 1 1	B
1 1 -1 –1	C
1 1 -1 1	D
1 1 1 –1	E
1 1 1 1	F

The mean degradations [2, equation (38)] which serve as a quality information of the periods $\underline{\mathbf{m}}_{\mathrm{P}}$ according to (6-5) and hence of the specified midamble code sets $\underline{\mathbf{m}}^{(k)}$; k=1,...,K; will be also given.

6.1.2.3.1 Sample midamble code set for burst type 1

In the case of burst type 1 (see section Bursts) the midamble has a length of L_{m}=512, which is corresponding to:

K=8; W=57; P=456

Table 11. Sample Periods $\underline{\mathbf{m}}_P$ according (6-5) for case of burst type 1.

Periods $\underline{\mathbf{m}}_P$ of length *P*=456	Degradation in dB
C482462CA7846266060D21688BA00B72E1EC84A3D5B7194C8DA39E21A3CE12BF512 C8AAB6A7079F73C0D3E4F40AC555A4BCC453F1DFE3F6C82	0.649471
56F3ACE0A65B96FC326A30B91665BD4380907C2B08DEC98C16A0B0339AEA855C3D 8BDD016E4C3E0F3DA5DF5C0891C851BA30A6C19ABE6C3ED4	0.695320
1D566C76440333CBF3CA2A405386068E19A2D6A53560CC50138B3A15BF7D9683F95F 66FF096431363E09A514D61099DD3EAD52903BF4A27D14	0.705751
9A0A349E49389CC184F7A3420D3FBE06B3A40BEE933D8E04E61FAA4A5214D918A1 ADD5BE25D833579FBCF17B422300D0CA1B419393F9722AA8	0.706513
B760E5694E49169C225A2FBCDACCCA8847F8486A6A351EB7D045BA2271B2A4CB90 0404C0D2BBA00F80F963861BD7DCE748F0F10AE6B785D0F0	0.707417
ECE93B83CE32E395405F7C889751970E84AFD632500B91E17C4E7846FE68D3C841013 5D3114D3281211214D1F5F1996A6B656259F11728AA52	0.708587
DE1B6F6219A0AD1A3EB5EEA02173D704C3340AAE7310B93A21BCF979BC7B6C081 7003AA300B1704BCE62524EC48C505977A1570F6C6BA1A2D8	0.711320

6.1.2.3.2 Sample midamble code set for burst type 2

In the case of burst type 2 (see section Bursts) the midamble has a length of L_m=256, which is corresponding to:

K=3; W=64; P=192

Table 12. Sample Periods $\underline{\mathbf{m}}_P$ according (6-5) for case of burst type 2.

Periods of length *P*=192	Degradation in dB
D4A124FE4D11BC14C258546A18C5DE0E3AA3F0617245DBFE	0.615566
48D76A687E21D22321C5201977F620D7A4CB5945F5693A1C	0.638404
9EEF5E79606DCAAB046769524691E09E816DC688ABC12030	0.663436
D2369A2B704878F55B58A300C853A2F62233E6207E39F944	0.677739
A26C7D9697B002714E9285D2AFC3AF1E233FC8C6C7486080	0.686287
8A615F5D7EE05668415E626482E90B11C95305E4707015B5	0.686660
5CC2D7409922FA463D2D14377EBCF0CC0E888426B06F0A82	0.688977
A68238D5BD37B2B4C48B466B9815087898409AFCB804FA0B	0.692613

6.1.2.3.3 Midamble transmit power

In the case of the downlink, 2*K* data blocks are transmitted in a burst simultaneously. Also in the uplink, if *K*' greater than one CDMA code are assigned to a single user, 2*K*' data blocks are transmitted in a burst simultaneously by this user. This is the so-called multi-code uplink situation. In the downlink and the multi-code uplink, the mean power used to transmit the midambles on the one hand and the 2*K* (or 2*K*') data blocks on the other hand shall be equal. This shall be achieved by multiplying the midamble codes $\underline{\mathbf{m}}^{(k)}$, *k*=1,...,*K*, with a proper real factor to achieve an attenuation or an amplification.

6.1.3 Mapping of transport channels to physical channels

This section describes the way in which transport channels are mapped onto physical channels as described in

Section Physical channels. A description of the multi-frame structure is given in Section Multi-frame structure.

6.1.4 Dedicated transport channels

A dedicated transport channel is mapped onto one or more sets of slots and codes within a frame. An interleaving period is associated with each allocation. The frame is subdivided into slots that are available for uplink and downlink information transfer. Each set of slots and codes over an interleaving period maps to a data unit and a data unit can correspond to one or more FEC code blocks and one or more RLC protocol data units dependent from the service being supported. The following diagram illustrates the mapping:

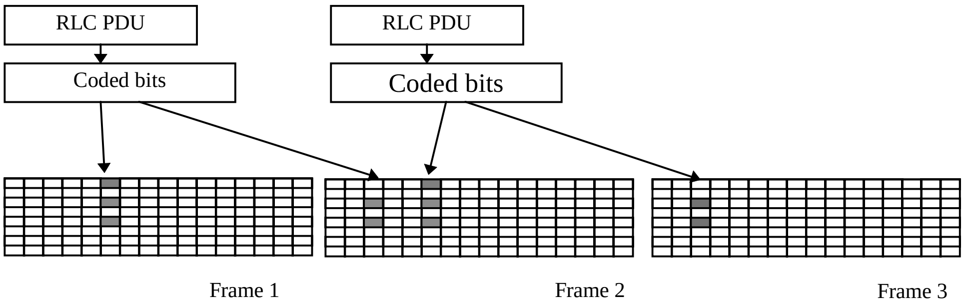


Figure 59. Mapping of PDU onto the physical bearer

For NRT packet data services an allocation is made only for a relatively short period of time. In general, for RT services an allocation is made for a certain time period and a release procedure is necessary to release the resource. For the efficient use of resources the slot/ code set allocated to a radio bearer may be changed from time to time and the resources allocated to a VBR service may increase or decrease along with the changes in the data rate. Traffic channels are power controlled, cf. Section Power Control.

6.1.5 Multi-frame structure

A strong requirement for the multi-frame structure comes from the realisation of low cost dual mode FDD-TDD terminals and from the GSM compatibility of the UTRA proposal. In this respect the super-frame and multi-frame structure for FDD and TDD mode have to be compatible and harmonised with GSM.

Thus in the proposed structure a multi-frame is composed by 24 frames each of length 10 ms. So the multi-frame period is 240 ms (twice the GSM TCH-F multi-frame).

All frames in the traffic channel multi-frames are used to carry both user data and dedicated signalling because:

- The use of in-band dedicated signalling or allocation of a dedicated signalling channel avoids the use of a signalling frame like SACCH frame in GSM.

  The most flexible method to distribute different user data blocks such as in-band signalling is under study.

- There is no need for an idle slot to read BCCH's of adjacent cells as in GSM

  Adjacent cells in the TDD network are frame-synchronised.

- The bursty nature of TD-CDMA transmission and reception allows the MS in idle time slots to make measurements on GSM and FDD networks. This is valid also for high bit rate users (BCCH and RACH slots could also be used to this purpose)

The multi-frame length is therefore given by the common channel with the lowest bit rate in the present case the SCH, if its multi-frame structure is compatible with the GSM TCH-F multi-frame. This leads to a multi-frame length of 240 ms. Three TDD multi-frames match exactly into a FDD multi-frame ensuring the compatibility of both components.

6.2 Multiplexing, channel coding and interleaving (TDD)

6.2.1 General

This section describes the services multiplexing, channel coding/interleaving and rate matching.

In the UTRA-TDD mode, the total number of basic physical channels (a certain time slot one spreading code on a certain carrier frequency) per frame is given by the maximum number of time slots which is 16 and the maximum number of CDMA codes per time slot. This maximum number of codes is 8 in case the different codes within one time slot are allocated to different users in the uplink and is higher than 8 (e.g. 9 or 10) in the downlink or if several codes are allocated to one single user in the uplink.

The service classes given in the following represent only a selection of all possibilities that are conceivable.

Two types of traffic bursts are used. They are described in section Physical channels.

6.2.2 Multiplexing

In a same connection, multiple services could be treated with separate channel coding/interleaving and mapping to different basic physical channels (slot/code), see Figure 60. In this way QoS can be separately and independently controlled.

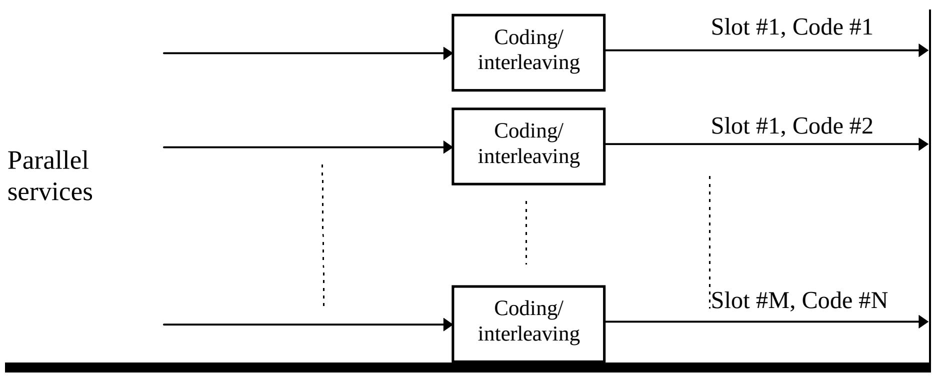


Figure 60. Service multiplexing (a)

A second alternative is time multiplexing at different points of the channel coding scheme, as shown in Figure 61.

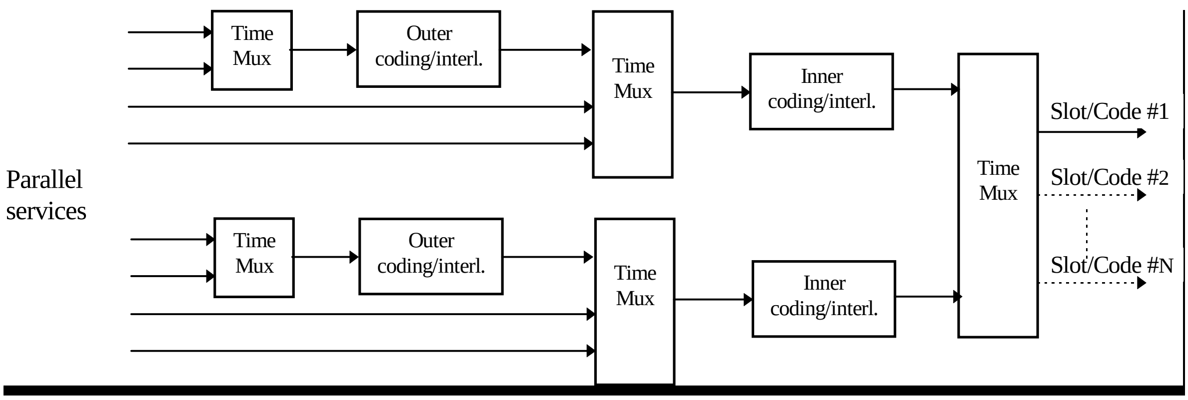


Figure 61. Service multiplexing (b)

After service multiplexing and channel coding, the multi-service data stream is mapped to one or, if the total rate exceeds the upper limit for single-code transmission, several resource units.

6.2.3 Channel coding and interleaving

In Real Time (RT) services a FEC coding is used, instead Non Real Time (NRT) services could be well managed with a proper combination of FEC and ARQ.

For the RT services two levels of QoS (10^{-3}, 10^{-6}) have been considered as examples in Figure 62.

Only convolutional coding is used in case of BER=10^{-3}, while a concatenated code scheme (Reed-Solomon, outer interleaving and convolutional coding) or Turbo codes could be used to achieve BER=10^{-6}.

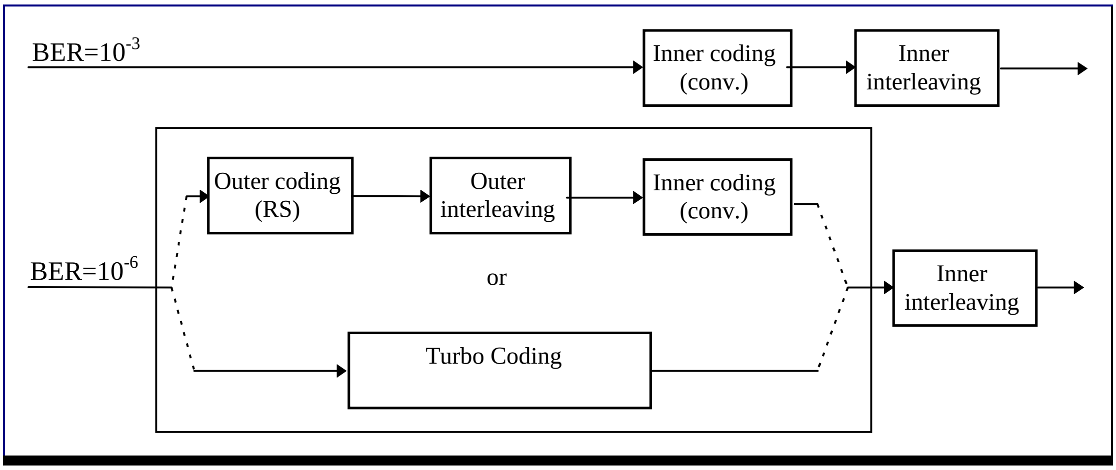


Figure 62. FEC coding

6.2.3.1 Inner coding/interleaving

The convolutional coding rates change according to the rates of different services. The convolutional coding rates from 1/4 to 1 have been chosen such that the complete system will be able to use as much as possible the same decoding structure.

After convolutional coding, interleaving is used. For LDD services, inter-frame interleaving over two 10 ms frames is applied. For LCD services an interleaving over 300 ms is applied.

6.2.3.2 Outer coding/interleaving

The outer RS coding, on GF(2^8) has different rate for different services. An outer interleaver to break the error burst at the output of the Viterbi decoder is needed in addition to an inner interleaver for breaking the error bursts due to fading.

6.2.4 Rate matching

To map the services on the air interface either puncturing or unequal repetition is used after channel coding.

This rate matching is performed considering both bursts:

- burst 1 (long midamble) used in uplink;
- burst 2 (short midamble) used in downlink as well as for uplink transmission in the case of multi-code transmission.

6.3 Spreading and modulation (TDD)

6.3.1 General

In this chapter, there has been made a separation between the data modulation and the spreading modulation. The data modulation is defined in Section Data modulation and the spreading modulation in Section Spreading modulation.

Table 13. Basic modulation parameters

Chip rate	Same as FDD basic chiprate: 4.096 Mchip/s
Carrier spacing	5.0 MHz
Data modulation	QPSK
Chip modulation	Same as FDD chip modulation: root raised cosine roll-off α = 0.22
Spreading characteristics	Orthogonal Q chips/symbol, where $Q = 2^p$, 0<= p <= 4

6.3.2 Data modulation

6.3.2.1 Symbol rate

The symbol rate and duration are indicated below.

Ts = Q. T_c, where $T_c = \frac{1}{chiprate}$ = 0.24414 µs, reflecting the dependence of the symbol time T_s upon the spreading factor Q.

6.3.2.2 Mapping of bits onto signal point constellation

A certain number K of CDMA codes can be assigned to either a single user or to different users who are simultaneously transmitting bursts in the same time slot and the same frequency. The maximum possible number of CDMA codes, which is smaller than or equal to 16, depends on the individual spreading factors, the actual interference situation and the service requirements. In Section Bursts examples of bodies of such spread bursts associated with a particular user are shown. Each user burst has 2 data carrying parts termed data blocks

$$\underline{\mathbf{d}}^{(k,i)} = (\underline{d}_1^{(k,i)}, \underline{d}_2^{(k,i)}, \ldots, \underline{d}_N^{(k,i)})^{\mathrm{T}} \qquad \mathrm{i} = 1,\ 2;\ \mathrm{k} = 1,\ldots,\mathrm{K}. \tag{0-7}$$

N_k is the number of symbols per data field for the user k. This number is linked to the spreading factor Q_k as described in Table 7.

Data block $\underline{\mathbf{d}}^{(k,1)}$ is transmitted before the midamble and data block $\underline{\mathbf{d}}^{(k,2)}$ after the midamble. Each of the *N* data symbols $\underline{d}_n^{(k,i)}$; i=1, 2; k=1,...,K; n=1,...,N; of (6-7) has the symbol duration $T_s^{(k)} = Q_k . T_c$ as already given.

The data modulation is QPSK; thus the data symbols $\underline{d}_n^{(k,i)}$ are generated from two interleaved and encoded data bits

$$b_{l,n}^{(k,i)} \in \{0,1\} \qquad \mathrm{l} = 1,2;\ \mathrm{n} = 1,\ldots,\mathrm{N};\ \mathrm{k} = 1,\ldots,\mathrm{K};\ \mathrm{i} = 1,\ 2 \tag{0-8}$$

using the equation

$$\mathrm{Re}\left\{\underline{d}_n^{(k,i)}\right\} = \frac{1}{\sqrt{2}}(2b_{1,n}^{(k,i)} - 1)$$
$$\mathrm{Im}\left\{\underline{d}_n^{(k,i)}\right\} = \frac{1}{\sqrt{2}}(2b_{2,n}^{(k,i)} - 1) \qquad \mathrm{n} = 1,\ldots,\mathrm{N};\ \mathrm{k} = 1,\ldots,\mathrm{K};\ \mathrm{i} = 1,\ 2. \tag{0-9}$$

Equation (6-9) corresponds to a QPSK modulation of the interleaved and encoded data bits $b_{l,n}^{(k,i)}$ of (6-8).

6.3.2.3 Pulse shape filtering

The pulse shape filtering is applied to each chip at the transmitter. In this context the term chip represents a single element $\underline{c}_q^{(k)}$ with k=1,...,K; q=1,...,Q_k; of a spreading code $\underline{\mathbf{c}}^{(k)}$; see also Section Spreading codes.

The impulse response of the above mentioned chip impulse filter $Cr_0(t)$ shall be a root raised cosine. The corresponding raised cosine impulse $C_0(t)$ is defined as

$$C_0(t) = \frac{\sin \pi \frac{t}{T_C}}{\pi \frac{t}{T_C}} \cdot \frac{\cos \alpha\pi \frac{t}{T_C}}{1 - 4\alpha^2 \frac{t^2}{T_C{}^2}} \tag{0-10}$$

The roll-off factor shall be $\alpha = 0.22$. T_C is the chip duration:

$$T_C = \frac{1}{\mathrm{Chiprate}} = 0.24414\mu\mathrm{s}$$

The impulse response $C_0(t)$ according to (6-10) and the energy density spectrum $\Phi_{C0}(f)$ of $C_0(t)$ are depicted in the figure below:

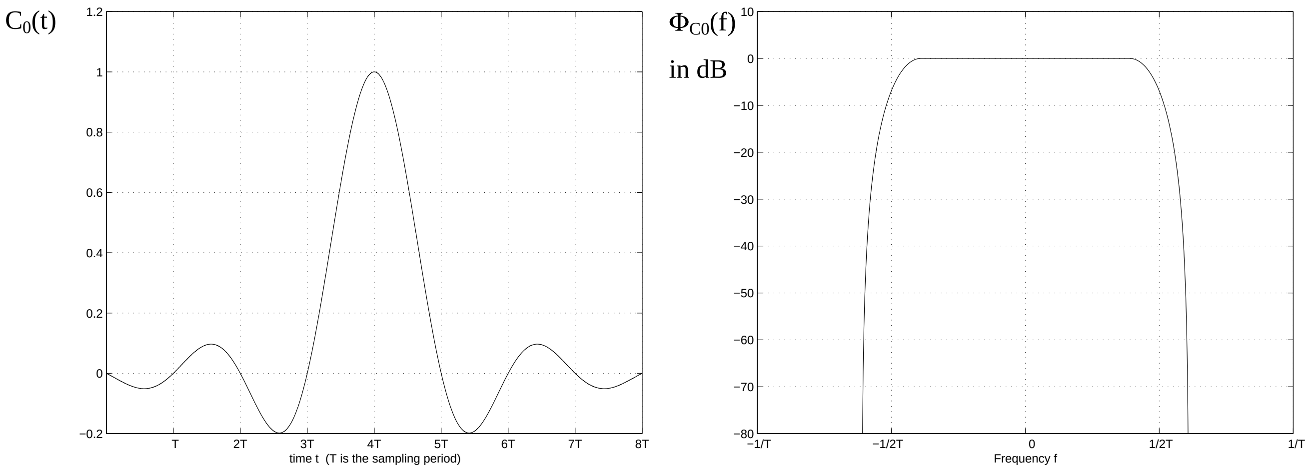


Figure 63. Basic impulse $C_0(t)$ and the corresponding energy density spectrum $\Phi_{C0}(f)$ of $C_0(t)$

6.3.3 Spreading modulation

6.3.3.1 Basic spreading parameters

Each data symbol $\underline{d}_n^{(k,i)}$ of (6-7) is spread with a spreading code $\underline{\mathbf{c}}^{(k)}$ of length. $Q_k \in \{1,2,4,8,16\}$. The resulting sequence is then scrambled by a sequence ν of length 16.

6.3.3.2 Spreading codes

The elements $\underline{c}_q^{(k)}$; k=1,...,K; q=1,...,Q_k; of the spreading codes $\underline{\mathbf{c}}^{(k)}=(\underline{c}_1^{(k)},\underline{c}_2^{(k)},...,\underline{c}_{Q_k}^{(k)})$; k=1,...,K; shall be taken from the complex set

$$\underline{\mathrm{V}}_{\mathrm{c}} = \{1,\ \mathrm{j},\ -1,\ -\mathrm{j}\}. \qquad (0\text{-}11)$$

In equation (6-11) the letter j denotes the imaginary unit. The spreading code $\underline{\mathbf{c}}^{(k)}$ = $\underline{\mathbf{c}}^{(k)}=(\underline{c}_1^{(k)},\underline{c}_2^{(k)},...,\underline{c}_{Q_k}^{(k)})$ is generated from the binary CDMA codes $\mathbf{a}^{(k)}$ of length Q_k shown in Figure 64allocated to the k^{th} user. . The relation between the elements $\underline{c}_q^{(k)}$ and $\underline{a}_q^{(k)}$ is given by:

$$\underline{c}_q^{(k)} = (\mathrm{j})^q \cdot a_q^{(k)} \qquad a_q^{(k)} \in \{1,\ -1\};\ \mathrm{q}=1,\ldots,\mathrm{Q};\ \mathrm{k}=1,\ldots,\mathrm{K}. \qquad (0\text{-}12)$$

Hence, the elements $\underline{c}_q^{(k)}$ of the CDMA codes $\underline{\mathbf{c}}^{(k)}$ are alternating real and imaginary. The $\mathbf{a}^{(k)}$ are Orthogonal Variable Spreading Factor (OVSF) codes, allowing mixing in the same timeslot channels with different spreading factors while preserving the orthogonality. The OVSF codes can be defined using the code tree of Figure 64.

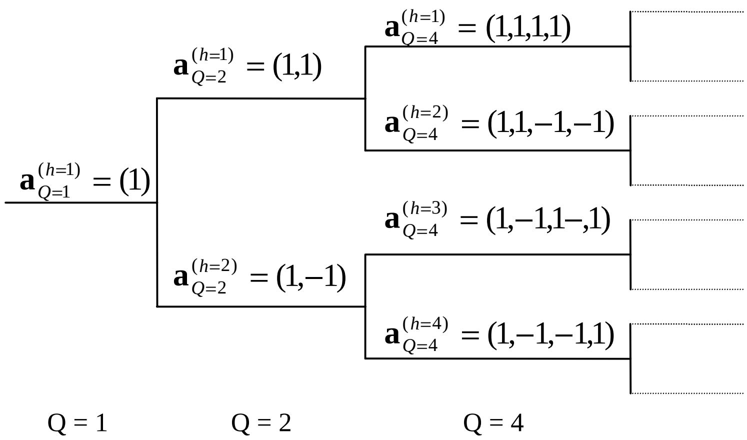


Figure 64. Code-tree for generation of Orthogonal Variable Spreading Factor (OVSF) codes.

Each level in the code tree defines spreading factors indicated by the value of Q in the figure. All codes within the code tree cannot be used simultaneously in a given timeslot. A code can be used in a timeslot if and only if no other code on the path from the specific code to the root of the tree or in the sub-tree below the specific code is used in this timeslot. This means that the number of available codes in a slot is not fixed but depends on the rate and spreading factor of each physical channel.

The spreading factor goes up to Q_{MAX}=16.

6.3.3.3 Scrambling codes

The spreading of data by a code $\mathbf{a}^{(k)}$ of length Q_k is followed by a cell specific scrambling sequence ν=(ν1, ν2, … ν_{QMAX}). The length matching is obtained by concatenating Q_{MAX}/Q_k spread words before the scrambling. The scheme is illustrated in Figure 65 below and is described in more detail in Section Spread and scrambled signal of data symbols and data blocks.

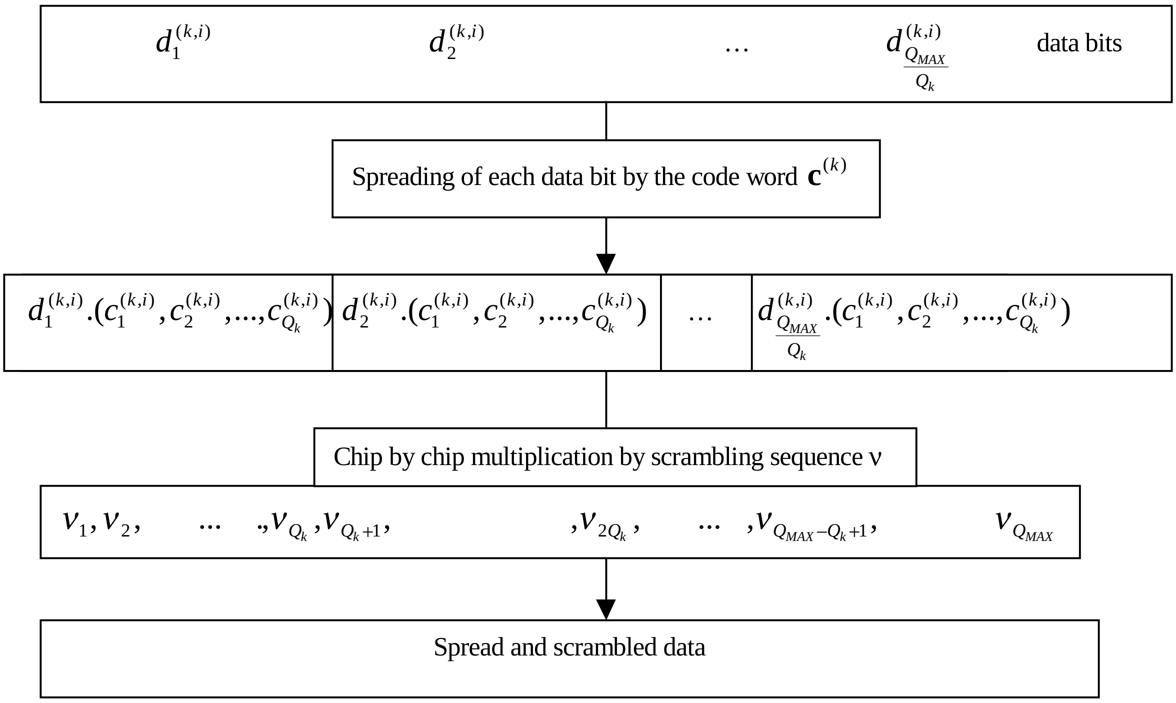


Figure 65. Spreading and subsequent scrambling of data bits.

6.3.3.4 Spread and scrambled signal of data symbols and data blocks

The combination of the spreading and cell specific scrambling codes can be seen as a user and cell specific spreading code $\mathbf{s}^{(k)} = \left(s_p^{(k)}\right)$ with $s_p^{(k)} = c^{(k)}_{1+[(p-1)\bmod Q_k]} .\acute{\imath}_{1+[(p-1)\bmod Q_{MAX}]}$, k=1,…,K, p=1,…,$N_k Q_k$. The transmitted signal belonging to the data block $\underline{\mathbf{d}}^{(k,1)}$ of (6-7) transmitted before the midamble is

$$\underline{d}_n^{(k,1)}(t) = \sum_{n=1}^{N} \underline{d}_n^{(k,1)} \sum_{q=1}^{Q} \underline{c}_q^{(k)} \cdot Cr_o(t-(q-1)T_c - nT_c) \qquad (0\text{-}13)$$

and for the data block $\underline{\mathbf{d}}^{(k,2)}$ of (6-7) transmitted after the midamble

$$\underline{d}_n^{(k,2)}(t) = \sum_{n=1}^{N} \underline{d}_n^{(k,2)} \sum_{q=1}^{Q} \underline{c}_q^{(k)} \cdot Cr_0(t-(q-1)T_C - nT_C - NQT_c - L_m T_c). \qquad (0\text{-}14)$$

where L_m is the number of midamble chips.

6.4 Radio transmission and reception (TDD)

6.4.1 Frequency bands and channel arrangement

6.4.1.1 Proposed frequency bands for operation

UTRA/TDD is designed to operate in any frequency band that will accommodate at least one 4,096 Mcps carrier.

6.4.1.2 Carrier raster

The channel raster is 200 kHz.

6.4.1.3 Tx - Rx Frequency Separation

Tx and Rx are not separated in frequency.

6.4.2 Service Class

See relevant chapter for FDD mode

6.4.3 Transmitter characteristics

6.4.3.1 Output power

The mobile station and base station output power profiles would be used to define a range of output powers for the use in different system scenarios. The power class would be based on the peak power, e. g. 30 dBm for the terminals.

6.4.3.2 Output power dynamics

The transmitter uses fast closed-loop carrier/interference based power control and slow quality based power control on both the up- and downlink. The step size is variable and in the range 1.5 ...3 dB with 100-800 steps/s. The power control dynamic is 80 dB on the uplink and 30 dB on the downlink.

6.4.3.3 Output RF spectrum emissions, adjacent channel power, occupied bandwidth, frequency stability

See relevant Sections for FDD mode (Section Radio transmission and reception (FDD)).

6.4.4 Receiver characteristics

The receiver is typically a Joint Detection Receiver. Except for this the relevant chapters for the receiver characteristic of the FDD system apply also for the TDD system.

6.5 Physical layer procedures (TDD)

6.5.1 Synchronisation of the TDD base stations

It is required that BTS supporting the TDD mode, are operated in synchronised mode, so far the coverage area of the cells are overlapping, i.e. we have contiguous coverage for a certain area. The nature of the TDD operation requires BTS frame synchronisation, to achieve good spectral efficiency. The fact that MS and BTS are receiving and transmitting on the same frequency makes it desirable, that in the reuse cell the same TX / RX timing get used.

The lack of a frame synchronisation can cause, depending on the actual time slip, interference events that will effect several time slots.

By means of a frame synchronisation this effect should be minimised. However, it will be necessary for a cost efficient solution to allow some slip. The tolerance of the frame synchronisation shall be such, that the affected timeslots receive only minor performance degradation. I.e. only some of the symbols shall be corrupted by the frame slip, rather than a full slot. However synchronisation on a chip level is not required.

6.5.2 Channel Allocation

For the UTRA-TDD mode a physical channel is characterised by a combination of its carrier frequency, time slot, and spreading code as explained in the chapter on the physical channel structure

Channel allocation covers both:

- resource allocation to cells (slow DCA)
- resource allocation to bearer services (fast DCA)

6.5.2.1 Resource allocation to cells (slow DCA)

Channel allocation to cells follows the rules below:

- A reuse one cluster is used in the frequency domain. In terms of an interference-free DCA strategy a timeslot-to-cell assignment is performed, resulting in a time slot clustering. A reuse one cluster in frequency domain does not need frequency planning. If there is more than one carrier available for a single operator also other frequency reuse patters >1 are possible.

- Any specific time slot within the TDD frame is available either for uplink or downlink transmission. UL/DL resources allocation is thus able to adapt itself to time varying asymmetric traffic.
- In order to accommodate the traffic load in the various cells the assignment of the timeslots (both UL and DL) to the cells is dynamically (on a coarse time scale) rearranged (slow DCA) taking into account that strongly interfering cells use different timeslots. Thus resources allocated to adjacent cells may also overlap depending on the interference situation.

Due to idle periods between successive received and transmitted bursts, mobiles can provide the network with interference measurements in time slots different from the currently used one. The availability of such information enables the operator to implement the DCA algorithm suited to the network.

For instance, the prioritised assignment of time slots based on interference measurements results in a clustering in the time domain and in parallel takes into account the demands on locally different traffic loads within the network.

6.5.2.2 Resource allocation to bearer services (fast DCA)

Fast channel allocation refers to the allocation of one or multiple physical channels to any bearer service Resource units (RUs) are acquired (and released) according to a cell-related preference list derived from the slow DCA scheme.

The following principles hold for fast channel allocation:

1. The basic RU used for channel allocation is one code / time slot / (frequency).
2. Multi-rate services are achieved by pooling of resource units. This can be made both in the code domain (pooling of multiple codes within one time slot = **multi-code** operation) and time domain (pooling of multiple time slots within one frame = **multi-slot** operation). Additionally, any combination of both is possible.
3. Since the maximal number of codes per time slot in UL/DL depends on several physical circumstances like, channel characteristics, environments, etc. (see description of physical layer) and whether additional techniques to further enhance capacity are applied (for example smart antennas). The DCA algorithm has to be independent of this number. Additionally, time hopping can be used to average inter-cell interference in case of low-medium bit rate users.
4. Channel allocation differentiates between RT and NRT bearer services:
    - RT services: Channels remain allocated for the whole duration the bearer service is established. The allocated resources may change because of a channel reallocation procedure (e.g. VBR).
    - NRT services: Channels are allocated for the period of the transmission of a dedicated data packet only UDD channel allocation is performed using 'best effort strategy', i.e. resources available for NRT services are distributed to all admitted NRT services with pending transmission requests. The number of channels allocated for any NRT service is variable and depends at least on the number of current available resources and the number of NRT services attempting for packet transmission simultaneously. Additionally, prioritisation of admitted NRT services is possible.
5. Channel reallocation procedures (intra-cell handover) can be triggered for many reasons:
    - To cope with varying interference conditions.
    - In case of high rate RT services (i.e. services requiring multiple resource units) a 'channel reshuffling procedure' is required to prevent a fragmentation of the allocated codes over to many timeslots. This is achieved by freeing the least loaded timeslots (timeslots with minimum used codes) by performing a channel reallocation procedure.
    - When using smart antennas, channel reallocation is useful to keep spatially separated the different users in the same timeslot.

6.5.3 Power Control

Power control is applied for UTRA/TDD to limit the interference level within the system thus reducing the inter-cell interference level and to reduce the power consumption in the MS.

As mandatory power control scheme, a slow C-level based power control scheme (similar to GSM) is used both for up- and downlink. Power control is made individually for each resource unit (code) with the following characteristics:

Table 14. PC characteristics

	Uplink	**Downlink**
Dynamic range	80 dB	30 dB
Power control rate	variable; 1-800 cycles / second	variable; 1-800 cycles / second
Step size	1.5 … 3 dB	1.5 … 3 dB
Remarks	A cycle rate of 100 means that every frame the power level is controlled	within one timeslot the powers of all active codes are balanced to be within a range of 20 dB

- All codes within one timeslot allocated to the same bearer service use the same transmission power.
- For RT services, in UL and DL a closed loop power control is used
- For NRT services, both open loop power control and closed loop power control are used according to the MS state and the operators' needs
- The initial power value is based on the path-loss estimate to the serving BS
- In case of one user with simultaneous RT and NRT bearer service, the closed loop power control is used both for RT and NRT bearer service. However, depending on the current services different power levels are used.

Optional enhancements concerning power control for further study:

- Introduction of quality based power control

6.5.4 Cell Search

"Cell Search" is the procedure activated by the MS to find out suitable BS, which it can synchronise to.

Depending on the MS state, the cell search procedure can be performed in one of the following ways:

- Initial Mode Cell Search
- Idle Mode Cell Search
- Active Mode Cell Search

6.5.4.1 Initial Mode Cell Search

The Initial Mode Cell Search procedure is activated by the MS at the power on.

As soon as the MS has been powered on, it tries to find suitable BS to synchronise to. With "Suitable" means a BS broadcasting the identities of the system/network the MS has access rights to and whose reference power level is detected with the lowest path loss.

From the selected BS, the MS shall derive all the TDD-TDMA timings (i.e. chip, slot, frame configuration, multi-frame, super-frame synchronism), the frequency synchronisation and all the system information, which are required to access to the network services.

During the first step of the procedure, the MS scans, over an open time window of 10 ms, for the synchronisation pattern that a BS transmits on the Synchronisation Channel (SCH).

From the detection of the auto-correlation pulse, frequency; chip; time slot; frame and multi-frame synchronism and path loss measurement can be derived.

As a result of this first step, the MS has registered the TDMA timings from the strongest base station.

In a following step, the MS tries to detect from the Broadcast Control Channel (BCCH) of the locked BS all the other information (e.g. the switching point synchronism; the system identities; the RACH position etc.) which allows accessing the network services.

For this second step, the closed time window (i.e. a time window centred around the midamble field of the time slot) is used.

6.5.4.2 Idle Mode Cell Search

The Idle Mode Cell Search procedure is activated by the MS when it is already synchronised to a BS but has no physical channel allocated, that is when the MS is in the “Idle” state.

This procedure is activated when the locked BS is detected under a predetermined power threshold, which could also depend on the link quality.

In the Idle state, the MS has all the TDD-TDMA timings from the locked BS, but still monitors the radio environment in order to find out stronger BS.

In order to save power consumption, the MS can perform the radio monitoring periodically.

By receiving the SCH, BCCH of other BS, the MS learns the information for a possible new cell selection. Furthermore, the MS monitors all other time slots for interference measurements that are utilised by the BS later when the MS tries to get into the cell via the RACH mechanism.

When stronger BS is detected, the MS can lock to this new cell after checking access rights and aligning its TDD-TDMA timings (by the detection of the BCCH and SCH).

6.5.4.3 Active Mode Cell Search

The Active Mode Cell Search procedure is activated by the MS when it is already synchronised to a BS with which at least one physical channel is allocated, that is when the MS is in the “Active” State.

In the Active state, the MS periodically scans for the radio environment in order to keep updated the list of the strongest cells, in respect to the serving one, it can detects. The list, which contains the identity and the Received Signal Strength Intensity (RSSI) of each detected cell, is periodically forward to the serving BS (or it can be forwarded on demand) and can be used to perform inter-cell handovers of the allocated physical channels.

By receiving the SCH, BCCH of other BS, the MS learns about its radio environment. Furthermore, the MS monitors all time slots that are not occupied by the MS for interference measurements being utilised by the BS.

6.5.5 Random Access

The MS that needs to access the network services or needs more capacity shall transmit, to the selected BS, a random access burst on the Random Access Channel (RACH).

The RACH can be positioned in one or more time slots of the uplink part of the frame, as indicated by the Broadcast Control Channel (BCCH). The random access burst can be accommodated either in the first half or in the second half of the assigned time slot(s), so that the time slot capacity is doubled. A further improvement of the capacity and, as a consequence, a further reduction of collisions is achieved by allowing up to eight orthogonal codes per random access time slot.

The network can regulate the RACH use by allowing separate access groups of MS at a time.

Upon reception of a random access burst, the selected BS shall answer to the MS by sending an access grant message on the Forward Access Channel (FACH).

This message shall indicate the physical channels/time slots within the cell, which are assigned to the MS.

6.6 Additional features and options (TDD)

6.6.1 Joint detection

Joint detection of simultaneously active CDMA codes in the uplink as well as the downlink will already be performed in the introductory phase of the UTRA TDD mode. Therefore, this subject is treated in other sections of this system description.

6.6.2 Adaptive antennas

In the UTRA TDD-component, adaptive antennas are supported through the use of connection dedicated midamble sequences in both uplink and downlink (they are optional in the downlink). Moreover, the reciprocity between the uplink and the downlink channel facilitates an efficient implementation of smart antennas. Although the UTRA TDD component does not require the use of smart antennas, the resulting signal-to-interference-plus-noise-ratio (SINR) can significantly be improved by incorporating various smart antenna concepts at the base station on the uplink as well as the downlink.

These SINR gains may be exploited

- to increase the capacity,

e.g., by reducing the amount of interference suffered (BS receiver) and created (BS transmitter) in the system

- to increase the quality,
- to decrease the delay spread,
- to reduce the transmission powers,
- to reduce the electromagnetic pollution and user health hazards,
- to enhance spatial user location due to the estimation of the dominant directions of arrivals,

or a combination thereof. Three different smart antenna concepts, namely

- diversity antennas,
- sector antennas,
- and adaptive antenna arrays,

can be incorporated into the UTRA TDD mode.

6.6.3 Downlink transmit diversity

Downlink transmit diversity is supported by the UTRA TDD mode.

6.6.4 Locationing function support

The fact that the base stations in a local area are synchronised facilitates the implementation of mobile positioning algorithms in the UTRA TDD mode. Time delay or delay difference measurements to the base stations are obtained in a very efficient fashion. They are required as input for mobile positioning algorithm.

6.6.4.1 Relaying and ODMA

The UTRA TDD mode is a suitable platform for the support of relaying. Relaying is a widely used technique for radio packet data transmission both in commercial and military systems but it has so far not been widely used in cellular systems. Relaying has the potential among others

- to improve coverage and/or maximum user bit rates by reduced effective path loss, optimum link adaptation and link diversity and
- to increase capacity by lowering transmission powers and associated inter-cell interference.

The UTRA TDD design is sufficiently flexible to support both simple relaying and advanced relaying protocols such as Opportunity Driven Multiple Access (ODMA) with negligible increase to the MS complexity or cost.

ODMA supports packet data transfer between an origin and destination via a network of intermediate relay nodes (dedicated fixed relays or relaying enabled mobiles). TDD operation enables each node to receive other nodes‘ transmissions and build a connectivity table neighbours at each node exploiting path loss and delay information to. This table is subsequently used to route packets across a network in a dynamic manner without incurring a significant routing overhead.

6.6.4.2 Radio-Resource Organisation and Synchronisation

ODMA relaying requires MS to MS transmission allowing information to be sent from one mobile to another without passing via BS. Each MS can receive broadcast-signalling information over a large cell area. Reception of the broadcast information will allow frequency, chip and slot/frame synchronisation and determine connectivity/path loss to the BS. The BCCH will also indicate which physical channels are available for conventional use and which channels are reserved for MS-MS transmission. The MS-MS communications may use a different unpaired frequency channel to the one generating the BCCH. In fact it may be feasible for the broadcast cell to be FDD. The BS common channels will also be used for initial authentication and mobile location. An additional advantage of receiving a BCCH is that it avoids violating any RX before TX regulations that may apply to the mobile.

Figure 66 shows an example how conventional TDD and MS-MS can be incorporated in the same frame structure. The MS-MS resources are sub-divided into Calling and Traffic Channels. The Calling Channel is RACH like i.e. random access with collision risk and the Traffic Channels are used for MS-MS transfers after negotiation on the Calling Channel(s). Traffic Channels are preferably full time slots seized exclusively for one MS-MS communication. Multi-code transmission is used to achieve high throughput if necessary to avoid excessive delay supporting the ODMA operation. Use of higher order modulation would further assist even higher rate transmission.

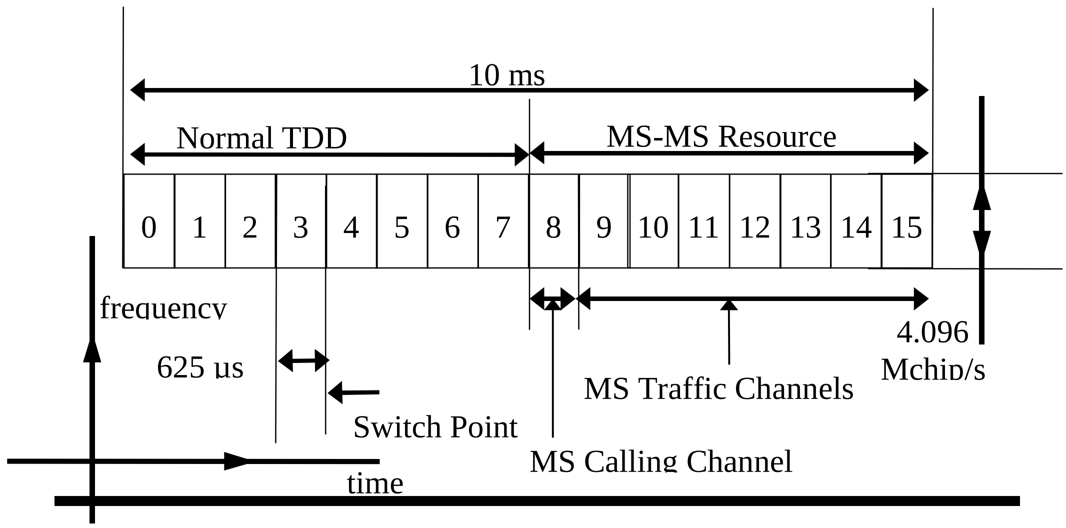


Figure 66. Example TDD Frame Structure Incorporating MS-MS Resources

An ODMA enabled MS will behave such that in MS-MS reserved slots it will be capable of receiving unless it has something to transmit and then it should be capable of transmitting in any of the MS-MS slots.

An MS can choose to just monitor the Calling Channel slot until it determines a need to also use Traffic Channels. This may be triggered by detecting its own address in a message field or by a requirement to source/sink data. To access a Calling Channel it is proposed that the MS transmits a type 1 or 2 traffic burst using a randomly chosen code. With joint detection the MS can simultaneously receive multiple signals with up to 30dB power difference and thus resolve collisions. It is assumed that MS-MS transmissions take place over micro-cellular range. Thus, the channel estimation can cope with the slight asynchronism between MS synchronised to the central BS.

The access to the traffic channels is based on a dynamic channel selection scheme based on interference measurements by the seizing MS. The seizure of channels by MS can be indicated in the calling channel burst structure as described below under addressing.

6.6.4.3 Idle-Mode Procedure

Each mobile requires some knowledge about other mobiles that it may communicate with and their relative connectivity. How it acquires this information could be implementation specific. For example an ODMA system would generate probe (RACH-like) signals to determine its neighbours and find end to end addresses. Probing is a mechanism used to indicate mobile activity in the ODMA network. When a mobile station is switched on for the first time it has no information about its surroundings. In this case the mobile will camp on one of the MS-MS CCH (after establishing synchronisation with the central BS) which are used by all mobile stations to receive and broadcast probes. With no ODMA system information stored in memory, the MS will begin a probing session, where the mobile initially camps on a CCH and periodically broadcasts a probe packet. The neighbour list will initially be empty. If another MS receives the broadcast packet it will register the probing MS as a neighbour and sends an addressed probe in response. The response probe is transmitted at random to avoid contention with other mobiles and typically one response is sent for every n broadcast probes received from a particular MS. The probe-response mechanism enables each MS to build a neighbour list that should contain at least 5 MS.

When using the probing approach initially there is no connectivity information and so the probe power must start low and ramp up until the required number of neighbouring MSs are determined. Additionally link adaptation mechanisms could be used for setting the local connectivity area to contain at least 5 other MS. Probe acknowledgements will appear on the Calling Channel (RACH-like). The acknowledgement will contain information to help refine the power control.

If the probing mechanism is allowed to occur at any time the MSs must RX continuously which may reduce battery life. To avoid this, a low duty cycle probing window, co-ordinated by BS, broadcast information can be used, i.e. the sleeping MSs wake up periodically to send and receive probes (e.g. every minute) and then go back to sleep. The window could be of the order of 0.5 seconds long. The BS has the capability to send a wakeup page to all the MSs via the BS’s paging channel. A sleeping MS that is then paged awake will stay active whilst it can detect local ODMA transmissions. If it has not participated in such communication for a timeout period it will fall asleep. Similarly it may decide to sleep after a long period of activity.

An alternative approach, if feasible, would be for some central intelligence to determine where all the mobiles are located, their relative connectivity and somehow pass this information in an efficient manner to the MS.

Other MSs monitoring the probe/acks will determine connectivity between the nodes and themselves and refine their own knowledge for future communications.

6.6.4.4 Addressing

There are 2 types of addressing to be considered, Relay-Relay and End-End i.e. the former manages a particular relay hop and the latter identifies the origin and destination of the relayed transmission - within the cell. Note that all messages will have a BTS as one end of a transmission - and so a BTS should have a special generic address e.g. 0. It is assumed that each mobile has some unique end-to-end address e.g. MSISDN. The MSISDN should not be used to address MS-MS transmissions, as these fields are unencrypted (or use encryption common to the cell). When an MS registers onto the network it may be given a temporary identity (like a TMSI) which can be used for relaying purposes. For efficiency the size of this identity (or derived version for relaying) should be kept to a minimum.

The probe information mapped onto traffic burst and transmitted on a Calling Channel contains in the first half of the transmission a message independent header and enables a relay transmission to be identified. The header reveals source and destination addressing, link quality and power control parameters and which resources (Traffic Channels) to be used next. The second part of the burst is message type dependent consisting of message type, source and destination and flow information as message number, creation time, time to die and time elapsed.

6.6.4.5 Call Set-up

When a MO wishes to start a call it makes a conventional RACH access to the BS. A conventional authentication/call set-up will take place but during the negotiation of resource it will be decided to use ODMA mode. Firstly the BS will send a broadcast wakeup page to the MS relays. The BS will then ask the originating MS to send a message to it via ODMA relaying which it then acknowledges. The initial route for these messages will be based on knowledge acquired from the background probing. Alternatively, the BS could indicate the route to be used to the mobile. The transmissions will be monitored by relays not directly involved in the link. These relays then determine connectivity routes between the MO and BTS and are available to make further transmissions more optimum and reliable. Other mobiles will fall asleep using the page-awake rules. A similar procedure is used for MT calls.

6.7 System scenarios

6.7.1 Uncoordinated operation

A system requirement for uncoordinated residential operation is that systems can be bought and installed independently. The reference points for power control will be different for the different systems and their spatial separation can be arbitrarily small. Also, time synchronisation is very difficult to obtain leading to the loss of orthogonality in the code domain not only in the uplink but also on the downlink. For these reasons it is has been considered a requirement that time orthogonality is achieved between residential systems operating in close proximity.

The consequence is that contrary to public systems that are synchronised and seek to maximise the interleaving gain and hence performance and capacity, residential systems need to occupy as few slots as possible. In this way, the scope for interference avoidance increases and more systems can be accommodated.

The unsynchronised base stations, upon installation and in periodic intervals thereafter, measure interference on all slots and transmit the common control slot in the optimal position with regard to the slots in the frame used by other systems.

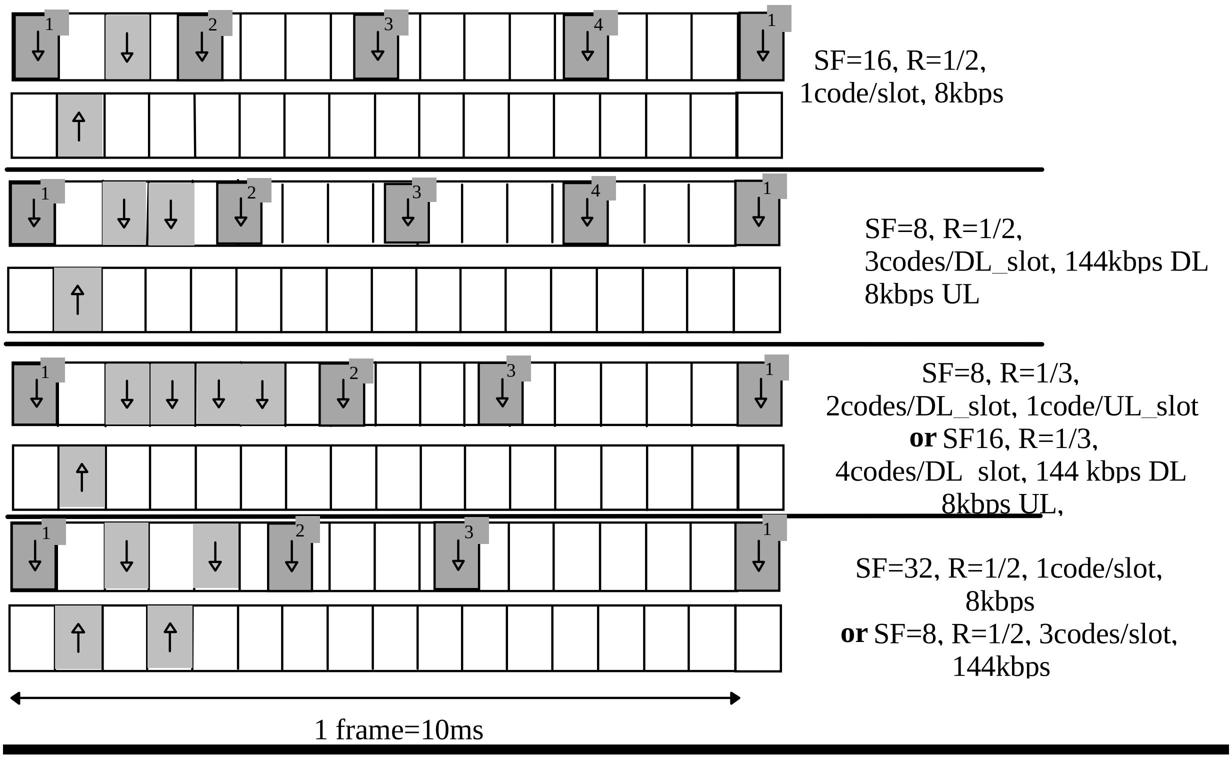


Figure 67. Frame structure and timing relation example for more than one system and several asymmetry patterns. The numbers in the grey boxes differentiate the common control channels of the different systems.

7. HANDOVER

7.1 Overview of handover types

The mobile station will support three types of handover:

- *Soft Handover*: A handover in which the mobile station communicates with a new base station without interrupting communications with the current serving base station. Soft handover can only be performed with base stations having identical frequency assignments.

- *UTRA to UTRA Hard Handover*: A handover in which the mobile is transitioned between disjoint sets of base stations, either because those base stations are operating on a different frequency assignment or in a different mode, (UTRA/FDD to UTRA/TDD or UTRA/FDD to UTRA/FDD or UTRA/TDD to UTRA/FDD or UTRA/TDD to UTRA/TDD handover), or same frequency when soft handover is either not possible or not needed.
- *UTRA to GSM Hard Handover:* A handover in which the mobile is directed from a UTRA traffic channel to a GSM traffic channel.

The soft handover and UTRA internal hard handover are briefly outlined below.

7.2 Soft and softer handover

7.2.1 Soft handover

When in active mode, the mobile station continuously searches for new base stations on the current carrier frequency. During the search, the mobile station monitors the received signal level from neighbouring base stations, compares them to a set of thresholds, and reports them accordingly back to the base station. Based on this information the network orders the mobile station to add or remove base station links from its *active set*. The *active set* is defined as the set of base stations from which the same user information is sent, simultaneously demodulated and coherently combined, i.e. the set of base stations involved in the soft handover.

From the cell-search procedure, the mobile station knows the frame offset of the Primary CCPCH of potential soft-handover candidates relative to that of the source base station(s) (the base stations currently within the active set). When a soft handover is to take place, this offset together with the frame offset between the downlink DPCH and the Primary CCPCH of the source base station, is used to calculate the required frame offset between the downlink DPCH and the Primary CCPCH of the destination base station (the base station to be added to the active set). This offset is chosen so that the frame offset between the downlink DPCH of the source and destination base stations at the mobile-station receiver is minimised. Note that the offset between the downlink DPCH and Primary CCPCH can only be adjusted in steps of one downlink DPCH symbol in order to preserve downlink orthogonality.

7.2.2 Softer handover

Softer handover is the special case of a soft handover between sectors/cells belonging to the same base station site. Conceptually, a softer handover is initiated and executed in the same way as an ordinary soft handover. The main differences are on the implementation level within the network. For softer handover, it is e.g. more feasible to do uplink maximum-ratio combining instead of selection combining as the combining is done on the Node B level rather than on the RNC level.

7.3 UTRA to UTRA hard handover

UTRA to UTRA inter-frequency hard handover may typically occur in the following situations:

- Handover between cells to which different number of carriers have been allocated, e.g. due to different capacity requirements (hot-spot scenarios).
- Handover between cells of different overlapping orthogonal cell layers using different carrier frequencies.
- Handover between different UTRA operators/systems using different carrier frequency.

A key requirement for the support of seamless inter-frequency handover is the possibility for the mobile station to carry out cell search on a carrier frequency different from the current one, without affecting the ordinary data flow. UTRA/FDD and UTRA/TDD supports inter-frequency cell search in two different ways, a dual-receiver approach and a slotted-downlink-transmission approach (see Section Coding for slotted mode for details).

7.4 UTRA/FDD - UTRA/TDD handover

For terminals with both FDD and TDD capability the handover between the UTRA modes can be used. Both modes use the same 10 ms frame length and can perform measurements on each other. The UTRA FDD mode can use the slotted mode or other measurement ways described in Section Coding for slotted mode to perform measurements on the UTRA TDD mode. The UTRA FDD mode must search first the downlink activity part(s) in the 10 ms frame. As the UTRA TDD cells within the area are frame synchronised, the downlink/uplink timing obtained for a single TDD cell is also valid for other cells belonging to the same network in the same area.

For the UTRA TDD mode, measurement time can be obtained between the activity periods (between uplink/downlink transmission) to facilitate sufficient measurement frequency from UTRA FDD cells.

In the FDD mode, the mobile is continuously transmitting and receiving information. In order to perform a handover to the TDD mode, it should be able to make measurements on TDD carriers. However, the spectral separation between FDD carriers and TDD carriers may not be sufficient in some cases to be able to implement a filter to protect the TDD receiver making the measurements. Therefore, the mobile might need to interrupt FDD transmission in order to perform measurements in the TDD band. This can be implemented through a slotted mode in the uplink direction similar to the one defined for the downlink transmission.

For both modes it is expected that the UTRA base station is able to indicate the channel numbers used for the FDD and TDD cells in the area as well as the base station spreading/scrambling codes used. This does not cover the unlicensed TDD use where handovers are not likely to happen as the networks are not likely to be inter-connected.

7.4.1 UTRA - GSM handover

The handover between UTRA and GSM system offering world-wide coverage already today has been one of the main design criteria taken into account in the UTRA frame timing definition. The GSM compatible multi-frame structure, with the super-frame being multiple of 120 ms, allows similar timing for inter-system measurements as in the GSM system itself. The compatibility in timing is important, that when operating in UTRA mode, a multi-mode terminal is able to catch the desired information from the synchronisation bursts in the synchronisation frame on a GSM carrier with the aid of the frequency correction burst. This way the relative timing between a GSM and UTRA carriers is maintained similar to the timing between two asynchronous GSM carriers.

7.4.1.1 UTRA/FDD to GSM handover

UTRA/FDD-GSM dual mode terminals can be implemented without simultaneous use of two receiver chains. Although the frame length is different from GSM frame length, the GSM traffic channel and UTRA FDD channels use similar 120 ms multi-frame structure. Similar timing can be naturally done with UTRA TDD mode as well.

A UTRA terminal can do the measurements either by requesting the measurement intervals in a form of slotted mode where there are breaks in the downlink transmission or then it can perform the measurements independently with a suitable measurement pattern. Independent measurements do not use slotted mode, but use dual receiver approach, where the GSM receiver branch can operate independently of the UTRA FDD receiver branch.

For smooth inter-operation between the systems, information needs to be exchanged between the systems, in order to allow UTRA base station to notify the terminal of the existing GSM frequencies in the area. Further more integrated operation is needed for the actual handover where the current service is maintained, taking naturally into account the lower data rate capabilities in GSM when compared to UMTS maximum data rates reaching all the way to 2 Mbits/s.

Measurements of GSM using slotted mode

6 ms idle periods (similar to that of GSM) can be created by using double-frame idle periods, as described in Section Coding for slotted mode. Therefore, it is possible to capture the GSM FCCH and SCH in the same way as in GSM-to-GSM handover. The GSM Frequency Correction Channel (FCCH) and GSM Synchronisation Channel (SCH) use one slot out of the eight GSM slots in the indicated frames with the FCCH frame with one time slot for FCCH always preceding the SCH frame with one time slot for SCH. The principle is indicated in Figure 68.

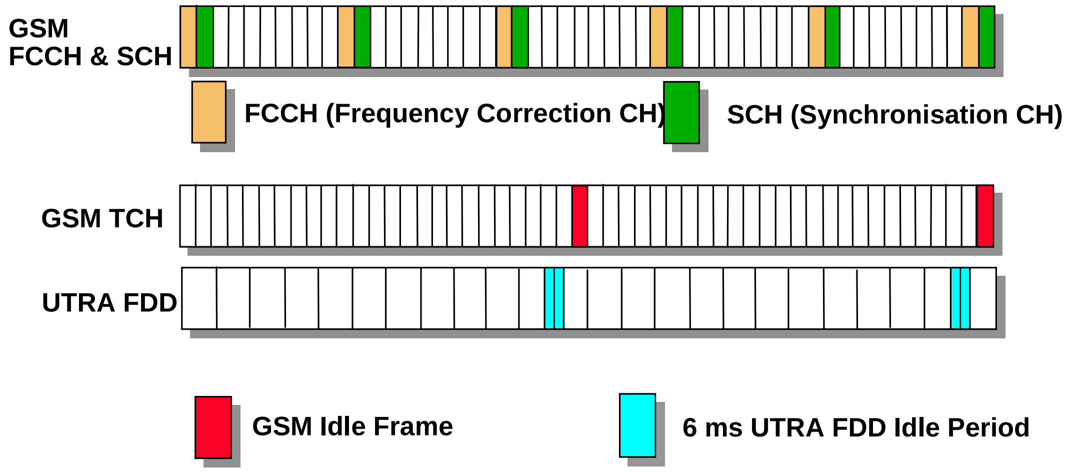


Figure 68. Example of GSM measurement timing relation between UTRA/FDD and GSM frame structures.

Alternatively, several shorter mid-frame idle periods (as described in Section Coding for slotted mode) with a certain spacing and every GSM super-frame, can be used to capture the GSM FCCH and SCH. For instance, two 3 ms idle periods every 120 ms, offset from each other by 30 ms, as illustrated in Figure 69.

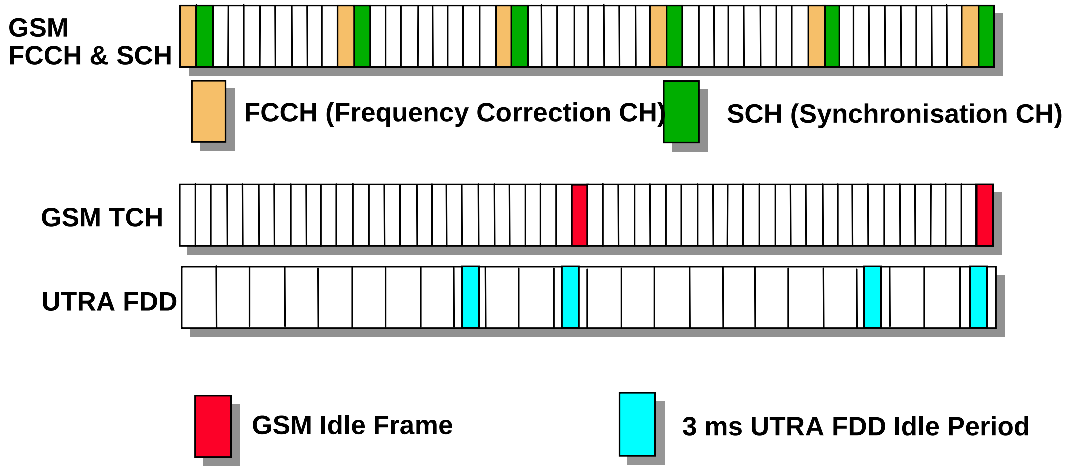


Figure 69. Another example of measurement timing relation between UTRA/FDD and GSM frame structures.

For the power measurements of GSM carriers, additional slotted frames will be used for single receiver FDD/GSM mobiles. Requirements concerning the number of power measurements per slotted frame are for further study.

Attachment 3

Updated Technology Description Template

A1.1	Test environment support
A1.1.1	In what test environments will the SRTT operate? **Answer:** Indoor office (I), Outdoor to indoor and pedestrian (P), Vehicular (V), and Mixed-cell pedestrian/vehicular (M)
A1.1.2	If the SRTT supports more than one test environment, what test environment does this technology description template address? **Answer:** The template addresses all four test environments listed in A.1.1.
A1.1.3	Does the SRTT include any feature in support of FWA application? Provide detail about impact of those features on the technical parameters provided in this template, stating whether the technical parameters provided apply for mobile as well as for FWA applications. **Answer:** The proposal can be used for FWA applications. There are no differences in the radio transmission technology parameters with respect to FWA than what is being used for the cellular applications. The flexibility of the RTT allows for an optimisation of the transmission and receiver chains according to the specific deployment scenario such as cellular or FWA. Indeed, the RTT is designed to be future proof taking advantage of extended range technologies such as adaptive antennas and antenna diversity in the downlink but also interference cancellation techniques. It is also possible to include those kinds of techniques later, if necessary, without requiring any frequency reconfiguration nor does it preclude the use of user equipment not supporting those techniques like antenna diversity in the downlink.
A1.2	Technical parameters Note: Parameters for both forward linear and reverse link should be described separately, if necessary.

A1.2.1	What is the minimum frequency band required to deploy the system (MHz)? **Answer:** FDD mode: 2×5 MHz TDD mode: 1×5 MHz With these spectrum allocations, up to 2 Mbps user rate is possible. However, note that these are the minimum spectrum requirements. Larger spectrum allocation is recommended for more efficient operation. A larger spectrum allocation supporting two or more 5 MHz carriers would e.g. allow for more efficent trunking or multiple cell layers.
A1.2.2	What is the duplex method: TDD or FDD? **Answer:** Both FDD and TDD modes are specified.

A1.2.2.1	What is the minimum up/down frequency separation for FDD? **<u>Answer:</u>** 130 MHz for the UMTS/IMT-2000 band. A different minimum up/down frequency separation may be applied for other frequency bands, e.g. for the American PCS band a separation of 80 MHz would apply...
A1.2.2.2	What is the requirement of transmit/receive isolation? Does the proposal require a duplexer in either the mobile or base station. **<u>Answer:</u>** FDD mode: Duplexer needed in mobile station. Required transmit/receive isolation: 50 dB (MS), 80 dB (BS). The required isolation in the BS can be achieved by separating the transmitter and receiver antennas together with an appropriate receiver filter. TDD mode: No duplexer needed.
A1.2.3	Does the RTT allow asymmetric transmission to use the available spectrum? Characterize. **<u>Answer:</u>** In both TDD and FDD modes asymmetric connections can be supported since it is possible to set uplink and downlink bearer service characteristics independently. <u>In the FDD mode:</u> On an overall system level, it is possible, with the FDD mode to assign more carriers to the downlink than uplink or vice versa. <u>In TDD mode:</u> The ratio of uplink to downlink capacity of a carrier can be adjusted by changing the ratio of the number of uplink and downlink time slots within the frame.
A1.2.4	What is the RF channel spacing (kHz)? In addition, does the SRTT use interleaved frequency allocation? Note: Interleaved frequency allocation; allocating the 2nd adjacent channel instead of adjacent channel at neighboring cluster cell is so called "interleaved frequency allocation". If a proponent is going to employ this allocation type, proponent should be stated at A1.2.4 and fill A1.2.15 of protection ratio for both of adjacent and 2nd adjacent channel. **<u>Answer:</u>** The RTT uses an RF channel raster of 200 kHz. A fixed RF channel spacing is not defined. The carrier spacing can be flexibly chosen, typically in the range 4.2-5.0 MHz (for 4.096 Mcps carrier) depending on the specific deployment scenario. The SRTT does not use interleaved frequency allocation.
A1.2.5	What is the bandwidth per duplex RF channel (MHz) measured at the 3 dB down points? It is given by (bandwidth per RF channel) x (1 for TDD and 2 for FDD). Please provide detail. **<u>Answer:</u>** FDD mode: ≈8.2 MHz, (≈16.4 MHzand ≈32.8 MHz for higher chip rates which are not yet described) TDD mode: 4.1 MHz

A1.2.5.1	Does the proposal offer multiple or variable RF channel bandwidth capability? If so, are multiple bandwidths or variable bandwidths provided for the purposes of compensating the transmission medium for impairments but intended to be feature transparent to the end user? **<u>Answer:</u>** The basic chip rate of the RTT is 4.096 Mcps corresponding to a channel bandwidth of approximately 5 MHz. Additional chip rates 8.192 Mcps and 16.384 Mcps, corresponding to bandwidths of approximately 10 MHz and 20 MHz respectively, are also specificied for the FDD mode. These bandwidths are seen as future evolution of the RTT towards even higher user rates (>2 Mbps). The different bandwidths are not used to compensate for transmission medium impairments. The different bandwidths are transparent to the end user.
A1.2.6	What is the RF channel bit rate (kbps)? NOTE 1 – The maximum modulation rate of RF (after channel encoding, adding of in-band control signalling and any overhead signalling) possible to transmit carrier over an RF channel, i.e. independent of access technology and of modulation schemes. **<u>Answer:</u>** FDD mode UL: 16/32/64/128/256/512/1024 kbps (BPSK, chip rate = 4.096 Mcps, variable SF = 4-256) FDD mode DL: 32/64/128/256/512/1024/2048 kbps (QPSK, chip rate = 4.096 Mcps, variable SF = 4-256) TDD mode UL/DL: 512/1024/2048/4096 kbps (QPSK, chip rate = 4.096 Mcps, variable SF = 2-16) Note 1: Multi-code transmission can be used to create composite RF channels with a combined bit rate exceeding the maximum values given above

A1.2.7	Frame Structure : Describe the frame structure to give sufficient information such as; - frame length - the number of time slots per frame - guard time or the number of guard bits - user information bit rate for each time slot - channel bit rate (after channel coding) - channel symbol rate (after modulation) - associated control channel (ACCH) bit rate - power control bit rate. Note 1: Channel coding may include FEC, CRC, ACCH, power control bits and guard bits. Provide detail. Note 2: Describe the frame structure for forward link and reverse link, respectively. Note 3: Describe the frame structure for each user information rate **Answer:** Refer to system description Frame length: 10 ms Number of time slots per frame: 16 (time slot = power control period for FDD) Guard time FDD mode: No guard time needed in FDD Guard time TDD mode: 23.4 μs User information bit rate for each time slot: Variable Channel bit rate (after channel coding and rate matching): FDD mode: UL: per IQ/branch (16/32/64/128/256/512/1024 kbps) DL: 32/64/128/256/512/1024/2048 kbps TDD mode: 512/1024/2048/4096 kbps Channel symbol rate (after modulation): FDD mode: (16/32/64/128/256/512/1024 ksps TDD mode: 256/512/1024/2048 ksps Associated control channel bit rate: Variable. From an SRTT point-of-view, associated control is not distinguished from traffic data. Power-control bit rate: The power control command rate is 1.6 kHz for FDD and 100-800 Hz for TDD See the detailed description of the proposal for more information.
A1.2.8	Does the RTT use frequency hopping? If so characterize and explain particularly the impact (e.g. improvements) on system performance. **Answer:** The RTT does not use frequency hopping.
A1.2.8.1	What is the hopping rate? **Answer:** N/A
A1.2.8.2	What is the number of the hopping frequency sets? **Answer:** N/A

A1.2.8.3	Are base stations synchronized or non-synchronized? **Answer: N/A**
A1.2.9	Does the RTT use spreading scheme? **Answer:** Yes, the RTT uses Direct-Sequence spreading.
A1.2.9.1	What is the chip rate (Mchip/s): Rate at input to modulator. **Answer:** FDD: 4.096 Mcps (8.192 Mcps, 16.384 Mcps) TDD: 4.096 Mcps
A1.2.9.2	What is the processing gain: 10 log (Chip rate / Information rate). **Answer:** The processing gain depends on the specific service. Assuming a span of the information rate of 100 bps to 2.048 Mbps, the processing gain varies in the range 46-3 dB for a 4.096 Mcps carrier.
A1.2.9.3	Explain the uplink and downlink code structures and provide the details about the types (e.g. PN code, Walsh code) and purposes (e.g. spreading, identification, etc.) of the codes. **Answer:** FDD mode: Channelisation codes (UL & DL): Orthogonal Variable Spreading Factor codes of length 2^k (for a 4.096 Mcps carrier) Short scrambling codes (UL): Complex MS-specific code of length 256 chips (based on extended Very-Large Kasami set). Long scrambling code (UL): Complex MS-specific code of length 10 ms (40960 chips for a 4.096 Mcps carrier). Segment of different long Gold codes. Scrambling code (DL): Real cell-specific code of length 10 ms (40960 chips for a 4.096 Mcps carrier). Segment of different long Gold codes. TDD mode: Channelisation codes (UL & DL) : Orthogonal codes of length 2-16 Scrambling codes (UL & DL): Cell-specific PN codes of length 2-16 See the detailed description of the proposal for more information.
A1.2.10	Which access technology does the proposal use: TDMA, FDMA, CDMA , hybrid, or a new technology? In the case of CDMA which type of CDMA is used: Frequency Hopping (FH) or Direct Sequence (DS) or hybrid? Characterize. **Answer:** FDD mode: Wide-band CDMA (Direct-Sequence). TDD mode: Wide-band TDMA/CDMA (Direct-Sequence).

A1.2.11	What is the baseband modulation technique? If both the data modulation and spreading modulation are required, please describe detail. What is the peak to average power ratio after baseband filtering (dB)? **Answer:** FDD mode: Data modulation: Dual-channel QPSK (UL), QPSK (DL) Spreading modulation: QPSK (UL), BPSK (DL). TDD mode: Data modulation: QPSK (UL & DL) Spreading modulation: QPSK (UL & DL) Root raised cosine pulse shaping, roll-off factor 0.22 for FDD and TDD. Resulting crest factor in the order of 5 dB for a single code case and up to 9 dB (TDD-mode) for the multi-code transmission.

A1.2.12	What are the channel coding (error handling) rate and form for both the forward and reverse links? e.g. - Does the SRTT adopt FEC (Forward Error Correction) or other schemes? **<u>Answer:</u>** Default FEC: • Convolutional inner code (rate 1/3 or rate 1/2, constraint length K=9). • Optional outer Reed-Solomon code (rate TBD) for BER=10^{-6} circuit-switched services. • The use of Turbo codes for high-rate services is under consideration and will most likely be adopted. However, the current evaluation results do not include Turbo codes. • Special FEC schemes, e.g. unequal error protection can be applied. - Does the SRTT adopt unequal error protection? Please provide details. **<u>Answer:</u>** Unequal error protection can be applied (see above). - Does the SRTT adopt soft decision decoding or hard decision decoding? Please provide details. **<u>Answer:</u>** The decoding scheme of the SRTT is a receiver implementation issue and is not covered by the RTT description. There is nothing in the RTT that prevents the use of either soft or hard decision decoding. - Does the SRTT adopt iterative decoding (e.g. turbo codes)? Please provide details. **<u>Answer:</u>** Turbo codes are under consideration (see above). See the detailed description of the proposal for more information.
A1.2.13	What is the bit-interleaving scheme? Provide detailed description for both up link and down link. **<u>Answer:</u>** Inner interleaving: Block interleaving with different interleaver spans (10 ms, 20 ms, 40 ms, 80 ms). Inter-frame interleaving (>10 ms) is applied on a transport-channel basis. Final intra-frame interleaving (10 ms) is applied after transport-channel multiplexing. Optional outer interleaving: Block interleaving with different interleaver spans (10 ms, 20 ms, 40 ms, 80 ms).

A1.2.14	Describe the taken approach for the receivers (MS and BS) to cope with multipath propagation effects (e.g. via equalizer, RAKE receiver, etc.). **<u>Answer:</u>** FDD: A RAKE receiver or any other suitable receiver structures can coherently combine multiple paths and give diversity gains (the detailed receiver structure is implementation dependent). Phase reference in the form of pilot symbols is available on both transmission directions. TDD: Typically, joint detection is used to coherently detect the data corresponding to different CDMA codes and copes with multipath propagation effects at the MS as well as the BS (the detailed receiver structure is implementation dependent). Phase reference in the form of a pilot sequence is available in both transmission directions.
A1.2.14.1	Describe the robustness to intersymbol interference and the specific delay spread profiles that are best or worst for the proposal. **<u>Answer:</u>** Within that range, the *size* of the delay spread does not, in itself, have any impact on the performance. On the other hand, the *shape* of the delay spread may have an impact on the performance. <u>FDD:</u> The interference due to time dispersion is suppressed by the processing gain. The exact performance depends on the number of Rake fingers and the search window. <u>TDD:</u> Intersymbol interference is eliminated in the data detection process typically due to the application of a Joint detection equalizer (the detailed detector structure is implantation dependent)
A1.2.14.2	Can rapidly changing delay spread profiles be accommodated? Please describe. **<u>Answer:</u>** <u>FDD:</u> Variations in the delay spread profiles in terms of amplitude and phase variations can be tracked at least on a slot-by-slot (0.625 ms) basis. Additional variations in the delay spread profile, such as the appearance/disappearance of rays can be tracked at least on a frame-by-frame basis. <u>TDD:</u> All variations in the delay spread profile, including amplitude and phase variations as well as the the appearance/disappearance of rays can be tracked at least on a slot-by-slot (0.625 ms) basis.
A1.2.15	What is the Adjacent channel protection ratio? In order to maintain robustness to adjacent channel interference, the SRTT should have some receiver characteristics that can withstand higher power adjacent channel interference. Specify the maximum allowed relative level of adjacent RF channel power in dBc. Please provide detail how this figure is assumed. **<u>Answer:</u>** Preliminary studies has indicated that Adjacent Channel Protection (ACP) in the order of 30 to 35 dB will be sufficient between the operators in an uncoordinated operation. For co-sited cells, the requirements are even less stringent.
A1.2.16	Power classes

A1.2.16.1	Mobile terminal emitted power: What is the radiated antenna power measured at the antenna? For terrestrial component, please give (in dBm). For satellite component, the mobile terminal emitted power should be given in EIRP (dBm). **<u>Answer:</u>** Not limited by the RTT.
A1.2.16.1.1	What is the maximum peak power transmitted while in active or busy state? **<u>Answer:</u>** Not limited by the RTT, typically 30 dBm and less.
A1.2.16.1.2	What is the time average power transmitted while in active or busy state? Provide detailed explanation used to calculate this time average power. **<u>Answer:</u>** <u>FDD mode:</u> Activity is 100 % if a mobile operates a dedicated channel. For packet transmission of the common channels smaller TX active cycles possible. <u>TDD mode:</u> 1 code (peak/average ratio 3.2 dB): Min. 14.8 dBm (1 timeslot), Max. 26.5 dBm (15 timeslots)8 codes (peak/average ratio 8.7dB): Min. 9.2 dBm (1 timeslot used), Max. 21 dBm (15 timeslots used) Calculation: Time average power =30dBm-peak/average ratio+10*log10(used timeslots/frame/16).
A1.2.16.2	Base station transmit power per RF carrier for terrestrial component
A1.2.16.2.1	What is the maximum peak transmitted power per RF carrier radiated from antenna? **<u>Answer:</u>** Not limited by the RTT.
A1.2.16.2.2	What is the average transmitted power per RF carrier radiated from antenna? **<u>Answer:</u>** Not limited by the RTT.
A1.2.17	What is the maximum number of voice channels available per RF channel that can be supported at one base station with 1 RF channel (TDD systems) or 1 duplex RF channel pair (FDD systems), while still meeting G.726 performance requirements? **<u>Answer:</u>** FDD mode: There are a maximum of 256 orthogonal downlink channels available, some of which must be allocated for downlink control channels. This leaves approximately 250 orthogonal channels for user traffic, such as voice. Normally, the cell capacity is interference limited, i.e. the actual number of voice channels is lower than this number (exact number of voice channels depends on operational conditions). Uplink is never limited by number of orthogonal code channels, as the orthogonal code tree used is user specific in the uplink. In some cases, e.g. for the case when adaptive antennas are used, the number of voice channels per BS can be increased above 250 by applying multiple non-orthogonal code sets on the downlink. TDD mode: There are a maximum of 128 orthogonal downlink channels available, some of which are allocated for downlink control channels. This leaves approximately 120 orthogonal channels for user traffic. The reason why the maximum number of channels in TDD is only 50% of that in FDD is the UL/DL sharing of one 5 MHz carrier in TDD.

A1.2.18	Variable bit rate capabilities: Describe the ways the proposal is able to handle variable base band transmission rates. For example, does the SRTT use: -adaptive source and channel coding as a function of RF signal quality **<u>Answer:</u>** Source coding is not part of the RTT. Adaptive source coding as a function of RF quality is possible. Adaptive channel coding as a function of RF signal quality is possible. - Variable data rate as a function of user application? **<u>Answer:</u>** The user rate can vary on a 10 ms basis. See A1.2.18.1 - Variable voice/data channel utilization as a function of traffic mix requirements? **<u>Answer:</u>** The RTT allows for variable voice/data channel utilisation as a function of traffic mix requirements. Characterise how the bit rate modification is performed. In addition, what are the advantages of your system proposal associated with variable bit rate capabilities? **<u>Answer:</u>** FDD: Different channel bit rates are possible by changing the spreading factor in factors of 2 from 256 down to 4. For the highest rates, multi-code transmission, i.e. transmission on several parallel code channels, is used. TDD: Different channel bit rates are possible by allocating a variable number of timeslots and a variable number of codes to a connection and variable spreading codes. On the uplink an arbitrary user bit rate after channel coding is matched to the closest possible channel bit rate by code puncturing/repetition. On the downlink an arbitrary user bit rate is matched to the chosen channel bit rate by discontinuous transmission. For variable-rate transmission, the rate can vary on a 10 ms basis. Explicit rate information, to simplify decoding, may be transmitted on a physical control channel. Multiple variable services can be multiplexed on one variable-rate physical channel or multiplexed on different variable-rate physical channels. The advantages with this approach are that the bit rate can be varied on a frame-by-frame basis without any explicit resource allocation and negotiation. It also caters for the independent quality control of each service on a multi-service connection.

A1.2.18.1	What are the user information bit rates in each variable bit rate mode? **<u>Answer:</u>** <u>FDD mode</u> Variable user bit rates between 0 and 2.048 Mbit/s can be supported with 100 bit/s granularity, with adjustments possible on a frame by frame basis. For a given connection, a sub-set of these rates is chosen at call set-up. During the call, the rate can be varied between the rates within the sub-set on a frame by frame basis. The sub-set of rates can also be changed during a call, e.g. due to the addition or removal of services. <u>TDD mode</u> Variable user bit rates between 0 and 2.048 Mbit/s can be supported with a high degree of flexibility by adjusting the number of codes and time slots used as well as by adjusting the channel coding and burst types used.

A1.2.19	What kind of voice coding scheme or codec is assumed to be used in proposed RTT? If the existing specific voice coding scheme or codec is to be used, give the name of it. If a special voice coding scheme or codec (e.g. those not standardized in standardization bodies such as ITU) is indispensable for the proposed RTT, provide detail, e.g. scheme, algorithm, coding rates, coding delays and the number of stocastic code books. **Answer:** Different voice coding schemes can be supported since the supported RTT has a flexible bearer capability supporting different bit rate allocation and voice coding frame length (e.g. 10 ms and 20 ms). Voice coding schemes envisaged to be used are the voice codecs used in the GSM system, e.g. EFR and AMR coding schemes. The AMR is to be finalised end'98.
A1.2.19.1	Does the proposal offer multplex voice coding rate capability? Provide detail. **Answer:** The RTT supports multiple voice coding rates through the chosen subset of the possible user bit rates as indicated in A1.2.18.1.*}*
A1.2.20	Data services: Are there particular aspects of the proposed technologies which are applicable for the provision of circuit-switched, packet-switched or other data services like asymmetric data services? For each service class (A, B, C and D) a description of SRTT services should be provided, at least in terms of bit rate, delay and BER/FER. Note 1: See [draft new] Recommendation [FPLMTS.TMLG] for the definition of - "circuit transfer mode" - "packet transfer mode" - "connectionless service" and for the aid of understanding "circuit switched" and "packet switched" data services Note 2: See ITU-T Recommendation I.362 for details about the service classes A, B, C and D **Answer:** All service classes can be supported with the proposed RTT. The pooling of resource units bearer services at the radio interface with various data rates can be achieved. Further, by variation of the spreading factor, power, coding rate and interleaving depth various BER and delay requirements can be met. For each service class dedicated bearer services at the radio interface are defined. The bearer services at the radio interface are separated into low delay data (LDD), long constrained delay (LCD) and unconstrained delay data (UDD) bearer services. The LDD bearer is characterised by stringent delay (and stringent delay variation) requirements. In contrary, the LCD bearer is characterised by less stringent delay (and delay variation) requirements but more stringent BER requirements. Both LDD and LCD bearers can have a constant or variable bit rate. Finally, the UDD bearer is characterised by unconstrained delay requirements. The following mapping may be used: • Class A: LDD • Class B: LDD-VBR • Class C: LCD • Class D: UDD

A1.2.20.1	For delay constrained, connection oriented. (Class A) **Answer:** The RTT provides user bit rates up to 2048 kbps. It can be set to any required value needed by a particular service. For instance, the ISDN services N*64 kbps (up to 2048 kbps), where N is an integer, can easily be supported. Various bit error ratios and/or frame error ratios and/or delays (see other related items) are supported depending on what the service demands. The RTT thus provides a flexible bearer concept. For delay constrained, connection oriented (Class A). The following non-comprehensive list gives some example supported LDD services: • 8 kbit/s; Delay 20 ms; BER < 10^{-3} • 144 kbit/s; Delay 50 ms; BER < 10^{-6} • 384 kbit/s; Delay 50 ms; BER < 10^{-6}
A1.2.20.2	For delay constrained, connection oriented, variable bit rate (Class B) **Answer:** Connection oriented variable bit rate service is supported since the bit rate can be controlled both on the physical layer and on higher layers. Up to 2048 kbps is supported. See also A1.2.18.1. The required QoS can be set according to what the service requires. For delay constrained, connection oriented variable bit rate (Class B). The following non-comprehensive list gives some example supported LDD-VBR services: • Peak data rate 64 kbit/s; Delay 50 ms; BER < 10-6; Granularity: 16 kbit/s • Peak data rate 144 kbit/s; Delay 50 ms; BER < 10-6; Granularity: 16 kbit/s • Peak data rate 384 kbit/s; Delay 50 ms; BER < 10-6; Granularity: 16 kbit/s • Peak data rate 2048 kbit/s; Delay 50 ms; BER < 10-6; Granularity: 32 kbit/s
A1.2.20.3	For delay unconstrained, connection oriented. (Class C) **Answer:** Class C services, e.g. best effort type, are supported. Depending on the service definitions different QoS levels can be supported individually per user. For delay unconstrained, connection oriented (Class C). The following non-comprehensive list gives some example supported LCD services: • 64 kbit/s; Delay 300 ms; BER < 10^{-6} • 144 kbit/s; Delay 300 ms; BER < 10^{-6} • 384 kbit/s; Delay 300 ms; BER < 10^{-6} • 2048 kbit/s; Delay 300 ms; BER < 10^{-6}

A1.2.20.4	For delay unconstrained, connectionless. (Class D) **<u>Answer:</u>** The answer to A1.2.20.3 applies. Connectionless or connection oriented packet services are considered to be a higher layer issue since with connection less service it is assumed that no end-to-end connection exists, i.e. final address is included in each packet. With connection oriented service it is meant that an end-to-end connection is set up and then transferring the data without explicitly mentioning the final address. Looking at the individual radio link both modes can of course be supported by the RTT if the upper layers support both modes. For delay unconstrained, connectionless (Class D). The following non-comprehensive list gives some example supported UDD services: • 64 kbit/s; Delay unconstrained; BER < 10^{-8} • 144 kbit/s; Delay unconstrained; BER < 10^{-8} • 384 kbit/s; Delay unconstrained; BER < 10^{-8} • 2048 kbit/s; Delay unconstrained; BER < 10^{-8}
A1.2.21	Simultaneous voice/data services: Is the proposal capable of providing multiple user services simultaneously with appropriate channel capacity assignment? Note : The followings describe the different techniques that are inherent or improve to a great extent the technology described above to be presented: Description for both BS and MS are required in attributes from A2.22 through A1.2.23.2. **<u>Answer:</u>** Parallel services can be provided. The different services can have independent bit rate, bit-error rate, delay, etc., and can have different transfer modes (packet/circuit-switched).
A1.2.22	Power control characteristics: Is power control scheme included in the proposal? Characterize the impact (e.g. improvements) of supported power control schemes on system performance. **<u>Answer:</u>** 3 types of power control are employed. One is the ‘fast closed loop power control’ (FDD mode only) which counters fading on a slot basis (0.625 ms): it is based on measurements on SIR. The second one is the ‘open loop power control’, it is used only for the initial power setting. The third one is the ‘outer loop power control’: it is based on BER and FER measurements. It has the role to change the target C/I, when the situation of the mobile is changing or for power control planning. It is done on a longer period basis. The use of fast power control significantly improves the link-performance (BER as a function of E_b/N_0) especially in the case slow-moving mobile stations. For fast moving mobile stations (>100 km/h), there is less performance improvement due to fast power control.
A1.2.22.1	What is the power control step size in dB? **<u>Answer:</u>** FDD mode: UL: Variable on a cell basis in the range 0.25-1.5 dB DL: Variable on a cell basis in the range 0.25-1.5 dB TDD mode: The power control step size is variable, range 1.5 to 3 dB..

A1.2.22.2	What are the number of power control cycles per second? **<u>Answer:</u>** FDD mode: 1600 TDD mode: 100-800 depending on the exact UL/DL time slot allocation
A1.2.22.3	What is the power control dynamic range in dB? **<u>Answer:</u>** UL: 80 dB DL: 30 dB
A1.2.22.4	What is the minimum transmit power level with power control? **<u>Answer:</u>** <-50 dBm for MS with highest power class (Maximum MS power <30 dBm, Power-control dynamics > 80 dBm)
A1.2.22.5	What is the residual power variation after power control when RTT is operating? Please provide details about the circumstances (e.g. in terms of system characteristics, environment, deployment, MS-speed, etc.) under which this residual power variation appears and which impact it has on the system performance. **<u>Answer:</u>** The residual power variation depends on the channel conditions, (Doppler spread and frequency selectivity) and are difficult to specify in detail. The residual power variations are fully included in the link-budget and capacity evaluations.
A1.2.23	Diversity combining in mobile station and base station: Are diversity combining schemes incorporated in the design of the RTT? **<u>Answer:</u>** Yes

A1.2.23.1	Describe the diversity techniques applied in the mobile station and at the base station, including micro diversity and macro diversity, characterizing the type of diversity used, for example: - time diversity : repetition, RAKE-receiver, etc., - space diversity : multiple sectors, multiple satellite, etc., - frequency diversity : FH, wideband transmission, etc., - code diversity : multiple PN codes, multiple FH code, etc., - other scheme. Characterize the diversity combining algorithm, for example, switch diversity, maximal ratio combining, equal gain combining. Additionally, provide supporting values for the number of receivers (or demodulators) per cell per mobile user. State the dB of performance improvement introduced by the use of diversity. **Answer:** Time diversity: Channel coding and interleaving in both uplink and downlink. Multipath diversity: RAKE, joint detection or similar receiver structure with, typically, maximum ratio combining is used in both BS and MS (implementation dependent). Space diversity: Receive antenna diversity with, typically, maximum ratio combining can be used in both uplink and downlink. Transmit antenna diversity is under consideration for downlink. Macro diversity: Soft (inter-site) handover with, typically, maximum ratio combining in downlink, selection combining in uplink. Softer (inter-sector) handover with, typically. Maximum ratio combining in both uplink and downlink. Frequency diversity: Wideband carrier (equivalent to multi-path diversity). For the mobile station: what is the minimum number of RF receivers (or demodulators) per mobile unit and what is the minimum number of antennas per mobile unit required for the purpose of diversity reception? These numbers should be consistent to that assumed in the link budget template in Annex 2 and that assumed in the calculation of the “capacity” defined at A1.3.1.5. **Answer:** One RF receiver per mobile unit. One antenna per mobile unit.
A1.2.23.2	What is the degree of improvement expected in dB? Please also indicate the assumed condition such as BER and FER. **Answer:** For receiver antenna diversity the diversity gain is 2.5 - 3.5 dB in required E_b/N_o for BER=10^{-3}. If power control is disabled the gain is much higher for the low speed cases. On top of the gain in reduced required E_b/N_o there is a gain in decreased transmitted power. This gain can be up to 2.5 dB, depending on the environment. Transmit diversity can also be employed, especially in the downlink. A gain similar to the gain with receiver antenna diversity is expected. All other diversity methods are inherent parts of the RTT concept and therefore it is difficult to specify an explicit diversity gain figure in dB.

A1.2.24	Handover/Automatic Radio Link Transfer (ALT): Do the radio transmission technologies support handover? Characterize the type of handover strategy (or strategies) which may be supported, e.g. mobile station assisted handover. Give explanations on potential advantages, e.g. possible choice of handover algorithms. Provide evidence whenever possible. **Answer** The RTT supports automatic handover. FDD: The handover scheme is based on a mobile assisted soft/softer handover mechanism and hard handover. The mobile station (MS) monitors the pilot signal levels received from neighbouring base stations and reports to the network pilots crossing or above a given set of dynamic thresholds. Based on this information the network orders the MS to add or remove pilots from its *Active Set*. The *Active Set* is defined as the set of base station for which user signal is simultaneously demodulated and coherently combined. The same user information modulated by the appropriate base station code is sent from multiple base stations. Coherent combining of the different signals from different sectored antennas, from different base stations, or from the same antenna but on different multiple path components is performed in the MS by the usage of Rake receivers. Base stations with which the mobile station is in soft handover process the signal transmitted by a mobile station. The received signal from different sectors of a base station (cell) can be combined in the base station, and the received signal from different base stations (cells) can be combined at the radio network controller. Soft handover results in increased coverage range on the uplink. This soft handover mechanism results in truly seamless handover without any disruption of service. The spatial diversity obtained reduces the frame error rate in the handover regions and allows for improved performance in difficult radio environment. Furthermore, the RTT support various types of hard handover, e.g. inter-frequency handover. The measurements to detect other frequency carriers available are made possible through the use of measurement slots. TDD: TDD provides two different handover mechanisms depending on the service type: connection oriented or packet services. For connection oriented services, the basic HO scheme is a mobile assisted, network evaluated and decided hard handover using backward signalling. Appropriate measures are provided to accelerate the HO procedure, e.g. in case of a corner effect. Furthermore the proposed RTT does not prevent the introduction of soft handover. The support of soft handover is for further study. For packet services, the basic HO scheme is a mobile evaluated and decided hard handover with background network control using forward signalling (cell reselection). Potential advantages: • Seamless HO for connection oriented, loss-less HO for packet bearer services • Mobile assisted network evaluated and decided handover scheme is most appropriate for RT services since it allows for both high flexibility in HO-algorithm design and implementation, e.g. to meet operator specific requirements in various deployment scenarios and system stability at high capacity. • Mobile evaluated and decided handover with network background control is most appropriate for packet services since it allows for both resource savings on the air interface by decentralised decision making and system integrity at high capacity by network co-ordinating measures.

A1.2.24.1	What is the break duration (sec) when a handover is executed? In this evaluation, a detailed description of the impact of the handover on the service performance should also be given. Explain how the estimate derived. **Answer:** FDD: Soft handover: No break duration (make before break) Hard handover: no loss for packet services due to ARQ TDD: For the basic scheme of hard HO the break duration on HO execution is the time interval between suspension of transmission on the traffic and signalling channels of the serving cell and the successful establishment of these on the new target cell. This time is mainly dependent on the access procedure to the target cell. Since cells are assumed to be synchronised on a frame basis a synchronous handover is executed, i.e. the MS performs a HO access onto the traffic channels of the new cell with known synchronisation resulting in a very short HO execution time. Impact of HO on service performance: • RT services: since the break duration is very short, seamless HO is possible • NRT services: ARQ mechanism ensures loss-less HO

A1.2.24.2	For the proposed SRTT, can handover cope with rapid decrease in signal strength (e.g. street corner effect)? Give a detailed description of - the way the handover detected, initiated and executed, - how long each of this action lasts (minimum/maximum time in msec), - the timeout periods for these actions. **Answer:** FDD: The MS continuously searches for signal from new and existing BS. It also maintains two thresholds (e.g. pilot Ec/Io) based on current combined quality of the down link soft handover legs to add newly detected BS or to drop existing BS from its soft handover 'active' set. The need to add or drop is sent in a message to the network, which determines whether or not to execute the addition or deletion. The time it takes to perform the above actions depends on the searcher and fixed infrastructure. When compared to the initial cell access the procedure is much faster as only the base stations indicated in the neighbour set need to be searched and thus the search time is greatly reduced and thus dependent on the size of the base station set to be searched. There is no time out period when soft or softer handover is performed. TDD: The HO functionality can successfully cope with rapid field drop effects like e.g. the street corner effect. Special means are introduced to speed up the process during each phase of the HO: 1. Detection and initiation: • Fast measurement acquisition and neighbour cell identification due to synchronised network • Signal strength trend analysis based on variable averaging window size and threshold comparison as well as MS speed and MS moving direction estimates 2. Decision: • Network handles this HO type with highest priority • default (hot standby target cells) may be used 3. Execution: For the TDD mode the network may be synchronised on frame basis. The handover execution procedures can take this in account to execute the HO in a very short time.
A1.2.25	Characterize how does the proposed SRTT react to the system deployment in terms of the evolution of coverage and capacity (e.g. necessity to add new cells and/or new carriers) particularly in terms of frequency planning. **Answer:** No frequency planning needed due to frequency reuse 1.

A1.2.26	Sharing frequency band capabilities: To what degree is the proposal able to deal with spectrum sharing among IMT-2000systems as well as with all other systems: - spectrum sharing between operators, - spectrum sharing between terrestrial and satellite IMT-2000 systems, - spectrum sharing between IMT-2000 and non-IMT-2000 systems, - other sharing schemes. **Answer:** For both FDD-mode and TDD-mode, sharing is always possible through frequency division. Furthermore, for the TDD-mode, sharing the same frequency with another TDD system including non-UMTS/IMT-2000 systems such as DECT and PHS is also possible due to the TDMA component. Both fixed time division and interference avoidance in the time domain using Dynamic Channel Allocation (DCA) can be used for this purpose. In addition, since ODMA relays do not own dedicated radio resource but share it in an asynchronous fashion with neighbouring nodes, they are tolerant to spectrum sharing.

A1.2.27	Dynamic channel allocation: Characterize the DCA schemes which may be supported and characterize their impact on system performance (e.g. in terms of adaptability to varying interference conditions, adaptability to varying traffic conditions, capability to avoid frequency planning, impact on the reuse distance, etc.) **Answer:** FDD mode: DCA not generally needed for the FDD mode. TDD mode: Slow DCA (allocation of time slots to cells) and fast DCA (allocation of a channel to a certain call) can be distinguished. Additionally for the TDD mode the allocation of a slot in a cell to uplink or download traffic is managed by the slow DCA, too. The interference on different slots in the time frame may be different. Therefore, a DCA algorithm that allocates the least interfered slots to ongoing calls with high QoS requirements results in a considerable gain in quality and/or capacity. The capability to vary the ratio of slots allocated in the uplink and in the downlink allows an optimal adaptation to the traffic asymmetry. Since synchronised base stations are used, advanced combinations of fast and slow DCA can be implemented in order to allocate the maximum amount of resources to the cell with the momentarily highest amount of traffic.
A1.2.28	Mixed cell architecture: How well do the technologies accommodate mixed cell architectures (pico, micro and macrocells)? Does the proposal provide pico, micro and macro cell user service in a single licensed spectrum assignment, with handoff as required between them? (terrestrial component only) Note: Cell definitions are as follows: pico - cell hex radius (r) < 100 m micro - 100 m < (r) < 1000 m macro - (r) > 1000 m **Answer:** Seamless handover is possible between cell layers.

A1.2.29	Describe any battery saver / intermittent reception capability **Answer:** In dedicated mode, i.e. during calls and in circuit oriented operation, the transmitter/receiver is continuously on (FDD) and transmits/receives only in allocated timeslots (TDD), respectively. Variable rate transmission is utilised whenever possible to reduce needed transmitted power. Employing power control methods also reduces the transmitted power levels to a minimum. With packet traffic, depending on the packet-access mode, the receiver and transmitter can be used only periodically, i.e. being switched off until data is available for transmission or the base station indicates the mobile station that is to be received. In the latter case, the polling is done according to A1.2.29.1, which also explains the power saving during idle mode operation of the terminal.
A1.2.29.1	Ability of the mobile station to conserve standby battery power: Please provide details about how the proposal conserve standby battery power. **Answer:** In idle mode, the mobile station uses a sleep mode that permits that e.g. most of the circuits can be turned off during the periods when the mobile station is not receiving. The mobile station is only awakened for short periods to listen to e.g. the paging channel or the broadcast channel. See the detailed description of the proposal for more information.
A1.2.30	Signaling transmission scheme: If the proposed system will use radio transmission technologies for signaling transmission different from those for user data transmission, describe details of signaling transmission scheme over the radio interface between terminals and base (satellite) stations. **Answer:** The signalling scheme for the RTT is basically the same as for user data. User data and signalling are using the same L1 services.
A1.2.30.1	Describe the different signaling transfer schemes which may be supported, e.g. in connection with a call, outside a call. Does the SRTT support new techniques? Characterize. Does the SRTT support signalling enhancements for the delivery of multimedia services? Characterize. **Answer:** The RTT does not limit the use of any advanced techniques. The physical layer provides means for transmission rate signalling which can be used also to indicate which services are active and thus introduction of an associated control channel with service negotiation is supported by the RTT.

A1.2.31	Does the SRTT support a Bandwidth on Demand (BOD) capability? Bandwidth on Demand refers specifically to the ability of an end-user to request multi-bearer services. Typically this is given as the capacity in the form of bits per second of throughput. Multi bearer services can be implemented by using such technologies as multi carrier, multi time slot or multi codes. If so, characterize these capabilities. Note: BOD does not refer to the self-adaptive feature of the radio channel to cope with changes in the transmission quality (see A1.2.5.1). **Answer:** Bandwidth on demand is supported in the range 0 bps to 2.048 Mbps user bit rate. FDD mode: The bandwidth on demand possibility is implemented by one or more multi-codes and variable spreading factor (4-256). TDD mode: The bandwidth-on-demand possibility is implemented by multi-code (assigning more than one code) and multi-slot (assigning more than one time slot) transmission.
A1.2.32	Does the SRTT support channel aggregation capability to achieve higher user bit rates? **Answer:** FDD mode: Channel aggregation to achieve higher rates is normally not needed for the FDD mode, due to different bit rates of the physical channels (maximum 1024 kbps). Channel aggregation (multi-code transmission) is supported and used for the highest user rates (up to 2 Mbps). TDD mode: Channel aggregation (multi-code and multi-slot) is used, see A1.2.31.
A1.3	Expected Performances
A1.3.1	for terrestrial test environment only
A1.3.1.1	What is the achievable BER floor level (for voice)? Note: BER floor level under BER measuring condition defined in Annex 2 using the data rates indicated in section 1 of Annex 2. **Answer:** Significantly below BER = 10^{-3}
A1.3.1.2	What is the achievable BER floor level (for data)? Note: BER floor level under BER measuring condition defined in Annex 2 using the data rates indicated in section 1 of Annex 2. **Answer:** Significantly below BER = 10^{-6} for circuit-switched data

A1.3.1.3	What is the maximum tolerable delay spread (in nsec) to maintain the voice and data service quality requirements? Note: The BER is an error floor level measured with the Doppler shift given in the BER measuring conditions of ANNEX 2. **Answer:** Implementation dependent Exact requirement is for further study but at least 50000 ns should be tolerated.

A1.3.1.4	What is the maximum tolerable doppler shift (in Hz) to maintain the voice and data service quality requirements? Note: The BER is an error floor level measured with the delay spread given in the BER measuring conditions of ANNEX 2. **Answer:** Implementation dependent. Exact requirement is for further study but at least 500 Hz should be tolerated.
A1.3.1.5	Capacity: The capacity of the radio transmission technology has to be evaluated assuming the deployment models described in ANNEX 2 and technical parameters from A1.2.22 through A1.2.23.2.
A1.3.1.5.1	What is the voice traffic capacity per cell (not per sector): Provide the total traffic that can be supported by a single cell in Erlangs/MHz/cell in a total available assigned non-contiguous bandwidth of 30 MHz (15 MHz forward/15 MHz reverse) for FDD mode or contiguous bandwidth of 30 MHz for TDD mode. Provide capacities considering the model for the test environment in ANNEX 2. The procedure to obtain this value in described in ANNEX 2. The capacity supported by not a standalone cell but a single cell within contiguous service area should be obtained here. **Answer:** See Simulation Results.
A1.3.1.5.2	What is the information capacity per cell (not per sector): Provide the total number of user-channel information bits which can be supported by a single cell in Mbps/MHz/cell in a total available assigned non-contiguous bandwidth of 30 MHz (15 MHz forward / 15 MHz reverse) for FDD mode or contiguous bandwidth of 30 MHz for TDD mode. Provide capacities considering the model for the test environment in ANNEX 2. The procedure to obtain this value in described in ANNEX 2. The capacity supported by not a standalone cell but a single cell within contiguous service area should be obtained here. **Answer:** See Simulation Results. Furthermore, for the TDD mode, ODMA can increase capacity by reducing effective path loss, optimum link adaptation and link diversity thus lowering transmission power and associated inter-cell interference.
A1.3.1.6	Does the SRTT support sectorization? If yes, provide for each sectorization scheme and the total number of user-channel information bits which can be supported by a single site in Mbps/MHz (and the number of sectors) in a total available assigned non-contiguous bandwidth of 30 MHz (15 MHz forward/15 MHz reverse) in FDD mode or contiguous bandwidth of 30 MHz in TDD mode. **Answer:** The RTT supports use of the sectorisation. See also simulation results.
A1.3.1.7	Coverage efficiency: The coverage efficiency of the radio transmission technology has to be evaluated assuming the deployment models described in ANNEX 2.
A1.3.1.7.1	What is the base site coverage efficiency in km^2/site for the lowest traffic loading in the voice only deployment model? Lowest traffic loading means the lowest penetration case described in ANNEX 2. **Answer:** See Link Budget Template.

A1.3.1.7.2	What is the base site coverage efficiency in km^2/site for the lowest traffic loading in the data only deployment model? Lowest traffic loading means the lowest penetration case described in ANNEX 2. **Answer:** See Link Budget Template. Furthermore, for the TDD mode, ODMA can increase coverage efficiency by reducing effective path loss, optimum link adaptation and link diversity. This is particularly useful for high rate data services.

A1.3.2	For satellite test environment only
A1.3.2.1-4	N/A
A1.3.3	Maximum user bit rate (for data): Specify the maximum user bit rate (kbps) available in the deployment models described in ANNEX 2. **Answer:** Protocols are designed in such a way that user bit rate at least up to 2048 kbps.
A1.3.4	What is the maximum range in meters between a user terminal and a base station (prior to hand-off, relay, etc.) under nominal traffic loading and link impairments as defined in Annex 2? **Answer:** See Link Budget Template.
A1.3.5	Describe the capability for the use of repeaters **Answer:** Repeaters can be used. In addition ODMA supports data transfer via a network of intermediate relaying nodes (dedicated fixed relays or relaying enable mobiles).

A1.3.6	Antenna Systems : Fully describe the antenna systems that can be used and/or have to be used; characterize their impacts on systems performance, (terrestrial only) e.g., does the RTT have the capability for the use of : - remote antennas: Describe whether and how remote antenna systems can be used to extend coverage to low traffic density areas. - distributed antennas: describe whether and how distributed antenna design are used, and in which IMT-2000 test environments - Smart antennas (e.g. switched beam, adaptive, etc.): describe how smart antennas can be used, and in which IMT-2000 test environments - other antenna systems **Answer:** Both FDD and TDD operating modes of UTRA are able to use all the standard types of Base Station antennas. This includes those that provide omni-directional, sectored, fixed or variable patterns. Directive Antennas decrease the interference, leading to an increase in system capacity. Both FDD and TDD mode support remote antenna systems, distributed antenna systems, and smart antenna systems. Signal-to-interference-plus-noise-ratio (SINR) can be improved significantly by incorporating various smart antenna concepts on the uplink as well as the downlink. These SINR gains may be exploited • to increase the capacity (mainly in urban areas), e.g. by reducing the interference, • to increase the coverage (mainly in rural areas), e.g., by increasing the cell size (range extension) or by improving the edge coverage, • to increase the link quality, • to decrease the delay spread, • to reduce the transmission powers, or a combination thereof.
A1.3.7	Delay (for voice) **Answer:** It depends on which speech codec is used. Different voice bearers can be supported. See answer on A1.3.7.1 below.
A1.3.7.1	What is the radio transmission processing delay due to the overall process of channel coding, bit interleaving, framing, etc., not including source coding? This is given as transmitter delay from the input of the channel coder to the antenna plus the receiver delay from the antenna to the output of the channel decoder. Provide this information for each service being provided. In addition, a detailed description of how this parameter was calculated is required for both the up-link and the down-link. **Answer:** Service specific delay (depends on interleaving/channel-coding setting). Minimum delay: 12 ms for 10 ms interleaving in FDD mode and 13 ms for interleaving over 2 frames in TDD mode. Processing time of 2 ms included.

A1.3.7.2	What is the total estimated round trip delay in msec to include both the processing delay, propagation delay (terrestrial only) and vocoder delay? Give the estimated delay associated with each of the key attributes described in Figure 1 of Annex 3 that make up the total delay provided. **Answer:** The round trip delay including source coding is implementation dependent. However, the approximate delay is 13 ms excluding the speech codec delays as indicated by the answer on A1.3.7.1. . In addition the speech coding delay should be added which is around 12 ms for speech framing and processing assuming a 10 ms frame length speech codec. Other delays such as delays on the interface between the BTS and the transcoder are not included.
A1.3.7.3	Does the proposed RTT need echo control? **Answer:** Echo control can be taken care of by mobile station design.
A1.3.8	What is the MOS level for the proposed codec for the relevant test environments given in Annex 2? Specify its absolute MOS value and its relative value with respect to the MOS value of ITU-T Recommendation G.711 (64 k PCM) and ITU-T Recommendation G.726 (32 k ADPCM). NOTE 1 - If a special voice coding algorithm is indispensable for the proposed RTT, the proponent should declare detail with its performance of the codec such as MOS level. (See § A1.2.19) **Answer:** For the suggested AMR speech codec no exact values are available yet. For clean speech a MOS value of 4.1 (+0.1 compared to G.726 and –0.1 compared to G.711) can be expected depending on the source coding bit rate.
A1.3.9	Description on the ability to sustain quality under certain extreme conditions.

A1.3.9.1	System overload (terrestrial only) : Characterize system behavior and performance in such conditions for each test services in Annex 2, including potential impact on adjacent cells. Describe the effect on system performance in terms of blocking grade of service for the cases that the load on a particular cell is 125%, 150%, 175%, and 200% of full load. Also describe the effect of blocking on the immediate adjacent cells. Voice service is to be considered here. Full load means a traffic loading which results in 1% call blocking with the BER of 10^{-3} maintained. **Answer:** FDD mode: Overload causes graceful degradation of system performance, e.g. by decreasing the speech codec bit rate or increasing the BER. TDD mode: Under overload conditions DCA can be used to increase the allocated resources to the overloaded cell at the cost of the capacity of the neighbouring cells.
A1.3.9.2	Hardware failures: Characterize system behavior and performance in such conditions. Provide detailed explanation on any calculation. **Answer:** Mostly implementation dependent. Radio bearer re-establishment is supported.

A1.3.9.3	Interference immunity: Characterize system immunity or protection mechanisms against interference. What is the interference detection method? What is the interference avoidance method? **Answer:** FDD mode: Interference is suppressed by the processing gain. Multi-user detection and/or interference cancellation can be used but is not required. TDD mode: Multi-user detection by means of joint detection is typically used as protection against interference.
A1.3.10	Characterize the adaptability of the proposed SRTT to different and/or time varying conditions (e.g. propagation, traffic, etc.) that are not considered in the above attributes of the section A1.3. **Answer:** Adaptive transmit power is used. Adaptive channel coding can be used for TDD mode.
A1.4	Technology Design Constraints
A1.4.1	Frequency stability : Provide transmission frequency stability (not oscillator stability) requirements of the carrier (include long term - 1 year – frequency stability requirements in ppm).
A1.4.1.1	For Base station transmission (terrestrial component only) **Answer:** 0.05 ppm
A1.4.1.2	For Mobile station transmission **Answer:** 3 ppm (unlocked), 0.1 ppm (locked)
A1.4.2	Out of band and spurious emissions: Specify the expected levels of base or satellite and mobile transmitter emissions outside the operating channel, as a function of frequency offset Δf. **Answer:** The limits for spurious emissions at frequencies greater than +/-250% of the necessary bandwidth would be based on the applicable tables from the ITU-R Recommendation SM.329. further guidance would be taken from the CEPT ERC recommendations that are currently under progress.

A1.4.3	Synchronisation requirements: Describe SRTT's timing requirements , e.g. - Is BS-to-BS or satellite land earth station (LES)-to-LES synchronisation required? Provide precise information, the type of synchronisation, i.e., synchronisation of carrier frequency, bit clock, spreading code or frame, and their accuracy. **Answer:** FDD-mode: not required. TDD mode: Timeslot synchronisation is recommended to decrease the effort for neighbour cell interference suppression. Frame synchronisation is recommended to speed up listening of neighbour cell beacon information. -Source stability of external reference frequency = $\pm 2 \cdot 10^{-8}$ or higher. -24 hours timing accuracy of base stations ≈ 1µs. - Is BS-to-network synchronisation required? (Terrestrial only). **Answer:** Yes. On a transmission level the base stations are synchronised to the transmission network. The different base stations involved in soft handover (macro diversity) are synchronised per active mobile station connection in soft handover. An accuracy of +/- 2 msec is acceptable for the fixed transmission between base stations-to-network (radio network controller (RNC)) and can be obtained through prioritised Iub (RNC-BS) and Iur (RNC-RNC) signalling - State short-term frequency and timing accuracy of BS (or LES) transmit signal. **Answer:** A BS short-term frequency/timing accuracy of 0.05 ppm can be considered. - State source of external system reference and the accuracy required, if used at BS (or LES) (for example: derived from wire-line network, or GPS receiver). **Answer:** FDD mode: A non-dedicated Synchronisation Network is used for this BS clock reference generation where the synchronisation network is superimposed on a PDH/SDH traffic network. A dedicated synchronisation network (e.g. GPS based) is thus not needed as a source for the BS synchronisation. Long-term stability is set by a Stratum 1 PRC (Primary Reference Clock) contained in or supplied to the core network (1×10^{-11} or better is expected). For the radio interface clock generation at the base station a PRC accuracy of Stratum 2 is more than sufficient, (1.6×10^{-8}), both for long and short term. TDD mode: Source stability of external reference frequency = $\pm 2 \cdot 10^{-8}$ or higher. - State free run accuracy of MS frequency and timing reference clock. **Answer:** 3 ppm - State base-to-base bit time alignment requirement over a 24 h period (µs). **Answer:** FDD mode: The base-to-base bit time alignment over the radio interface is set by the individual BS frequency/timing accuracy of 0.05 ppm as indicated above. There is no further synchronisation needed between the different base stations from the radio transmission point of view. The base stations involved in macro diversity/soft handover to a specific mobile station are synchronised to that mobile station . That synchronisation is done individually per mobile station when entering macro diversity mode. TDD mode: see BS-BS synchronisation above.

A1.4.4	Timing jitter : For base (or LES) and mobile station give: - the maximum jitter on the transmit signal, - the maximum jitter tolerated on the received signal. Timing jitter is defined as RMS value of the time variance normalized by symbol duration. **Answer:** TBD
A1.4.5	Frequency synthesizer : What is the required step size, switched speed and frequency range of the frequency synthesizer of mobile stations? **Answer:** - Step size: 200 kHz - Switched speed: 250 µs - Frequency range: 60 MHz
A1.4.6	Does the proposed system require capabilities of fixed networks not generally available today? **Answer:** No special requirements on transmission for the fixed network.
A1.4.6.1	Describe the special requirements on the fixed networks for the handover procedure. Provide handover procedure to be employed in proposed SRTT in detail. **Answer:** N/A. Since the answer was no to the question above.
A1.4.7	Fixed network feature transparency

A1.4.7.1	Which service(s) of the standard set of ISDN bearer services can the proposed RTT pass to users without fixed network modification. **Answer:** The RTT can provide bit rates up to 2048 kbps.
A1.4.8	Characterize any radio resource control capabilities that exist for the provision of roaming between a private (e.g., closed user group) and a public IMT-2000operating environment. **Answer:** There are no additional radio resource control capabilities foreseen to support roaming between private and public environments, as indicated above, than what already exists for normal roaming support. To differentiate between different type of users/networks is seen as a higher layer signalling issue which then will be used by the normal radio resource control management functions to select the appropriate base stations.

A1.4.9	Describe the estimated fixed signaling overhead (e.g., broadcast control channel, power control messaging). Express this information as a percentage of the spectrum which is used for fixed signaling. Provide detailed explanation on your calculations. **<u>Answer:</u>** In downlink, system and cell specific information are broadcasted on the broadcast control channel. FDD mode: Reference (pilot) symbols for coherent detection, power control commands, and rate information are provided in dedicated physical control channel (DPCCH). In uplink, DPCCH uses fixed 16 kbps Q-branch channel, while DPDCH uses variable rate I-branch channel. In downlink, DPCCH is time multiplexed with DPDCH, and its rate can be variable depending on the DPDCH rate. The signalling overhead of DPCCH in dedicated physical channel is ranging from 2.8% up to 25% in downlink, while 5.9% - 33% in uplink. See the detailed description of the proposal for more information. TDD mode: This is mainly dependent on layer 2 and 3 protocol architecture and on the used layer2 and 3 algorithms. Layer 2 and 3 signalling overhead is currently being optimised. See the detailed description of the proposal for more information.
A1.4.10	Characterize the linear and broadband transmitter requirements for base and mobile station. (terrestrial only) **<u>Answer:</u>** <u>Bandwidth:</u> MS and BTS: 60 MHz, depending on band allocation <u>Linearity</u>: Mobile Station: For 2 equal power signals being separated by 200 kHz leading to an output level of 21 dBm each the resulting intermodulation spectrum shall not exceed relative to peak spectrum: -38 dB at 200 kHz to -90 dB at 800 kHz offset from higher/lower frequency signal (linear decrease) <=-95 dB at 1 MHz from higher/lower frequency signal and above Base Station: For 2 equal power signals being separated by 200 kHz leading to an output level of 6 dB below nominal output level the resulting intermodulation spectrum shall not exceed relative to peak spectrum: -38 dB at 200 kHz to -90 dB at 800 kHz offset from higher/lower frequency signal (linear decrease) <=-95 dB at 1 MHz from higher/lower frequency signal and above
A1.4.11	Are linear receivers required? Characterize the linearity requirements for the receivers for base and mobile station. (terrestrial only) **<u>Answer:</u>** Linear receivers are needed both for BS and MS. The 3rd order intercept point will be specified between -10 dBm and -5 dBm.
A1.4.12	Specify the required dynamic range of receiver. (terrestrial only) **<u>Answer:</u>** 80 dB for Automatic Gain Control.

A1.4.13	What are the signal processing estimates for both the handportable and the base station? - MOPS (Mega Operation Per Second) value of parts processed by DSP - gate counts excluding DSP - ROM size requiements for DSP and gate counts in kByte - RAM size requirements for DSP and gate counts in kByte Note 1: At a minimum the evaluation should review the signal processing estimates (MOPS, memory requirements, gate counts) required for demodulation, equalization, channel coding, error correction, diversity processing (including RAKE receivers), adaptive antenna array processing, modulation, A-D and D-A converters and multiplexing as well as some IF and baseband filtering. For new technologies, there may be additional or alternative requirements (such as FFTs etc.). Note 2 : The signal processing estimates should be declared with the estimated condition such as assumed services, user bit rate and etc. **Answer:** FDD mode: 8 kbps – 2048 kbps: 5 – 86 (valid for both UL and DL) Answer is given in million real multiplications with DSP and correlators included taken the word length requirements relative to the DSP operation into account. TDD mode: It depends on the implementation, e.g. for a 110 kbps service around 15-20 MIPS is needed. The convolutional encoding/decoding is not included in the figures as it is the same regardless of the multiple access for the same data rate(s).
A1.4.14	*Dropped calls*: describe how the RTT handles dropped calls. Does the proposed RTT utilize a transparent reconnect procedure – that is, the same as that employed for handoff? **Answer:** The RTT supports the transparent reconnect procedure for handling dropped calls.

A1.4.15	Characterize the frequency planning requirements: - Frequency reuse pattern: given the required C/I and the proposed technologies, specify the frequency cell reuse pattern (e.g. 3-cell, 7-cell, etc.) and, for terrestrial systems, the sectorization schemes assumed; **Answer:** 1-cell reuse Different sectorizations possible, e.g. 3 sectors/site, 6 sectors/site - Characterize the frequency management between different cell layers; **Answer:** Frequency-separated cell layers. Additionally, reuse of frequencies of the macro layer possible for TDD due to TDMA component. - Does the SRTT use interleaved frequency allocation? **Answer:** No - Are there any frequency channels with particular planning requirements? **Answer:** No - all other relevant requirements. NOTE 1 - The use of the second adjacent channel instead of the adjacent channel at a neighbouring cluster cell is called "interleaved frequency planning". If a proponent is going to employ an interleaved frequency plan, the proponent should state so in § A1.2.4 and complete § A1.2.15 with the protection ratio for both the adjacent and second adjacent channel.
A1.4.16	Describe the capability of the proposed SRTT to facilitate the evolution of existing radio transmission technologies used in mobile telecommunication systems migrate toward this SRTT. Provide detail any impact and constraint on evolution. **Answer:** The detailed parameters of the RTT have been chosen with the easy implementation of dual-mode UMTS/GSM.
A1.4.17	Are there any special requirements for base site implementation? Are there any features, which simplify implementation of base sites? (terrestrial only) **Answer:** The base station configuration can be modular thus the number of user supported can be increased modularly if desired, similar to introducing new TX/RX units to a GSM base station with the difference being that RF hardware is not effected as only single RX/TX per base station is required.
A1.5	Information required for terrestrial link budget template: Proponents should fulfill the link budget template given in Table 1.3 of Annex 2 and answer the following questions.

A1.5.1	What is the base station noise figure (dB)? **Answer:** See Link Budget Template.
A1.5.2	What is the mobile station noise figure (dB)? **Answer:** See Link Budget Template.
A1.5.3	What is the base station antenna gain (dBi)? **Answer:** See Link Budget Template.
A1.5.4	What is the mobile station antenna gain (dBi)? **Answer:** See Link Budget Template.
A1.5.5	What is the cable, connector and combiner losses (dB)? **Answer:** See Link Budget Template.
A1.5.6	What are the number of traffic channels per RF carrier? **Answer:** Variable (depends on the rate of each traffic channel).
A1.5.7	What is the SRTT operating point (BER/FER) for the required E_b/N_0 in the link budget template? **Answer:** For speech BER = 10^{-3}, for LCD BER = $10^{-6,}$ for UDD BLER = 10%
A1.5.8	What is the ratio of intra-sector interference to sum of intra-sector interference and inter-sector interference within a cell (dB)? **Answer:** Depends on the environment, in addition intra-cell interference cancellation, e.g. by means of joint detection can be applied, especially for the TDD mode.
A1.5.9	What is the ratio of in-cell interference to total interference (dB)? **Answer:** Depends on the environment. In addition intra-cell interference cancellation, e.g. by means of joint detection, can be applied and is typically used for the TDD mode.
A1.5.10	What is the occupied bandwidth (99%) (Hz)? **Answer:** Approximately 4.4 MHz
A1.5.11 A1.6 A1.6.1-10	What is the information rate (dBHz)? **Answer:** Service dependent. *Satellite system configuration* (applicable to satellite component only): Configuration details in this subsection are not to be considered as variables. They are for information only. N/A

Attachment 4

Updated Requirements and Objective Template

Table I. Technical Requirements and Objectives Relevant to the Evaluation of Candidate Radio Transmission Technologies

IMT-2000 Item Description	Comments	Obj/Req	Source	Meets?*
Voice and data performance requirements				
One-way end to end delay less than 40 ms**	*The one-way end to end delay depends on whether or not a CODEC is used, and if it is used it also depends on the specific CODEC used. Assuming a 10 ms CODEC, it can use a 10 ms interleaving radio bearer giving a one-way end-to-end delay in the order of 25 ms allowing for a 5 ms processing delay. The figure takes into account the delays imposed by the radio transmission technology parts such as channel encoding, interleaving, channel decoding and de-interleaving as well as the speech encoding and decoding.*	Req	G.174, § 7.5	❐ Yes Y ❐ No
For mobile videotelephone services, the IMT-2000 terrestrial component should operate so that the maximum overall delay (as defined in ITU-T Rec. F.720) should not exceed 400 ms, with the one way delay of the transmission path not exceeding 150 ms	*Excluding the source codec delay, the delay achievable by UTRA for radio transmission related parts is ranging from 10 ms to 80 ms, depending on what type of bearer you select*	Req	Suppl. F.720, F.723, G.114	❐ Yes Y ❐ No
Speech quality should be maintained during ≤3% frame erasures over any 10 second period. The speech quality criterion is a reduction of ≤0.5 mean opinion score unit (5 point scale) relative to the error-free condition (G.726 at 32 kb/s)	*CODEC dependent but the proposed RTT provides flexible and high quality bearers. The speech quality will be defined as to meet appropriate recommendations.*	Req	G.174, § 7.11 & M.1079 § 7.3.1	❐ Yes Y ❐ No
DTMF signal reliable transport (for PSTN is typically less than one DTMF errored signal in 10^4)	*Achieved with 64kbit/s PCM channel or out-band signalling messages*	Req	G.174, § 7.11 & M.1079 § 7.3.1	❐ Yes Y ❐ No
Voiceband data support including G3 facsimile	*Achieved with 64kbit/s PCM channel or inter-working functions*	Req	M.1079 § 7.2.2	❐ Yes Y ❐ No

* Explanation is requested when the candidate SRTT checks the No box.

** The source Recommendation suggests numerical limits for the overall delay, but provides no guidance about the measurement techniques. Moreover there is an apparent inconsistency with ITU-T Rec. G.114, where the value of 40 ms is indicated as the 'objective' value. These issues are addressed in a liaison statement sent to the relevant ITU groups. Until TG 8/1 receives a response and resolves this issue, proponents should submit candidates providing delay values using the methodology specified in Rec. ITU-R M.1225.

Support packet switched data services as well as circuit switched data; requirements for data performance given in ITU-T G.174	*As specified in the Minimum Performance Capabilities table*	Req	M.1034-1 §§ 10.1.5-, 10.2.4	❒ Yes Y ❒ No
Radio interfaces and subsystems, network related performance requirements				
Network interworking with PSTN and ISDN in accordance with Q.1031 and Q.1032	*This requirements refers to the network capabilities; however the proposed RTT does not poses any constraints to this respect.*	Req	M.687-2 § 5.4	❒ Yes Y ❒ No
Meet spectral efficiency and radio channel performance requirements of M.1079	*The spectral efficiency of the proposed RTT exceeds the one of current systems*	Req	M.1034-1 § 11.3.3/4	❒ Yes Y ❒ No
Provide phased approach with data rates up to 2 Mbit/s in phase 1	*2 Mbit/s are available from the beginning*	Obj	M.687-2, § 1.1.6	❒ Yes Y ❒ No
Maintain bearer channel bit-count integrity (e.g. synchronous data services and many encryption techniques)	*Frame/simbol/bit synchronization is realized with bearer channel bit count integrity*	Obj	M.1034-1, § 10.2.5	❒ Yes Y ❒ No
Support for different cell sizes, for example - Mega cell Radius ~100-500 km Macro cell Radius ≤35 km, Speed ≤500 km/h Micro cell Radius ≤1 km, Speed ≤100 km/h Pico cell Radius ≤50m, Speed ≤10 km/h	*The proposed RTT can support all cell size. The support of Mega cells depends on the radio propagation conditions, base station equipment like antennas, etc.*	Obj	M.1035 § 10.1	❒ Yes Y, except MEGA cells ❒ No
Application of IMT-2000 for fixed services and developing countries				
Circuit noise - idle noise levels in 99% of the time about 100 pWp	*CODEC dependent: it can be fulfilled easily by choosing appropriate coding*	Obj	M.819-2, § 10.3	❒ Yes Y ❒ No
Error performance - as specified in ITU-R F.697	*The quality of services can be as good as the one of the fixed network*	Obj	M.819-2, § 10.4	❒ Yes Y ❒ No
Grade of service better than 1%	*It depends on market conditions and strategies; however the proposed RTT does not poses any constraint to this respect*	Obj	M.819-2, § 10.5	❒ Yes Y ❒ No

Table II.Generic Requirements and Objectives Relevant to the Evaluation of Candidate Radio Transmission Technologies

IMT-2000 Item Description		Obj/Req	Source	Meets?*
Radio interfaces and subsystems, network related performance requirements				
Security comparable to that of PSTN/ISDN	*It depends on network capabilities; however, the proposed RTT does not poses any constraint to this respect*	Obj	M.687-2 § 4.4	❒ Yes Y ❒ No
Support mobility, interactive and distribution services	*It depends on network capabilities; however, the proposed RTT does not poses any constraint to this respect*	Req	M.816-1 § 6	❒ Yes Y ❒ No
Support UPT and maintain common presentation to users	*It depends on network capabilities; however, the proposed RTT does not poses any constraint to this respect*	Obj	M.816 § 4	❒ Yes Y ❒ No
Voice quality comparable to the fixed network (applies to both mobile and fixed service)	*CODEC dependent but the proposed RTT provides flexible and high quality bearers*	Req	M.819-2 Table 1, M.1079 § 7.1	❒ Yes Y ❒ No
Support encryption and maintain encryption when roaming and during handover	*It depends on network capabilities; however, the proposed RTT does not poses any constraint to this respect*	Req	M.1034-1 § 10.3.1/2	❒ Yes Y ❒ No
Network access indication similar to PSTN (e.g. dialtone)	*This can be implemented very easily*	Req	M.1034-1 § 10.1.9	❒ Yes Y ❒ No
Meet safety requirements and legislation	*The CDMA technology can inherently keep the peak power low*	Req	M.1034-1 § 10.6.1	❒ Yes Y ❒ No
Meet appropriate EMC regulations	*The CDMA technology can inherently use continuous transmission*	Req	M.1034-1 § 10.6.2	❒ Yes Y ❒ No
Support multiple public/private/ residential IMT-2000 operators in the same locality	*Frequency co-ordination may be needed*	Req	M.1034-1 § 11.1.2	❒ Yes Y ❒ No
Support multiple mobile station types	*The proposed RTT does not poses any constraint on the types of mobile stationn*	Req	M.1034-1 § 11.1.4	❒ Yes Y ❒ No
Support roaming between IMT-2000 operators and between different IMT-2000 radio interfaces/ environments	*It depends on network capabilities; however, the proposed RTT does not poses any constraint to this respect*	Req	M.1034-1 § 11.2.2	❒ Yes Y ❒ No
Support seamless handover between different IMT-2000 environments such that service quality is maintained and signalling is minimized	*Soft handover technique is applied*	Req	M.1034-1 § 11.2.3	❒ Yes Y ❒ No
Simultaneously support multiple cell sizes with flexible base location, support use of repeaters and umbrella cells as well as deployment in low capacity areas	*The proposed RTT can support simultaneously various cell size as well as repeaters*	Req	M.1034-1 § 11.2.5.1/2/3/6	❒ Yes Y ❒ No
Support multiple operator coexistence in a geographic area	*Frequency co-ordination may be needed*	Req	M.1034-1 § 11.2.5.4	❒ Yes Y ❒ No

Support different spectrum and flexible band sharing in different countries including flexible spectrum sharing between different IMT-2000 operators (see M.1036) *The co-ordination of TDD and FDD mode can achieve this requirements*		Req	M.1034-1 § 11.2.8.1/2	❒ Yes Y ❒ No
Support mechanisms for minimizing power and interference between mobile and base stations	*The proposed RTT uses the SIR-based power control mechanism*	Req	M.1034-1 § 11.2.8.3	❒ Yes Y ❒ No
Support various cell types dependent on environment (M.1035 § 10.1)	*The proposed RTT can support different cell size*	Req	M.1034-1 § 11.2.9	❒ Yes Y ❒ No
High resistance to multipath effects	*The proposed RTT uses Rake receiver*	Req	M.1034-1 § 11.3.1	❒ Yes Y ❒ No
Support appropriate vehicle speeds (as per § 7) NOTE: applicable to both terrestrial and satellite proposals	*Vehicle speeds up to 500 Km/h can be supported*	Req	M.1034-1 § 11.3.2	❒ Yes Y ❒ No
Support possibility of equipment from different vendors	*The relevant interfaces are defined allowing the possibility of equipment from different vendors*	Req	M.1034-1 § 11.1.3	❒ Yes Y ❒ No
Offer operational reliability as least as good as 2nd generation mobile systems	*Implementation dependent; however the proposed RTT does not pose any particular constraint on reliability*	Req	M.1034-1 § 11.3.5	❒ Yes Y ❒ No
Ability to use terminal to access services in more than one environment, desirable to access services from one terminal in all environments	*Implementation dependent; however the proposed RTT can support such terminals*	Obj	M.1035 § 7.1	❒ Yes Y ❒ No
End-to-end quality during handover comparable to fixed services	*The quality is maintained during the handover thank to handover diversity techniques*	Obj	M.1034-1 § 11.2.3.4	❒ Yes Y ❒ No
Support multiple operator networks in a geographic area without requiring time synchronization	*The proposed RTT does nor require time synchronization between base stations*	Obj		❒ Yes Y ❒ No
Layer 3 contains functions such as call control, mobility management and radio resource management some of which are radio dependent. It is desirable to maintain layer 3 radio transmission independent as far as possible	*Network dependent; the proposed RTT is expected to affect only the radio resource management functions*	Obj	M.1035 § 8	❒ Yes Y ❒ No
Desirable that transmission quality requirements from the upper layer to physical layers be common for all services	*It is possible to trade transmission quality against capacity. Furthermore, different services have different inherent requirements on the transmission quality. This is in line with the discussion which took place within ITU-R TG 8/1 WG4 on this topic.*	Obj	M.1035 § 8.1	❒ Yes Y ❒ No
The link access control layer should as far as possible not contain radio transmission dependent functions	*Radio dependent part is minimized*	Obj	M.1035 § 8.3	❒ Yes Y ❒ No
Traffic channels should offer a functionally equivalent capability to the ISDN B-channels	*The proposed RTT can support up to 2 Mbit/s*	Obj	M.1035 § 9.3.2	❒ Yes Y, ❒ No

Continually measure the radio link quality on forward and reverse channels	*The proposed RTT can support frame error measurements and bit error estimation on both downlink and uplink*	Obj	M.1035 § 11.1	❒ Yes Y ❒ No
Facilitate the implementation and use of terminal battery saving techniques	*Intermitting receiving and technologies to reduce the output power are part of the proposed RTT*	Obj	M.1035 § 12.5	❒ Yes Y ❒ No
Accommodate various types of traffic and traffic mixes	*The proposed RTT can support various types of bearer services simultaneously*	Obj	M.1036 § 1.10	❒ Yes Y ❒ No
Application of IMT-2000 for fixed services and developing countries				
Repeaters for covering long distances between terminals and base stations, small rural exchanges with wireless trunks etc.	*The proposed RTT can support repeaters and wireless trunks*	Req	M.819-2 Table 1	❒ Yes Y ❒ No
Withstand rugged outdoor environment with wide temperature and humidity variations	*Implementation dependent; the proposed RTT does not pose any particular constraints*	Req	M.819-2 Table 1	❒ Yes Y ❒ No
Provision of service to fixed users in either rural or urban areas	*This feature is supported by the proposed RTT*	Obj	M.819-2 § 4.1	❒ Yes Y ❒ No
Coverage for large cells (terrestrial)	*The proposed RTT can support macro cell easily*	Obj	M.819-2 § 7.2	❒ Yes Y ❒ No
Support for higher encoding bit rates for remote areas	*The proposed RTT supports various types of data rates*	Obj	M.819-2 § 10.1	❒ Yes Y ❒ No
Additional satellite- component specific requirements and objectives				
Links between the terrestrial and satellite control elements for handover and exchange of other information		Req	M.818-1 § 3.0	❒ Yes ❒ No
Take account for constraints for sharing frequency bands with other services (WARC-92)		Obj	M.818-1 § 4.0	❒ Yes ❒ No
Compatible multiple access schemes for terrestrial and satellite components		Obj	M.818-1 § 6.0	❒ Yes ❒ No
Service should be comparable quality to terrestrial component as far as possible		Obj	M.818-1 § 10.0	❒ Yes ❒ No
Use of satellites to serve large cells for fixed users		Obj	M.819-2 § 7.1	❒ Yes ❒ No
Key features (e.g. coverage, optimization, number of systems)		Obj	M.1167 § 6.1	❒ Yes ❒ No
Radio interface general considerations		Req	M.1167 § 8.1.1	❒ Yes ❒ No
Doppler effects		Req	M.1167 § 8.1.2	❒ Yes ❒ No

Table III.Subjective Requirements and Objectives Relevant to the Evaluation of Candidate Radio Transmission Technologies[*]

IMT-2000 Item Description		Obj/Req	Source
Fixed Service – Power consumption as low as possible for solar and other sources	*The RTT uses a power control scheme in order to reduce the output power*	Req	M.819-2 Table 1
Minimize number of radio interfaces and radio sub-system complexity, maximize commonality (M.1035 § 7.1)	*The RTT is applicable to all terrestrial environments*	Req	M.1034-1 § 11.2.1
Minimize need for special interworking functions	*Inter-working functions may be needed only when connecting to other systems*	Req	M.1034-1 § 11.2.4
Minimum of frequency planning and inter-network coordination and simple resource management under time-varying traffic	*No frequency planning is required within an operators network, New frequencies can be inserted without any frequency replanning. The RTT provides different means to handle radio resources in a flexible and spectrum efficient way, e.g.if a user reduces its bit rate the radio resource portion not used can be immediately used for another user since the radio resource is the emitted power. Base stations do not need to be synchronised to each other on the radio interface.*	Req	M.1034-1 § 11.2.6
Support for traffic growth, phased functionality, new services or technology evolution	*The RTT is designed to be future proof. New service can be easily introduced since a flexible radio bearer concept is defined. New frequencies can be inserted without re-planning, Techniques that improve link level performance such as down link antenna diversity can be introduced and will improve the system performance for all users immediately and does not need any re-planning. The RTT has been designed to provide possibilities to use adaptive antennas.*	Req	M.1034-1 § 11.2.7

[*] Descriptive information should be provided explaining how the candidate SRTT supports the concept specified in the Recommendation

Facilitate the use of appropriate diversity techniques avoiding significant complexity if possible	*Several diversity tecniques can be easily used (e.g., time, multipath, space diversity, etc), The RTT has been designed to support adaptive antennas, if required. Any radio performance enhancement technique can be readily used to enhance system performance without needing to change existing planning*	Req	M.1034-1 § 11.2.10
Maximize operational flexibility	*The RTT has got capabilities to support this flexibility*	Req	M.1034-1 § 11.2.11
Designed for acceptable technological risk and minimal impact from faults	*This risks are minimized*	Req	M.1034-1 § 11.2.12
When several cell types are available, select the cell that is the most cost and capacity efficient	*This can be supported by the resource management function*	Obj	M.1034-1 § [9.2] M.1035 § 10.3.3
Minimize terminal costs, size and power consumption, where appropriate and consistent with other requirements	*The RTT has the capabilities to achieve this objective, which may depend on manufactures' design*	Obj	M.1036 § 2.1.12

Attachment 5

UPDATED PERFORMANCE RESULTS

1. INTRODUCTION

The performance of the proposed RTT has been evaluated by means of computer simulations. This evaluation is carried out based on the methods and conditions described in ETSI Technical Report 101 112, Annex B. The evaluated test cases are presented in Table 1. The values have been taken from Attachment 7 of ITU-R Circular Letter 8/LCCE/47. In that table it is said that the lowest bit rate and the highest bit rate possible by the RTT under test should be evaluated. The following test data rates have been chosen from the exhaustive list of test cases in the Circular Letter:

- For the speech service the 8 kbps bearer have been selected.
- Only the Long Constrained Delay (LCD) data service has been evaluated. The short delay data service is thus excluded. The reason is that there are no specific delay requirements defined in the ITU-R test services which could give any guidance about what level of delay those definitions assume.
- The data packet services are called UDD (Unconstrained Delay Data) in Table 1 and is modelled as a packet service with ARQ protection and no loss of data is expected over the radio link. Hence, there is no need to specify a Bit Error Ratio (BER) requirement.

The performance evaluation consists of two stages: the link level simulation and the system level simulation. Each stage includes both uplink (UL) simulations and downlink (DL) simulations. This document describes the detailed conditions and assumptions for each stage. Note that the results provided will be subject for further refinement and analysis during the evaluation phase and may also be subject to the ongoing development of the radio interface specification.

The performance results provided are valid only for the defined test scenarios including any additional assumption that have been made, and are intended to be used for RTT evaluation purposes only.

Table 1. Simulation cases. All cases have not been simulated for the TDD mode, see Table 6 for TDD details.

	Indoor, 3km/h	Pedestrian, 3 km/h	Vehicular, 120 km/h
Speech BER = 10^{-3}	8 kbps	8 kbps	8 kbps
LCD BER = 10^{-6}	64 kbps 2048 kbps	64 kbps 384 kbps	64 kbps 144 kbps 384 kbps (Not required by ITU)
UDD	64 kbps 2048 kbps	64 kbps 384 kbps	64 kbps 144 kbps 384 kbps (Not required by ITU)

Abbreviations used when presenting the results are:
N/A = Not applicable
NA = Not available

2.

3. IMPLEMENTATION OF SIMULATIONS

3.1 FDD Link-Level Simulations

3.1.1 Simulation Model

In the simulations, sampling was made at chip level. Fast power control is included in all simulations. In actual systems, power control commands are sent on the return channel, i.e. the uplink power control commands are sent on the downlink and vice versa. In the simulations, random errors with a certain error probability are added to the power control commands. To find the appropriate values for this error probability of (Transmit Power Control) TPC symbols, the errors of TPC symbols are collected under the simulation condition that provides predetermined BER for information bit stream, e.g. BER of 10^{-3} for speech. The error probabilities of 4% and 1% are used for speech and high-speed data respectively for the UL and DL. The lower power control error rate on the high rate services can be used due to that the performance on the TPC can be improved and still keeping the overhead due to the DPCCH low. The power amplifier is not modelled; i.e. an ideal power amplifier is assumed.

A fixed searcher is used in the receiver; i.e. the receiver knows the delay of all rays and picks up the energy of some rays using a fixed set of fingers in the RAKE. This is further discussed in the section describing the channel models.

The channel estimation is based on several pilot groups and the different groups are multiplied by a weighting factor. There are several possible choices of weighting factor and all choices have not been evaluated for all test cases. Further studies will be undertaken in order to determine if one of the choices is superior to the others.

All interference is modelled as additive white Gaussian noise.

Table 2. Simulation parameters and methods for UL

Channel estimation method	Channel estimation value is based on the present pilot group and pilot groups before and after the present slot. The different pilot groups are multiplied by a weighting factor. In the parameter list for the different test cases the vector alpha contains the weight factors for the pilot groups. The weighting factor for the present slot corresponds to the element in the middle of the alpha vector.
SIR estimation method	S: Channel estimation value per Rake finger is calculated as average of pilot symbols from one pilot group. S is the sum of the powers of channel estimates from different fingers. I: Interference is assumed constant in the link level simulations
Channel model	According to M.1225 Annex 2 (Annex B of ETSI TR 101 112). v=3,120 km/h (c=3*10^8, fc=2 GHz)
Number of RAKE fingers	Indoor, Outdoor to indoor and pedestrian A: 2 fingers/branch Vehicular, Outdoor to indoor and pedestrian B: 4 fingers/branch
Searcher	Fixed delays, see Figure 1
Sampling rate	Chip level sampling (1 sample/chip)
PC dynamic range	80 dB
PC step size / delay	1 dB / 1 slot (0.625 ms)
PC symbol error	4% random error for speech 1% random error for high speed data
Interference from other users	Modelled as AWGN
Eb/No scaling	Eb will be calculated as the received power for each information bit. The following items will be calculated as overhead: pilot, TPC, TFI, CRC, tail, Convolutional Coding, RS coding, repetition, block number in the case of packet.

Table 3. Simulation parameters and methods for DL

Channel estimation method	Channel estimation value is based on the present pilot group and pilot groups before and after the present slot. The different pilot groups are multiplied by a weighting factor. In the parameter list for the different test cases the vector alpha contains the weight factors for the pilot groups. The weighting factor for the present slot corresponds to the first element that is one in the alpha vector.
SIR estimation method	Each pilot symbol will be processed through coherent detection and maximum ratio combining with the channel estimation value of its own block and several blocks before own block. The blocks are multiplied by a weighting factor and the different weights are collected in the vector alpha. Vehicular A: 2 blocks are used, alpha = (0.6,1.0) Indoor A: 3 blocks are used alpha = (0.3, 0.8, 1.0) Pedestrian A: the same as Indoor A S: S is calculated as square of the average of the results from the coherent detection and the maximum ratio combining of pilot symbols within the pilot block. I: Interference is calculated as an exponentially weighted average of the variance of the results from coherent detection and maximum ratio combining of the pilot symbols within the pilot block (A forgetting factor of 0.99 is used in the exponentially weighted averaging.)
Channel model	According to M.1225 Annex 2 (Annex B of ETSI TR 101 112). v=3,120 km/h ($c=3*10^8$, fc=2GHz)
Number of RAKE fingers	Indoor , Outdoor to indoor and pedestrian A: 2 Vehicular, Outdoor to indoor and pedestrian B: 4
Sampling rate	Chip level sampling
PC dynamic range	Unlimited[1]
PC step size / delay	1 dB / 1 slot (0.625 ms)
Searcher	Fixed delays, see Figure 1
Interference from other users	Modelled as AWGN
TPC bit error	4% random error for speech 1% random error for high speed data
Eb/No scaling	Eb will be calculated as the received power for each information bit. Following items will be calculated as overhead: pilot, TPC, TFI, CRC, tail, Convolutional Coding, RS coding, repetition, block number in the case of packet.

3.1.1.1 Channel Models

The channel models given in M.1225 Annex 2, also used in ETSI TR 101 112, cannot be used right away, since the time resolution of the simulation model is one sample. For the simulations the following model was used:

Each ray is split into two rays, one to the sample to the left and one to the sample to the right. The power of these new rays is such that the sum is equal to the original power, and the power of each of the new rays is

[1] Simulations have shown nearly no difference between limited and unlimited dynamic range.

proportional to the (1-normalised distance to the original ray). Finally, the power of all rays on one sample are added up and normalised. This yields a model with a number of independently Rayleigh fading rays on the sampling instants.

In the simulations the sampling time is equal to the chip time, resulting in the channel models in Figure 1.

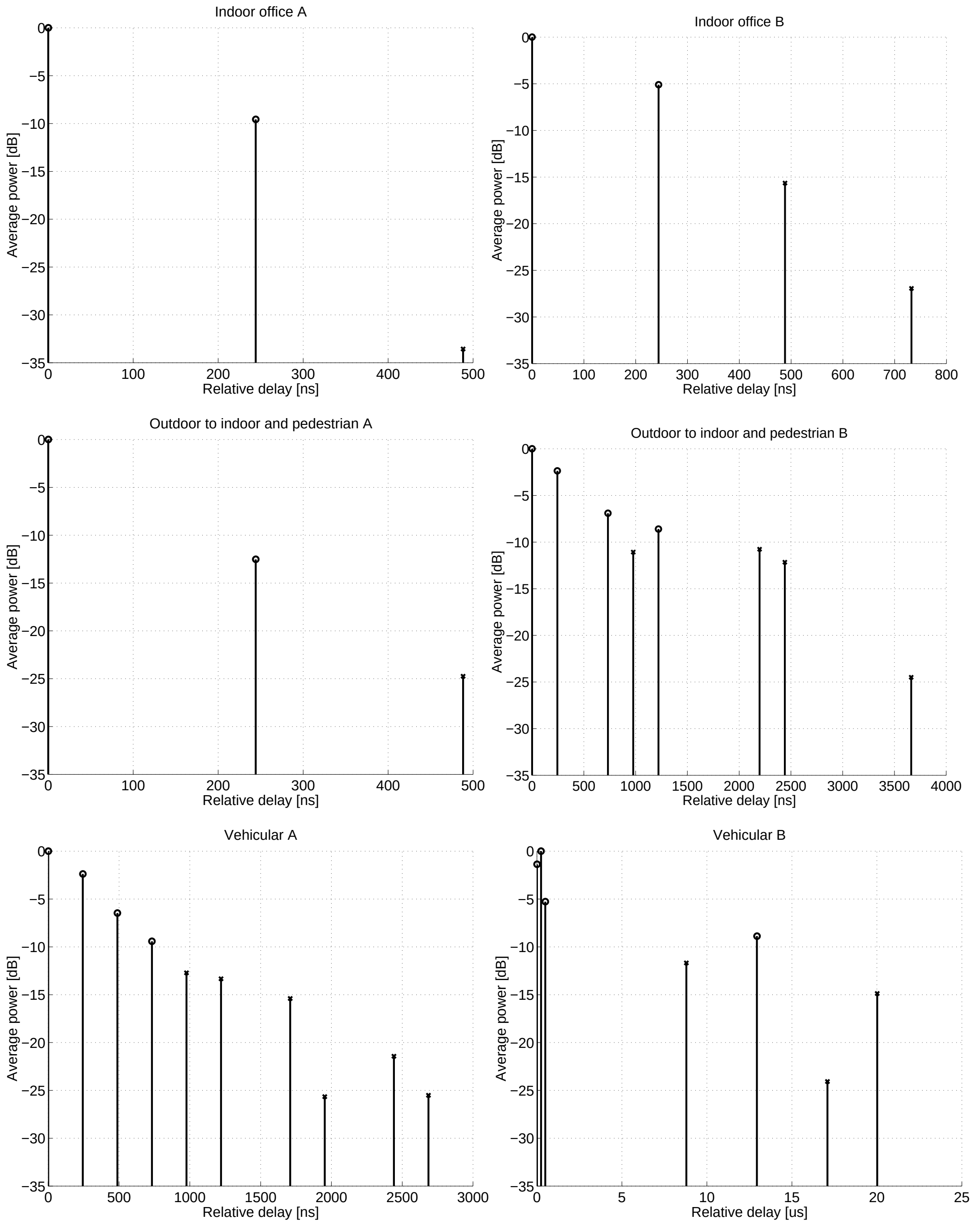


Figure 1. Modified channel models used in the simulations.

The rays picked up by the RAKE receiver are marked with “o” in Figure 1, while other rays are marked with “x”. No special link simulations were made for soft handover situations. In a soft handover the result from two single connection RAKEs are combined. For the Vehicular case this would mean 8 RAKE fingers. However, the number of RAKE fingers can be lowered in soft handover without affecting the performance, so 4 - 6 fingers should be sufficient.

3.1.2

3.1.3 TDD Link-Level Simulations

In the following, link level simulation results for UTRA TDD Mode are presented. The circuit switched services, i.e. speech and LCD services, cf. Table 1, are implemented with forward error correction (FEC) and the packet services, i.e. UDD services, use automatic repeat request (ARQ) together with FEC. The basic assumptions and technical choices for the link level simulations are summarised in Table 4.

Table 4. Basic assumptions and technical choices for the link level simulations

carrier frequency	2 GHz
carrier spacing	5 MHz
Chip rate	4.096 Mchip/s
duration of a TDMA frame	10 ms
duration of a time slot	625 µs
data modulation	QPSK
Chip modulation	root raised cosine roll-off α=0.22
Spreading characteristics	Orthogonal Q chips/symbol
number of chips per symbol	16
chip duration	0.24414 µs
channel coding	convolutional coding + puncturing for rate matching
outer coding	RS coding
Interleaving	block interleaving
data detection	joint detector: minimum mean square error block linear equaliser
channel estimation	joint channel estimator according to [4] based on correlation
power control	frame based C-level power control

In the simulations, all intra cell interferers are modelled completely with their whole transmission and reception chains. Inter cell interference is modelled as white Gaussian noise. In the following, bit error rates (BER) are given as a function of the average E_b/N_0 in dB (E_b is the energy per bit and N_0 is the one-sided spectral noise density) with the intra cell interference, i.e. the number K of active users per time slot as a parameter. The relation between the E_b/N_0 and the carrier to interference ratio C/I, with C denoting the carrier power per CDMA code and with I denoting the inter cell interference power, is given by

$$\frac{C}{I} = \frac{E_b}{N_0} \cdot \frac{R_c \cdot \log_2 M}{B \cdot Q \cdot T_c}$$

with

R_c the rate of the channel encoder (depends on the service),

M the size of the data symbol alphabet (4),

B the user bandwidth

Q the number of chips per symbol (16) and

T_c the chip duration (0.24414 µs).

The expression $\log_2 M$ is the number of bits per data symbol and $Q.T_c/\log_2 M$ is the bit duration at the output of the encoder. One net information bit is transmitted in a duration of $Q.T_c/(R_c.\log_2 M)$. Therefore, (1-1) is equivalent to $C/I = (E_b/T_b)/(N_0.B)$, i.e., $C = E_b/T_b$ and $I = N_0.B$ with T_b the duration of a net information bit.

The carrier to interference ratio per user is K_c times the carrier to interference ratio per CDMA code, with K_c denoting the number of CDMA codes per time slot per user.

3.1.4 FDD System-Level Simulations

3.1.4.1 Simulation Environment

The simulation environments are described in ETSI TR 101 112 Annex B. Implementation assumptions are described below.

The Indoor office environment characterises a three floors office building where users are moving (3 km/h) between an office room to the corridor or vice versa. The base stations (60 base stations all using Omni-directional antennas) are deployed in every second office room. No wall propagation loss was assumed, only between floors.

The Outdoor to indoor and pedestrian deployment environment is a Manhattan-like environment with the block size of 200 m and low speed (3 km/h) users. The environment consists of 72 base stations and are located as described in Annex B of ETSI TR 101 112. The base stations are using Omni-directional antennas and are deployed 10 m above ground, which is below the rooftops. The radio propagation going above rooftops is also included in the system simulation model. The street width is 30 m and it is assumed that the pedestrians are moving in the middle of the street.

The Vehicular environment is classic macro environments with site-to-site distance of 6 km (1.5 km site-to-site is used for UDD144 and LCD144). Three-sectored sites are used, i.e. each site is serving three sectors (cells). The speed of the mobile stations is 120 km/h. Wrap around is used in order to make an infinite cell plan, i.e. there are no border effects in the simulations. The BS transmit power is limited to 20 W, including common control channels.

3.1.4.2 Downlink Orthogonality

The downlink will not be perfectly orthogonal due to multipath propagation. The downlink orthogonality factor, i.e. the fraction of the total output power that will be experienced as intra-cell interference, has been calculated for the different environments and is presented in Table 5. An orthogonality factor of zero corresponds to a perfectly orthogonal downlink, while a factor of one is a completely non-orthogonal downlink. As seen in the table below, 40% of the power transmitted from the own cell will act as intra-cell interference in the Vehicular environment.

Table 5. Orthogonality factor for the environments' different propagation models.

Propagation model	**Orthogonality factor**
Indoor office A	0.10 (simulated)
Outdoor to indoor and pedestrian A	0.06 (simulated)
Vehicular A	0.40 (simulated)
Indoor office B	0.25 (calculated)
Outdoor to indoor and pedestrian B	0.44 (calculated)
Vehicular B	0.64 (calculated)

The orthogonality factor has been derived in the following way:

Two simulations were made one with white Gaussian noise and one with intra cell interference. The BER was then plotted as a function of E_b/N_o and E_b/I_o respectively. These curves may differ significantly, where the E_b/I_o curve is to the left of the E_b/N_o curve. A difference of 10 dB means that a given E_b/I_o gives the same BER as $E_b/N_o = E_b/I_o + 10$. Consequently, a certain I_o in the system simulations is equivalent to having 10 dB less N_o in the link-level simulations. Hence, it is possible to say that the orthogonality removes 90% of the interference, or in other words an orthogonality factor of 10% is obtained (10% of the interference remains).

3.1.4.3 Multiple downlink scrambling codes

In the downlink all users in a cell (sector) use the same scrambling code. Hence, all users share the available channelisation codes in the OVSF code-tree. This means that the channelisation codes in the downlink is a much more limited resource in the downlink than in the uplink, where each user has its own scrambling code and therefore can utilise all the codes in the code-tree. On the other hand, with only one scrambling code in the downlink, the degree of orthogonality between users (and physical channels) will be high. The orthogonality depends on the multipath profile of the channel.

Since orthogonality in the downlink helps increasing downlink capacity, one could draw the conclusion that the current solution with one orthogonal code set in the downlink is the right one. Also, since CDMA systems are interference limited, the limited number codes should probably not be a problem. However, there are situations where the limited number of codes in the downlink can be too small.

The are two ways to increase the number of downlink channelisation codes:

1. Use multiple downlink scrambling codes (use multiple code-trees as described above).

2. Change the channel coding scheme in order to increase the spreading factor (the number of channelisation codes is equal to the spreading factor).

The current simulation results utilise the first approach only. The users have been allocated uniformly between different code sets within the cell. Further, it is assumed that the orthogonality factor within the code set is same as the values stated in Table 5 and that no orthogonality between different code sets exists.

3.1.4.4 Soft / Softer Data Combining

For the Indoor office and the Outdoor to indoor and pedestrian environment soft handover is used between base stations. This means that the uplink C/I (or SIR = PG×C/I) is calculated as selection diversity and the downlink as maximum ratio combining (a sum of the received C/I from each base station). For the Vehicular environment softer handover is used, i.e. the mobile is connected to several sectors belonging to the same site, which will affect the calculation of the uplink C/I. Therefore the uplink C/I for all sectors belonging to one site is calculated as maximum ratio combining. Soft handover in the Vehicular environment is treated as regular selection diversity.

The softer handover data combining (maximum ratio combining) is performed on Layer 1 in the UTRA/FDD mode. Softer handover is used only in the Vehicular environment. In the uplink (within one site) and downlink the SIR during *softer* handover is modelled as:

$$SIR_{combined} = \sum_{\sec tors} SIR_{\sec tor}$$

The combined downlink (maximum ratio combining) SIR during *soft* handover is modelled as:

$$SIR_{DL,combined} = \sum_{\sec tors} SIR_{\sec tors}$$

The combined uplink (selection diversity) SIR during *soft* handover is modelled as:

$$SIR_{UL,combined} = \max_{\sec tors}(SIR_{\sec tor})$$

Note that no diversity gain (against fast fading) is assumed for soft handover data combining.

3.1.4.5 Increase in TX Power due to Power Control

One effect of the fast power control is that the transmitted power from each mobile will vary with time, and this can cause an increase in the average background interference power.

For the speech service the average transmitted powers increase is used when calculating the interference to other cells (the power increase will not affect the own cell). A good model of the power increase is perfect tracking of the fast fading. This assumption is valid only for the 3 km/h cases (Indoor office and Outdoor to indoor and pedestrian). The power increase in the Vehicular environment is negligible since the power control cannot track the fading, and is therefore not included in the system simulations.

For the UDD simulations fast fading values from the link level simulations are used in the system level simulator to adjust the output power of the transmitters for each frame. This means that for each frame a new fading value will be used when calculating the gain matrix (including path loss, shadow fading and fast fading).

3.1.4.6 Radio Resource Management

Fast SIR based power control is assumed in both uplink and downlink, and the powers of the transmitters are balanced to meet the averaged SIR during one frame.

Soft/softer handover is used for the circuit-switched services. The soft/softer handover algorithm simply connects the strongest, based on path loss (excluding fast fading), base stations within the handover window. The soft/softer handover window threshold is set to 3 dB and the algorithm is executed every 0.5 second and the maximum active set size is two. No significant performance improvement is expected by having an active set size of three or more in these environments. Measurement errors are not included. No soft handover is currently used in the packet simulations; the user simply connects to the strongest base station. Some simulation cases also use C/I based soft handover, which means that the handover decision is based on path loss and uplink interference, i.e. the algorithm tries to minimise the MS transmit power.

About 5% of the total BS power is allocated to downlink common control channels, which will interfere all users in the system. I.e. we assume in the system level simulations that all common channels are acting as non-orthogonal channels, which is a bit pessimistic since only the SCH is non-orthogonal to the users within the same cell.

For the UDD service dedicated channel packet transmission is used. No random access / forward access signalling is included in the results.

We assume that a RLC block can be re-transmitted in the next frame, i.e. that the ACK/NACK channel is error free and infinitely fast.

3.1.4.7 Performance Measures

Circuit-Switched Services

The circuit-switched services have been evaluated by means of dynamic system simulations. The performance measure of the speech (8 kbps, 50% voice activity) and LCD services is that 98% of the users are *satisfied*. A user is satisfied if all three of the following constraints are fulfilled:

1. The user does not get blocked when arriving to the system.
2. The user has sufficiently good quality more than 95% of the session time. The quality threshold is defined as BER $=10^{-3}$ (speech) or BER $= 10^{-6}$ (LCD).
3. The user does not get dropped. A speech user is dropped if BER $> 10^{-3}$ during 5 s and e.g. for the LCD 384 a user is dropped if BER $> 10^{-6}$ during 26 s.

Packet Services

The performance measure of the packet services is that 98% of the users are *satisfied*. A user is satisfied if all three of the following constraints are fulfilled:

1. The user does not get blocked.
2. The user does not get dropped.

The *active session throughput*[1] shall not be below 14.4 kbps (UDD 144), 38.4 kbps (UDD 384) or 204.8 kbps (UDD 2048) if a user should be satisfied. Taking the invert will show the delay per bit that is allowed.

The time waiting on ACK/NACK (i.e. when the transmitter buffer is empty) is not included when calculating the active session throughput. If the data packet that shall be transmitted has fewer bits than can be transmitted in a frame, dummy bits (or rather dummy blocks) are added in the system level simulations. These dummy bits are not included when calculating the session throughput, however they will increase the interference in the system. A data packet will be divided into data blocks of 340 bits (304 information bits) for the uplink. Several blocks are then put into a frame, e.g. 8 blocks per frame for the UDD 384 service.

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[1] The active session throughput is defined as the ratio of correctly received bits during the entire session and the session length excluding the time when there is nothing to transmit (i.e. empty input buffer).

3.1.5 TDD System-Level simulations

3.1.5.1 General

This section describes evaluated system performance resulting from system level simulations of the proposed RTT for UTRA-TDD. The simulations have been performed for several services and environments according to ETSI TR 101 112 Annex 2 ETSI TR 101 112, “Selection procedures for the choice of radio transmission technologies of the Universal Mobile Telecommunications System UMTS (UMTS 30.03)“, version 3.2.0, April 1998.. The evaluated test cases shown in Table 6 represent the most relevant applications of those proposed in Attachment 7 of ITU-R Circular Letter ITU-R Circular Letter, „Request for Submission of Candidate Radio Transmission Technologies (RTTs) for the IMT-2000/FPLMTS Radio Interface,“ Circular Letter 8/LCCE/47 with updates.. This selection results from the limited amount of time.

The UTRA-TDD approach uses a 3-dimensional resource space (consisting of frequency, time and code dimension). When taking only one carrier (5MHz) into account the resource space is reduced to the time and code dimension. Soft blocking is based upon single 5 MHz carrier. Trunking efficiency is based upon 30 MHz spectrum.

Since it was found that the downlink is limiting the system capacity, only the *downlink* direction has been considered. For instance, antenna diversity can be used in the uplink to improve the soft blocking limit significantly. In case of data services (LCD and UDD) also in downlink direction antenna diversity is assumed.

The simulation results presented in this document rely on statistics gathered from about 5000 calls in the reference cells.

Table 6. Simulation cases

	Vehicular, 120 km/h	Pedestrian, 3 km/h	Indoor, 3km/h
Speech BER = 10^{-3}	8 kbps	8 kbps	8 kbps
Circuit switched data services (LCD) BER = 10^{-6}	144 kbps	384 kbps	2 048 kbps
Packet data (UDD) BER = 10^{-6}	144 kbps	384 kbps	2 048 kbps

3.1.5.2 Used Models

All test environments including network structure, cell shape, antenna pattern, propagation models (path loss and shadowing, channel model), mobility models, traffic models and quality of service (QoS) criteria for Real Time (RT) and UDD users are modelled according to ETSI TR 101 112, “Selection procedures for the choice of radio transmission technologies of the Universal Mobile Telecommunications System UMTS (UMTS 30.03)“, version 3.2.0, April 1998..

In the vehicular environment (macro environment) three-sectored sites are used with cell radius 2 km, i.e. site-to-site distance is 6 km. Statistics are collected within the central site (3 reference cells) surrounded by 54 interfering cells. A frequency re-use scheme of 1 is used.

In the pedestrian deployment (micro) environment based on a Manhattan grid given in ETSI TR 101 112, “Selection procedures for the choice of radio transmission technologies of the Universal Mobile Telecommunications System UMTS (UMTS 30.03)“, version 3.2.0, April 1998. with low speed users (3 km/h) a frequency re-use scheme of 1 is applied. Statistics are gathered from the 6 cells according toETSI TR 101 112, “Selection procedures for the choice of radio transmission technologies of the Universal Mobile Telecommunications System UMTS (UMTS 30.03)“, version 3.2.0, April 1998..

For the results (given in Table 32) in the indoor office (pico) environment which characterises a three floors office building where users are entering/leaving the office rooms a frequency re-use scheme of 1 is used.

Statistics are gathered from the 6 cells in the middle floor according to ETSI TR 101 112, “Selection procedures for the choice of radio transmission technologies of the Universal Mobile Telecommunications System UMTS (UMTS 30.03)“, version 3.2.0, April 1998.. Traffic and quality of service (QoS) criteria for Real Time (RT) and UDD users are modelled according to ETSI TR 101 112, “Selection procedures for the choice of radio transmission technologies of the Universal Mobile Telecommunications System UMTS (UMTS 30.03)“, version 3.2.0, April 1998..

3.1.5.3 Performance Measures

The evaluation criterion is in accordance with the satisfied user criterion. For speech (8 kbps, 50% voice activity) and LCD services a user is satisfied if all three constraints described in ETSI TR 101 112, “Selection procedures for the choice of radio transmission technologies of the Universal Mobile Telecommunications System UMTS (UMTS 30.03)“, version 3.2.0, April 1998. are fulfilled.

As well, the three different packet data services (UDD144, UDD384 and UDD2048) have been evaluated according to the satisfied user criterion described in ETSI TR 101 112, “Selection procedures for the choice of radio transmission technologies of the Universal Mobile Telecommunications System UMTS (UMTS 30.03)“, version 3.2.0, April 1998..

3.1.5.4 Resource Allocation

A resource unit (RU) in UTRA-TDD is a triple consisting of frequency channel, time slot and code. For services that require more than one RU:

- a number of codes (multi-code), or
- a number of time-slots (multi-slot), or
- a combination of both (mixed allocation)

may be allocated.

For RT services the resources are allocated at session set-up and are kept unchanged till session end. On inter-cell handover the same type of resource is allocated in the new cell.

For UDD services the allocation and de-allocation is done on block level driven by the number of data a user has in the buffer.

The resource allocation tries to distribute the allocated codes homogeneously over all frequencies and timeslots, i.e. it is searched for the time slot with minimum number of codes.

Channel prioritisation based upon interference level is applied. Channels with low interference have been assigned to users in weak conditions.

3.1.5.5 Power Control

A slow (0.5 sec control interval) level based power control is used within high-speed (vehicular) environment. Enhanced PC is used within slow speed deployment environments (indoor and pedestrian). According to the actual value interface time series of Rx levels resulting from link level simulations are used to incorporate the fast fading conditions. Hence the transmitting power is controlled by tracking the needed power gain (5 ms dead time in average due to open loop power control). Transmitting power is limited to maximum TX power. The dynamic range is 30 dB in downlink and 80 dB in uplink direction for both types of control mechanisms.

For UDD services no enhanced PC is used.

3.1.5.6 Handover

The handover is based on power budget (path loss difference between serving and neighbour cells) with a handover margin of 3 dB.

3.1.5.7 ARQ

For UDD the Hybrid II ARQ is used. Here the first transmission of a block is done nearly uncoded. If the first transmission fails a second transmission contains the coding bits in a way that all blocks of the first and second transmission together result in a lower coding rate (i.e. better coding).

If the second transmission also fails the worst (raw-BER) burst is retransmitted and a maximum ratio combining is done between the original and the retransmitted burst. The code rate is not decreased in this step. For all further retransmissions this maximum ratio combining is used.

If the number of retransmissions exceeds a given threshold (10 - 20) the session is dropped.

3.1.5.8 DTX

For speech services voice activity detection (VAD) and discontinuous transmission (DTX) are used with an activity factor of 0.5 and mean speech (activity) periods of 3 seconds according to ETSI TR 101 112, “Selection procedures for the choice of radio transmission technologies of the Universal Mobile Telecommunications System UMTS (UMTS 30.03)“, version 3.2.0, April 1998.. During the silent periods the transmit power is switched off, but the channels are not released, i.e. DTX is used to reduce interference and thereby to relieve the soft blocking limit.

3.1.5.9 Interface between System and Link Level Simulation

Link level simulations have been carried out with fast (multipath) fading channels, and their results describe the relationship between CIR and BER needed in the system level simulations.

In general, radio transmission systems are using a variety of diversity features. Thus, system simulations based on an average value interface result in a very low accuracy. Figure 2 shows the actual value interface (AVI) which has been taken into account in the system level simulator to calculate the actual values of CIR experienced on each burst. The results presented in this document rely on the AVI.

Due to the code dimension in UTRA-TDD the CIR values are distinguished between inter-cell CIRinter and intra-cell CIR_{intra}.

The CIR_{inter} is the ratio between the wanted signal in the reference cell and the sum of interfering signals from all other cells at the same frequency and time slot.

The CIR_{intra} is the ratio between the wanted signal in the reference cell and the sum of the signals from all other users in the same cell at the same frequency and timeslot. Simulations have shown that the impact of the intra-cell interference is negligible (due to joint detection).

Within the AVI the burst CIR values are mapped on a raw BER on a burst. Beside on the CIR_{intra} and CIR_{inter} the raw BER also depends upon

- the number of codes K_C per user and timeslot used with a certain service (code pooling)
- the number of users K per time slot.

Depending on the interleaving depth assumed for each service, the average raw BER on a corresponding number of bursts constituting one block is calculated and subsequently mapped on a user BER value of the received block. For ARQ there is an additional interface function that gives the relationship between average raw BER and BLER (block erasure rate).

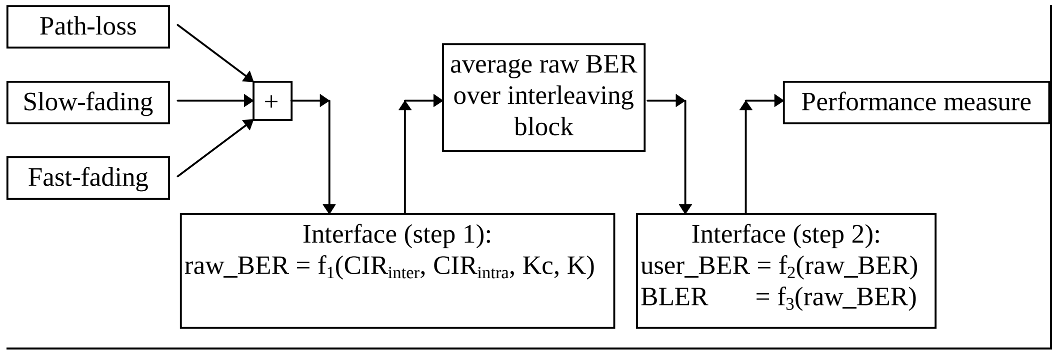

Figure 2. Actual Value Interface at system level

3.2

3.3 UTRA Results

3.3.1 FDD Link-Level Simulations

The E_b/N_o values presented here are the actual E_b/N_o values that are needed in the receiver to achieve the corresponding BER, FER and BLER. The E_b/N_o values include all overhead, i.e. the DPCCH (Dedicated Physical Control Channel: pilot symbols, power control bits, TFI) and overhead on the DPDCHs (Dedicated Physical Data Channels) such as CRCs, block numbers and tail bits for the convolutional code. In other words, the E_b value contains all energy needed to receive one information bit. Energy from common broadcast channels is not included in the link-level results.

After coding of the DPDCH rate matching is applied, using puncturing or repetition. On the DPCCH rate matching is always performed using repetition. The rate matching used for the different services are given below. Note that discontinuous transmission is used in the downlink. The interleaving for the different services is specified in the tables below as x*y. This should be interpreted as vectors of x bits are read in and vectors of y bits are read out at the transmitter according to a conventional block-interleaver scheme.

The detailed simulation parameters are shown in Table 7 to Table 12, and the link results are presented in Table 13 to Table 15.

Reed-Solomon and convolutional codes have been used for services that require bit error rates of around 10^{-6} after forward error correction. Turbo coding has been studied as an alternative and results and parameters for some of the services are presented in Section UTRA/FDD Results with turbo codes.

Table 7. Parameters for Speech (8kbps) in UL

Physical channel rate	32 kbps	32 kbps
Info/CRC/tail bits per frame	80/16/8	160/16/8 20ms
Convolutional coding rate	1/3	1/3
Repetition	8 bits/10 ms (312 ->320)	88 bits/20ms (552->640)
Interleaver	10 ms, 16*20	20 ms, 20*32
Pilot/TPC/TFI bits per slot	6/2/2 or 7/3/0	6/2/2 or 7/3/0
Antenna receiver diversity	On	On
Channel estimation	Indoor, Outdoor to indoor and pedestrian: 5 pilot groups, alpha = (1,1,1,1,1) Vehicular: 3 pilot groups, alpha = (0.4,1,0.4)	Indoor, Outdoor to indoor and pedestrian: Present slot and 8 previous averaged Vehicular: Present slot and the previous one averaged
DPCCH/DPDCH power [dB]	-3	-3

Table 8. Parameters for LCD Services in UL

Source rate	64 kbps	144 kbps	384 kbps	2048 kbps
Physical channel rate	256 kbps	512 kbps	1024 kbps	1024 kbps * 6
Information bits	5120 bits (80 ms)	11520 bits (80ms)	30720 bits (80 ms)	163840 bits (80 ms)

RS coding	(36,32)	(36, 32)	(36,32)	(36,32)
Symbol Interleaver	80 ms, 36*20	80 ms, 36*45	80 ms, 36*120	80 ms, 36*640
Outer coding process synch. Info	3 bits per sub-frame (subframe=720 bits)	3 bits per sub-frame (sub-frame = 810 bits)	3 bits per sub-frame (subframe=720 bits)	3 bits per sub-frame (subframe=720 bits)
CRC	13 bits per sub-frame (subframe=720 bits)	13 bits per sub-frame (sub-frame = 810 bits)	13 bits per sub-frame (subframe=720 bits)	13 bits per sub-frame (subframe=720 bits)
Tail	8 bits per frame (1 sub-frame = 1 frame, 10 ms)	8 bits per frame (2 sub-frames = 1 frame, 10 ms)	8 bits per frame (6 sub-frames = 1 frame, 10 ms)	8 bits per frame (32 sub-frames = 1 frame, 10 ms)
Convolutional coding rate	1/3	1/3	1/3	1/3
Repetition	328 bits/10 ms (2232 -> 2560)	140 bits/10 ms (4980 -> 5120)	3032 bits/10 ms (13272 -> 10240)	9240 bits/10 ms (70680 -> 61440 per 6 code channels)
Bit Interleaver	10 ms, 16*160 bits	10 ms, 16*320	10 ms, 16*640	10 ms, 16*640 per one code channel
Pilot/TPC/TFI bits per slot	6/2/2	6/2/2	6/2/2	6/2/2
Antenna receiver diversity	On	On	On	On
Channel estimation	Indoor, Outdoor to indoor and pedestrian: 5 pilot groups, alpha = (1,1,1,1,1) Vehicular: 3 pilot groups, alpha = (0.4,1,0.4)			
DPCCH/DPDCH power [dB]	-6	-6	-9	-6 relative one DPDCH code

Table 9. Parameters for UDD Services in UL

Source rate	64 kbps	144 kbps	384 kbps	2048 kbps
Information bit rate	30.4 kbps	60.8 kbps	243.2 kbps	486.4 kbps
Physical channel rate	128 kbps (sf_DPDCH = 32)	256 kbps (sf_DPDCH = 16)	1024 kbps (sf_DPDCH = 4)	1024 kbps (sf_DPDCH = 4)
Block size	304 bits	304 bits	304 bits	304 bits
# blocks per frame	1	2	8	16

CRC bits per block	16	16	16	16
Block number bits	12	12	12	12
Tail bits per block	8	8	8	8
Convolutional coding rate	1/3	1/3	1/3	1/3
Rate matching	Repetition 260 bits/10ms (1020 -> 1280)	Repetition 520 bits/10ms (2040 -> 2560)	Repetition 2080 bits/10ms (8160 -> 10240)	Puncturing 6080 bits/10ms (16320 -> 10240)
Interleaving	10 ms or 20ms	10 ms or 20 ms	10 ms	10 ms
Pilot/TPC/TFI bits per slot	6/2/2	6/2/2	6/2/2	6/2/2
Antenna receiver diversity	On	On	On	On
Channel estimation	Indoor, Outdoor to indoor and pedestrian: Present slot and 7 previous averaged Vehicular: Present slot and the previous one averaged			
DPCCH/DPDCH power [dB]	-4.3	-5.1	Pedestrian: -12 Vehicular: -9	-12

Table 10. Parameters for Speech (8 kbps) in DL

Physical channel rate	32 ksps	32 ksps
Info/CRC/tail bit per frame	80/16/8	160/16/8 20ms
Convolutional coding rate	1/3	1/3
Repetition	8 bits/10 ms (312 -> 320)	24 bits/20ms (552 -> 576)
Interleaver	10 ms, 16*20	20 ms, 32*18
Pilot/TPC/TFI bits per slot	8/2/0	8/2/0
Antenna receiver diversity	Off	Off
Channel estimation	Indoor, Outdoor to indoor and pedestrian: 4 pilot groups alpha = (0.6,1.0,1.0,0.6) Vehicular: 6 pilot groups alpha = (0.3,0.8,1.0,1.0,0.8,0.3)	

Table 11. Parameters for LCD Services in DL

Source rate	64 kbps	144 kbps	384 kbps	2048 kbps
Physical channel rate	128 ksps	256 ksps	1024 ksps	1024 ksps x 4
Information bits	5120 (80 ms)	11520 (80 ms)	5120 (1B)x6 (80 ms)	5120 (1B)x32 (80 ms)
RS coding	(36, 32)	(36, 32)	(36, 32) per 1B	(36, 32) per 1B
Symbol interleaver	36x20	36x45	36x20 per 1B	36x20 per 1B
CRC	13 bits	13 bits per subframe (subframe= 6480bit)	13 bits per subframe (1B=1subframe)	13 bits per subframe (1B=1subframe)
Tail	8 bits	8 bits per 8frame (2subframe=8 frame)	8 bits per 8frame (6subframe=8 frame)	8 bits per 8frame (32subframe=8 frame)
Convolutional coding rate	1/3	1/3	1/3	1/3
Rate matching	65 bits repetition (80 ms)	326 bits puncturing (80 ms)	254 bits repetition (80 ms)	776 bits repetition (80 ms)
Bit interleaver	80 ms, 128x136	80 ms, 128x302	80 ms, 128x814	80 ms, 128x4336
Pilot/TPC/TFI bit per slot	8/2/0		16/2/0	
DPCCH/DPDCH power [dB]	0	0	0	3
Channel estimation	Indoor, Outdoor to indoor and pedestrian: 4 pilot groups alpha = (0.6,1.0,1.0,0.6) Vehicular: 6 pilot groups alpha = (0.3,0.8,1.0,1.0,0.8,0.3)			
Antenna receiver diversity	On	On	On	On

Table 12. Parameters for UDD Services in DL

Source rate	64 kbps	144 kbps	384 kbps	2048 kbps
Information bit rate	30.4 kbps	60.8 kbps	243.2 kbps	486.4 kbps
Physical channel rate	64 ksps	128 ksps	512 ksps	1024 ksps
Block size	304 bits	304 bits	304 bits	304 bits
#blocks per frame	1	2	8	16
CRC bits per block	16	16	16	16
Block number bits	12	12	12	12
Tail bits per block	8	8	8	8

Convolutional coding rate	1/3	1/3	1/3	1/3
Rate matching	Repetition 4 bits/10 ms (1020 ->1024)	Repetition 8 bits/ 10ms (2040 -> 2048)	-	-
Interleaver	10 ms 16x64	10 ms 16x128	10 ms 16x510	10 ms 16x1020
Pilot/TPC/TFI bit per slot	8/2/0		16/2/0	
Channel estimation	Indoor, Outdoor to indoor and pedestrian: 4 pilot groups alpha = (0.6,1.0,1.0,0.6) Vehicular: 6 pilot groups alpha = (0.3,0.8,1.0,1.0,0.8,0.3)			
Antenna receiver diversity	On	On	On	On

Table 13. Link results for speech.

Service	**Environment**	**E_b/N_o @ BER = 10^{-3} [dB] (UL / DL)**
Speech (8 kbps, 50% VA) 10 ms interleaving	Indoor A, 3km/h	4.8 / 6.7
	Indoor B, 3km/h	5.2 / 6.7
	Pedestrian A, 3km/h	4.8 / 6.8
	Pedestrian B, 3km/h	6.1 / 7.7
	Vehicular A, 120km/h	6.8 / - 6.4 / 8.8 (no TFI)
	Vehicular B, 120km/h	6.9 / 8.9
Speech (8 kbps, 50% VA) 20 ms interleaving	Indoor A, 3km/h	3.5 / - 3.2 / 6.0 (no TFI)
	Indoor B, 3km/h	3.3 / 6.1 (no TFI)
	Pedestrian A, 3km/h	3.6 / - 3.3 / 6.1 (no TFI)
	Pedestrian B, 3km/h	4.3 / 7.0 (no TFI)
	Vehicular A, 120km/h	6.3 / - 6.1 / 7.9 (no TFI) 5.4 / - (no TFI & PC step = 0.25 dB)
	Vehicular B, 120km/h	6.0 / 8.0 (no TFI) 5.3 / - (no TFI & PC step = 0.25 dB)

Table 14. Link results for LCD services.

Service	Environment	Eb/No @ BER = 10^{-6} [dB] (UL / DL ant. div.)
LCD 64	Indoor A, 3km/h	2.3 / 1.9
	Pedestrian A, 3km/h	2.4 / 1.9
	Vehicular A, 120km/h	3.8 / 3.7
LCD 144	Vehicular A, 120km/h	3.1 / 2.5
	Vehicular B, 120km/h	3.6 / 2.6
LCD 384	Pedestrian A, 3km/h	1.3 / 1.1
	Pedestrian B, 3km/h	2.6 / 2.5
	Vehicular A, 120km/h	3.1 / 3.2
LCD 2048	Indoor A, 3km/h	1.8 / 1.6
	Indoor B, 3km/h	2.9 / 2.6

Table 15. Link results for UDD services. (NA = Not available).

Service	Environment	Eb/No @ BLER = 10^{-6} [dB] (UL / DL ant. div.)
UDD 64	Indoor A, 3km/h	1.5 / 1.2
	Indoor B, 3km/h	1.6 / NA
	Pedestrian A, 3km/h	1.5 / 1.2
	Pedestrian B, 3km/h	2.2 / NA
	Vehicular A, 120km/h	3.8 / 3.0 3.5 / - (20 ms interl.) 3.3 / - (20 ms interl. & PC step = 0.25 dB)
	Vehicular B, 120km/h	3.6 / NA
UDD 144	Vehicular A, 120km/h	3.0 / 2.9 2.7 / - (20 ms) 2.6 / - (20 ms interl. & PC step = 0.25 dB)
	Vehicular B, 120km/h	2.9 / 2.9

UDD 384	Pedestrian A, 3km/h	0.4 / 0.1
	Pedestrian B, 3km/h	1.4 / 1.2
	Vehicular A, 120km/h	2.4 / 2.0
	Vehicular B, 120km/h	2.3 / NA
UDD 2048	Indoor A, 3km/h	0.6 / 0.1
	Indoor B, 3km/h	0.9 / 0.3

3.3.2 TDD Link-Level Simulations

3.3.2.1 Speech service

In this section, link level simulation results for the speech service are given. The system parameters for implementing the speech service are summarised in Table 16.

Table 16. System parameters for the speech service

Service	**speech, 8 kbit/s, 20 ms delay**
user bit rate	8000 bit/s
number of time slots per frame per user	1
number of codes per time slot per user	1
burst type	spread speech/data burst 1 for the uplink and downlink
data modulation	QPSK
convolutional code rate	0.31 for the spread speech/data burst 1
interleaving depth	2 frames = 2 bursts
user block size	160 bits
antenna diversity	uplink: yes (2 branches), downlink: no

The required values for E_b/N_0 and C/I in order not to exceed a BER of 10^{-3} as defined for the speech service are summarised in Table 17. The values of C/I are obtained from the values of Eb/N_0 according to (0-1) by subtracting 13.9 dB for the spread speech/data burst 1.

Table 17. Required values for Eb/N0 and C/I for the speech service

Speech 8 kbit/s	$10 \log_{10} (E_b/N_0)$ in dB @ BER = 10^{-3}			$10 \log_{10}$ (C/I) in dB @ BER = 10^{-3}		
$K_c = 1$	K = 1 UL / DL	K = 4 UL / DL	K = 8 UL / DL	K = 1 UL / DL	K = 4 UL / DL	K = 8 UL / DL
Vehicular A, 120 km/h; without power control	5.3/-	5.8/8.3	6.1/8.5	-8.6/-	-8.1/-5.6	-7.8/-5.4
Outdoor to Indoor and Pedestrian A, 3 km/h; with power control	3.8/-	3.7/6.1	4.0/6.1	-10.1/-	-10.2/-7.8	-9.9/-7.8
Indoor 3 km/h; with power control	3.2/-	3.6/6.0	4.0/5.7	-10.7/-	-10.3/-7.9	-9.9/-8.2

Kc = number of codes per time slot per user, K = number of users per time slot, UL/DL = uplink/downlink

The required values for E_b/N_0 for speech 8 kbit/s in order not to exceed a BER of 10^{-3} are given as a function of the number K of active users per time slot for Indoor A, Pedestrian A and Vehicular A in the downlink and uplink. Values of K between 1 and 8 are taken into account. There is a slight degradation with increasing number K of active users per time slot. This is due to the increase of intra cell interference with increasing K. The degradation is less for the Indoor and Pedestrian channels which have less multipaths than for the Vehicular channel with more multipaths.

3.3.2.2 LCD services

In this section, link level simulation results for the LCD services are given. The system parameters for implementing the LCD 64 kbit/s service are summarised in Table 18, the system parameters for implementing the LCD 144 kbit/s service are summarised in Table 19 and for implementing the LCD 384 kbit/s service in. Furthermore, an LCD 2048 kbit/s service is investigated, for which the system parameters are given in Table 21. Antenna diversity in the uplink and downlink are also included in the results.

For the LCD 64kbit/s service, 4 codes in 1 of the 16 time slots are allocated to a user (LCD 64), with outer coding. For the LCD 144 kbit/s service, 9 codes in 1 of the 16 time slots are allocated to a user (LCD 144), with outer coding. For the LCD 384 kbit/s service, 9 codes are allocated in 3 of the 16 time slots to a user (LCD 384), with outer coding. For the LCD 2084 kbit/s service, 9 codes are allocated in 13 of the 16 time slots to a user (LCD 2048), with outer coding.

Table 18. System parameters for the LCD 64 kbit/s service

service	**LCD, 64 kbit/s, 300 ms delay**
	LCD 64; 2 users
user bit rate	64 kbit/s
number of time slots per frame per user	1
number of codes per time slot per user	4
burst type	burst type 2
data modulation	QPSK
convolutional code rate (inner code)	0.61
Reed Solomon code rate (outer code)	120/127
total code rate	0.58
interleaving depth	30 frames = 30 bursts
user block size	4800 bits

Table 19. System parameters for the LCD 144 kbit/s service

service	**LCD, 144 kbit/s, 300 ms delay**
	LCD 144
user bit rate	144 kbit/s
number of time slots per frame per user	1
number of codes per time slot per user	9
burst type	burst type 2
data modulation	QPSK
convolutional code rate (inner code)	0.61
Reed Solomon code rate (outer code)	120/127
total code rate	0.58
interleaving depth	30 frames = 30 bursts
user block size	4800 bits

Table 20. System parameters for the LCD 384 kbit/s service

Service	**LCD, 384 kbit/s, 300 ms delay**
	LCD 384
user bit rate	388.9 kbit/s
Number of time slots per frame per user	3
Number of codes per time slot per user	9
burst type	burst type 2
data modulation	QPSK
Convolutional code rate (inner code)	0.57
Reed Solomon code rate (outer code)	108/118
total code rate	0.52
Interleaving depth	30 frames = 30 bursts
user block size	4320 bits

Table 21. System parameters for the LCD 2048 kbit/s service

Service	**LCD, 2048 kbit/s, 300 ms delay**
	LCD 2048
user bit rate	2059.0 kbit/s
Number of time slots per frame per user	13
Number of codes per time slot per user	9
burst type	burst type 2
data modulation	QPSK
Convolutional code rate (inner code)	0.72
Reed Solomon code rate (outer code)	66/75
total code rate	0.64
Interleaving depth	30 frames = 30 bursts
user block size	5280 bits

To reach the required BER of 10^{-6}, LCD services use a concatenated coding scheme with an inner convolutional code and an outer Reed Solomon code. The required values for Eb/N0 and C/I are summarised in Table 22 for LCD 64 kbit/s, in Table 23 for LCD 144 kbit/s, in Table 23 for LCD 384 kbit/s and in Table 24 for LCD 2048 kbit/s. The values of C/I are obtained from the values of Eb/N0 according to (0-1) by subtracting 11.4 dB for LCD 64, 11.4 dB for LCD 144, 11.9 dB for LCD 384 and 11.0 dB for LCD 2048. The required Eb/N0 and C/I values for the downlink when using antenna diversity are identical to those given in the uplink except for the case LCD 64 Kbit/s, K=2. Antenna diversity would be a reasonable assumption for those applications that are executed e.g. on a notebook.

Table 22. Required values for Eb/N0 and C/I for the LCD 64 kbit/s service

LCD 64	10 $\log_{10}$ (Eb/N0) in dB @ BER = 10^{-6} RS	10 $\log_{10}$ (C/I) in dB @ BER = 10^{-6} RS
$K_c = 4$	K = 2 UL / DL	K = 2 UL / DL
Indoor (A), 3km/h; with power control	3.2/3.1	-8.2/-8.3
Outdoor to Indoor and Pedestrian (A), 3 km/h; with power control	3.3/3.1	-8.1/-8.3
Vehicular (A) 120 km/h; without power control	3.9/3.7	-7.5/-7.7

Kc = number of codes per time slot per user, K = number of users per time slot, UL/DL = uplink/downlink, RS = at the output of the outer Reed Solomon decoder

Table 23. Required values for Eb/N0 and C/I for the LCD 144 kbit/s service

LCD 144	10 $\log_{10}$ (Eb/N0) in dB @ BER = 10^{-6} RS	10 $\log_{10}$ (C/I) in dB @ BER = 10^{-6} RS
$K_c = 9$		
Vehicular (A), 120 km/h; without power control	4.1	-7.3

Kc = number of codes per time slot per user, K = number of users per time slot, RS = at the output of the outer Reed Solomon decoder

Table 24. Required values for Eb/N0 and C/I for the LCD 384 kbit/s service

LCD 384	10 $\log_{10}$ (Eb/N0) in dB @ BER = 10^{-6} RS	10 $\log_{10}$ (C/I) in dB @ BER = 10^{-6} RS
$K_c = 9$		
Outdoor to Indoor and Pedestrian (A), 3 km/h; with power control	1.4	-10.5

Kc = number of codes per time slot per user, K = number of users per time slot, RS = at the output of the outer Reed Solomon decoder

Table 25. Required values for Eb/N0 and C/I for the LCD 2048 kbit/s service

LCD 2048	10 $\log_{10}$ (Eb/N0) in dB @ BER = 10^{-6} RS	10 $\log_{10}$ (C/I) in dB @ BER = 10^{-6} RS
K_c = 9		
Indoor (A), 3km/h; with power control	2.8	-8.2

Kc = number of codes per time slot per user, K = number of users per time slot, RS = at the output of the outer Reed Solomon decoder

The LCD results were achieved without Turbo Codes, however turbo coding is likely to be used for LCD services. According to first simulations, the implementation of Turbo Codes would lead to an improvement of about 2 dB over the used convolutional and RS codes.

3.3.2.3 UDD services

In this section, link level simulation results for the UDD services are given. The UDD services are implemented by using a type II hybrid ARQ scheme. This ARQ scheme is explained in the following for improving the code rate from one transmission to the next from 1 to 1/2 and to 1/3. Some of these steps can also be omitted, for instance the 1/3 code rate can be omitted. In one of the used ARQ schemes, the user data is encoded with a 1/3 rate convolutional code and interleaved over 3 bursts. Rate compatible punctured convolutional (RCPC) codes are used [6]. The coding and interleaving are done in such a way that decoding is possible after one of three bursts has been received. Thus, the effective code rate is 1 after the reception of these one burst and the packets to be transmitted are divided into blocks of 240-8 bits each, which constitutes the user block including data, CRC, block number, and encoder tail. If the decoding is not successful, the second burst is sent and decoding is reattempted. After the second burst, the code rate is 1/2. If the decoding is still not successful, the third burst is sent and decoding is done again, now with the code rate of 1/3. If the decoding is not successful, the burst with the lowest signal to noise-and-interference value is resent and the original burst and the retransmitted burst are combined by maximum ratio combining. This repetition coding is repeated until the decoding is successful. The system parameters for implementing the UDD services are summarised in Table 26. Both code pooling and time slot pooling are considered.

Table 26. System parameters for the UDD services

Service	**UDD, 144 kbit/s, 384 kbit/s and 2048 kbit/s, no delay constraint**
user bit rate	variable
Number of time slots per frame per user	variable
Number of codes per time slot per user	variable
burst type	spread speech/data burst 2
data modulation	QPSK
Convolutional code rate (inner code)	variable, 1, 1/2, 1/3
Interleaving depth	1,2,3 frames = 1,2,3 bursts
user block size	232 bits for QPSK
Antenna diversity	yes (2 branches)

In the link level simulations, an ideal CRC (cyclic redundancy check) is modelled. The effects of ARQ are included in the system level simulations. The aim of the link-level simulations is to find the required E_b/N_0 values to achieve certain BERs and BLERs. The following alternatives have been simulated as being extreme cases:

- allocating 1 code to a user in the uplink, with 1 user being active per time slot.
- allocating 9 codes to a user in the uplink, with 1 user being active per time slot.

- allocating 1 code to a user in the downlink, with 4 users being active per time slot.
- allocating 9 codes to a user in the downlink, with 1 user being active per time slot.
- allocating 1 code to a user in the uplink, with 8 user being active per time slot.
- allocating 3 codes to a user in the uplink, with 1 user being active per time slot.
- allocating 3 codes to a user in the downlink, with 2 users being active per time slot.
- allocating 3 codes to a user in the downlink, with 3 user being active per time slot.
- allocating 4 codes to a user in the uplink, with 1 user being active per time slot.

These cases are extreme cases with respect to code pooling. The performance when pooling other numbers of codes is between the extreme cases given here. Since it is likely that these applications will be executed on a notebook, antenna diversity in the downlink is also included in the results.

Based on a throughput analysis, the bit rates achievable depending on the average C/I are determined. The achievable bit rates are determined by taking into account the necessary retransmissions due to block errors and the related decrease of the effective information bit rate. The considered ARQ scheme is improving the code rate from one transmission to the next from 1 to 1/2 and to 1/3.

The required C/I values in order to achieve the required nominal bit rate are summarised in Table 27 for UDD 64 kbit/s, in Table 28 for UDD 144 kbit/s, in Table 29 for UDD 384 kbit/s and in Table 30 for UDD 2048 kbit/s.

Table 27. Required value for C/I for the UDD 64 kbit/s service, code rates 1, 1/2 and 1/3.

UDD 64	10 $\log_{10}$ C/I in dB @ 64 kbit/s
	K_c = 4, K = 1, TS = 1 (UL / DL)
Indoor Office A, 3 km/h	-6.2 / -5.8
Outdoor to Indoor and Pedestrian A, 3 km/h	-5.9 /-6.5
Vehicular A, 120 km/h	-6.7 /-7.1

Kc = number of codes per time slot per user, K = number of users per time slot, UL/DL = uplink/downlink, TS = number of time slots per user

Table 28. Required value for C/I for the UDD 144 kbit/s service, code rates 1, 1/2 and 1/3.

UDD 144	10 $\log_{10}$ C/I in dB @ 144 kbit/s
	K_c = 9, K = 1, TS = 1 (UL / DL)
Vehicular A, 120 km/h	-4.8 / -4.8

Kc = number of codes per time slot per user, K = number of users per time slot, UL/DL = uplink/downlink, TS = number of time slots per user

Table 29. Required value for C/I for the UDD 384 kbit/s service, code rates 1, 1/2 and 1/3.

UDD 384	10 $\log_{10}$ C/I in dB @ 384 kbit/s
	K_c = 9, K = 1, TS = 3, (UL / DL)
Outdoor to Indoor and Pedestrian A, 3 km/h	-8.1 / -8.1

Kc = number of codes per time slot per user, K = number of users per time slot, UL/DL = uplink/downlink, TS = number of time slots per user

Table 30. Required value for C/I for the UDD 2048 kbit/s service, code rates 1, 1/2 and 1/3.

UDD 2048	10 $\log_{10}$ C/I in dB @ 2048 kbit/s
	K_c = 9, K = 1, TS = 13, (UL / DL)
Indoor Office A, 3 km/h	-5.0 / -5.0

Kc = number of codes per time slot per user, K = number of users per time slot, UL/DL = uplink/downlink, TS = number of time slots per user

3.3.3 FDD System-Level Simulations

Dynamic system simulations have been performed for three different types of services in three different environments described in Annex B of ETSI TR 101 112. In these simulations all base stations are assumed to be equipped with *one* 4.096 Mcps UTRA/FDD carrier using 5 MHz carrier spacing (assuming 3 carriers within 15 MHz). It is likely that the concept will perform better if a larger bandwidth is used for higher data rates due to a better interference averaging. Therefore all results of higher data rate services shall be regarded as pessimistic results. Also, the simulations of the UDD services have only used a *fixed* bit rate radio bearer, which will also decrease the performance of the UDD services.

3.3.3.1 Circuit-Switched Services

Four circuit-switched services, speech, LCD 144, LCD 384 and LCD2048, have been evaluated by means of dynamic system simulations. The performance measure of the speech (8 kbps, 50% voice activity) and LCD services is that 98% of the users are *satisfied*. No admission control has been used; therefore no users are blocked due to high interference level. Also, the simulation results show that cell capacity in all cases is limited by the requirement that a satisfied user must have sufficiently good quality more than 95% of the session time or the blocking criteria has been reached and not by the dropping criteria.

The UTRA/FDD concept uses fast power control also in downlink. This means that slow moving users can compensate for the fast channel fading, hence no substantial diversity gain from connecting more base stations (i.e. increase the maximum number of active set) is seen. Connecting more base stations will only increase the required capacity of base station to base station controller transmission. Users moving with high speed do not require good tracking of the fast channel fading due to the gain from coding and interleaving.

Speech and LCD results are presented in Table 31. Please note that in Table 31 the term “cell” is defined as an area covered by a sector.

The speech service is evaluated using 50% voice activity. However, the DPCCH is transmitted with constant bit rate (and energy) independent of the speech user information rate (8 kbps or 0 kbps information bit rate). Therefore, the spectrum efficiency (measured in kbps/MHz/cell) will increase more than 25-30% when an equivalent 8kps DPCH is transmitted with 100% duty cycle (e.g. 8kbps circuit switched data). This is due to the decreased DPCCH overhead as a percentage of the total DPCH energy.

3.3.3.2 Packet Services

Three different packet data services have been evaluated: UDD 144, UDD 384 and UDD 2048. The performance measure of the packet services is that 98% of the users are *satisfied*.

As mentioned before the UDD services are evaluated using a fix rate bearer, e.g. 60.8 kbps for UDD 144 and 486.4 kbps for UDD 2048. Better interference averaging will be achieved if higher chip rate or variable rate is used, i.e. this will improve the results for the high data rates.

The results for the UDD services are shown Table 31.

3.3.3.3 Capacity results

Table 31 shows the capacity results. The table also includes the link level results that have been used in the system simulations.

Table 31. Summary of simulation results. The voice activity is 50% for the speech service. Note that UDD bearer bit rates are not the same as the source bit rates specified for the UDD service. (NA = Not available, N/A = Not applicable)

Service	Environment	Source bit rate	E_b/N_o [dB] (UL / DL)	Cell capacity [Erlang/carrier/cell] (UL / DL)	Spectrum efficiency [kbps/MHz/cell] (UL / DL)
Speech 20 ms interl.	Indoor A, 3km/h	8 kbps	3.2 / 6.0	169 / 92[1]	135 / 74
	Indoor B, 3km/h	8 kbps	3.3 / 6.1	189 / 85	151 / 68
	Pedestrian A, 3km/h	8 kbps	3.3 / 6.1	154 / 157[1]	123 / 125
	Pedestrian B, 3km/h	8 kbps	4.3 / 7.0	129 / 95	103 / 76
	Vehicular A, 120km/h	8 kbps	5.4 / 7.9	112 / 89[1]	90 / 71
	Vehicular B, 120km/h	8 kbps	5.3 / 8.0	115 / 69	92 / 55
LCD	Indoor A, 3km/h	64 kbps	2.3 / 1.9	18.1 / 21.3[1]	231 / 273
		2048 kbps	1.8 / 1.6	0.43 / 0.12[1] 0.58[2] / (N/A)	176 / 47 238 / (N/A)
	Pedestrian A, 3km/h	64 kbps	2.4 / 1.9	16.4 / 33.7[1]	210 / 431
		384 kbps	1.3 / 1.1	3.5 / 6.0[3] 4.9[2] / (N/A)	269 / 461 379 / (N/A)
	Vehicular A, 120km/h	64 kbps	3.8 / 3.7	14.0 / 12.4	179 / 158
		144 kbps	3.1 / 2.5	7.3 / 7.3[1]	210 / 210

[1] Two code sets are used.

[2] C/I based soft handover.

[3] Four code sets are used.

		384 kbps	3.1 / 3.2	2.4 / 2.3 2.4 / 2.1[1]	183 / 177 183 / 161
UDD	Indoor A, 3km/h	64 kbps	1.5 / 1.2	96 / 154 (parallel sessions)	224 / 372
		2048 kbps	0.6 / 0.1	50 / 82[1] (parallel sessions)	273 / 453
	Indoor B, 3km/h	64 kbps	1.6 / NA	110 / NA (parallel sessions)	252 / NA
		2048 kbps	0.9 / 0.3	56 / NA (parallel sessions)	306 / 336
	Pedestrian A, 3km/h	64 kbps	1.5 / 1.2	146 / 231 (parallel sessions)	340 / 523
		384 kbps	0.4 / 0.1	91 / 135[1] (parallel sessions)	449 / 668
	Pedestrian B, 3km/h	64 kbps	2.2 / NA	144 / NA (parallel sessions)	335 / NA
		384 kbps	1.4 / 1.2	79 / 82 (parallel sessions)	388 / 415
	Vehicular A, 120km/h	64 kbps	3.8 / 3.0	65 / NA (parallel sessions)	171 / 337
		144 kbps	3.0 / 2.9	52 / 85 (parallel sessions)	202 / 290
		384 kbps	2.4 / 2.0	42 / NA (parallel sessions)	216 / 289
	Vehicular B, 120km/h	64 kbps	3.6 / NA	68 / NA (parallel sessions)	179 / NA
		144 kbps	2.9 / 2.9	53 / NA (parallel sessions)	207 / 278
		384 kbps	2.3 / NA	43 / NA (parallel sessions)	220 / NA

3.3.4

3.3.5 TDD System-Level simulations

3.3.5.1 Capacity Results

The simulation results are summarised in Table 32. Two circuit switched services, speech and LCD, and one packet service have been evaluated for each environment. The performance measure for circuit switched and packet services is in accordance with Annex D of ETSI TR 101 112 (UMTS 30.03) 7. The spectrum efficiency or cell capacity, respectively, is derived from the interference limited system load which fulfils the satisfied user criterion, i.e. for RT the requirement that a satisfied user must have sufficiently good quality more than 95% of the session time.

Table 32. Summary of the system simulation results (downlink)

Environment	Service	Spectrum Efficiency [kbps/MHz/ cell]
Vehicular	Speech	70
	LCD144	201
	UDD 144	320
Outdoor to indoor pedestrian	Speech	148
	LCD 384	330
	UDD384	642
Indoor office	Speech	73
	LCD 2048	62
	UDD2048	400

Note: For LCD and UDD services antenna diversity is used.

3.3.5.2 Summary

Simulation results concerning spectrum efficiency of UTRA-TDD have been presented for real time and non real time services. The presented values do not include the impact of signalling which however is estimated to be about 10%.

On the other hand there are some options which may significantly improve the spectrum efficiency values:

- enhanced link adaptation
- power control based on quality
- turbo coding
- smart antennas.

These options and the corresponding trade-off between capacity increase and complexity issues are for further study.

3.4 Coverage Analysis (Link Budget Calculation)

In the following pages link budgets are presented for the simulated test cases. The link budgets follows the link budget template in UMTS 30.03, and also presents some range calculations using concept optimised parameters.

3.4.1 FDD Coverage Analysis

Link budgets are only presented for the A-type of channel models. The B-type channel link budgets are easily derived by just changing the required E_b/N_0 values and the TX power increase values (see Table *34*).

3.4.1.1 Basic Assumptions

Since it is the average transmitter power per traffic channel that is specified in UMTS 30.03, power control is included in the link-level simulations to find the coverage. However, this means that the transmitted power can be increased due to the power control, and this is compensated for in the row "Power control TX power increase".

The TX power increase is dependent on the environment and service. For speech and LCD soft handoff is assumed, while UDD services assume no soft handoff. The values used are presented in Table 33 and Table 34.

Table 33. TX power increase in different Channel A environments.

Environment	Speech & LCD Uplink [dB]	Speech & LCD Downlink [dB]	UDD Uplink [dB]	UDD Downlink [dB]
Indoor office	0	2	2	4
Outdoor to indoor and pedestrian	0	2	2	4.5
Vehicular	0	0	0	0

Table 34. TX power increase in different Channel B environments.

Environment	Speech & LCD Uplink [dB]	Speech & LCD Downlink [dB]	UDD Uplink [dB]	UDD Downlink [dB]
Indoor office	-0.3	1.4	1.4	2.9
Outdoor to indoor and pedestrian	-0.4	0.6	0.6	1.2
Vehicular	0	0	0	0

The handoff gain and log-normal fade margin were calculated for 95% area coverage with a shadowing correlation of 50%. Values can be found in Table 35.

Table 35. Log-normal fade margins and handoff gains.

Environment	σ [dB]	α	Log-normal fade margin [dB]	Handoff gain (soft handoff) [dB]	Handoff gain (hard handoff) [dB]
Indoor office	12	3.0	15.4	6.1	5.9
Outdoor to indoor and pedestrian	10	4.0	11.3	5.0	4.7
Vehicular	10	3.76	11.3	5.0	4.7

Please note that the total TX EIRP is not computed (only one user is assumed). Also, the coverage analysis is done for an unloaded system. This means that the RX interference density is zero (set to -1000 dBm/Hz in the tables).

3.4.1.2 Concept Optimised Parameters

An alternative link budget is presented below the link budget according to UMTS 30.03, in which the antenna gains and TX powers are modified to more reasonable values.

The specified three-sector antenna in the Vehicular environment has a gain of only 13 dBi, which is rather low. A more reasonable value of 17 dBi has been used. The mobile antenna gain is specified as 0 dBi for all services and environments. It is expected that a mobile station handling the high bit rates will not be used next to the ear. This is taken into account by increasing the gain to 2 dBi.

The average TX powers specified in UMTS 30.03 are quite low, especially for high bit rate services. Higher values are proposed (DL/UL): Indoor office A 13/10 dBm, Outdoor to indoor and pedestrian A 23/20, Vehicular A 30/24 dBm.

		Downlink	Uplink	Downlink	Uplink	Downlink	Uplink
Test environment		Indoor	Indoor	Pedestr.	Pedestr.	Vehicular	Vehicular
Multipath channel class		A	A	A	A	A	A
Mobile speed		3 km/h	3 km/h	3 km/h	3 km/h	120 km/h	120 km/h
Test service		Speech	Speech	Speech	Speech	Speech	Speech
Note		20 ms int	20 ms int	20 ms int	20 ms int	20 ms int	20 ms int
Bit rate	bit/s	8000	8000	8000	8000	8000	8000
Average TX power per traffic ch.	dBm	10	4	20	14	30	24
Maximum TX power per traffic ch.	dBm	10	4	20	14	30	24
Maximum total TX power	dBm	10	4	20	14	30	24
Cable, conn. and combiner losses	dB	2	0	2	0	2	0
TX antenna gain	dBi	2	0	10	0	13	0
TX EIRP per traffic channel	dBm	10	4	28	14	41	24
Total TX EIRP	dBm	10	4	28	14	41	24
RX antenna gain	dBi	0	2	0	10	0	13
Cable and connector losses	dB	0	2	0	2	0	2
Receiver noise figure	dB	5	5	5	5	5	5
Thermal noise density	dBm/Hz	-174	-174	-174	-174	-174	-174
RX interference density	dBm/Hz	-1000	-1000	-1000	-1000	-1000	-1000
Total effect. noise + interf. density	dBm/Hz	-169	-169	-169	-169	-169	-169
Information rate	dBHz	39.0	39.0	39.0	39.0	39.0	39.0
Required Eb/(No+Io)	dB	6.0	3.2	6.1	3.3	7.9	5.4
RX sensitivity	dB	-124.0	-126.8	-123.9	-126.7	-122.1	-124.6
Power control TX power increase	dB	2.0	0.0	2.0	0.0	0.0	0.0
Handoff gain	dB	6.1	6.1	5.0	5.0	5.0	5.0
Explicit diversity gain	dB	0	0	0	0	0	0
Other gain	dB	0	0	0	0	0	0
Log-normal fade margin	dB	15.4	15.4	11.3	11.3	11.3	11.3
Maximum path loss	dB	122.7	121.5	143.6	142.4	156.8	153.3
Maximum range	m	717.2	654.1	773.5	721.9	5787.3	4670.8
Coverage efficiency	km^2/cell	1.6	1.3	1.9	1.6	21.8	14.2
Concept optimised parameters							
Maximum TX power per traffic ch.	dBm	13	10	23	20	30	24
TX antenna gain	dBi	2	0	10	0	17	0
RX antenna gain	dBi	0	2	0	10	0	17
Maximum path loss	dB	125.7	127.5	146.6	148.4	160.8	157.3
Maximum range	m	902.9	1036.7	919.3	1019.7	7393.7	5967.2
Coverage efficiency	km^2/cell	2.6	3.4	2.7	3.3	35.5	23.1

Table 36. Link budgets for speech service.

		Downlink	Uplink	Downlink	Uplink
Test environment		Indoor	Indoor	Indoor	Indoor
Multipath channel class		A	A	A	A
Mobile speed		3 km/h	3 km/h	3 km/h	3 km/h
Test service		LCD 64	LCD 64	LCD 2048	LCD 2048
Note					
Bit rate	bit/s	64000	64000	2048000	2048000
Average TX power per traffic ch.	dBm	10	4	10	4
Maximum TX power per traffic ch.	dBm	10	4	10	4
Maximum total TX power	dBm	10	4	10	4
Cable, conn. and combiner losses	dB	2	0	2	0
TX antenna gain	dBi	2	0	2	0
TX EIRP per traffic channel	dBm	10	4	10	4
Total TX EIRP	dBm	10	4	10	4
RX antenna gain	dBi	0	2	0	2
Cable and connector losses	dB	0	2	0	2
Receiver noise figure	dB	5	5	5	5
Thermal noise density	dBm/Hz	-174	-174	-174	-174
RX interference density	dBm/Hz	-1000	-1000	-1000	-1000
Total effect. noise + interf. density	dBm/Hz	-169	-169	-169	-169
Information rate	dBHz	48.1	48.1	63.1	63.1
Required Eb/(No+Io)	dB	1.9	2.3	1.6	1.8
RX sensitivity	dB	-119.0	-118.6	-104.3	-104.1
Power control TX power increase	dB	2.0	0.0	2.0	0.0
Handoff gain	dB	6.1	6.1	6.1	6.1
Explicit diversity gain	dB	0	0	0	0
Other gain	dB	0	0	0	0
Log-normal fade margin	dB	15.4	15.4	15.4	15.4
Maximum path loss	dB	117.7	113.3	103.0	98.8
Maximum range	m	491.2	350.4	158.3	114.7
Coverage efficiency	km^2/cell	0.8	0.4	0.1	0.0
Concept optimised parameters					
Maximum TX power per traffic ch.	dBm	13	10	13	10
TX antenna gain	dBi	2	2	2	2
RX antenna gain	dBi	2	2	2	2
Maximum path loss	dB	122.7	121.3	108.0	106.8
Maximum range	m	721.0	647.6	232.4	211.9
Coverage efficiency	km^2/cell	1.6	1.3	0.2	0.1

Table 37. Link budgets for LCD services in Indoor Office A environment.

		Downlink	Uplink	Downlink	Uplink
Test environment		Pedestr.	Pedestr.	Pedestr.	Pedestr.
Multipath channel class		A	A	A	A
Mobile speed		3 km/h	3 km/h	3 km/h	3 km/h
Test service		LCD 64	LCD 64	LCD 384	LCD 384
Note					
Bit rate	bit/s	64000	64000	384000	384000
Average TX power per traffic ch.	dBm	20	14	20	14
Maximum TX power per traffic ch.	dBm	20	14	20	14
Maximum total TX power	dBm	20	14	20	14
Cable, conn. and combiner losses	dB	2	0	2	0
TX antenna gain	dBi	10	0	10	0
TX EIRP per traffic channel	dBm	28	14	28	14
Total TX EIRP	dBm	28	14	28	14
RX antenna gain	dBi	0	10	0	10
Cable and connector losses	dB	0	2	0	2
Receiver noise figure	dB	5	5	5	5
Thermal noise density	dBm/Hz	-174	-174	-174	-174
RX interference density	dBm/Hz	-1000	-1000	-1000	-1000
Total effect. noise + interf. density	dBm/Hz	-169	-169	-169	-169
Information rate	dBHz	48.1	48.1	55.8	55.8
Required Eb/(No+Io)	dB	1.9	2.4	1.1	1.3
RX sensitivity	dB	-119.0	-118.5	-112.1	-111.9
Power control TX power increase	dB	2.0	0.0	2.0	0.0
Handoff gain	dB	5.0	5.0	5.0	5.0
Explicit diversity gain	dB	0	0	0	0
Other gain	dB	0	0	0	0
Log-normal fade margin	dB	11.3	11.3	11.3	11.3
Maximum path loss	dB	138.7	134.2	131.8	127.6
Maximum range	m	585.7	452.1	391.9	307.7
Coverage efficiency	km^2/cell	1.1	0.6	0.5	0.3
Concept optimised parameters					
Maximum TX power per traffic ch.	dBm	23	20	23	20
TX antenna gain	dBi	10	2	10	2
RX antenna gain	dBi	2	10	2	10
Maximum path loss	dB	143.7	142.2	136.8	135.6
Maximum range	m	781.1	716.5	522.6	487.7
Coverage efficiency	km^2/cell	1.9	1.6	0.9	0.7

Table 38. Link budgets for LCD services in Outdoor to Indoor and Pedestrian A environment.

		Downlink	Uplink	Downlink	Uplink	Downlink	Uplink
Test environment		Vehicular	Vehicular	Vehicular	Vehicular	Vehicular	Vehicular
Multipath channel class		A	A	A	A	A	A
Mobile speed		120 km/h	120 km/h	120 km/h	120 km/h	120 km/h	120 km/h
Test service		LCD 64	LCD 64	LCD 144	LCD 144	LCD 384	LCD 384
Note							
Bit rate	bit/s	64000	64000	144000	144000	384000	384000
Average TX power per traffic ch.	dBm	30	24	30	24	30	24
Maximum TX power per traffic ch.	dBm	30	24	30	24	30	24
Maximum total TX power	dBm	30	24	30	24	30	24
Cable, conn. and combiner losses	dB	2	0	2	0	2	0
TX antenna gain	dBi	13	0	13	0	13	0
TX EIRP per traffic channel	dBm	41	24	41	24	41	24
Total TX EIRP	dBm	41	24	41	24	41	24
RX antenna gain	dBi	0	13	0	13	0	13
Cable and connector losses	dB	0	2	0	2	0	2
Receiver noise figure	dB	5	5	5	5	5	5
Thermal noise density	dBm/Hz	-174	-174	-174	-174	-174	-174
RX interference density	dBm/Hz	-1000	-1000	-1000	-1000	-1000	-1000
Total effect. noise + interf. density	dBm/Hz	-169	-169	-169	-169	-169	-169
Information rate	dBHz	48.1	48.1	51.6	51.6	55.8	55.8
Required Eb/(No+Io)	dB	3.7	3.8	2.5	3.1	3.2	2.9
RX sensitivity	dB	-117.2	-117.1	-114.9	-114.3	-110.0	-110.3
Power control TX power increase	dB	0.0	0.0	0.0	0.0	0.0	0.0
Handoff gain	dB	5.0	5.0	5.0	5.0	5.0	5.0
Explicit diversity gain	dB	0	0	0	0	0	0
Other gain	dB	0	0	0	0	0	0
Log-normal fade margin	dB	11.3	11.3	11.3	11.3	11.3	11.3
Maximum path loss	dB	151.9	145.8	149.6	143.0	144.7	139.0
Maximum range	m	4305.2	2963.2	3734.6	2492.9	2756.4	1944.2
Coverage efficiency	km^2/cell	12.0	5.7	9.1	4.0	4.9	2.5
Concept optimised parameters							
Maximum TX power per traffic ch.	dBm	30	24	30	24	30	24
TX antenna gain	dBi	17	2	17	2	17	2
RX antenna gain	dBi	2	17	2	17	2	17
Maximum path loss	dB	157.9	151.8	155.6	149.0	150.7	145.0
Maximum range	m	6216.8	4278.9	5392.9	3599.9	3980.3	2807.5
Coverage efficiency	km^2/cell	25.1	11.9	18.9	8.4	10.3	5.1

Table 39. Link budgets for LCD services in Vehicular A environment.

		Downlink	Uplink	Downlink	Uplink
Test environment		Indoor	Indoor	Indoor	Indoor
Multipath channel class		A	A	A	A
Mobile speed		3 km/h	3 km/h	3 km/h	3 km/h
Test service		UDD 64	UDD 64	UDD 2048	UDD 2048
Note					
Bit rate	bit/s	30400	30400	486400	486400
Average TX power per traffic ch.	dBm	10	4	10	4
Maximum TX power per traffic ch.	dBm	10	4	10	4
Maximum total TX power	dBm	10	4	10	4
Cable, conn. and combiner losses	dB	2	0	2	0
TX antenna gain	dBi	2	0	2	0
TX EIRP per traffic channel	dBm	10	4	10	4
Total TX EIRP	dBm	10	4	10	4
RX antenna gain	dBi	0	2	0	2
Cable and connector losses	dB	0	2	0	2
Receiver noise figure	dB	5	5	5	5
Thermal noise density	dBm/Hz	-174	-174	-174	-174
RX interference density	dBm/Hz	-1000	-1000	-1000	-1000
Total effect. noise + interf. density	dBm/Hz	-169	-169	-169	-169
Information rate	dBHz	44.8	44.8	56.9	56.9
Required Eb/(No+Io)	dB	1.2	1.5	0.1	0.6
RX sensitivity	dB	-123.0	-122.7	-112.0	-111.5
Power control TX power increase	dB	4.0	2.0	4.0	2.0
Handoff gain	dB	5.9	5.9	5.9	5.9
Explicit diversity gain	dB	0	0	0	0
Other gain	dB	0	0	0	0
Log-normal fade margin	dB	15.4	15.4	15.4	15.4
Maximum path loss	dB	119.5	115.2	108.5	104.0
Maximum range	m	561.1	403.4	242.3	171.5
Coverage efficiency	km^2/cell	0.99	0.51	0.18	0.09
Concept optimised parameters					
Maximum TX power per traffic ch.	dBm	13	10	13	10
TX antenna gain	dBi	2	2	2	2
RX antenna gain	dBi	2	2	2	2
Maximum path loss	dB	124.5	123.2	113.5	112.0
Maximum range	m	823.6	745.4	355.6	317.0
Coverage efficiency	km^2/cell	2.13	1.74	0.40	0.32

Table 40. Link budgets for UDD services in Indoor Office A environment.

		Downlink	Uplink	Downlink	Uplink
Test environment		Pedestr.	Pedestr.	Pedestr.	Pedestr.
Multipath channel class		A	A	A	A
Mobile speed		3 km/h	3 km/h	3 km/h	3 km/h
Test service		UDD 64	UDD 64	UDD 384	UDD 384
Note					
Bit rate	bit/s	30400	30400	243200	243200
Average TX power per traffic ch.	dBm	20	14	20	14
Maximum TX power per traffic ch.	dBm	20	14	20	14
Maximum total TX power	dBm	20	14	20	14
Cable, conn. and combiner losses	dB	2	0	2	0
TX antenna gain	dBi	10	0	10	0
TX EIRP per traffic channel	dBm	28	14	28	14
Total TX EIRP	dBm	28	14	28	14
RX antenna gain	dBi	0	10	0	10
Cable and connector losses	dB	0	2	0	2
Receiver noise figure	dB	5	5	5	5
Thermal noise density	dBm/Hz	-174	-174	-174	-174
RX interference density	dBm/Hz	-1000	-1000	-1000	-1000
Total effect. noise + interf. density	dBm/Hz	-169	-169	-169	-169
Information rate	dBHz	44.8	44.8	53.9	53.9
Required Eb/(No+Io)	dB	1.2	1.5	0.1	0.4
RX sensitivity	dB	-123.0	-122.7	-115.0	-114.7
Power control TX power increase	dB	4.5	2.0	4.5	2.0
Handoff gain	dB	4.7	4.7	4.7	4.7
Explicit diversity gain	dB	0	0	0	0
Other gain	dB	0	0	0	0
Log-normal fade margin	dB	11.3	11.3	11.3	11.3
Maximum path loss	dB	139.9	136.1	131.9	128.1
Maximum range	m	625.2	502.4	396.1	318.2
Coverage efficiency	km^2/cell	1.23	0.79	0.49	0.32
Concept optimised parameters					
Maximum TX power per traffic ch.	dBm	23	20	23	20
TX antenna gain	dBi	10	2	10	2
RX antenna gain	dBi	2	10	2	10
Maximum path loss	dB	144.9	144.1	136.9	136.1
Maximum range	m	833.7	796.2	528.2	504.4
Coverage efficiency	km^2/cell	2.18	1.99	0.88	0.80

Table 41. Link budgets for UDD services in Outdoor to Indoor and Pedestrian A environment.

		Downlink	Uplink	Downlink	Uplink	Downlink	Uplink
Test environment		Vehicular	Vehicular	Vehicular	Vehicular	Vehicular	Vehicular
Multipath channel class		A	A	A	A	A	A
Mobile speed		120 km/h	120 km/h	120 km/h	120 km/h	120 km/h	120 km/h
Test service		UDD 64	UDD 64	UDD 144	UDD 144	UDD 384	UDD 384
Note							
Bit rate	bit/s	30400	30400	60800	60800	243200	243200
Average TX power per traffic ch.	dBm	30	24	30	24	30	24
Maximum TX power per traffic ch.	dBm	30	24	30	24	30	24
Maximum total TX power	dBm	30	24	30	24	30	24
Cable, conn. and combiner losses	dB	2	0	2	0	2	0
TX antenna gain	dBi	13	0	13	0	13	0
TX EIRP per traffic channel	dBm	41	24	41	24	41	24
Total TX EIRP	dBm	41	24	41	24	41	24
RX antenna gain	dBi	0	13	0	13	0	13
Cable and connector losses	dB	0	2	0	2	0	2
Receiver noise figure	dB	5	5	5	5	5	5
Thermal noise density	dBm/Hz	-174	-174	-174	-174	-174	-174
RX interference density	dBm/Hz	-1000	-1000	-1000	-1000	-1000	-1000
Total effect. noise + interf. density	dBm/Hz	-169	-169	-169	-169	-169	-169
Information rate	dBHz	44.8	44.8	47.8	47.8	53.9	53.9
Required Eb/(No+Io)	dB	3.0	3.8	2.9	3.0	2.0	2.4
RX sensitivity	dB	-121.2	-120.4	-118.3	-118.2	-113.1	-112.7
Power control TX power increase	dB	0.0	0.0	0.0	0.0	0.0	0.0
Handoff gain	dB	4.7	4.7	4.7	4.7	4.7	4.7
Explicit diversity gain	dB	0	0	0	0	0	0
Other gain	dB	0	0	0	0	0	0
Log-normal fade margin	dB	11.3	11.3	11.3	11.3	11.3	11.3
Maximum path loss	dB	155.6	148.8	152.7	146.6	147.5	141.1
Maximum range	m	5378.0	3546.2	4500.0	3097.3	3288.8	2222.4
Coverage efficiency	km^2/cell	18.79	8.17	13.15	6.23	7.03	3.21
Concept optimised parameters							
Maximum TX power per traffic ch.	dBm	30	24	30	24	30	24
TX antenna gain	dBi	17	2	17	2	17	2
RX antenna gain	dBi	2	17	2	17	2	17
Maximum path loss	dB	161.6	154.8	158.7	152.6	153.5	147.1
Maximum range	m	7765.9	5120.9	6498.2	4472.6	4749.0	3209.2
Coverage efficiency	km^2/cell	39.17	17.03	27.43	12.99	14.65	6.69

Table 42. Link budgets for UDD services in Vehicular A environment.

3.4.2 TDD Coverage Analysis

All E_b/N_0 values derived by the link level simulations are associated with the energy per bit needed to achieve the corresponding BER/FER. Thus the midamble is excluded in the E_b/N_0 and therefore included in the information rate as explained hereafter. Power control is applied on a frame by frame basis (thus every 10 ms power is changed if necessary).

QPSK modulation leads to the gross bit rate on the air of 512 kbps including midamble and guard period and 492.8 kbps including midamble and excluding the guard period. Since no energy is needed to transmit the guard period, 492.8 kbps is used to determine the information bit rate in the template. Multiplying 492.8 kbps with the total coding rate used by the service derives the information bit rate. That means the information rate becomes

- 492.8 kbps · 0.330 = 162.1 kbps for speech, data burst 1 (long midamble),
- 492.8 kbps · 0.580 = 285.8 kbps for LCD 64, data burst 2 (short midamble),
- 492.8 kbps · 0.580 = 285.8 kbps for LCD 144, data burst 2 (short midamble),
- 492.8 kbps · 0.522 = 257.2 kbps for LCD 384, data burst 2 (short midamble),
- 492.8 kbps · 0.638 = 314.4 kbps for LCD 2048, data burst 2 (short midamble),

As mentioned above, the E_b/N_0 figures are related to one code. This is referred to by multiplying the information rate with the number of codes used for the particular service.

The log-normal fade margin and the handover gain in the subsequent templates are obtained by independent quasi-static simulations. Basis of the simulations is a hexagonal cell structure, where mobiles are randomly distributed over the cell area. Each mobile is subject of fading effects according to the log-normal conditions specified for each test environment (e.g. standard deviation of 10/12 dB for outdoor/indoor environments). Thus the log normal fade margin is determined regarding one single cell. In a multiple hexagonal cell layout one gains from the possibility of maintaining the connection due to several potential serving cells. According to the pre-set conditions for the Link Budget Template, the hand off gain is calculated assuming 50% shadowing correlation. The handover gain depends on the handover margin used in the simulations. Exactly the same figures as stated in the evaluation reports of the last year are used:

Environment	log-normal fade margin	hard hand-off gain
Indoor Office	15.4	5.9
Outdoor to Indoor and Pedestrian	11.3	4.7
Vehicular	11.3	4.7

For the following reason no link budget templates are calculated for UDD services. The proposed Type II Hybrid ARQ scheme allows for retransmission in case of unsuccessful data detection. Therefore no fixed E_b/N_0 or C/I values required to achieve the QoS at the cell border can be defined. However the range of UDD services is larger than the range of the corresponding LCD service due to ARQ retransmissions.

		Downlink	Uplink	Downlink	Uplink	Downlink	Uplink
Test environment		Indoor A	Indoor A	Pedestr. A	Pedestr. A	Vehicular A	Vehicular A
Test service		speech	speech	speech	speech	speech	speech
Note	1 TS, 1 Code	burst1	burst1	burst1	burst1	burst1	burst1
bit rate per traffic channel (incl. midamble)	bps	10133	10133	10133	10133	10133	10133
average TX per traffic channel	dBm	10	4	20	14	30	24
max. TX power per traffic channel	dBm	22,0	16,0	32,0	26,0	42,0	30,0
max. total TX power	dBm	31,1	16,0	41,1	26,0	51,1	30,0
cable, conn. and combiner losses	dB	2	0	2	0	2	0
TX antenna gain	dBi	2	0	10	0	13	0
TX eirp per traffic channel	dBm	22,0	16,0	40,0	26,0	53,0	30,0
total TX eirp	dBm	31,1	16,0	49,1	26,0	62,1	30,0
RX antenna gain	dBi	0	2	0	10	0	13
cable and connector losses	dB	0	2	0	2	0	2
RX noise figure	dB	5	5	5	5	5	5
thermal noise density	dBm/Hz	-174	-174	-174	-174	-174	-174
RX interference density	dBm/Hz	-1000	-1000	-1000	-1000	-1000	-1000
total effect. noise plus interf. density	dBm/Hz	-169	-169	-169	-169	-169	-169
information rate Rb	dbHz	52,1	52,1	52,1	52,1	52,1	52,1
required Eb/(N0+I0)	dB	6	3,6	6,1	3,7	8,3	5,8
receiver sensitivity	dBm	-110,9	-113,3	-110,8	-113,2	-108,6	-111,1
hand off gain	dB	5,9	5,9	4,7	4,7	4,7	4,7
explicit diversity gain	dB	0	0	0	0	0	0
other gain	dB	0	0	0	0	0	0
log-normal fade margin	dB	15,4	15,4	11,3	11,3	11,3	11,3
maximum path loss	dB	123,4	119,8	144,2	140,6	155,0	145,5
maximum range	m	**761,1**	**577,3**	**805,5**	**654,7**	**5206,7**	**2902,7**
coverage efficiency	km²/site	1,8	1,0	2,0	1,3	17,6	5,5
Concept optimised parameters							
average TX per traffic channel	dBm	13	10	23	20	30	24
max. TX power per traffic channel	dBm	25,0	22,0	35,0	30,0	42,0	30,0
TX antenna gain	dBi	2	0	10	0	17	0
RX antenna gain	dBi	0	2	0	10	0	17
maximum path loss	dB	126,4	125,8	147,2	144,6	159,0	149,5
maximum range	m	**958,1**	**915,0**	**957,3**	**822,3**	**6651,8**	**3708,4**
coverage efficiency	km²/site	2,9	2,6	2,9	2,1	28,7	8,9

Table 43. Link budgets for speech service in channel A environment.

		Downlink	Uplink	Downlink	Uplink	Downlink	Uplink
Test environment		Indoor A	Indoor A	Pedestr. A	Pedestr. A	Vehicular A	Vehicular A
Test service		LCD 64	LCD 64	LCD 64	LCD 64	LCD 64	LCD 64
Note	1 TS, 4 codes	burst2	burst2	burst2	burst2	burst2	burst2
bit rate per traffic channel (incl. midamble)	bps	71456	71456	71456	71456	71456	71456
average TX per traffic channel	dBm	10	4	20	14	30	24
max. TX power per traffic channel	dBm	22,0	16,0	32,0	26,0	42,0	30,0
max. total TX power	dBm	22,0	16,0	32,0	26,0	42,0	30,0
cable, conn. and combiner losses	dB	2	0	2	0	2	0
TX antenna gain	dBi	2	0	10	0	13	0
TX eirp per traffic channel	dBm	22,0	16,0	40,0	26,0	53,0	30,0
total TX eirp	dBm	22,0	16,0	40,0	26,0	53,0	30,0
RX antenna gain	dBi	0	2	0	10	0	13
cable and connector losses	dB	0	2	0	2	0	2
RX noise figure	dB	5	5	5	5	5	5
thermal noise density	dBm/Hz	-174	-174	-174	-174	-174	-174
RX interference density	dBm/Hz	-1000	-1000	-1000	-1000	-1000	-1000
total effect. noise plus interf. density	dBm/Hz	-169	-169	-169	-169	-169	-169
information rate Rb	dBHz	60,6	60,6	60,6	60,6	60,6	60,6
required Eb/(N0+I0)	dB	3,1	3,2	3,1	3,3	3,7	3,9
receiver sensitivity	dBm	-105,3	-105,2	-105,3	-105,1	-104,7	-104,5
hand off gain	dB	5,9	5,9	4,7	4,7	4,7	4,7
explicit diversity gain	dB	0	0	0	0	0	0
other gain	dB	0	0	0	0	0	0
log-normal fade margin	dB	15,4	15,4	11,3	11,3	11,3	11,3
maximum path loss	dB	117,9	111,8	138,8	132,6	151,2	138,9
maximum range	m	**495,8**	**310,4**	**587,5**	**411,1**	**4104,8**	**1939,7**
coverage efficiency	km²/site	0,8	0,3	1,1	0,5	52,9	11,8
Concept optimised parameters							
average TX per traffic channel	dBm	13	10	23	20	30	24
max. TX power per traffic channel	dBm	25,0	22,0	35,0	30,0	42,0	30,0
TX antenna gain	dBi	2	0	10	0	17	0
RX antenna gain	dBi	0	2	0	10	0	17
maximum path loss	dB	120,9	117,8	141,8	136,5	155,2	142,9
maximum range	m	**624,2**	**492,0**	**698,2**	**516,4**	**5244,1**	**2478,0**
coverage efficiency	km²/site	1,2	0,8	1,5	0,8	86,4	19,3

Table 44. Link budgets for LCD64 service in channel A environments.

		Downlink	Uplink
Test environment		Vehicular A	Vehicular A
Test service		LCD 144	LCD 144
Note	1 TS, 9 codes	burst2	burst2
bit rate per traffic channel (incl. midamble)	bps	160776	160776
average TX per traffic channel	dBm	30	24
max. TX power per traffic channel	dBm	42,0	30,0
max. total TX power	dBm	42,0	30,0
cable, conn. and combiner losses	dB	2	0
TX antenna gain	dBi	13	0
TX eirp per traffic channel	dBm	53,0	30,0
total TX eirp	dBm	53,0	30,0
RX antenna gain	dBi	0	13
cable and connector losses	dB	0	2
RX noise figure	dB	5	5
thermal noise density	dBm/Hz	-174	-174
RX interference density	dBm/Hz	-1000	-1000
total effect. noise plus interf. density	dBm/Hz	-169	-169
information rate Rb	dBHz	64,1	64,1
required Eb/(N0+I0)	dB	4,1	4,1
receiver sensitivity	dBm	-100,8	-100,8
hand off gain	dB	4,7	4,7
explicit diversity gain	dB	0	0
other gain	dB	0	0
log-normal fade margin	dB	11,3	11,3
maximum path loss	dB	147,2	135,2
maximum range	m	**3228,4**	**1544,3**
coverage efficiency	km²/site	32,7	7,5
Concept optimised parameters			
average TX per traffic channel	dBm	30	24
max. TX power per traffic channel	dBm	42,0	30,0
TX antenna gain	dBi	17	0
RX antenna gain	dBi	0	17
maximum path loss	dB	151,2	139,2
maximum range	m	**4124,5**	**1973,0**
coverage efficiency	km²/site	53,4	12,2

Table 45. Link budgets for LCD144 service in channel A environments.

		Downlink	Uplink
Test environment		Pedestr. A	Pedestr. A
Test service		LCD 384	LCD 384
Note	3 TS, 9 codes	burst2	burst2
bit rate per traffic channel (incl. midamble)	bps	434095	434095
average TX per traffic channel	dBm	20	14
max. TX power per traffic channel	dBm	27,3	21,3
max. total TX power	dBm	27,3	21,3
cable, conn. and combiner losses	dB	2	0
TX antenna gain	dBi	10	0
TX eirp per traffic channel	dBm	35,3	21,3
total TX eirp	dBm	35,3	21,3
RX antenna gain	dBi	0	10
cable and connector losses	dB	0	2
RX noise figure	dB	5	5
thermal noise density	dBm/Hz	-174	-174
RX interference density	dBm/Hz	-1000	-1000
total effect. noise plus interf. density	dBm/Hz	-169	-169
information rate Rb	dBHz	63,6	63,6
required Eb/(N0+I0)	dB	1,4	1,4
receiver sensitivity	dBm	-104,0	-104,0
hand off gain	dB	4,7	4,7
explicit diversity gain	dB	0	0
other gain	dB	0	0
log-normal fade margin	dB	11,3	11,3
maximum path loss	dB	132,6	126,6
maximum range	m	**412,7**	**292,1**
coverage efficiency	km²/site	0,5	0,3
Concept optimised parameters			
average TX per traffic channel	dBm	23	20
max. TX power per traffic channel	dBm	30,3	27,3
TX antenna gain	dBi	10	0
RX antenna gain	dBi	0	10
maximum path loss	dB	135,6	132,6
maximum range	m	**490,5**	**412,7**
coverage efficiency	km²/site	0,8	0,5

Table 46. Link budgets for LCD384 service in channel A environments.

		Downlink	Uplink
Test environment		Indoor A	Indoor A
Test service		LCD 2048	LCD 2048
Note	13 TS, 9 codes	burst2	burst2
bit rate per traffic channel (incl. midamble)	bps	2299096	2299096
average TX per traffic channel	dBm	10	4
max. TX power per traffic channel	dBm	10,9	4,9
max. total TX power	dBm	10,9	4,9
cable, conn. and combiner losses	dB	2	0
TX antenna gain	dBi	2	0
TX eirp per traffic channel	dBm	10,9	4,9
total TX eirp	dBm	10,9	4,9
RX antenna gain	dBi	0	2
cable and connector losses	dB	0	2
RX noise figure	dB	5	5
thermal noise density	dBm/Hz	-174	-174
RX interference density	dBm/Hz	-1000	-1000
total effect. noise plus interf. density	dBm/Hz	-169	-169
information rate Rb	dBHz	64,5	64,5
required Eb/(N0+I0)	dB	2,8	2,8
receiver sensitivity	dBm	-101,7	-101,7
hand off gain	dB	5,9	5,9
explicit diversity gain	dB	0	0
other gain	dB	0	0
log-normal fade margin	dB	15,4	15,4
maximum path loss	dB	103,1	97,1
maximum range	m	**159,5**	**100,7**
coverage efficiency	km²/site	0,1	0,0
Concept optimised parameters			
average TX per traffic channel	dBm	13	10
max. TX power per traffic channel	dBm	13,9	10,9
TX antenna gain	dBi	2	0
RX antenna gain	dBi	0	2
maximum path loss	dB	106,1	103,1
maximum range	m	**200,8**	**159,5**
coverage efficiency	km²/site	0,1	0,1

Table 47. Link budgets for LCD2048 service in channel A environments.

3.5

3.6 UTRA/FDD Results with turbo codes

3.6.1 Link-Level Simulations

Turbo codes have been studied as an alternative to convolutional and concatenated coding. Table 48 to Table 51 show the simulation parameters that have been used for these simulations. The selection process of turbo codes is still ongoing in ETSI and the parameters are therefore tentative. The MIL (Multi-stage interleaving) method is explained in Multi-stage interleaving (MIL) methods for Turbo codes, Tdoc SMG2 UMTS-L1 273/98..

Table 52 shows the link results. For comparison, the result when using convolutional and concatenated coding (convolutional coding and Reed-Solomon coding; cc+RS) is also shown.

Table 48. Parameters for LCD Services in UL

	LCD64	LCD144	LCD384	LCD2048
Physical channel rate	256 ksps	512 ksps	1024 ksps	1024 ksps * 6
Information bits	640 (10 ms)	1440 (10 ms)	3840 (10 ms)	20480 (10 ms)
CRC	16 bits per subframe (subframe=640 bits)	16 bits per subframe (subframe=720 bits)	16 bits per subframe (subframe=640 bits)	16 bits per subframe (subframe=640 bits)
Tail	2 bits per frame (1 frame= 1 subframe)	2 bits per frame (1 frame=2 subframes)	2 bits per frame (1 frame = 6 subframes)	2 bits per frame (1 frame = 32 subframes)
Turbo coder interleaving	10ms, 658 bits random interleaver	10ms, 1474 bits random interleaver	10ms, 3938 bits random interleaver	10ms, 20994 bits random interleaver
Coding rate	1/3	1/3	1/3	1/3
Rate matching	repetition 586 bits/10ms (1974=>2560)	repetition 698 bits/10ms (4422=>5120)	Puncturing 1574 bits/10ms (11814=>10240)	puncturing 1542 bits/10ms (62982=>61440)
Bit interleaving	10 ms block interleaving, 16*160	10 ms block interleaving,16*320	10 ms block interleaving, 16*640	10 ms block interleaving, 16*640 per code channel
Pilot/TPC/TFI bits per slot	6/2/2	6/2/2	6/2/2	6/2/2
Antenna receiver diversity	On	On	On	On
Channel estimation	Indoor, Outdoor to indoor and pedestrian: 5 pilot groups, alpha = (1,1,1,1,1) Vehicular: 3 pilot groups, alpha = (0.4,1,0.4)			
DPCCH/DPDCH power [dB]	-6 dB	-6 dB	Pedestrian: -9 Vehicular: -6	-6 dB relative one DPDCH code

Table 49. Parameters for UDD Services in UL

	UDD64	UDD144	UDD384	UDD2048
Information bit rate	30.4 kbps	60.8 kbps	243.2 kbps	486.4 kbps
Physical channel rate	128 ksps (sf_DPDCH=32)	256 ksps (sf_DPDCH=16)	1024 ksps (sf_DPDCH=4)	1024 ksps (sf_DPDCH=4)
Block size	304 bits	304 bits	304 bits	304 bits
CRC bits per block	16	16	16	16
Block number bits	12	12	12	12
# of blocks per frame	1	2	8	16
Turbo coder interleaving	10 ms random interleaver	10 ms random interleaver	10 ms random interleaver	10 ms random interleaver
Coding rate	1/3	1/3	1/3	1/3
Tail bits (after coding)	8	8	8	8
Rate matching	Repetition 276 bits/10ms (1004 -> 1280)	Repetition 552 bits/10ms (2008 -> 2560)	Repetition 2208 bits/10ms (8032 -> 10240)	Puncturing 5824 bits/10ms (16064 -> 10240)
Pilot/TPC/TFI bits per slot	6/2/2	6/2/2	6/2/2	6/2/2
Antenna receiver diversity	On	On	On	On
Channel estimation	Indoor, Outdoor to indoor and pedestrian: Present slot and 7 previous averaged Vehicular: Present slot and the previous one averaged			
DPCCH/DPDCH power [dB]	-4.3	-5.1	Pedestrian: -12 Vehicular: -9	-12

Table 50. Parameters for LCD Services in DL

	64 kbps	144 kbps	384 kbps	2048 kbps
Physical channel rate	128 ksps	256 ksps	1024 ksps	1024 ksps x 4
Information bits	1 subframe (10 ms) 1 subframe= 640 bits	2 subframes (10 ms) 1 subframe= 720 bits	6 subframes (10 ms) 1 subframe= 640 bits	32 subframes (10 ms) 1 subframe= 640 bits
CRC	13 bits per subframe (10 ms)	13 bits per subframe (10 ms)	13 bits per subframe (10 ms)	13 bits per subframe (10 ms)
Coding rate	1/3 (80 ms)			1/3 per code (80 ms)

Tail	8 bits (80 ms)	8 bits (80 ms)	8 bits (80 ms)	8 bits per code (80 ms)
Rate matching	15680>15872 repetition (80 ms)	35192>35328 repetition (80 ms)	94040>94208 repetition (80 ms)	125384>125440 per code repetition (80 ms)
MIL size	80 ms	80 ms	80 ms	80 ms
Pilot/TPC/TFI bits per slot	8/2/0	16/2/0		
Pilot/TPC bit power rate [dB]	0	0	0	3
Antenna receiver diversity	On			
Channel estimation	Indoor, Outdoor to indoor and pedestrian: 4 pilot groups alpha = (0.6,1.0,1.0,0.6) Vehicular: 6 pilot groups alpha = (0.3,0.8,1.0,1.0,0.8,0.3)			

Table 51. Parameters for UDD Services in DL

	64 kbps	144 kbps	384 kbps	2048 kbps
Information bit rate	30.4 kbps	60.8 kbps	243.2 kbps	486.4 kbps
Physical channel rate	64 ksps	128 ksps	512 ksps	1024 ksps
Block size	304 bits	304 bits	304 bits	304 bits
CRC bits per block	16	16	16	16
Block number bits	12	12	12	12
# of blocks per frame	1	2	8	16
Coding rate	1/3	1/3	1/3	1/3
Tail	8 bit (10 ms)	8 bit (10 ms)	8 bit (10 ms)	8 bit (10 ms)
Rate matching	1004>1024 repetition	2000>2016 repetition	7976>8000 repetition	15944>15968 repetition
MIL size	10 ms	10 ms	10 ms	10 ms
Pilot/TPC/TFI bits per slot	8/2/0		16/2/0	
DPCCH/DPDCH power [dB]	0	0	0	0
Antenna receiver diversity	On	On	On	On
Channel estimation	Indoor, Outdoor to indoor and pedestrian: 4 pilot groups alpha = (0.6,1.0,1.0,0.6) Vehicular: 6 pilot groups alpha = (0.3,0.8,1.0,1.0,0.8,0.3)			

Table 52. Turbo code simulation Results

		UL			DL		
		Required E_b/N_0 [dB]		Gain [dB]	Required E_b/N_0 [dB]		Gain [dB]
		cc+RS	turbo		cc+RS	turbo	
LCD 64	Indoor A 3km/h	2.3	2.1	0.2	1.9	1.0	0.9
	Pedestrian A 3km/h	2.4	2.2	0.2	1.9	1.1	0.8
	Vehicular A 120km/h	3.8	3.8	0	3.7	3.3	0.4
LCD144	Vehicular A 120km/h	3.1	2.9	0.2	2.5	2.0	0.5
	Vehicular B 120km/h	3.6	3.2	0.4	2.6	2.0	0.6
LCD384	Pedestrian A 3km/h	1.3	0.6	0.7	1.1	0.1	1.0
	Pedestrian B 3km/h	2.6	1.8	0.8	2.5	1.6	0.9
	Vehicular A 120km/h	3.1	2.7	0.4	3.2	2.4	0.8
LCD2048	Indoor A 3km/h	1.8	0.25	1.55	1.6	0.4	1.2
	Indoor B 3km/h	2.9	2.1	0.8	2.6	1.0	1.6
UDD64	Indoor A 3km/h	1.5	1.4	0.1	1.2	1.1	0.1
	Pedestrian A 3km/h	1.5	1.4	0.1	1.2	1.1	0.1
	Vehicular A 120km/h	3.8	3.6	0.2	3.0	3.0	0.0
UDD144	Vehicular A 120km/h	3.0	2.9	0.1	2.9	2.8	0.1
	Vehicular B 120km/h	2.9	2.8	0.1	2.9	2.8	0.1
UDD384	Pedestrian A 3km/h	0.4	0.2	0.2	0.1	-0.1	0.2
	Pedestrian B 3km/h	1.4	1.1	0.3	1.2	1.1	0.1
	Vehicular A 120km/h	2.4	2.3	0.1	2.0	1.9	0.1
UDD2048	Indoor A 3km/h	0.6	0.2	0.4	0.1	0.0	0.1
	Indoor B 3km/h	0.9	0.5	0.4	0.3	0.0	0.3

3.6.2

3.6.3 System-Level Simulations

The system simulations were performed as described in Section FDD System-Level Simulations, and the results are presented in Table 53.

Table 53. Summary of turbo code simulation results. Note that UDD bearer bit rates are not the same as the source bit rates specified for the UDD service. (NA = Not available.)

Service	Environment	Source bit rate	E_b/N_o [dB] (UL / DL)	Cell capacity [Erlang/carrier/cell] (UL / DL)	Spectrum efficiency [kbps/MHz/cell] (UL / DL)
LCD	Indoor A, 3km/h	64 kbps	2.1 / 1.0	18.9 / NA	241 / NA
		2048 kbps	0.25 / 0.4	0.6 / NA	240 / NA
	Indoor B, 3km/h	2048 kbps	2.1 / 1.0	0.46/ NA	188/ NA
	Pedestrian A, 3km/h	64 kbps	2.2 / 1.1	17.1 / NA	219 / NA
		384 kbps	0.6 / 0.1	4.0 / NA	307 / NA
	Pedestrian B, 3km/h	384 kbps	1.8 / 1.6	3.5/ NA	265/ NA
	Vehicular A, 120km/h	64 kbps	3.8 / 3.3	14.0 / 14.8	179 / 189
		144 kbps	2.9 / 2.0	7.6 / NA	219 / NA
		384 kbps	2.7 / 2.4	2.6/ NA	203/ NA
	Vehicular B, 120km/h	144 kbps	3.2 / 2.0	7.2 / NA	205 / NA
UDD	Indoor A, 3km/h	64 kbps	1.4 / 1.1	99 / NA	229 / NA
		2048 kbps	0.2 / 0.0	55 / NA	302 / 366
	Indoor B, 3km/h	2048 kbps	0.5 / 0.0	62 / NA	335 / NA
	Pedestrian A, 3km/h	64 kbps	1.4 / 1.1	149 / NA	348 / NA
		384 kbps	0.2 / -0.1	96 / NA	470 / NA
	Pedestrian B, 3km/h	384 kbps	1.1 / 1.1	83 / NA	413 / NA
	Vehicular A, 120km/h	64 kbps	3.6 / 3.0	68 / NA	181 / 337
		144 kbps	2.9 / 2.8	53 / NA	206 / 322
		384 kbps	2.3 / 1.9	42 / NA	220 / 300
	Vehicular B, 120km/h	144 kbps	2.8 / 2.8	54 / NA	211 / 287

3.7 References

[1] UMTS TR 30.03, "Universal Mobile Telecommunications System (UMTS); Selection procedures for the choice of radio transmission technologies of the UMTS," version 3.0.0.

[2] ETSI/SMG2 Tdoc 260/97, "Common Workplan of SMG2 UTRA Concept Groups".

[3] A. Klein, G. K. Kaleh, and P. W. Baier, "Zero forcing and minimum mean-square-error equalization for multiuser detection in code-division multiple-access channels," IEEE Transactions on Vehicular Technology, vol. 45, May 1996, pp. 276-287.

[4] B. Steiner and P. W. Baier, "Low cost channel estimation in the uplink receiver of CDMA mobile radio systems," Frequenz, vol. 47, Nov./Dez. 1993, pp. 292-298.

[5] ETSI/SMG2 Tdoc 329/97, "Next simulation test cases".

[6] P. Frenger, P. Orten, T. Ottosson, and A. Svensson, "Rate matching in multichannel systems using RCPC-codes," in Proc. IEEE Vehicular Technology Conference, Arizona, 1997, pp. 354-357.

[7] ETSI TR 101 112, “Selection procedures for the choice of radio transmission technologies of the Universal Mobile Telecommunications System UMTS (UMTS 30.03)“, version 3.2.0, April 1998.

[8] ETSI SMG2 UMTS ad hoc #3 TDoc 73, Rennes, 1997.

[9] ITU-R Circular Letter, „Request for Submission of Candidate Radio Transmission Technologies (RTTs) for the IMT-2000/FPLMTS Radio Interface,“ Circular Letter 8/LCCE/47 with updates.

[10] Multi-stage interleaving (MIL) methods for Turbo codes, Tdoc SMG2 UMTS-L1 273/98.

Attachment 6

Minimum Performance Capabilities for IMT-2000 Candidate Radio Transmission Technologies

Test environments	**Reference**	**Indoor Office**	**Outdoor to Indoor and Pedestrian**	**Vehicular**
Mobility considerations		mobility type (low)	mobility type (medium)	mobility type (high)
Handover	A1.2.24, A1.2.24.1, A1.2.24.2	Yes	Yes	Yes
Support of general service capabilities				
Packet data	A1.2.20, A1.2.20.1, A1.2.20.2, A1.2.20.3, A1.2.20.4	Yes	Yes	Yes
Asymmetric services	A1.2.3	Yes	Yes	Yes
Multimedia	A1.2.21, A1.2.30.1, A1.2.31	Yes	Yes	Yes
Variable bit rate	A1.2.18, A1.2.18.1	Yes	Yes	Yes
Data services key capabilities		user bit rates BER	user bit rates BER	user bit rates BER
Circuit-switched low and long delay	Capacity and Coverage Results	2 048 kbit/s $\leq 10^{-6}$	384 kbit/s $\leq 10^{-6}$	144 kbit/s $\leq 10^{-6}$
Packet	Capacity and Coverage Results	2 048 kbit/s $\leq 10^{-6}$	384 kbit/s $\leq 10^{-6}$	144 kbit/s $\leq 10^{-6}$

Attachment 7

Summary tables of key characteristics for RTT proposals based on CDMA technology and TDMA technology

ETSI has considered other terrestrial RTT proposals, and a list of the key characteristics is contained in the tables in the two Appendix to this Attachment. In the light of examining these other RTTs, and of previous experience (the ETSI concept group analysis during the development of UTRA) it is considered inappropriate to make comments or other subjective assessments of the RTT proposals, due to the difficulties inherent in the comparison of different systems, especially those with which the analysts are not familiar.

The tables are intended to be informative and the aim is to:

- facilitate the understanding of the various RTT proposals;
- highlight the commonalities between the various proposals;
- highlight the differences between the various proposals;
- assist in identifying areas where clarification or additional information is needed.

The parameters listed are relevant to layer 1 as it was felt that this is where the potential or limitations towards achieving good performance lie. Additionally, it was felt that not enough information was available for a meaningful comparison at layers 2 and 3.

It should be recognised that these tables present a snapshot taken on 15 September, of documents that were frozen before that date - they do not represent a definitive position but are living items that will change with time as the technologies develop.

It should also be stressed that these tables, in which the information has been drawn from a variety of sources, may need additional review to verify compliance with the source documentation. It is not considered appropriate to attempt to achieve a “perfect” summary at this time.

APPENDIX 1

Summary of ETSI UTRA and other RTT proposals based on W-CDMA technology							
	ARIB(W-CDMA)	ETSI UTRA	TTA CDMA II	T1P1 WCDMA/NA	WIMS	TIA TR45.5 cdma2000	TTA CDMA I
Multiple Access	DS-CDMA	DS-CDMA	DS-CDMA	DS-CDMA	DS-CDMA	DS-CDMA or multi-carrier CDMA forward link DS-CDMA for reverse link	DS-CDMA
Chip Rate	4.096Mcps (1.024/8.192/16.384)	4.096Mcps (8.192/16.384)	4.096Mcps (1.024/8.192/16.384)	4.096Mcps (8.192/16.384)	4.096, 8.192, 16.384Mcps	3.6884Mcps (1.2288xN, N=3) (Other chip rates : Nx1,2288, N=1, 6, 9, 12)	3.6884Mcps (0.9216/14.7456)
Carrier Spacing	Flexible with 200kHz carrier raster	Flexible with 200kHz carrier raster	?	Flexible with 200kHz carrier raster	Flexible, with 200 kHz raster, 30 kHz raster under study	Not described in the RTT proposal, however in IS-95 raster equals 30 kHz carrier raster in Cellular band (800MHz) and 50 kHz in the PCS band.	1.25 MHz
Inter BS timing	Asynchronous (Sync. Possible)	Asynchronous (Sync. Possible)	Asynchronous (Sync. Possible)	Asynchronous (Sync. Possible)	Asynchronous (Sync. Possible)	Synchronous.	Sync. (Async. Is possible)
Cell Search Scheme	3 step code acquisition based on non-scrambled symbols	3 step code acquisition based on non-scrambled symbols	Two-Pilot scheme	3 step code acquisition based on non-scrambled symbols	3 step with search code	Pilot channel 1 step search of broadcast common pilot	Pilot channel 1 step search of broadcast common pilot (optional asynchronous operation adopts 2 pilot scheme)
Frame Length	10ms	10ms	10ms	10ms	10ms	5 or 20ms	10 or 20 ms
VSF(spreading code)	1-512	4-256	4-256	4-256	64	4-256 N=3 4-512 N=6 4-1024 N=9, 12	16 - 1024 8??
Intra-carrier HO	SHO	SHO	SHO	SHO	SHO	SHO	SHO

Summary of ETSI UTRA and other RTT proposals based on W-CDMA technology								
		ARIB(W-CDMA)	ETSI UTRA	TTA CDMA II	T1P1 WCDMA/NA	WIMS	TIA TR45.5 cdma2000	TTA CDMA I
Inter-carrier HO		HHO with compressed transmission	HHO with compressed transmission	HHO reduced transmission rate	HHO with compressed transmission	HHO	HHO	HHO
Transmitter diversity		TDTD under consideration	OTD/TDTD under consideration	OTD/TSTD under consideration	OTD/TSTD under consideration	OTD/TSTD under consideration	Multi-carrier Transmit Diversity OTD for direct spread TSTD under consideration	――――――
DL	Data mod.	QPSK	QPSK	QPSK	QPSK	QPSK	QPSK	QPSK
	Spreading mod.	QPSK	BPSK	QPSK	BPSK	QPSK	QPSK	QPSK
	Spreading code	1 symbol length	1 symbol length	1 symbol length	1 symbol length	1 symbol length	1 symbol length	1 symbol length
	Scrambling code	10ms	10ms	10ms	10ms	1 symbol length (one 625us time-slot, Cell specific))	2^{15} chips 26.7 ms	20ms
	Pilot structure	TCH dedicated Pilot sym. Common pilot symbol	TCH dedicated Pilot sym. Common pilot symbol	TCH dedicated Pilot sym. Common pilot symbol	TCH dedicated Pilot sym. Common pilot symbol	TCH dedicated Pilot sym	Common Pilot symbols/ Auxiliary PL	Common Pilot
		Time multiplexed	Time multiplexed	Time multiplexed	Time multiplexed	Time mux.	Code multiplexed	Code Mux.
	Detection	coherent	coherent	coherent	coherent	coherent	Coherent	coherent
	Power control	Closed-loop based on dedicated ch SIR - 1.6kbps	Closed-loop based on dedicated ch SIR - 1.6kbps	Closed-loop based on Fund. Ch SIR – 1.6kbps	Closed-loop based on dedicated ch SIR - 1.6kbps	Closed-loop (0.8-0.1kbps cycles/sec) 1.6 kbps	Closed-loop based on fund. Ch SIR – 0.8kbps ***or DCCH***	Closed-loop (1.6kbps SIR based)

Summary of ETSI UTRA and other RTT proposals based on W-CDMA technology

		ARIB(W-CDMA)	ETSI UTRA	TTA CDMA II	T1P1 WCDMA/NA	WIMS	TIA TR45.5 cdma2000	TTA CDMA I
UL	Variable rate accommodation	Orthogonal VSF + Multi-Code (MC)+DTX	Orthogonal VSF + Multi-Code (MC)***+DTX***	Orthogonal VSF + Multi-code	Orthogonal VSF + Multi-Code (MC)+DTX	Orthogonal VSF + VTS+VMC+ DTX ***Orthogonal MC + DTX***	Orthogonal VSF + Repetition + ***Multicode + DTX***	Orthogonal VSF + Rep. ***+ MC***
	Data mod.	BPSK	BPSK	BPSK	BPSK	QPSK	BPSK	BPSK
	Spreading mod.	HPSK (=OCQPSK)	HPSK	HPSK (=OCQPSK)	QPSK	QPSK	QPSK	OCQPSK
	Spreading code	1 symbol length	1 symbol length	1 symbol length	1 symbol length	1 symbol length	1 symbol length	1 symbol length
	Scrambling code	2^{16}x10ms	10ms/256chips	10ms	10ms/256chips	1 symbol length	2^{42}-1 chips 2^{42} chips	2^{42}-1 chips ***??***
	Pilot structure	IQ/code multiplexed	IQ/code multiplexed	Code multiplexed	IQ/code multiplexed	Time mux. Time and code mux	**IQ/**code multiplexed	IQ/code mux.
	Detection	coherent	coherent	coherent	coherent	coherent	Coherent	coherent
	Power control	Open-loop(initial, RACH), Closed-loop (1.6kbps DCH SIR based)	Open-loop (initial, RACH) Closed-loop (1.6kbps DCH SIR based)	Open-loop + Closed-loop (1.6kbps Pilot CH SIR based)	Open-loop (initial) + Closed-loop (1.6kbps DCH SIR based)	Open-loop(initial), Closed-loop (0.8-0.1k cycles/sec) **1.6 kbps ??**	Open-loop + Closed-loop (0.8kbps Pilot CH SIR based)	Open-loop + Closed-loop (1.6 kbps Pilot CH SIR based
	Variable rate concept	VSF+ Rate Matching + Multi-code	VSF+ Rate Matching + Multi-code	VSF+ Multi-code	VSF+ Rate Matching + Multi-code	**VSF**+VTS+VMC MC + Rate matching?	VSF + Rate Matching (Repetition/Puncturing) **+** Multicode	VSF + Multicode
Channel Coding		Convolutional codes Turbo codes	Convolutional codes RS codes (Turbo codes under study)	Convolutional codes Turbo codes	Convolutional codes RS codes Turbo under study	Convolutional. Code RS code Turbo under study	Convolutional codes Turbo codes	Convolutional. Code RS code

Summary of ETSI UTRA and other RTT proposals based on W-CDMA technology							
	ARIB(W-CDMA)	ETSI UTRA	TTA CDMA II	T1P1 WCDMA/NA	WIMS	TIA TR45.5 cdma2000	TTA CDMA I
Interleaving periods	10/20/40/80ms	10/20/40/80ms		10/20/40/80ms	10/20/40/80ms	5/20ms	Outer interleaver 20 ms. Inner interleaver 10 ms. (----------)
Rate Detection	Explicit rate detection and Blind rate detection	Explicit rate detection and Blind rate detection	Rate Info. On signaling channel	Explicit rate detection and Blind rate detection	Explicit rate information	Fund. CH : Blind. Supp. CH : Scheduled No Blind detection for rates > 14.4.	Voice: Blind rate Non-voice: - Scheduled
Random Access mechanism	Message(I-ch)+ Signature(Q-ch) SF of Q-ch = 128,32	Preamble(1ms) + Message(10ms, I-ch:data,Q-ch : PL+RI), SFof I-ch =256,128,64,32		Preamble 1 ms + Message variable length SF to be determined	Preamble and various message lengths	Preamble(Nx1.25ms)+ Message (Nx5(10,20)ms)	
Power control steps	1dB	0.25-1.5	1dB	0.25 - 1.5 dB	1 dB	1dB (0.5,0.25 option)	step: 0.5 or 1dB
Super Frame Length	720ms	720ms	640ms	720 ms	720 ms	N/A	N/A

PROPOSALS BASED ON W-CDMA TECHNOLOGY INCORPORATING A TDD MODE						
	ARIB (W-CDMA)	**ETSI UTRA**	**T1P1 WCDMA/NA**	**WIMS**	**TIA TR45.5 cdma2000**	**China**
Multiple Access	TDMA/CDMA	TDMA/CDMA	TDMA/CDMA	TDMA.CDMA	DS-CDMA or multi-carrier CDMA forward link DS-CDMA for reverse link	TDMA/CDMA
Chip Rate	4.096Mcps (1.024/8.192/16.384)	4.096Mcps	4.096Mcps	4.96 Mcps 4.97(8.192//16.384)	3.6884Mcps (1.2288xN, N=3) (Other chip rates : Nx1,2288, N=1, 6, 9, 121.2288/7.3728/ 11.0592/14.7456)	1.1136Mcps
Carrier Spacing	Flexible with 200kHz carrier raster	Flexible with 200kHz carrier raster	Flexible with 200kHz carrier raster	200 kHz 30 kHz raster under consideration	1.25 MHz with 50 kHz carrier spacing	
Inter BS Sync.	Synchronous	Synchronous	Synchronous	Synchronous	Synchronous	Synchronous
Cell Search Scheme	3 step code acquisition based on non-scrambled symbols	Time multiplexed SCH in dedicated beacon slot (N repeats in 240ms)	Time multiplexed SCH in dedicated beacon slot (N repeats in 240ms)	Time mux SCH in dedicated beacon slots (in 240 ms)	Pilot channel	
Frame Length	10ms	10ms	10ms	10 ms	5 or 20ms	5ms (8 slots)
VSF (spreading code)	1-512	2-16	2-16	64	4-256 N=3 4-512 N=6 4-1024 N=9, 12	16-64
HO	SHO (DHO)	HHO	HHO		SHO	HHO
Intra-carrier HO	SHO	HHO	HHO	SHO	SHO	
Inter-carrier HO	HHO	HHO	HHO	HHO	HHO	
DL Data mod	QPSK.	QPSK	DQPSK 8PSK or 16QAM	QPSK	QPSK	QPSK

PROPOSALS BASED ON W-CDMA TECHNOLOGY INCORPORATING A TDD MODE							
		ARIB (W-CDMA)	**ETSI UTRA**	**T1P1 WCDMA/NA**	**WIMS**	**TIA TR45.5 cdma2000**	**China**
	Spreading mod.	QPSK	QPSK	QPSK	QPSK	QPSK	BPSK
	Spreading code	1 symbol length	1 symbol length	1 symbol length	1 symbol length	1 symbol length	1 symbol length
	Scrambling code	10ms	1 symbol length 10 ms	1 symbol length	1 symbol length	2^{15} chips	
	Pilot structure	TCH dedicated Pilot sym	TCH dedicated pilot sequence	TCH dedicated pilot sequence	Common pilot sequence	Common Pilot symbols/ Auxiliary PL	Common pilot symbols
		Time multiplexed	Time multiplexed	Time multiplexed	Time multiplexed	Code multiplexed	Time multiplexed
	Detection	coherent	coherent	coherent	coherent	coherent	coherent
	Power control	Fast Closed-loop (0.8-0.1kbps DCH SIR based)	Fast Closed-loop (0.8-0.1kbps cycles/sec)	Fast Closed-loop (0.8-0.1kbps cycles/sec)	___________	Fast Closed-loop (0.8kbps Fund. CH SIR based)	

	Variable rate accommodation	Orthogonal VSF + Multi-code + Variable_num_slots + DTX **+ Rate matching**	Orthogonal VSF + Multi-code + Variable_num_slots + DTX **+ Rate matching**	Orthogonal VSF + Multi-code + Variable_num_slots + DTX **+ Rate matching**	Orthogonal MC + DTX	Orthogonal VSF + Repetition **+ MC + DTX**	Variable_num_slots + Multi-code + Variable SF option (32 or 64)
UL	Data mod.	QPSK	QPSK	QPSK	QPSK	BPSK	DQPSK 8PSK or 16QAM
	Spreading mod.	QPSK	QPSK	QPSK	QPSK	QPSK	BPSK
	Spreading code	1 symbol length	1 symbol length	1 symbol length	1 symbol length	1 symbol length	1symbol length
	Scrambling code	2^{16}x10ms	1 symbol length (Cell specific – for data only)	1 symbol length (Cell specific)	1 symbol length	2^{42}-1 chips	
	Pilot structure	TCH dedicated pilot symbols	TCH dedicated pilot sequence	TCH dedicated pilot sequence	TCH dedicated pilot sequence Time mux	**IQ/**code multiplexed **Code mux**	TS0: common PL sym TS1-3: dedicated time mux PL sym
	Detection	coherent	coherent	coherent	coherent	coherent	coherent
	Power control	Fast Open-loop (Perch CH based) + Closed -loop (0.8-0.1kbps DCH SIR based)	open-loop (initial), Enhanced Closed-loop (0.8-0.1k cycles/sec)	Open-loop (initial), Enhanced Closed-loop (0.8-0.1k cycles/sec)	Open and closed loop (values under study)	Fast open-loop + Closed-loop (0.8kbps Pilot CH SIR based)	Closed loop (200 cycles/sec)
	Variable rate accommodation	Orthogonal VSF (any) + Rate Matching + Multi-code	Orthogonal VSF (1-16) +Multi-code +Variable_num_slots + DTX **+ Rate matching**	Orthogonal VSF (1-16) +Multi-code +Variable_num_slots + DTX **+ Rate matching**	Orthogonal MC + DTX	Rate Matching (Repetition/Puncturing) **VSF + RM + MC**	Variable_num_slots + Multi-code + Variable SF option (32, 64)

Channel Coding	Convolutional codes Turbo codes	Convolutional codes RS codes Turbo codes	Convolutional codes RS codes Turbo codes	Convolutional codes Turbo codes under consideration	Convolutional codes Turbo codes	Convolutional codes RS codes Turbo codes
Interleaving	10/20/40/80ms	10/20/40/80ms	10/20/40/80ms	10/20/40/80 ms	5/20ms **(---)**	10/300ms
Rate Detection	Variable length RI (with/without Blind detection)		(----)	___________	Blind **Blind rate detection for voice;** **Blind for data**	
Random Access	Message(10ms) SF=128,32	RACH dedicated slot	RACH dedicated slot	RACH dedicated slot preamble and messages	Preamble(Nx1.25ms) +Message (Nx5(10,20)ms)	Training sequence + control fields
Power control steps	1dB (DL) 0.25dB (UL)	2dB (1.5-3dB)	2dB (1.5-3dB)	1 dB	1 dB (0.25, 0,5)	1dB
Super Frame Length	720ms	720 ms	720 ms	720 ms	N/A	

APPENDIX 2

The evaluation group notes that the UWC-136 community has adopted a new modulation for 136 HS Outdoor and Indoor modes (per TR45.3.AHIC/98.08.17.04R1). This modulation is 8PSK and represents a harmonisation decision with ETSI EDGE program.

Note: In order to avoid subjective input, the list of the key characteristics provided in this document consists only of a table without any comments. Since the focus of the UWC-136 harmonisation work in progress is GSM Radio Interface based, we included a GSM Radio Interface description for reference.

Key Characteristics		TIA UWC-136	GSM Radio Interface (for reference only)
Multiple Access		TDMA	TDMA
Band Width		30/200/1600 kHz	200 kHz
Bit Rate		48.6 kbps 72.9 kbps 270.8 kbps 361.1 kbps 722.2 kbps 2.6 Mbps 5.2 Mbps	270.8 kbps for EDGE 812.5 kbps
Duplexing		FDD/TDD	FDD
Carrier Spacing		30/200/1600 kHz	200 kHz
Inter BS timing		Asynchronous (Sync. Possible)	Asynchronous (Sync. Possible)
Inter-cell Synchronization		not required	not required
Base Station synchronization		not required	not required
Cell Search Scheme		L1 power based, L2 parameter based, L3 service/network/operator based.	L1 power based, L2 parameter based, L3 service/network/operator based
Frame Length		40/40/4.6/4.6 ms	4.6 ms
HO		HHO	HHO
DL	**Data mod.**	$\pi/4$ DPSK $\pi/4$ coherent QPSK 8 PSK GMSK Q-O-QAM[1] B-O-QAM[1]	GMSK 8 PSK[2]
DL	**Power control**	Per slot and/or per carrier.	Per slot
DL	**Variable rate Accommodation**	slot aggregation	slot aggregation

[1] Proponent notes plans to change 136 HS to 8 PSK for harmonization with ETSI EDGE.

[2] EDGE program

Key Characteristics		TIA UWC-136	GSM Radio Interface (for reference only)
UL	**Data mod.**	π/4 DPSK π/4 coherent QPSK 8 PSK GMSK Q-O-QAM[1] B-O-QAM[1]	GMSK 8 PSK[2]
UL	**Power control**	BS directed MS power control	BS directed MS power control
UL	**Variable rate Accommodation**	slot aggregation.	slot aggregation.
Channel Coding		Punctured Convolutional code (R=1/2, 2/3, 3/4, 1/1) Soft or hard decision decoding	Convolutional coding. Rate dependent on service.
Interleaving periods		0/20/40/140/240 ms	Dependent on service
Rate Detection		via L3 signaling	Via stealing flags
Other Features		Space and frequency diversity; MRC/"MRC-like" Support for Hierarchical structures	MRC
Random Access mechanism		Random Access with Shared Control Feedback (SCF), also Reserved Access	Random
Power control steps		4 dB	2 dB
Super Frame Length		720 ms/640ms (Hyperframe is 1280ms)	720 ms
Slots/Frame		6 per 30 kHz carrier 8 per 200 kHz carrier 16-64 per 1.6 Mhz carrier	8
Focus of backward compatibility		AMPS/IS54/136/GSM	GSM